Synclavier

REGEN

USER MANUAL

Thank-you for purchasing the Synclavier Regen. You now have on your desk, one of the most powerful digital synthesizers ever conceived. You also own a piece of synthesizer history, this being the first new hardware Synclavier in 30 years.

In the intervening years, there have been software emulations that have largely sought to recreate the experience and feature-set of the Synclavier II. This product is more than that. It contains the same Synclavier II DSP engine, but it's been extended and enhanced to take advantage of the increased computing power available today, and to make it compelling in a modern setting, just as the original was compelling for the many happy owners of its era.

In short, a real Regen-eration of Synclavier.

Notes

WARNINGS

Synclavier Regen comes with a 12 month warranty on manufacturing defects. If you look after your Regen, it is going to be operational for many years to come.

Safety instructions

Please read and follow these safety instructions. Treat the unit itself and the power adapter with equal care.

- This product is not designed for use by young children. Adult supervision is suggested.

- If the AC power adapter is damaged, do not use it. Instead, contact Synclavier Digital Corporation Ltd for a replacement.

- The AC adapter is in standby mode when connected. The primary circuit is always active as long as the cord is connected to the outlet. Do not cover the adapter while it is connected. If the product malfunctioning, remove the adapter from the outlet. If the adapter is malfunctioning, first cut the power to the receptacle in use before removing the adapter.

- Do not expose this product to water, direct sunlight, moist, dusty, or other extreme environments.

- This product is not safety grounded and only approved for indoor use or in similar environments.

- Do not use this product near heat sources such as radiators, ovens, or amplifiers.

- Do not obstruct the ventilation slots on the sides of this unit. Fire hazard.

- Do not cover, enclose, or place things on top of the unit. This unit is not designed to operate in an enclosed space. Fire hazard.

- Pull out all cables including power during thunderstorms.

- Do not use flammable substances such as alcohol or gasoline near the product.

- For service and repairs, use only shops authorized by Synclavier Digital Corporation Ltd.

- Do not open the casing on this product. There are no user repairable parts inside.

- Use this product on a stable, flat surface. If the product is mounted on a VESA mount, ensure that the proper VESA hardware is used an all screws are tightened. Do not drop this unit from a height.

- Connect the AC adapter to a power outlet located near the product.

- Do not use this product or the sounds you create with it in such a way that it risks harming your hearing, or risks harming the hearing of others. Ensure the headphone volume is set to a low level before putting on headphones.

- Always transport or ship this product in the packaging provided. The packaging provides support. If you use standard bubble-wrap for packaging, the product will be inadequately supported and damage may occur.

- Clean using only a lint-free soft cloth, no sprays or liquids. The screen can be cleaned with a micro-fibre lens-cleaning cloth, such as the Hoya premium.

WARNING: Cancer and Reproductive Harm — www.P65Warnings.ca.gov

This product must be recycled as Electronic Waste. The packaging and any printed material should be sorted and recycled.

SYNCLAVIER REGEN DECLARATION OF CONFORMITY

This unit complies with the RoHS (Restriction of Hazardous Substances) directive.

USA

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his own expense.

Properly shielded and grounded cables and connectors must be used in order to meet FCC emission limits. Synclavier Digital is not responsible for any radio or television interference caused by using other than recommended cables and connectors or by unauthorized changes or modifications to this equipment. Unauthorized changes or modifications could void the user's authority to operate the equipment.

This device complies with Part 15 of the FCC rules. Operation is subject to the following two conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Canada

NOTICE: This class A digital apparatus meets all the requirements of the Canadian Interference-Causing Equipment Regulation.

AVIS: Cet appareil numérique de la classe A respecte toutes les exigences du Règlement sur le matériel brouilleur du Canada.

This product complies with the requirements of European Directive 89/336/EEC.

This product may not work correctly by the influence of an electrostatic discharge. If an electrostatic discharge disrupts the operation of your Regen, simply restart the unit.

EU

This product complies with the requirements of European Directive 2014/30/EU.

This product may not work correctly by the influence of an electrostatic discharge. If an electrostatic discharge disrupts the operation of your Regen, simply restart the unit.

#

CONTENTS

INTRODUCTION

1.1. Using this Manual

Synclavier Regen was designed to be intuitive, so users with previous synthesizer experience should become productive without extensive reference to this manual, but at some point in your journey, it's advisable to read it. Regen has also been designed to be a deep sound design platform, and some techniques and shortcuts will not be self-evident solely from experimentation with the machine. **Please at least read the Cheat Sheet section at the end of this chapter that covers things that will not be self-evident from exploring the unit without documentation.** Perhaps even print out the cheat sheet section as well the pages on multi-coloured buttons if that's how you work.

People learn in different ways, and while lots of the text within these pages is necessarily dry, we have broken this up with tables, diagrams, walkthroughs and so on. Walkthroughs are good, because you can follow along on your Regen, rather than passively watch a video, and each walkthrough is self-contained, typically covering a single function.

We've also created a video tutorial series, divided into block of functionality. However you learn, our advice is to tackle a block of functionality at a time —

get comfortable before progressing to the next block — even for users with prior experience of Synclavier II, there's just too much in Regen to learn it all at once.

Additionally, we've included help pages within the product itself, accessible via the [?] button. Their aim is to act as a reminder to the various parameters and functions of Regen rather than as an exhaustive explanation. The system schematic diagram at the end of the this chapter and the session memory map in the Sessions section of the LIBRARY chapter should help to "glue it all together."

So, as for this manual, as you read further the content becomes deeper. It's written as a top-down. So first covering basic operation, such as loading and saving timbres using the Library, then assembling sessions and tweaking tracks, going deeper into sound design, starting with modifying timbre-level parameters and finally, partials. This is to make the content more digestible, but also allows for different types of users that will require different levels of knowledge, to read only as much as required. For example, the gigging musician may only be interested in loading timbres and playing them, while the music producer may be interested in arranging many timbres in sessions, and finally the sound designer will want to know every in and out of the product. Of course, the lines are often blurred and you may be wearing one hat one day and another hat another day. Just know that as you read further, you will be going deeper into Pandora's box.

Updates

Synclavier Digital intends to support this product many years into the future. From time to time, obscure bugs may need fixing and new features may be added. It's worth checking periodically that you have the latest firmware. With Regen powered-on, if you tap the **Home** button then the third **Option** button, **Setup**, you will see the firmware version you are running is shown at the top of the main display. Visit the Regen Downloads page here – www.synclavier.com/regen-downloads – and near the top the latest firmware version will be stated along with some version notes and install instructions.

Likewise, this manual will get an update occasionally, so if you come back to the manual after a period, it's worth checking you're reading the latest text: the release date is shown on the title page; compare that to the one shown on the Synclavier Digital Support page – www.synclavier.com/manuals – to know if you have the latest.

Conventions in this Manual

The User interface is divided into panels. Each is surrounded by a perimeter border and has a clear title. The exception to this rule is the SELECTOR panel which has no border and its title is, in fact, 'PARTIALS or TRACKS'.

The convention for this manual is that panel titles are written in ALL CAPS. Another convention is button legends, including screen-printed hard buttons and soft button legends for the 3 option buttons, are in boldface and have the first letter capitalized, like the legends themselves. Labels are titles that appear on the main display, and sometimes they will be selectable from a list. Labels also appear capitalized in the manual, as they are on Regen,.

Pro tips and asides are written in text surrounded by a border.

> Walkthrough sections are written in indented block quotes. You should read the walkthroughs as you encounter them, even if you don't intend to follow them at that time, since they present information not necessarily covered elsewhere in the text.

Example timbres, samples, sessions, and reverbs, are often cited in the text. They will be capitalized and followed by square brackets [] containing their speed dial code (for timbres) or position number.

Tags are always shown in lower case, even at the start of a sentence, just like on the Regen displays, and prepended with the # symbol.

1.2. Getting around Regen

Rear Connections
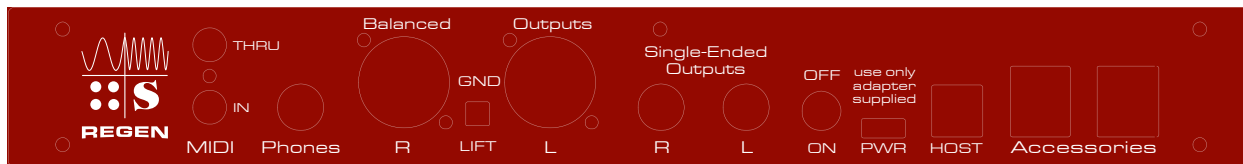
Synclavier Regen rear panel

MIDI

MIDI IN and MIDI THRU ports are provided on 1/8" TRS connectors. The connection scheme is Type A — the MIDI standard for TRS — see https://www.midi.org/midi-articles/trs-specification-adopted-and-released

Phones

A 1/4" TRS stereo headphone connector. NB: The headphone circuit in Regen has been designed to work for a range of headphones from 30 ohm commercial headphones to 300 ohms rated professional monitoring headphones. Correct performance cannot be guaranteed outside of that range. Note also, using headphone splitter cables so that two people can listen on headphones will half the effective impedance for equal-impedance headphones.

WARNING: A moderate volume setting on a pair of 300 ohms headphones would produce excessive power on headphones rated at 30 ohms, so care must be taken when first using a new pair of headphones that the **Phones** volume is initially set low, otherwise damage to headphones or hearing could result.

Balanced Outputs

Stereo pair of XLR connections for master audio output. DC coupled. +4 dBU with 11.4dB headroom.

GND LIFT

Synclavier Regen is designed to be connected to a variety of commercial and pro-level audio gear. Most of the time, you'll leave these set to GND. If you experience ground-loop feedback (hum) while using the balanced outputs you can try setting the switches to the LIFT positions. Use a small flat screwdriver to carefully change switch positions. One is for the left channel and one is for the

right, but you'd typically change both together. NB: If you are using the unbalanced connections, always set the lift switch to GND, since the unbalanced connections use the shield ground as a return for the signal.

> The outputs are DC-coupled, so if the internal ground is lifted, the signals will centre around the ground coming in on pin 1 of the XLR from the receiving equipment. If there is no ground coming in, then Regen will create a "virtual ground;" in this case, the receiving equipment would need to be AC-coupled to interpret the signal; in this case, there is likely not to be a ground loop in the first place.

Single-Ended Outputs

Stereo pair of 1/4" TS connectors for master audio output. DC coupled. -10dbV with 11.1dB headroom.

NB: Phones, Balanced, and Unbalanced outputs can all be used simultaneously with no ill effect as they each have their own driver-stage circuit.

ON-OFF

Push-button style power button. Start-up sequence takes around 22 seconds to complete. Before shutting off power, you can put your unit to sleep, by tapping the **Sleep** option button on the **Home** page. NB: If there is a change to the system state, you will have to first save it by tapping **Save State** then that soft button offers the **Sleep** function. When the unit is sleeping, you can be assured nothing will be writing to the internal disk, so it is completely safe to power off. And the current settings, such as tracks loaded, custom tuning, etc., will have been stored, for when you next power-up the unit.

PWR

USB-C connection for incoming power only. Use power adapter supplied, which is rated for 5.1V and max 3A. The included power supply has been selected for low ripple noise for minimal interference and therefore superior audio quality of the product. If you have damaged your power adapter, you can contact Synclavier Digital via the website for advice on a replacement.

HOST

USB-B connection. When Regen is in DAW mode, you can connect a USB cable here to a computer "host" (Mac, Windows, Linux). "Synclavier Regen" will show in your host machine's MIDI destinations and you can send MIDI data to it from a DAW or similar software. NB: When in Controller mode, the Accessories USB connectors cannot be used, so accessories such as a MIDI keyboard will have to be routed through your host computer.

Accessories

USB-A connections. You can connect USB accessories, such as MIDI keyboards, sequencers, and controllers, to these connections. Regen is also MPE compatible, which opens up the possibility of using this recent breed of expressive MIDI controller. NB: When Regen is in DAW mode, these ports do not function and all interfaces would have to connect to your host computer.

These ports would also be where you would connect a USB Audio Interface if you wanted to use your own, and a Synclavier® Knob, if you have one.

You can also connect a USB QWERTY keyboard for typing timbre names if you're engaged in sound design and will be saving a lot of timbre presets.

> Synclavier Regen is a synthesizer, but to get sounds out, you'll have to feed MIDI in. There are lots of MIDI keyboards and interfaces on the market that will do an admirable job of supplying MIDI information.

SD Card (not shown on diagram)

This is a slot in the side for a user's SD card. FAT32 format is recommended, but some other formats are recognized. This is to hold user samples and to save your own timbres. Each SD card can contain many "libraries" of timbres and samples, each one will be a different folder (directory).

NB: Never remove an SD Card if a warning message is showing or the BUSY light is lit.

Cooling Slots

There are arc-shaped cooling slots on the left side panel, above the SD card slot. There's also one large cooling slot that runs the length of the right side panel. There should be clearance around these slots to allow air flow in and out, so don't place your Regen up against a wall, other surface, or other device, without leaving adequate room to breathe. If these slots are blocked, devices inside the unit could overheat, causing a system malfunction.

VESA Mount

The bottom panel is equipped with a VESA mount — look for the four larger threaded holes in a square configuration, 100mm wide. These accept M4 screws, no more than 10mm long. VESA standard brackets are commonly sold to hold display monitors. Using one will mean you can get your Regen in the position and angle you desire for an ergonomic setup. Follow the instructions that come with the bracket. Most brackets will accommodate the 100 by 100 mm hole pattern and carry the 1.8kg Regen weight with attached cabling, but do check before purchasing a bracket.

User Interface
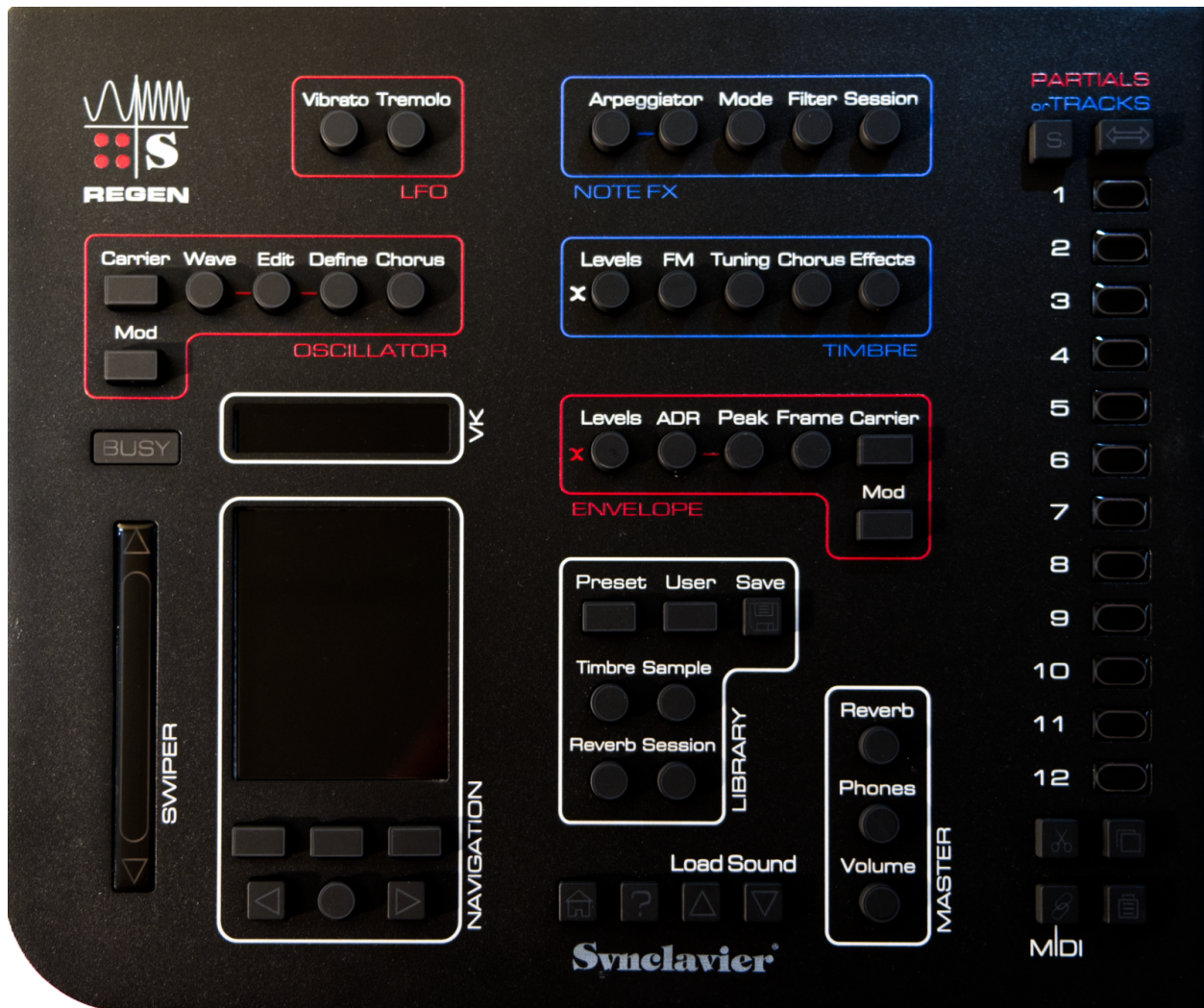
Synclavier Regen top panel

As in the original Synclavier II system, all buttons on Regen also provide annunciation feedback in the form of LED backlighting and their relative brightness dictates state information:

Brightness Level	State Meaning
Dim	Not Selectable
Medium	Selectable
Bright	Selected

In some environments, such as near a window, it may be difficult to discern the brightness of modes. For these environments, we have provided a brightness setting. Settings will be covered in the following chapter.

Single-coloured Buttons

The single-colour buttons are grouped as follows:

Colour	Meaning
Red	Parameters related to partials.
Blue	Parameters related to timbres.
White	Overall functions and session parameters.

The different colours are very useful. You'll know when you're changing a partial-level parameter or a timbre-level parameter by the colour of the selected button. The colours are also useful when loading and saving presets: you'll be reminded what bits are loaded or saved by the colour of the library button. All this and more will be covered later in the manual.

> The tracks can be thought of as layers. Each Track can be empty or contain 1 timbre. Other synthesizers may use the word preset for timbres. This is simply Synclavier's word for a sound preset. Since Synclavier was one of the earliest digital synthesizers, that over time grew into a foundational DAW, some of the terms used, such as timbre and soundfile, have been ingrained into our customers, so it might be confusing to update them to, for instance, preset and sample.

Multi-coloured Buttons

Most buttons are single-coloured, but two notable exceptions are the **Switcher** button and the **Enter/Back** button.

The **Enter**/**Back** button, in addition to being selectable or not, also serves two functions. Enter can confirm a selection or drill down into a submenu. Enter function is denoted by red. Back will cancel an operation or traverse from a subpage to a main page. Back is denoted by blue.

The Switcher button can also be red or blue and these colours carry a bit more meaning; they tie into two of the groupings of single-coloured buttons (mentioned above); and they denote the two main views of Regen. If the Switcher button is lit red, this means the machine is in PARTIALS view and the 12 selector buttons on the right-hand-side represent the 12 partials; when blue, it is in TRACKS view and the 12 selector buttons represent the 12 tracks. Pressing the switcher button changes views.

Selector button colours

The 12 **Selector** buttons are unique in that they're touch sensitive and and backlit by RGB (multicoloured) LEDs. There will be much more information on the two two main views later in the manual, but for completeness, here is the full table of colours for the selector buttons:

State View:	Tracks	Partials
Empty	Unlit	Unlit
Active	Blue	Red / White (LFO only)
Selected	Cyan	Magenta
Solo'd	Chartreuse	Green

That covers the two main views of tracks and partials. The advanced views are shown below; we suggest you don't concern yourself greatly with these views until you reach the section of the manual that discusses them.

State View:	Modulators	Harmonics (P1-8)
Unselected	blinking White	blinking yellow

Selected	blinking Magenta	blinking Magenta

1.3. Configurations

Synclavier Regen supports two mutually-exclusive modes: Controller and DAW.

Controller Mode

Select this mode from the **Home** page's **Settings** to operate Synclavier Regen as the MIDI host. An example connections configuration is shown below:
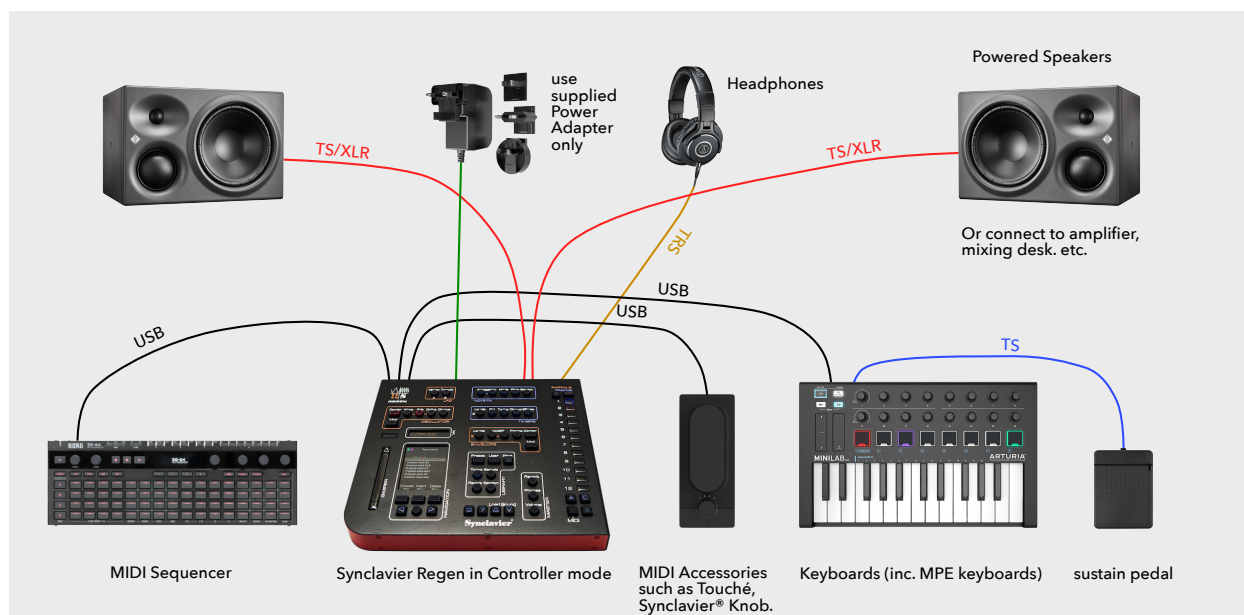
Of course, some stuff featured is optional: you don't need headphones and speakers; you may choose to use only a sequencer or keyboard; etc. And there's lots of stuff not featured that you may decide to use in your setup: you could send the MASTER audio out through effects boxes or a loop station; and since you have two sets of audio outputs that can be used simultaneously, you could send one to an amplifier and one to an audio recorder, for example.

Always use only the supplied power adapter, fitted with the correct spade for your country's power outlets. This power adapter has been selected for voltage and current ratings as well as low ripple noise.

In Controller mode, when you connect a MIDI controller, via USB, it should appear in the MIDI Inputs list (tap **Home**, **Setup,** then ⌨ to get there). To enable your controller, you may need to light the "On" green blob by tapping **Enter**. Only light the "MPE" blob for MPE keyboards.

Here's another set-up possibility, that uses a sequencing keyboard that allows for multiple MIDI tracks simultaneously. A QWERTY keyboard is also connected for easy entry of timbre names and descriptions. NB: When audio is sent over USB to an Audio Interface, as in this set-up, the internal DAC is disabled, so you can't also use the audio outputs or headphone output, but the MASTER **Volume** controls do still work.

It's likely you'll want to use the power adapter supplied with the audio interface even if it can be bus-powered, as this will mean less burden on the Regen power, and possibly less power-supply noise. Also, some keyboards that have large display of lots of LEDs will state in their manual that they only work at full brightness when connected to a dedicated power adapter.

DAW Mode

Select this mode from the **Home** page's **Settings** to operate Synclavier Regen as a MIDI client. An example connections configuration is shown below:
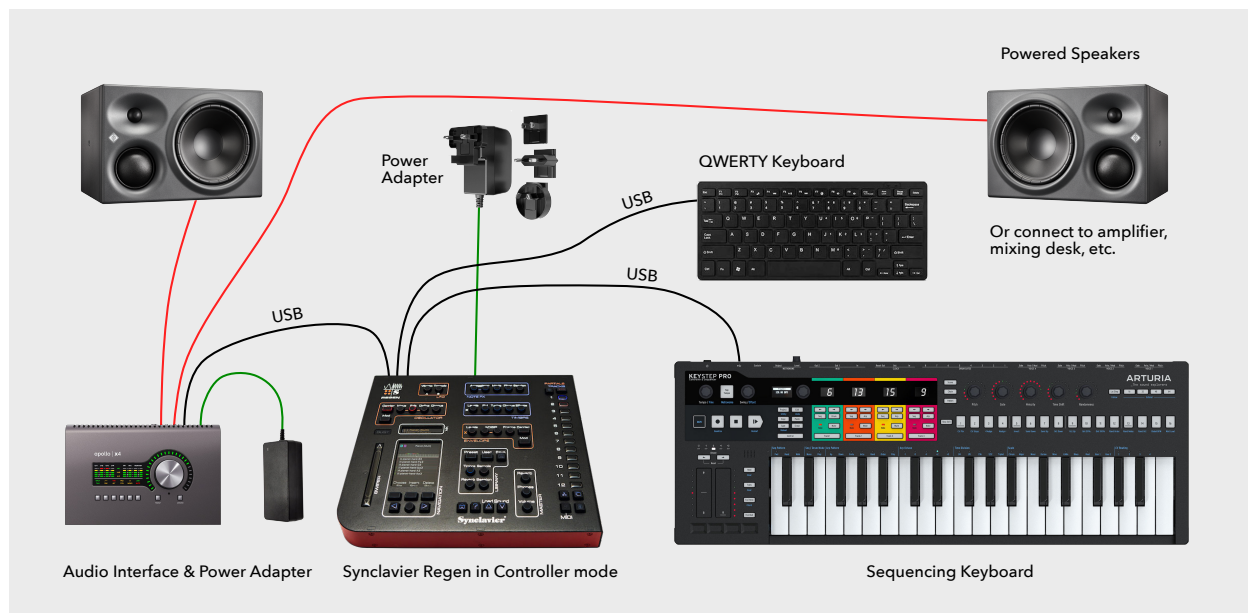

Power Amplifier

Audio Interface

TS/XLR

USB

USB

USB

into USB B
"Host"

Passive
Speakers

Synclavier Regen in DAW mode   Computer with DAW software   Keyboards (inc. MPE keyboards)
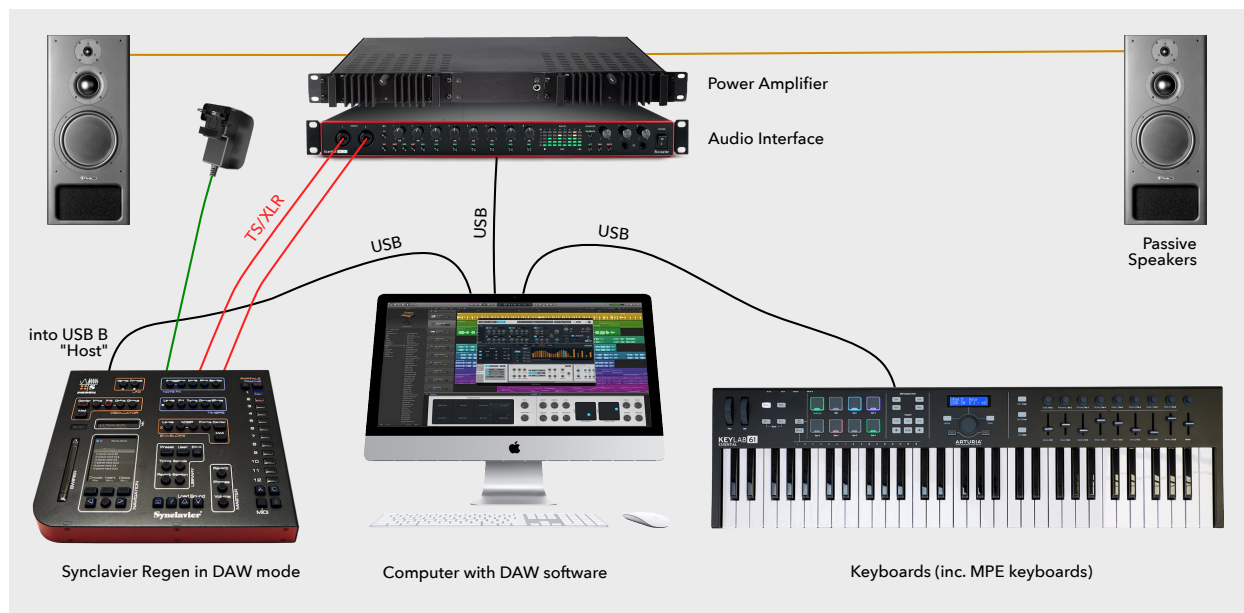
In this configuration, the computer takes up the host duty and routes MIDI messages in from keyboards and controllers. You can assign multiple **MIDI** tracks (up to 12) in your DAW for Regen and configure each to send data on a MIDI channel that you have a Regen instrument assigned. If you connect the audio from Regen to your Audio Interface, you should be able to record the audio mix back to your DAW on an **AUDIO** track.

NB: In DAW mode the USB Accessories ports do not work, so MIDI interfaces have to be connected to your host computer.

DAW mode is also also the one to use to connect Regen to an iPad. You can run a DAW application to route MIDI through an audio interface or HUB into Regen. See an example iPad set-up below:

into USB A "HUB"

Headphones

ARTURIA

Audio Interface

5-pin MIDI

TS/XLR

USB

into USB B "Host"

Camera Kit adapter

KEYLAB 49   ARTURIA

Synclavier Regen in DAW mode   iPad with DAW / Host app   Keyboard / Controller

1.4. Panels Summary

This summary section is worth reading before diving into the other sections of the manual that cover the functions in depth, so you, at minimum, have a broad understanding of all functions before zooming-in on one. Rarely does a function behave in isolation. For example, the FM modulation is controlled by parameters on the OSCILLATOR and ENVELOPE panels.

NAVIGATION

The NAVIGATION panel contains the main display. This display will present one of many "pages" containing useful information and feedback for the current function that the user is concerned with. In the top left corner of the main display, you'll find the track and partial blobs, cyan and magenta respectively; if present, they show what track and partial are currently selected.

There are 3 **Option** buttons below the display, numbered, for the purposes of this manual, left to right 1.,2.,3. If these are selectable, they will be dimly lit and a legend will appear on the display immediately above the button: legends for actions are left-aligned to more closely align to the physical button below; legends for parameters are center-aligned with the parameter control graphic above. These **Option** buttons typically portray different parameters of the function the user had selected — for instance, if you tap **Vibrato**, you will be

presented with **Wave**, **Rate**, and **Depth** — 3 vibrato parameters that can be adjusted by selecting the corresponding selector button and using the **Swiper** to change the value. **Option** buttons can also be used to access sub-pages.

Below the **Option 2** button is the **Enter**/**Back** button. When red, this button's function is Enter — used, for instance, to select a Timbre to load when browsing the library. When blue, this button's function is Back — used to cancel an operation or traverse up from a sub-page to a page.

The Left and Right arrows either side of the Enter/Back button will be dimly lit if selectable. They typically navigate to other available pages — for instance, if you go right from the main **Vibrato** page, you get access to **Invert**, **Quantize**, and **Bias** parameters. In some cases **Arrow Left** and **Right** are used to navigate within a page, such as in the case of changing the brightness on the **Settings** page.

> The philosophy of the **Option** buttons is that their legends depict what the button does, not what the state is. For example, when changing the sort method of timbres in a Library, Option 1's legend is "**Change Sort**" but the current state — alphabetical, by tag — is portrayed on the main display, discerned by the formatting of the timbres.

VK DISPLAY

The VK DISPLAY is so called, because it mimics the display from the Synclavier II VPK edition (velocity pressure keyboard). As per its namesake, it's the area that displays the most pertinent information: for instance, when you load a timbre, its name will appear on the VK display; if you copy or paste a partial, the VK will provide confirmation messages; when you change a parameter value using the **Swiper**, a real readout of the value is given in the VK in a zoomed view; and so on.

SWIPER

The **Swiper** is a touch sensitive strip with 7 bits accuracy (resolution 128 positions). To use it, lightly swipe your finger or thumb up and down the strip.

It was placed on the left-hand-side with a 1.3" sweep of the bottom left corner of the enclosure, to facilitate easy swiping with a left thumb, but you can of course use whichever digit is most convenient.

The **Swiper** is used to scroll lists or modify parameter values. For most parameters, 2 full swipes is enough for full scale change — for example, the **Vibrato Depth** will change from 0 to 25 semitones with two full upward swipes.

The **Swiper** is intelligent and changes its resolution depending on the parameter being changed.

Nudge Up and **Nudge Down** buttons are provided — tapping those will always increment or decrement the parameter value by the smallest unit possible.

NB: If, during a swipe action, your finger covers the **Nudge Up** or **Nudge Down** areas, it will not be recognized as a tap, but the swipe action will continue.

The **Swiper** supports Fine mode: hold one finger on **Nudge Down** and then use another finger to swipe, and the rate of change will be lower than normal.

OVERALL

Buttons not surrounded by a panel border contain controls and parameters that pertain to system-level settings.

 The **Home** button, no matter what state the system is in, will return you to the home page. From here, you can view or change settings, or save state and put the unit to sleep. Press **Home** twice and any playing sounds will be muted (sometimes called a MIDI panic feature).

 The **Help** button, when dimly lit, can be activated to dispense help about the current page in the main display.

 Load Sound Up and **Down** buttons will always switch to the current Library page and call up the previous or next timbre or sample to the selected track. Very useful when first exploring the sounds, and also to navigate around your own timbre libraries, as you create them. The exception to the rule is

when you're designing a patchlist — then these buttons navigate the patchlist instead of the timbre/sample lists.

[BUSY] is the only indicator not associated with a button. It will light when a new sound is called up to a track, indicating that MIDI notes sent to that track will be ignored until the sound has fully loaded. In the vast majority of cases, you'll barely notice the indicator flash, but timbres containing many long samples may take a moment to load. When a new timbre is called up to a track, playing audio on that track is gracefully muted and audio from the other tracks is unaffected.

You'll also see the **BUSY** light on during system start-up for around 22 seconds, indicating that the system is not ready for use. Also, during a firmware update or reconfiguration, the **BUSY** indicator may light or flash.

MASTER Panel

The MASTER **Volume** affects everything — all tracks — for balanced, unbalanced, and headphone outputs. Press this button and then use the **Swiper** to adjust the level. You will see VU style meters for: Master Volume setting, live master out levels (left and right channels), and the third set of VUs show current CPU and memory usage levels.

There's only one parameter — MASTER **Volume** — but this is a very useful page. If you're playing many tracks at once and suspect that you may have some clipping distortion, you can check the VUs and if you're in the red, dial back the **Volume** a notch or two. Similarly, if you think you're overdriving a pre-amp with the Regen output, this is where you would dial the volume back (although the VU meter is of less use in this scenario). There's also a live time-domain display, sometimes referred to as an oscilloscope, in the top third..

Also, if you're playing a lot of tracks at once and there are a lot of notes playing on each track, it's possible you may overload the CPU and some of those voices would be dropped. This would be the place to check the CPU meter.

If you **Arrow Right**, you'll get to the multiscope page, divided into thirds:
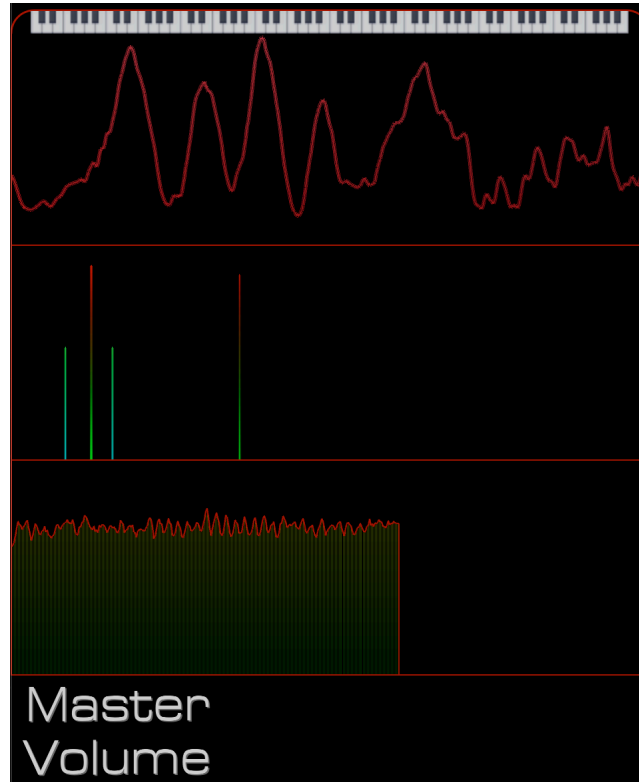
Master
Volume

- The top third has a live time-domain display of the master out, with normalised amplitude. It also contains a graphical representation of a keyboard; the corresponding keys light up when Regen receives MIDI note information. NB: On this page, the VK Display will show the MIDI note on information as it comes in.

- The middle third contains a frequency-domain display which shows the current voices being produced by Regen, and where they lie on the frequency spectrum; the height of the bar providing an indication of intensity.

- The final third is a plot of the master out over time. The amplitude is not normalised.

The one parameter on this page is again MASTER **Volume**.

Synclavier Regen contains a 4-core CPU and all loads are summed and presented on the CPU level, accessible from the MASTER **Volume** page. This is a useful page to check if certain combinations of timbres and TIMBRE **Effects** (reverbs and filters) will cause a high CPU usage: the meter will roam into the yellow region.

In extreme load scenarios, the meter will *redline*, voices may be dropped and you'll be shown an audio overrun warning, but this level of CPU usage is not common under normal load conditions.

The Synclavier Regen DSP Engine is highly-optimized to use only as many CPU cycles as necessary to process audio. Unlike some fixed cycle systems, where a hard limit of voices can be determined, this architecture means the total available voices actually depends on the burden of system configuration at the time.

We have conservatively stated that we can support 98 simultaneous voices, based on our experimentation, but you may achieve many more voices (and potentially fewer) before your CPU meter *redlines*.

Features that more substantially affect CPU usage are: number of operational filters, number of track reverbs, and partial frame count.

Phones provides volume adjustment of the headphones connection. The output amplitude will be this parameter compounded with the MASTER **Volume** parameter. The live output VU is repeated here, but note that adjustment of **Phones** volume is not going to affect it. The CPU level is also repeated on this page for convenience.

On **Volume** and **Phones** pages, the top third of the main display houses a live oscilloscope-style graphic of time vs amplitude. This is provided because it looks cool.

Reverb in the MASTER panel concerns the reverb algorithm applied to the master signal chain (the sum of all tracks). Pressing this button will show the set

of pages with all **Reverb** parameters and associated MASTER effects parameters (namely, **Noise Floor**).

Reverb is new to Synclavier in this product. Our carefully tailored Schroeder reverb algorithm is not only highly efficient, but compliments many of our timbres sonically. Like many things in life, a little goes a long way. Moderate changes of reverb parameters dramatically change the overall sound.

We should mention here, that rather than tweak each parameter separately, you can call up complete **Reverb** settings to the MASTER reverb by selecting the **Reverb** button in the library section and browsing the preset reverbs. See manual section on Library.

The MASTER values — volume, phones level, and reverb settings — are stored when you tap **Save State** (accessible from the **Home** page) and remembered the next time the system is used. In the case of MASTER **Reverb** settings, they are also saved as part of a session (covered in the LIBRARY and SESSION chapters of this manual).

SELECTOR Panel

The twelve **Selector** buttons and co-located buttons make up the SELECTOR panel. The 12 **Selector** buttons contain touch sensitive pads and are epoxy-coated. This makes it easy to swipe a selection of more than one at once.

The **Switcher** button above the selector buttons indicates the view. Blue is for TRACKS and red is for PARTIALS. When blue, the 12 selector buttons represent the 12 tracks of Synclavier Regen. You can assign MIDI channels to the tracks using the **Track Crossfader** page (covered in the SESSION chapter). When red, the selector buttons represent the 12 partials of the last-selected track. Partials are "part" of a timbre.

To the left of the **Switcher** button is the **Solo** button. This also shows state. If it's lit, it means the system is in SOLO mode. Using solo mode, you can isolate the selected tracks to hear their audio — the other tracks will be muted.

Or, if you're working in PARTIAL view, you can select **Solo** to hear only the selected partials in isolation. When soloing partials, you're also soloing the track of the timbre the partials are contained in, you do not need to also solo the track.

- When in partials view, if tracks are in solo mode, the solo button will blink.

- When switching tracks, so you don't get tied into knots, Regen will cancel any solo'd partials on the track you've switched from.

The **Selector** buttons are back-lit with RGB LEDs. The increased palette of colours available afforded Regen designers the capability of representing different states to the user per partial or per track. These modes are helpful to the user, so we've conveyed this information in other methods additional to distinguishing colours, such as legends on the display, as well as the **Solo** and **Switcher** button states (as stated above).

There's a full table of colours in the introduction section of the manual, but the main thing to be aware of when first using Regen, is that red, magenta, and green pertain to partials — active, selected, and solo'd respectively. And blue, cyan, and chartreuse pertain to tracks — active, selected, and solo'd respectively.

The clipboard buttons work in the customary way to **Cut** or **Copy** tracks or partials to a clipboard and then **Paste** them back. You can copy more than one contiguous partial by swiping several and tapping **Copy**. You can then paste the partials (contiguous, in the same order) by swiping a selection of the same quantity and tapping **Paste**. Use the same procedure to copy multiple tracks at once.

The **MIDI-link** is used to reveal MIDI configuration pages. If you have a parameter selected from the TIMBRE panel, pressing the **MIDI-link** button will bring-up a MIDI Controller page. However, if you have a NOTE FX or partial parameter selected, the **MIDI-link** button will bring up the Expression Modulators page. More on that distinction later in the manual.

For the odd parameter that cannot map to a MIDI input, the button will be unlit, showing it can't be selected. Otherwise it will be dimly lit. While you're on a MIDI configuration page, it will be brightly lit. You can press **MIDI-link** again to return to the parameter page you were on, or press another button to leave the MIDI configuration page.

LIBRARY Panel

The LIBRARY panel is where you load and save presets for **Timbres**, **Samples**, **Reverbs**, and **Sessions**, using the corresponding buttons. **Preset**, **User** and **Save** are modal buttons that show and change the mode of the LIBRARY.

> Library was the original term Synclavier used for a disk of timbres. The old 5¼ floppy disks would hold 64 user timbres that were organized in 8 banks of 8. This was referred to as a library. We still use this term and capacity to denote a library. In regard to the panel title being singular, all panels are singular as a convention, but they provide access to many libraries, envelopes, LFOs etc.

OSCILLATOR Panel

All red buttons pertain to partials, which means that if you change the value of a red parameter, you will be changing it for the selected partial/s.

The OSCILLATOR panel contains parameters for the development of waveforms for a partial. Typically, you would start sound design of a new timbre here.

Each partial can have a carrier wave and modulator wave. The **Carrier** and **Mod** buttons are modal, so you work on one at a time. The carrier oscillator can be a sample (or patchlist of samples), a subtractive waveform, or an additive generator. You use the **Wave** parameter to select, and the other buttons to further refine the oscillator. The modulator wave is frequency-modulated (FM) with the carrier wave by a selected amount. The modulator oscillator can only be an additive generator.

ENVELOPE Panel

Synclavier Regen contains sophisticated envelope generators for use in your sound design. There's a volume envelope accessed from the **Carrier** button and an FM envelope accessed from the **Mod** button (sometimes referred to as a harmonic envelope). From this panel, you can also access the **Partial Crossfader** function, which is invaluable in creating complex, responsive sounds.

LFO Panel

Vibrato and **Tremolo** are oft-used LFO functions, so they have their own buttons. Many of Synclavier's signature sounds use one, or other, or both, extensively. **Vibrato** and **Tremolo** are mapped to pitch and volume/pan respectively, but can also be exported as LFOs to other parameters by use of the Expression Modulator pages accessed by pressing the **MIDI-link** button, covered in-depth later in this manual. Most LFOs act on a partial level, with a few exceptions.

TIMBRE Panel

The blue buttons in the TIMBRE panel are timbre controls, affecting the character of a timbre's sound. You could consider them as acting on all partials, and in some cases that is an accurate way to think about it. When you save a timbre, you're saving a timbre's set of partials plus the timbre parameters.

The white button on this panel is for **Track Levels**. A timbre cannot be heard without placing it on a track, and from this button you can modify the track's **Volume** and **Pan** settings, "placing" it in the mix. If you swap out the current timbre for another, the track settings won't change. Likewise, if you save a timbre, the track settings won't be saved with it.

Track settings are saved with a session. So, when you save a session, you save all active tracks, their timbres (containing timbre settings and partials) and session settings, such as MASTER **Reverb**.

NOTE FX Panel

The blue parameters on this panel are also saved in a timbre, but unlike the TIMBRE Panel, they're more concerned with the performance of a track than the character of the timbre. Here you switch on **Arpeggiate** and **Repeat** functions, for instance. Synclavier considers this an element of sound design and so this is part of a timbre. Likewise, the **Mode** of a timbre, can be polyphonic or monophonic and this behaviour decides when notes sound, glide, re-trigger, or mute on a given track. These parameters are saved within a timbre.

The white **Session** button on this panel brings up the current sessions's track list showing all the active tracks in the current session. The assigned MIDI channel for each track is shown on a green blob to the left of this list.

1.5. Cheat Sheet

Here are some useful shortcuts. They can all the be found elsewhere in the manual, but this section is a good quick reference / reminder.

To get a summary of a page and its parameters/functions
Tap the **Help** [?] button and a short help page appears in the main screen detailing the page you were perusing. Tap **Help** [?] again to dismiss this.
To quickly listen to different presets
If some tracks already contain timbres, reset the state of the machine (see below) then use the **Load Sound Up** and **Dn** buttons to call up different timbres. The **Load Sound** buttons will keep giving you new sounds from wherever you are in the library. When all sounds are exhausted, it will go around again. Sounds are generally arranged in banks of 8 that have something in common, such as all #percussion sounds, or all #keys sounds. If you prefer to go straight to a certain category: from the LIBRARY **Timbre** page, tap the **Change Sort** option twice to browse the list of tags then **Arrow Right** when you find a category you like.
To select multiple partials or tracks
Swipe several **Selector** buttons at once to select. If the desired buttons are not contiguous, tap and hold one button while tapping the other buttons you desire. The **Switcher** button atop the 12 selector buttons will be blue when the selector buttons represent tracks, red for partials. Tap to switch.
To view a summary of the tracks
Tap **Session** to get a track listing — each active track is shown with the name of the timbre it contains. You can **Arrow Right** from here to also see each track's crossfader settings (MIDI channel and keyboard range). If there is also midi sequence information in the session, you can **Arrow Left** to view the track listing with midi track names *as well as* the timbre names.

To mix track levels and panning
Tap the **Levels** button in the TRACKS panel and then navigate left with the **Left Arrow** (below the display) to reach the Tracks Crossfader pages. The x legends are reminders of where the Crossfaders live.
To engage fine mode on the **Swiper**
Tap and hold the bottom nudge button, then swipe as usual and the selected parameter values will increment in tiny intervals.
To engage scrub mode on the **Swiper**
For sample **Start Time**, **End Time**, **Loop Start** and **Loop End** parameters, you can activate scrub mode to coarsely change the parameter value — hold down the chosen parameter option button while using the **Swiper**.
To call up the Modulators list quickly
Rather than select a parameter that can be modulated and then tap the MIDI-link button, you can just tap the **Vibrato LFO A** and **Tremolo LFO B** buttons simultaneously and you'll be presented with the Modulators list.
To modify more than one parameter at once
If, on a page, you tap multiple option buttons at once, for example, **Vib/A Rate** and **Vib Depth**, both of those values can be changed in unison by the **Swiper** and **Nudge** buttons.
To quickly get to the volume envelope **Delay** and **Decay** parameters
Rather than arrowing right from ENVELOPE **Levels** or arrowing left from **ADSR 1**, you can press the **Levels** and **ADSR 1** simultaneously.
To quickly get to the layer parameters when your partial is in Samples mode
Tap Wave and Edit simultaneously
To reset the state of the machine

Selecting all twelve tracks and tapping cut will not only erase any timbre on the tracks, as well as MIDI assignment and track volume and panning settings, but also reset the MASTER Reverb to default. You may also have studio settings, to reset these, go to **Home** -> **Setup** -> **Studio** and tap the **Clear Studio** option.

Patching samples to keyboard keys or key ranges

When you tap the **Choose File** option under OSCILLATOR **Edit** you're presented with the Samples browser. When you find a sample you want, you can strike a key on your MIDI keyboard and that key will be assigned for the next sample you load. If you strike two keyboard keys a range will be assigned. Tap **Enter** to load the sound and your mapping will be set when the sample is loaded. This can be altered after the fact from the **Define Range** page.

Similarly, you can toggle the **Auto Tune** option from the Choose File page. If lit, when you tap Enter to load a sample, Regen will set the **Base Key** and **Tuning Offset**. This can be altered after the fact from the **Define Pitch** page.

To solo a Frame

When you are editing the Harmonics of a partial, you are solo'ing the currently selected frame.

To return to where you were working

Regen remembers which page you are on when switching back and forth between panel buttons. Additionally, when viewing the MASTER panel, the **Back** button will take you back to the previously viewed parameter page, so you can jump back and forth between setting MASTER controls and other levels or parameters.

1.6. System Schematic

REGEN

Synclavier Regen Signal Flow

Pitch Inputs

Keyboard Pitch Track
Track Transpose
Partial Transpose
Custom MTS Tuning
Pitch Class Tuning
Overall Octave Ratio
Timbre Octave Ratio
Partial Tuning
Timbre Tuning
Timbre Detune
Overall Tuning
Pitch Modulator(s)
Vibrato Generator
Frame Tuning
Pitch Quantizer

Partial (12)

Modulator Oscillator

FM Ratio

Pitch Computer

Carrier Oscillator

FM ADSR Generator
Partial FM %
Timbre FM %
Timbre FM +/-
Frame FM %
FM Modulator(s)

Chorus Generator

Time Ditherer

9

9

Chorus Summer

Envelope ADSR Generator

Partial Volume & Pan

Tremolo Generator

Stereo Partial Output

Bit Depth
Grunge Setting
Alias Filter

Wave Table Lookup
• 24 Harmonics
• Sample
• Synth Wave

9

Synclavier

12 Partials

12 Partial Summer

Timbre Volume, Leveling & Spread

Timbre (12)

ADSR Note Filter

Effects EQ Filter

Timbre Reverb

Track Volume & Pan

Stereo Track Output

12 Tracks

12 Track Summer

Master Controls

Master Reverb

Noise Floor

Master Volume

Headphone Volume

Headphone Output

Unbalanced ¼" Outputs

Balanced XLR Outputs

USER SETTINGS

2.1. Home & Settings

Home, Sleep, and Power Off

The ⌂ button will always take you to the home page. From here you can access user settings via the **Setup** option.

If the system has changed state from its currently saved state — e.g., you have loaded a new timbre into a track slot — you will have a Save State soft button on the home screen; if not, you will have the **Sleep** function. If you're not going to use Regen for some time, it's advised to put it in sleep mode. This will save power and reduce wear levels of many internal components. If you tap **Save State**, all user settings and the current system state are preserved, ready for when the system next wakes. NB: You do not need to save the state for the system to remember what USB mode it is in (Controller or DAW) or what audio mode it is in (Composite or Multi-Channel).

If the system is not going to be in use for more than a few hours, you should power-down the system. The best way to power-down your unit is via the ON/OFF button at the rear, ideally after you have tapped **Save State** then **Sleep**, so all settings are remembered.

Brightness

The main page of **Setup** contains the brightness setting. The default brightness level is 2. In certain scenarios, it may be difficult to see the contents of the screen or discern the difference between annunciator brightness modes. Turning this brightness level up increases the brightness of the LED backlighting on the buttons for all states and increases the screen brightness. Your brightness preference will be saved when you save state.

The main subpages of user settings are MIDI and Audio denoted by customary icons.

Audio

Top of this **Audio** subpage will be an entry for the SD101 Synclavier DAC. This is the inbuilt DAC (digital to audio conversion unit). This DAC feeds balanced and unbalanced (often referred to as single-ended) outputs, as well as headphones.

The inbuilt DAC circuitry has been designed for studio use. It's been locked to 50kHz to take full advantage of the Synclavier II DSP (Digital Signal Processing) synth engine, which was optimized around the 50kHz sampling rate of the original "poly-sampling" system and to give accurate reproduction of many of the included NED samples that were recorded at 50kHz.

The SD101 is a high-resolution 24-bit stereo DAC circuit with extremely low noise and negligible jitter designed for studio use. Research into human hearing shows humans with the best hearing can discern a maximum of 21 bit of dynamic range (2097152 levels of amplitude). A CD is mastered for 16-bit digital, so why go to 24-bit?

This level of redundancy is good. Regen can be used in different settings with different gain-staging and can play many voices (up to 98) simultaneously. With this level of performance built-in, Regen is never going to be the weakest link in your audio chain.

Having lots of digital levels (bits) is worthless if that resolution is lost later in the analogue circuitry. The balanced outputs have been measured to perform at

130db above the noise floor, exceeding studio requirements for useable dynamic range. The unbalanced outputs were measured at 110dB.

As good as the built-in DAC is, there may be times when you want to use your own converters and bypass it. Perhaps you want to plug into a studio's converters, or "house" sound at a venue, or use your own audio interface simply to try a different sample rate (see Pro-Tip below). As a bonus feature, Regen supports class-compliant USB Audio interfaces that are not composite devices. If you plug in a compatible USB Audio interface or USB DAC, Regen will recognize it and make it available on this screen. Only one audio interface can be used at one time, so when a selection is made, if another device is in use, it will be dropped (including the built-in DAC). Some DACs and interfaces support multiple channels, not just stereo. If you select the Multi mode, Regen will allow you to route different tracks to different channels. This is very useful if you want to feed Regen's tracks independently into a mixing desk, and could be another good reason to choose a USB Audio output over the built-in DAC. In this mode, the track volume and pan controls are not honoured and instead you would set those on your mixing desk outside of Regen.

Where supported by the interface, choices of sample rate and numbered output pairs will be selectable on the device. Navigate this menu with the **Nudge Up** and **Down** buttons and the **Enter** button. Use **Left** and **Right Arrow** buttons to switch between sample rate output choices.

Some USB audio interfaces are "bus" powered, meaning they draw power from the USB cable. Synclavier Regen supports this and can feed up to 1AMP of power shared across all USB peripherals. If this is exceeded, all ports will shutdown.

Regen's bus power has some level of power filtering and should be relatively clean, but since the USB Peripherals section of Regen is a USB Hub noise from other attached USB peripherals could bleed into a USB audio interface, so this is something to be aware of. Computer frequency noise is especially bad for audio, so if audio quality is paramount, stick to the onboard DAC, which has several layers of power conditioning, or use separate power supplies for USB audio interfaces where applicable (check the supplier's manual).

A **Test** function is available from the **Audio** settings menu. The option button will toggle this function on or off. When on, a test tone is output at the system volume level; 440Hz on the left and 220Hz on the right. In the case of the built-in SD101 DAC, the test tone is output on the balanced, unbalanced, and headphone outputs. This is a simple test, but it proves invaluable to track down audio problems if they do occur in your signal chain.

Humans can only hear up to 20KHz sounds in the prime of life and it deteriorates as we age. Nyquist theory says we only need a sample rate double 20KHz to reproduce the spectrum of human hearing, so CD's 44.1KHz or Regen's 50KHz is more than adequate. However, there are notable reasons why we may want to use a higher than Nyquist sample rate. Nyquist's mathematical theorem assumes a perfect reconstruction filter and there is no such thing as a perfect filter. The higher the sample rate, the more the specification of the reconstruction filter can be relaxed. This is especially true when it comes to sigma-delta DACs found in most modern audio convertors. Most high-end DAC architectures are aware of this, and select a combination of digital and analog filters to suit the incoming sample rate. Also, many DAC architectures up-sample the incoming audio, and if that's going to happen anyway, if Regen supplies those additional samples they will be truer to the sound intended than if the DAC interpolates them.

Another reason you may choose to use a higher sample rate is if you're extensively using FM on your sounds. The FM algorithm produces artifacts in higher frequency registers than an unmodulated wave. Normally, most are under the Nyquist frequency, but as you increase the amount of FM applied, you can push beyond it, and those artifacts will "fold over" into the audible frequency range, and are often undesirable. A higher sample rate will potentially avoid this effect. NB: You could also adjust this by mapping Keyboard to **FM Amount** such that higher keys will reduce the **FM Amount** (mapping sources to destinations are covered later in the manual in the Modulators section) or you could activate a TIMBRE **Effects** filter set to low pass with a cut-off just below the Nyquist frequency to eliminate those high frequency FM artifacts before they get to the DAC and are "folder over." .

So why didn't we use a higher sample rate on Regen? There's a trade off. The more audio samples Regen has to create per millisecond, the more it crunches the CPU. So the higher the sample rate, the fewer voices its cable of producing. We settled on 50Khz as a good sweet spot where you get lots of voices and great sound quality.

MIDI

This ⌨**MIDI** settings subpage is similar to the **Audio** subpage, but instead lists all available "control surface" interfaces. Regen supports class-compliant USB MIDI Interfaces, such as MIDI keyboards, sequencers, pedals, studio controllers, and the Synclavier knob. Anything plugged into a USB Peripheral port should show up on this page. From here you can elect to *listen* to or *mute* MIDI devices by toggling their "on" status. Also, if you have an MPE-compatible device, you can tell Regen to accept MPE messages for that unit by checking the "MPE" blob and electing which MIDI channel the MPE data for that unit will be assigned to.

NB: For info on how MPE information is interpreted by Regen, see the MPE Controllers and Keyboards subsection at the end of the Modulators section of the PARTIALS & SOUND DESIGN chapter.

As a convenience, the system will remember previously *listened* to MIDI devices and automatically *unmute* them.

Bluetooth

A ⬡ **Bluetooth** subpage is selectable from this menu. But be warned, bluetooth is not officially supported. We have deliberately left out bluetooth compatibility from our product specifications and any product literature. Bluetooth, at best, is a complex, temperamental protocol, and during our testing we found only a handful of bluetooth dongles to work reliably. The team debated whether to put this feature in the system or not, and, since we'd spent a lot of time on writing the controller software, the jury came down on the side of including it as a non-supported feature. We don't guarantee that it will work, and we unfortunately won't be able to answer any technical questions on it.

That said, if you have a USB dongle, you might get it to work for you. And at the time of writing this manual, new bluetooth dongles purporting ultra-low latency especially for musicians appear to be entering the market.

This feature will only work in Controller mode (not DAW mode). The basic procedure is to plug in your dongle and turn the feature on from the **Bluetooth**

subpage under **MIDI Settings**. If your dongle is supported, it will be recognized and start scanning for nearby bluetooth devices that support the MIDI-over-bluetooth protocol and list them (it shouldn't pick up your smart lights and washing machine). Pairing and passwords aren't supported, but this shouldn't be necessary for MIDI keyboards. You can select any devices that are found to create a connection. Once connected, the device should show up on the usual MIDI **Settings page** ready for *unmuting*.

2.2. Help

The **Help** button, when dimly lit, can be activated to dispense information about the current page in the main display. Depending on your user preference, you will be able to read the help text in the main display or hear the text spoken. The advantage of the latter method being that you can still see the page the help is referring to as you absorb the info.

This manual, of course, expands greatly on the help pages, whose primary role is to act as a reminder when a user has taken a break from Regen.

Tutorial videos, each focusing on a different part of Regen, and are available on our youtube channel: https://www.youtube.com/@Synclavier

LIBRARY

3.1. Preset Timbres

The LIBRARY is naturally the first panel of which to gain familiarity. The LIBRARY panel is where you load and save presets for **Timbres**, **Samples**, **Reverbs**, and **Sessions**, using the corresponding buttons.

Preset, **User** and **Save** are modal buttons that show the mode of the LIBRARY. I.e., one will be lit to show what mode the LIBRARY is in (NB: with **Save** lit, **User** is also lit). Each mode will remember the last library it was perusing. To change a library, for example, for **Preset**, you would tap **Preset** when **Preset** is already lit, which would be a double tap if the library was in another mode. Then you can scroll all the factory libraries: **Preset** pertains to the inbuilt factory presets, which cannot be changed and are always available. The currently selected library is shown with a highlighted background. When you have scrolled your chosen library, you would typically tap **Timbre** to view the timbres therein, or **Samples** to view the samples therein.

User pertains to user presets from the currently inserted SD card. When **User** is lit, the LIBRARY will show you user presets, assuming you have a correctly formatted SD card inserted. Again, if you wish to change the **User** library you're perusing, you would tap **User** a second time and scroll the list.

User and **Preset** cannot be lit at the same time. The third mode is Save. If you tap **Save**, **Save** and **User** will light, and, assuming you have an SD card ready, you can save your own **Timbre**, **Reverb,** or **Session**. If you have created multiple libraries on your SD Card, tapping **Save** a second time will get you to the library list to select a library. After selecting a library, you would tap **Timbre** to save the currently selected timbre.

Timbres

On some synths the word for timbre is preset, but Regen has factory presets for timbres, sessions, and reverbs. The word timbre has always been the term used on Synclavier machines.

We invite you to explore the Regen Warehouse library first, since this has a wide range of presets designed specifically to bring out the best of Synclavier Regen. In addition to this, Synclavier Digital partnered with accomplished sound designers Anthony Marinelli, Paul Schilling, Red Sky Lullaby, INHALT, Leith Fleming-Smith, Jamie Lidell, Gary Chang, and Drew A Neumann to bring world-class preset timbres and samples to your fingertips. As well as having a unique flavour, each artist's Library contains many readily useable sounds.

When you've gone through all the included factory content, you can find yet more after-market content on Regen Downloads web page – https://www.synclavier.com/regen-downloads

---

All Synclavier systems have been designed for left-side and right-side brain activities. What does this mean? Take the activity of auditioning preset timbres, as discussed below, for example. You can scroll through lists of timbres, organized canonically, alphabetically, or by curated tags, and select ones to audition — very left-side brain. Or you can use the **Load Sound Up/Dn** buttons, and play and listen to the timbres that are presented to you — very right-side brain. This ethos permeates every facet of Regen.

When first browsing timbres in a library new to you, it's advised to use the most detailed view of the library (**Arrow** rightmost), so you can read all the text:

there's frequently useful information in there, such as a note that the Mod Wheel opens the filter or Aftertouch adds vibrato; without this information, the full potential of the timbre being auditioned could easily be overlooked.

Loading Timbres

When you load a timbre you are replacing the timbre on that track, but not affecting any other track than that selected — or the last one selected in the case of multiple selections.

To load a new timbre (sound preset), typically, you will be in TRACKS view (press **Switcher** to switch from PARTIALS view, if this is the case). Tap a track button to select it. Now you can either press **Load Sound Up/Dn** button to pick the previous or next timbre in the current Library, or you can browse for a specific timbre by use of the LIBRARY.

Pressing **Preset** takes you to the factory presets, which are arranged in libraries. See image from main display below:

The list of Libraries can be scrolled with the **Swiper** or **Nudge** buttons and the highlighted one is automatically selected for you. There are different levels of zoom. The one shown above is from the *cover view* that shows 3 Libraries at a time. If you want more detail, press the **Right Arrow** and you'll be presented with the *inside cover view*. This only shows one Library at a time, but presents a

full description, and can still be scrolled. Once you're familiar with the preset libraries, you may prefer the *list view*, which lists 7 libraries on the screen at a time — **Arrow Left** to get to this view.

Libraries may have sub-folders, they will be denoted by a ellipsis at the end of the name. If you tap **Enter** while highlighting one of these libraries, it will *unfold*, revealing the sub-folders underneath, each having an ellipsis at the start of the name. Tapping **Enter** again will *fold* the library back.

Now, if you tap **Timbre**, you'll be presented with a list of timbres in the selected library. This also has 3 levels of zoom. The highlighted line isn't the current one (as in the case of the library views) but the candidate. If you tap **Enter**, that highlighted timbre will be loaded to the selected track. If the current timbre on the selected track is shown on the list it will be indicated with a light magenta blob.

Libraries are organized in 8 banks of 8, just like the original Synclavier II, and to the left of the entries you'll see the code. For example, 2-4 refers to bank 2 and entry 4. Dividing a 64 timbre library into 8 banks provides an opportunity for arranging your libraries in useful ways, for example, grouping instrument types together, or different versions of the same timbre could be in a bank, but with different expression mappings, or some kind of version control. Different sound designers will employ different policies (or none at all). The Regen Warehouse timbres are roughly grouped by instrument type.

Once you become familiar with the presets, you'll no-doubt have your favourite go-to sounds. Another way to call up a timbre is to use the **Speed Dial** feature. From the timbre list, press this **Option** button and the available **Selector** buttons pulse. Now enter the two digit code to call up that bank and entry to the currently selected track. Easy peasy. And as a further convenience, you can send MIDI Program Change messages to Synclavier Regen. A Program Change message will cause Regen to load the respective timbre in the currently active library to the selected track, irrespective of the channel number of the message.

The **Choose Partials** function enables you to mix and max. Each timbre is made up of 1-12 partials, and if you select the **Choose Partials** button, you can select which individual partials are loaded to the current timbre. Active partials in slots that are not selected will not be overwritten. If you're in the most zoomed-in view of the timbre list, you'll see a row of partials on the highlighted timbre at the bottom of the main display. Active partials in the candidate timbre will be highlighted with a green background. Now, if you select **Choose Partials**, you'll see that all partials available for recall will be pulsing magenta on the selector buttons. During the off-cycle of this pulse, you'll notice that currently active partials (that would be overwritten) stay dimly lit. To complete the procedure, select the partials you wish to recall and tap **Enter**. There is much more to say on partials in later sections of this manual.

When browsing timbres, there's an option to **Change Sort**. This button cycles between sort modes of canonical, alphabetical, and by tag. NB: In alphabetical or tags mode, all the preset libraries are collated and presented. In canonical mode, only timbres from selected highlighted library are presented.

In alphabetical sort, timbres are grouped according to the first letter of their title, with the special case * representing non-letter characters. The **Swiper** and **Nudge** buttons will scroll down this list from *-A-Z, also showing the first timbre (alphabetically) in the group. You can tap **Enter** to select the timbre highlighted, but more often, you'll tap **Select** and then you can scroll within that group of timbres to find the one you want and then tap **Enter** to load it.

You can **Arrow Right** to see more information on the timbres within the highlighted category, and also have access to the **Choose Partials** function.

NB: No concession is made for timbres that start with "The" — they're lumped in with "T."

Sorting by tag operates in a similar way. All timbres are "tagged" with meaningful keywords (covered further in the following section). In by tag sort mode, you're presented with an alphabetical list of tags and can scroll through them. If you **Arrow Right**, you'll be able to browse the timbres containing those

tags (from all libraries). You can further refine your search by adding another tag: if you tap **+Tag** you can then scroll within the first tag — for instance, you may want to peruse all #bass timbres that are also #vintage. If you **Arrow Right** with those tags highlighted you'll find that Regen offers Mild FM [D 2-1] and Oscar Bass [RW 5-4] as options for you.

3.2. Tags

Category Tags

As mentioned previously, every factory preset timbre is "tagged." Each timbre has at least one, and typically only one, category tag.

Category tags are somewhat synonymous with instrument type. The full list is:

#bass
#pad
#strings
#drone
#choral
#vox

#lead
#keys
#chime
#organ
#wind
#brass
#orchestral

#texture
#beatloop

#percussion
#sfx

#template

There's a few timbres that contain two category tags, for instance, Tongue Drum [RW 2-1] is tagged a #chime and also a #percussion, but most timbres have only one category tag.

Not everyone is going to agree with the categorization, but we've tried to make the groupings useable, so no one group has too few or too many, timbres: #vox and #choral are quite similar, but we had enough timbres of each to make the distinction; #chimes, in some synths, might be split into #vibes and #chimes, but we didn't feel we had enough timbres to warrant that; and so on.

In lists, category tags appear in green text, ordered as above, except special category tags, which appear yellow.

Special Category Tags

#template is a special category tag — its timbres will be an example of a technique or a starting point for sound design, and these are mainly found in the Synclavier Regen Tutorial library.

#textures and #beatloops are special categories of timbres worth describing. #textures are atmospheric and often ambient soundscapes, typically comprising many sounds working in harmony. Useful as a foundation of a music track or to use in film/tv soundtracks, each #texture evokes a different emotion.

Most textures change substantially with different mod wheel settings, crossfading between different sounds or scenes. This can, of course, be automated by an external sequencer, and the mapping could be changed from Mod Wheel to another CC controller easily enough.

#beatloops are predefined chunks of audio with a specified BPM and bar count. Unlike #textures, #beatloops are one-shot. Whereas #textures need to have a key held for the soundscape to build (unless using a sustain pedal), with a #beatloop, you would press a key and hear the full duration of the sound. Since they are locked to a BPM tempo, they will not change pitch depending on which key you press (like most other categories). In fact, with most #beatloops, different keys are mapped to trigger different versions of the same "beat" and the timbre description will have to be consulted to find out which keyboard keys do what.

Example sessions are provided that use the #beatloop timbres and they're a great place to start experimenting with #beatoops. See the Sessions section of the LIBRARY chapter for more info.

NB: #beatloops should not be confused with #loops. #loops, rather than chunks of sound, are simply samples that contain looping information, such as a flute that can be sustained.

Properties Tags

Unlike category tags, which are nouns, properties tags are adjectives that describe an aspect of the sound. A sound could be described as #bold or #vintage or #cinematic. Again, careful judgement has been exercised to make sure the list of properties isn't onerously large or unusably small.

In lists, properties tags appear in orange text, ordered alphabetically.

Special Properties Tags

#split means that the timbre is split into different sounds across the keyboard. This would've been done with the **Partial Crossfader** or sample patchlist, both of which are covered later, which can set a range of keyboard keys for a partial or sample. Traditional (Split) [RW 4-4] is a good example of this that has a harpsichord on bass duty with a violin sound on the treble keys. NB: This is not the same as Multi timbres, which employ different samples *patched* to keyboard regions, because the samples are all from the same instrument.

#multi means that a timbre has multiple samples for the same source. So this could mean many samples are different velocity levels or many samples for multiple keyboard ranges.

#mpe is another special tag that highlights those timbres as being designed such that using an MPE controller will be the best way to extract their full potential. #mpe tagged timbres will typically have modulation sources such as pressure (sometimes called aftertouch), velocity and slide (mod wheel) mapped to

interesting timbre parameters. See https://youtu.be/Q6bvRj4GTQ4 for an explanation of 5D touch.

#mpe timbres will sound best with an #mpe controller / keyboard, and if you have an #mpe controller lots of other timbres will be very playable; plus, using the synth engine, it's easy to map one of the MPE dimensions to a timbre parameter for more expression. This will be covered in the Modulators section in the Partials and Sound Design chapter.

#loop is a special properties tag that doesn't apply to timbres. You'll find this on samples, and it means that the sample has been designed to loop. Samples tagged with #loop have loop markers and attributes set to make long continuous loops of the sample as seamless as possible. But more on playing samples in the next section.

FYI: #stereo isn't really a special property tag — it simply describing that the timbre has a lot of movement in the stereo field. Most timbres are stereo, but if you're looking for some stereo width and movement for a sound, start by searching timbres with this tag.

And there's #virtuoso which simply means the timbre is effective across the keyboard range. Often a timbre designed for bass will not sound pleasing in the upper octaves, or a sound designed for the treble clef will *breakdown* in the bass region. If you're looking for sounds that present nicely over a large octave range, start with these #virtuoso timbres and, of course, the #multi and #split timbres.

3.3. Samples

Sample processing and playback are an integral part of the Regen synthesizer engine, and this is reflected by the LIBRARY panel having a section devoted to samples.

Synclavier Digital has been lucky enough to work with some of the world's leading sound experimenters during Regen's development: Jamie Lidell, Leith

Fleming-Smith, and Drew A Neumann have contributed many wonderful samples to this product. Additionally Anthony Marinelli has contributed high-fidelity samples of his pianos and synth collection that are instantly applicable. In addition to these artist sample libraries, the Selected Legacy Samples library includes over 350 samples taken from the archives of Synclavier, many in the glorious 50Khz sample rate of the Synclavier II machine.

All samples included as factory presets are completely royalty free, and can be used in the Regen user's composition whether for fun or for commercial release.

By means of an SD card, users can load their own samples and many common formats are recognized.

Loading Samples

The **Sample** section of the LIBRARY is much like the **Timbre** section, described above. There are also 3 levels of zoom, accessible using the **Arrow Left** and **Right** buttons. Pressing **Enter** will replace the current track's timbre with a fresh timbre containing only the sample highlighted. Before you do that, you have the opportunity to **Audition** the sample before loading it. Press and hold the **Audition** option button to hear the sample played at its native sampling rate.

Samples are used as an oscillator type on a partial, so one timbre can contain many samples. You can call up a sample to a freshly-created timbre on the current track with the **Enter** button, but more often than not, you'll want to choose a specific partial on your existing timbre to load a sample to. You can do this with the **Choose Partial** option, which naturally acts differently than for timbres: for samples you simply select the destination partial with a selector button and tap **Enter**.

There's a third way to load a sample — to a samples patchlist — and this will be covered in the Soundfiles section of the chapter Partials & Sound Design.

Using your own Samples

If you have samples, you can load them onto an SD card for use with Regen. Many common formats are recognized. You should be okay if your sample if one of the following formats, but beware that weird encodings such as 12 bit DWW may not work.

.wav

.aiff

.snd

.raw

.W64

.caf

.sd2

.iff

.flac

To use an SD card, we recommend formatting it as FAT32. Only the first partition of an SD card is read, but it's unusual to have more than one partition on an SD card. Regen looks for a specific directory structure. It will create this directory structure the first time you save a timbre to your SD Card. But if you want to load your samples first, you can create the directory structure yourself:

> So with a fresh SD card, you can create a top-level directory, `Libraries/`. In there, you can create many sub directories, and each one will be recognized as a user library. Once you create a library directory, e.g., `SD card:/Libraries/My Great Samples/` then you can add sample files to that directory.

When you insert an SD card, you should feel it latch into place and you'll be presented with a message, "Load SD Card?". Select the **Load** option, and the card will be indexed. You'll see the **BUSY** indicator light on for a moment or three, and a message stating, "Loading SD Card - Do Not Remove." It's very important to heed this warning — if a card is ejected during this process the system will hang and the only way to recover would be to shut-off power, losing any unsaved data. There's a possibility also, of damaging the contents of the SD card. So if you've suddenly realized you inserted the wrong card, just wait a few moments until it's loaded before ejecting it.

You can, for instance, have several cards, one for each project, and swap them in and out with relative ease. Other than during the warning message and

having the BUSY light lit, you can safely remove an SD card, at any time. However, you are advised to put the unit to sleep prior to removing an SD Card (from the **Home** page), as this will ensure its safely unmounted and the current system state will be saved to the internal state.

> Now, to view your Samples, tap **User** in the LIBRARY panel. If you have multiple libraries on that SD card, you'll be able to select them via the **Up** and **Down** Arrows. Then tap **Samples** to view your samples. You can load them using the same procedure for preset samples.

3.4. Reverbs

If you tap the **Reverb** button on the LIBRARY panel, you'll be presented with a list of preset reverbs. Unlike timbres and samples, there's only one library of reverb presets, so it doesn't matter which preset library you have selected, you'll always get the same list.

There are 3 levels of zoom, as usual, and you can load one of these reverb presets to the MASTER reverb via the **Enter** button. Since many Synclavier timbres benefit from a small amount of reverb, the default MASTER reverb is Studio A [1-1]. If you want a clean master output with no reverb select Zero Reverb [1-3].
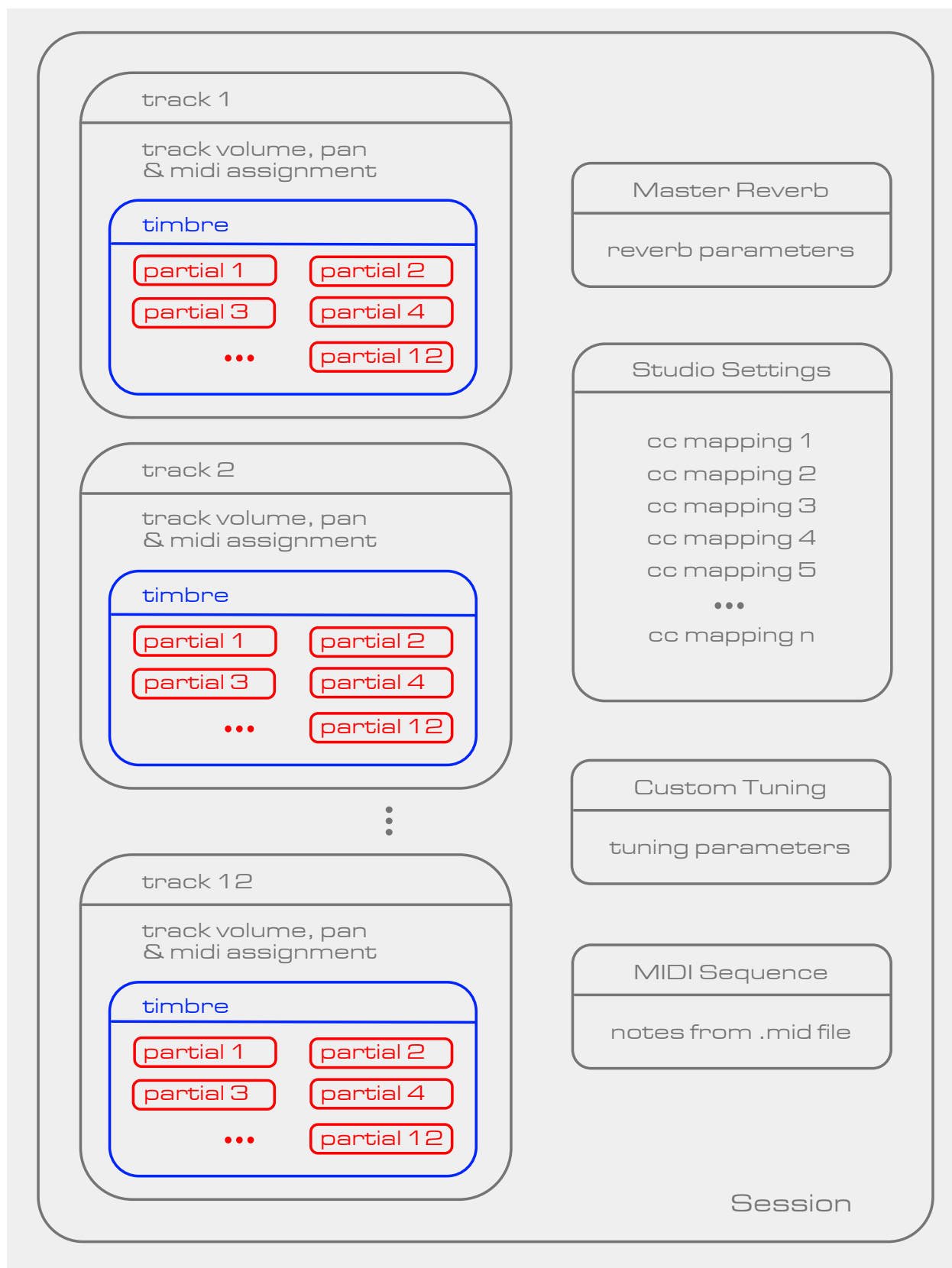
If a preset reverb is the current reverb on the MASTER output, it will be denoted in the list by a red blob. Reverbs can also be applied to individual tracks — covered later in the manual.

A reverb algorithm draws from many parameters, and these will be covered later in the manual — see the Timbre Effects section of the Playing With Timbres chapter. Now, it's sufficient to say that the reverb parameters are accessible from the **Reverb** button of the MASTER panel. When you modify one of these settings, you will have deviated from a preset reverb and be inputting a custom reverb. If you come up with something you like, you can save that to an SD card (covered

later). Since the MASTER reverb is part of the session, this is also saved when you save a session on an SD card (sessions covered in the next section).

It's sometimes difficult to evaluate reverbs against each other, as there's a lot going on, so one tip is to start with a percussive sound, such as Raindrops [RW 7-7], and flick through presets or change settings before going to something more harmonious like Hour Glass Transmission [RW 3-8].

3.5. Sessions

track 1

track volume, pan
& midi assignment

timbre

partial 1   partial 2

partial 3   partial 4

•••   partial 12

track 2

track volume, pan
& midi assignment

timbre

partial 1   partial 2

partial 3   partial 4

•••   partial 12

track 12

track volume, pan
& midi assignment

timbre

partial 1   partial 2

partial 3   partial 4

•••   partial 12

Master Reverb

reverb parameters

Studio Settings

cc mapping 1
cc mapping 2
cc mapping 3
cc mapping 4
cc mapping 5
•••
cc mapping n

Custom Tuning

tuning parameters

MIDI Sequence

notes from .mid file

Session

As can be seen from the memory map above, a session contains the entire sound design state of the machine. When you save a session, you save all the track information along with any timbres they contain, as well as all your studio settings (covered in the next section), Master Reverb settings, and any custom tunings. So when you later recall a session, you can continue where you left off.

Factory content for sessions is divided into 4 categories: Split, Stacks, Mixes, and Beatloops. None of these is meant to be a comprehensive suite, but they act as examples of what can be done with Regen and can be used as starting point or templates for your own work.

Unlike timbres and samples, there's only one library of session presets, so it doesn't matter which preset library you have selected, you'll always get the same list. There are 3 levels of zoom, as usual — the rightmost *inside cover view* view detailing useful information on the session — and you can call up one of these presets via the **Enter** button.

When you load a session, you'll be loading all the tracks (even empty ones) containing all active timbres(with one exception), volume and panning for each track, Studio mappings, and the MASTER **Reverb** algorithm. This will replace any current information. When you select a session for recall, you can elect to load or not load the Studio settings using the Change Recall option button — see screenshot below. Tapping **Enter** will confirm the recall.
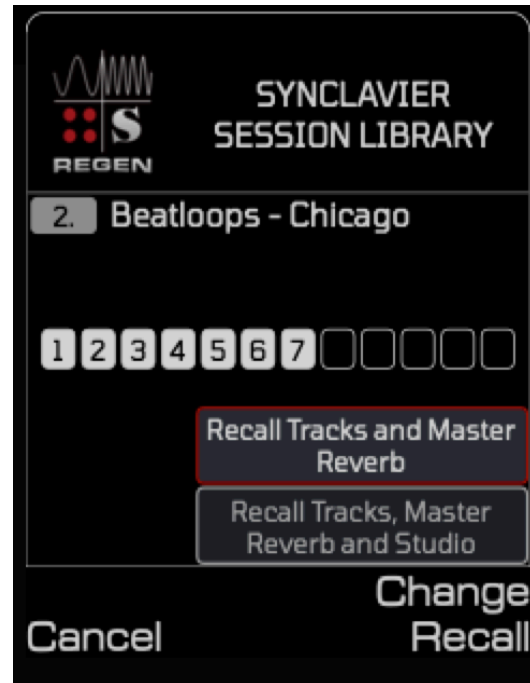
The exception noted above, is the **Choose Tracks** option. If you tap this **Option**, much like the **Choose Partials** function, your selector buttons will flash cyan, and on the off-cycle the currently active tracks will stay dimly lit. Now you can pick and choose which slots to load tracks into before tapping **Enter**, allowing you to preserve any current tracks you want to keep, and simply load the tracks into a session you intend to use. Selective recall will never overwrite the MASTER **Reverb** settings.

Once a session is loaded, when you select active tracks with the selector button, named timbres will have their title displayed in the VK Display, but if you want to see a full track listing, you can tap the **Session** button in the NOTE FX panel anytime and the session title and track-list will be displayed. The tracks are numbered and show their assigned MIDI channels on a green blob.

If you swipe all **Selector** buttons while in Tracks mode, and hit **Cut**, you'll clear the system of all current tracks as well as volume and pan settings. Master Reverb settings will remain intact.

Regards the factory content, Stacks are 2 or more timbres layered at different volume and pan levels. Combining, for instance, a synthesized piano and

sampled pianos, such as in the case of Stack - Triple Piano enables us to blend many similar sounds for a rich sonic texture. Other applications are to add a bed of background texture to a lead sound, or to add a percussive attack to an otherwise gentle sound. Splits are also 2 or more layers stacked, but the sounds are divided along the keyboard range rather than all playing all at once in the case of Stacks.

A Mix is a complete set of goodies needed to start making a song. Unlike Stacks, the individual tracks are set to listen to different MIDI channels. The convention is that channel 1 is for drums, channel 2 has bass, channel 3 will have a preset useful for chords, and channel 4 for a lead sound. That's enough to get a groove going, but some Mixes contain further channels with accompaniment sounds.

Beatloops are curated chunks of audio that are matched in tempo. In the description, you'll find BPM and bar count information as well as the trigger keys. NB: BPM assumes quarter note lengths. When you strike a key to play a beatloop it will play through in entirety. At the end of the beatloop, you can strike the same key to play it again, or you can use an external MIDI sequencer set to the same tempo to trigger it at the start each pattern, thus creating a loop. You can bring in and out the various beatloops whenever you want and mute them with the black key/s to the right of the trigger key.

The supplied beatloops are pedagogical in nature and intended to provide inspiration, but if you end up using a beatloop in a production of your own, that's fine, since all Regen content is royalty free.

Follow along with this walkthrough to get the idea:

> Load one of these beatloops. If you're on the highest zoom level, you'll see the description on the main display after loading, which will inform you what keys play what. For instance, in the case of **Beatloops - Paris**,

C3 and D3 are drums, A3 and B3 are drones, and accompaniments are on F4 and G4.

Strike and release C3 on your MIDI keyboard and you'll hear a drum track. To the beat, now strike and release D3, and you'll hear the drums replaced with an alternative version. It will play for 8 bars, unless you press a C#3 or D#3, which will mute the drums. Now press C3 and A3 simultaneously and you'll hear a drone loop accompanying the drum beat. This can be substituted with B3 or muted with A#3. Play around, pressing keys to the beat and bringing in the accompaniment with F4 or G4, and have yourself a jam.

So it's fun and easy to jam with beatloops, but for this to be a bedrock on which to design a full song, you'll likely want to sequence this with a external MIDI sequencer, such as the likes of an Arturia Keystep, Elektron Digitact, or Korg SQ-64. In which case, you'll want to select the tracks you're using for #beatloops: tap **Levels** in the TIMBRE panel, then **Arrow Left**; you'll see the **Track Crossfader**, and the first option is **MIDI Channel**. Rather than the default ALL value, change this to one particular MIDI channel that you'll use with your sequencer. If you set the same BPM and bar count on your sequencer, you can send notes at the start of each pattern. Now you have a beat going, you have the other MIDI channels free to send other instruments to Regen to play along on other tracks.
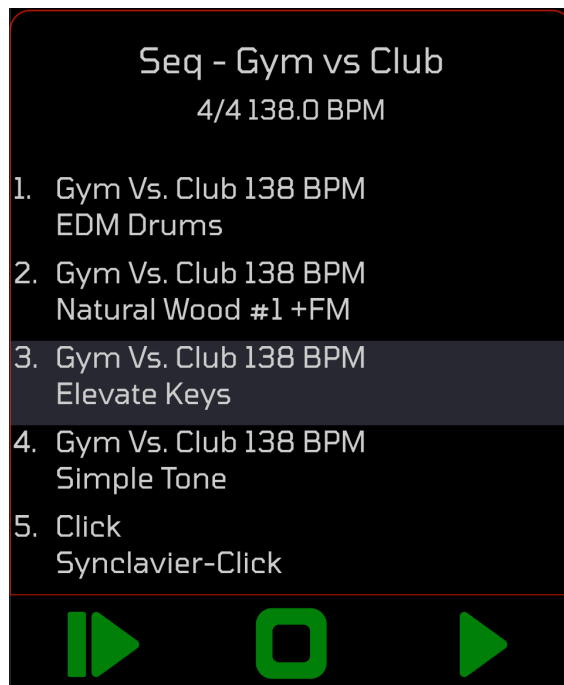
Sequences

Sequences is a bonus feature of Synclavier Regen, which allows you to load .mid MIDI sequence files and play them back to trigger Regen tracks, either for demonstration purposes or as a backing track for gigging musicians. If you have any on your SD Card, they will appear under the **User** list as an off-white

filenames — in either the **Timbres** list, if they're contained within your library, or the **Sessions** list if they live there on your SD Card.

When you select a sequence with the **Enter** button, you will see that you have an option button for various load settings. Pressing **Enter** a second time will load the sequence into the current Session. From then on, the sequence can be saved with the session.

You can load one sequence into the session at a time and easily clear the current sequence from the Session menu (via the NOTE FX panel). There are some sequences included in the factory content in the Mixes and Beatloops sessions.

For example, load the Mix - Gym vs Club session from Leith-Fleming Smith's Worn Jacket library. Since this session contains a sequence, Regen takes you straight to the sequence player page:

You'll notice there are 5 sequence tracks. The click track is auto-generated by Regen if required and can be muted. In this case, the other 4 tracks are simply named

as per the filename. The Regen track information is shown below the sequence track names, and if you just wanted to see the usual session track listing, you can press the **Session** button to get that. Then press the musical note icon ♫ to go back to the player.

 that 4 tracks have been recalled. If you tap **Sessions** from the NOTE FX panel, you'll see that the tracks are named, "EDM Drums, Natural Wood #1 +FM, Elevate Keys, and Simple Tone."

 Tap the **Start** option to hear the Gym Vs Club sequence. You'll also hear a 1 bar count-in metronome. Then tap **Stop** to pause the playback and **Continue** (option 3) to continue where you left off.

To load your own sequences, use the .mid format and copy your files to the `Sessions/` folder of your SD Card or one of your libraries. Most of the common .mid format functions are recognized. Only the first 12 MIDI tracks will be loaded and no tempo changes are supported.

   Track zero of a MIDI sequence file is traditionally a conductor track. If track zero has tempo/meter events and no notes it is turned into a metronome track when loaded by Regen and clicks are synthesized on the beats. You can elect to turn this option off from **MIDI File Options**. Also, under this page, you will find the Track Mapping relationship, which can be set thusly:
   "Map Tracks 1:1"
      - Track N of the MIDI file goes to Track N in Regen regardless of the Track Crossfader settings. This is the default and should work for most situations.
   "Use Track No. As MIDI channel"

- The Track number of the MIDI file is used as the 'MIDI Channel Number' into the Track Crossfader. That's to let you use 1 track in the MIDI file to trigger 2 or more Regen timbres, for example.

"Use File MIDI Channel"

- The MIDI channel number encoded in the MIDI messages on the track are used as input to the Track Crossfaders. Endless possibilities here. Each track in the MIDI file

could trigger 1 or more Regen timbres on any track.

"Use Track No with MPE"

- This should work for MPE data encoded on MIDI tracks and will work like "Use Track No. As MIDI channel."

3.6. Saving

Firstly, you'll need an SD card. Any size should work. FAT32 format is recommended, but some other formats are recognized.

The SD Card slot on the left side of the unit is of the spring-loaded style, so when you push your card in, you'll feel a satisfying resistive latch. The main display will prompt you to "Load SD Card?". Select the **Load** option, and the card will be indexed. You'll see the **BUSY** indicator light on for a moment or three, and a message stating, "Loading SD Card - Do Not Remove." It's very important to heed this warning — if a card is ejected during this process the system will hang and the only way to recover would be to shut-off power, losing any unsaved data. There's a possibility also, of damaging the contents of the SD card. So if you've suddenly realised you inserted the wrong card, just wait a few moments until it's loaded before ejecting it.

You can, for instance, have several cards, one for each project, and swap them in and out with relative ease. Other than during the warning message, you can safely remove an SD card, at any time. However, you are advised to put the unit to sleep prior to removing an SD card (from the **Home** page), as this will ensure its safely unmounted and the current system state will be saved to the internal state.

Let's assume you've made some changes to a factory preset timbre, or created your own from scratch and want to save it on your SD card. Enter your card and push so it locks in place. Tap **Load** and wait for the indexing to complete. Tap ⊞ **Save** and you will see the list of user libraries on this card. If this is the first time using this card, you'll be presented with a user library with the default name, "User Library 1".

 You can change the name of the default Library by selecting **Edit & Delete** and then **Rename Edit**. If you don't already have one connected, plug in a USB keyboard to a free USB Accessories socket. Now you can type your name for the new Library. You can also add a description if you wish; tap **Edit Info** or press tab on your USB keyboard. When done, tap **Save** and you'll be returned to the list of user libraries.
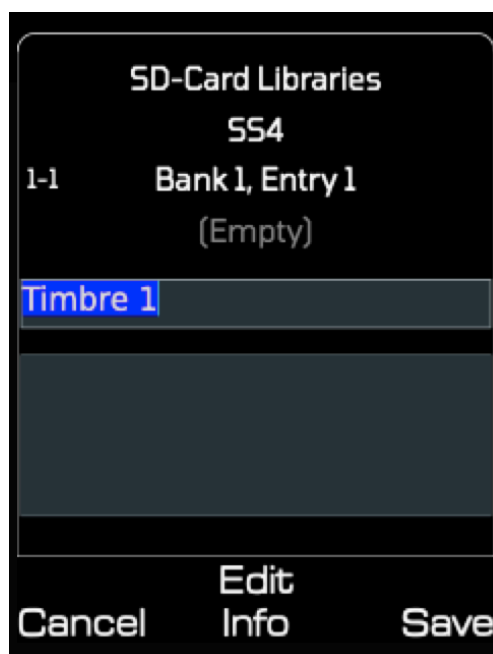
Now tap **Timbre** and you'll be presented with the familiar timbre slots in your first user library (8 banks of 8). Use the **Swiper** to scroll to a slot you wish to save the timbre to and hit **Save**. The timbre from the last-selected track will be cue'd to save in this slot and you'll see its title with a version number appended (since it's no longer the original). The timbre description is not copied across, since it's likely that the reason for saving this new timbre is that its properties have changed and the old description will be invalid. But if you save the same timbre to the same slot — say you've been working on it a while and have updates to it — the info will be preserved and a version number will not be appended.

With a USB keyboard attached, you can now change the title and optionally choose to add a description; type tab or tap **Edit Info**. Then tap the **Save** option and your timbre will be saved on your SD card.

Regen will create the necessary directory structure on the SD card as it needs to:

```
SD Card:/
    /Libraries
    /Reverbs
    /Sessions
    /Studios
    /Samples
    /Settings
```

So in the case of saving a new timbre in a new library, you would get a directory structure, `Libraries/User Library 1/New Timbre 1`, for instance.

You can have one reverb library per SD card and one session library, and on your SD card they will be created as `Reverbs/` and `Sessions/` top-level directories.

The procedure described above for saving timbres is the same for reverbs.

Samples cannot be saved directly. However, when you save a timbre it will save out any samples to the `/Samples` folder of your SD Card if the samples are not already on that card. This enables hot-swapping of cards. Factory samples, of course, do not need to be saved to an SD Card.

This top-level `/Samples` folder is only for Regen system use. If you wish to load your own samples onto a User card, you should save them in a library subfolder. That is the only way to view them from the Regen user interface.

You can perform file-management on a computer with the contents directly, such as renaming libraries and deleting or backing-up timbres and libraries. You can customize the presentation of your libraries with a cover image and presentation by adding `CDImage.png` and `Description.txt` to a library's folder. Both files are case-sensitive. `CDImage.png` expects an aspect ratio of 1:1 and `Description.txt` has a 220 character limit.

NB: An example User card structure is available as a download from the Regen Downloads page – synclavier.com/regen-downloads – this is useful if you want to prepare a blank SD card in advance and load it with samples and/or copy timbres from another project.

Sessions

With sessions, the saving procedure is a little different. Sessions don't have bank and entry slots, so you just start typing the name you want to save the session under, tap the **Save** option button, optionally edit the description, and hit **Save** option again. If you don't have a USB keyboard and don't want to overwrite a session with the same name, hit **Nudge Up** to add an appended version number and then tap **Save** option.

When you save a session, you're saving all the tracks, the master reverb setting, and any studio settings you have set. This means you can revisit the session anytime and your *workspace* will be restored for you. NB: The timbres in the tracks are not saved: i.e., if you load a timbre to track 3, say, and tweak a few parameters, when you save a session, this new tweaked timbre is saved in the session in track 3, but not updated in the timbre library; if you wanted to do that, you'd have to also save the tweaked timbre. Likewise, all your CC mappings are

saved in the session, but if you want to save those mappings for use as a studio, you would have to explicitly save it in the studio settings.

> Regen saves index files on your SD Card with the extension `.tsv`. These files help Regen load information, such as soundfile length and tags, much faster than examining the contents of the SD folder directly. Regen knows to update these files if it spots a new file or a file is removed from a folder, but if you rename files, maybe to change the ordering, you could trip this up; in which case, you would need to delete the `.tsv` file, so Regen is forced to recreate it. You can elect to not build the index files when you first insert an SD Card under the **Index Options** option, but it is a rare situation where that would be necessary.

Renaming Timbres

Timbres can be renamed by loading one and going through the save process again. With a session, you'd have to save it under your new name and then delete the old session.

Importing Go! V and Synclavier II timbres

Synclavier Go! timbres you may have created can be loaded to Regen. Simply copy them from your iCloud or iPad folder and put them in a library folder on an SD Card. Regen recognizes Go! timbres, but not the other way around — the synth engine on Regen is more sophisticated that that of Go! with lots of advanced features, so it makes little sense to load Regen timbres onto the Go! app — they just wouldn't sound or behave the way you've designed them even if they could be loaded.

And, if you own Synclavier[3], existing Synclavier II timbres can be exported to Go! format by following the instructions here: https://www.synclavier.com/ Synclavier3Documentation/Timbre-Library-Descriptions-and-Export.pdf

Importing timbres from Arturia's Synclavier V is currently supported, but we draw your attention to Arturia's "Rights of use" as laid out in the End User License Agreement, which can be found in the manual. Instructions for loading

your own users presets from Synclavier V to Regen is here: https://www.synclavier.com/wp-content/uploads/SYNX-Import.pdf

NB: Synclavier V contains a bank where effects, such as delay and chorus, can be added to the sound at the end of the processing chain. Since there is no one-to-one mapping for these effects on Regen, any sound converted from Synclavier V could potentially sound and behave different on Regen.

SESSION

4.1. Tracks

Synclavier Regen is multitimbral, meaning more than one timbre can sound simultaneously. This is made possible through the use of tracks. NB: In some synthesizers this feature may be referred to as layers, since you can layer many sounds.
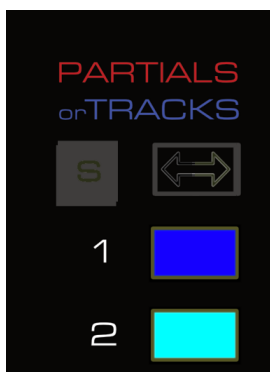
The image above shows the selector panel. The **Switcher** button at the top-right is lit blue, which means it's in **TRACKS** view. Each track can have 1 timbre loaded onto it. Active tracks are tracks that currently contain a timbre and they'll be lit blue — as per track 1 shown in the image. To select a track, tap its selector button, and it will turn cyan — as per track 2 shown in the image.

You can either tap a button or swipe multiple buttons to select tracks. If you want to select non-contiguous tracks, hold a finger down on a desired selector button, then tap further selector buttons, before removing the first finger.

To reset the state of the machine regarding sound design, you would select all 12 tracks and then tap the cut button.

A track doesn't have many parameters — one button's worth — if you tap the TRACKS **Levels** button on the TIMBRE panel, you'll have access to them on the option buttons:

**Track Volume | Track Pan | Track Xpos**

The VK display only accommodates room to display one value, so with multiple track selections the VK display is going to show the value of the last track selected.

Track Volume

Let's take **Volume**. If you swipe **Selector** buttons for tracks 4 to 6 and then tap **Levels**, you'll see something like "T6   0dB" in the VK display. It's showing the value of the last track selected, so if you want to see a particular value, you'd have to tap that track's **Selector** button.

However, in the case of **Levels**, the top-third of the main display actually has 12 little VU meters, so although you can't see each exact value, you get a sense of all track volumes and the overall mix. And the pan settings are represented by a small green bar at the top of the VU. Most pages have a neat graphical display in the top third that portrays useful information for the current page and parameters.

The last track selected* is shown on all parameter pages in the top-left of the main display on a cyan-coloured blob, so this is a good place to keep an eye on.
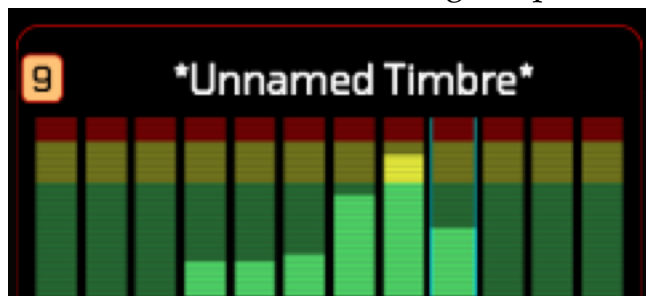
One of the time-saving features of Synclavier Regen is that you can change multiple values at once, even if you can't see each value when more than one is selected. And in solo mode, all solo'd partials will change. The way this works, is that the *last selected track can increase or decrease to its extents, but it drags the other selected tracks along with it — each other selected track will increase or decrease until either, they hit their extent, or the selected track hits its extent.

> Try playing with this now on the TIMBRE **Levels** Volume page**.** This is a good page to experiment, because observation of the VU meters will demonstrate how it works.

Track Pan

The **Pan** parameter is self-explanatory — Synclavier convention is that plus figures are right-panned and minus figures are left-panned, up to a magnitude of 63 corresponding to fully panned.

Track Xpos

Xpos is short for transposition. Here you can transpose a track up or down +/- 72 semitones. This not only retunes the additive and subtractive generators, but for sample patchlists (more on that later) it repatches: so rather than pitch-stretching to the desired transposition, it will select the sample range to play based on a modification of the incoming MIDI.

Track Crossfader

So with **Pan** and **Volume**, we can call up a bunch of timbres to some tracks and arrange a mix where the instruments are laid out at appropriate levels. But how do we play multiple tracks at once? This is where the Track Crossfader comes in. If you're coming from a background of using other multi-timbral instruments, you might be familiar with the term Keyboard Mapping or MIDI Mapping and it might be easier for you to think of the Track Crossfader as that function — the

Track Crossfader is where you filter to listen for certain MIDI channels and keyboard ranges — but that's not all it can do. You can apply **Fade In** and **Fade Out** values for your keyboard ranges, so if you have more than one track listening to the same MIDI channel, you will get *crossfading* of the tracks as you play.

That nomenclature out of the way, let's explore, If you tap **Arrow Left** from track **Levels (**on the TIMBRE panel), you'll get the main page of the **Track Crossfader** with the following options:

MIDI Channel | **Low Note** | **High Note**

So, it's easy to think of these as filter controls. The default MIDI Channel is ALL, so incoming MIDI is not filtered by channel. To set a track to a specific MIDI Channel, make sure that track is selected — refer to the top-left blob — tap the **MIDI Channel** button and swipe or nudge to your preferred selection.

MIDI parameters can also be set another way. A second pressing of MIDI Channel will toggle it into "arm"mode. You will see a thin green line to indicate the parameter is *armed* and listening to MIDI events. The channel number will now be set to the channel of the next MIDI note on event to be received by Synclavier Regen.

You can further define the track to a keyboard range, filtering-out incoming notes that don't fall within that range. In this way, you can quickly and easily split your keyboard across multiple tracks and have different keyboard ranges sound different instrument timbres.

The range defaults to every key. Tap **Low Note** or **High Note** and swipe or nudge to a specific key, or tap the option button again to *arm* it. When *armed*, Synclavier Regen is going to listen for the next MIDI note on event to set the selected range limit. The graphic of the keyboard employs shading to highlight the chosen range for the current track. The line graphic shows transfer functions of each track, with the keyboard range on the x-axis and volume on the y-axis.

Double-tapping **Low Note** or **High Note** options will set that parameter to its default — a shortcut available on many Regen parameters.

- If, when the **Low Note** is *armed*, you press and hold your chosen MIDI key for low range and while its depressed press your chosen key for high range, both notes are set. Likewise, if you have **High Note** armed, you can press and hold your chosen high note and then press your chosen low note and both will be set in one action.

- NB: Keyboard ranges can also be set from the session pages, and some will find this easier, as you can see the range of each track at the same time on one list. Tap **Session** to get the track list and then **Arrow Right** to see each track's range, and you can modify them from there.

If you **Arrow Left** from the Track Crossfader page, you'll see further controls:

Fade Shape | Fade In | Fade Out

You can adjust the fade so that a given track isn't simply "on inside the range and off outside it", but it gradually fades in and out. This is best seen on the graphical display. If you have a range set of, say C4-B4, and then apply a **Fade In** amount of, say 12, the octave below will also sound the track, and the volume increases from C3-B3. Using fades is one technique to make keyboard splits more natural sounding, having adjacent tracks crossfade each other. Other techniques for making multi-sampled instruments more natural sounding are covered in later sections of this manual.

It should be noted that there are lots of applications for the Track Crossfader. Maybe you want multiple tracks to sound at once when you press a range of MIDI keys, because you're combining a piano and pad sound with a long release for a harmonizing effect on your melody line. The pad sound's volume is lower in the mix, so it backs up the piano, giving it a bed of sound. Adding many timbres to sound at once is sometimes referred to as layering. Or maybe you want different #percussion timbres on different keys, but you extend the ranges to the sharp keys between, so that two percussion timbres can sound at once, for instance a bass drum and hi-hat cymbal. So, in summary, there are lots of uses for this feature and easy to configure.

Saving Track Info

So all this track information — levels, panning, crossfading — is part of a session. When you save a session, all the track information, along with the MASTER **Reverb** settings, are saved within that session. If you load a saved session, all this information is overwritten.

Conversely, if you simply load a new timbre to a track, the other session information is not overwritten. So if you have a snare drum sound on track 1, for example, configured for one particular key of your MIDI keyboard (say C2), panned slightly to the left, and set the volume to be congruent with other sounds in your mix; then you decide to try a different type of snare; you can load a new snare timbre to track 1, and haven't disturbed the placement of the snare in the mix or the key that sounds it.

Mixing

It might seem silly to state, but getting the levels of instruments correct in a mix is paramount. Things that are *fixed* with EQs and effect plugins during the mixing and mastering stage, often could have been addressed earlier by tweaking mix levels.

However, in some cases, two or more instruments present energy in the same frequency range, and even with level adjustment, it can still present a problem. If these clashes can't be addressed in arrangement, and can't be fixed with levels, the mixing engineer will next try to EQ the sounds to remove or lower some of the frequency content from the sounds that clash.

In a rock band set-up, for example, you might have vocals, drums, bass guitar, acoustic rhythm guitar, and lead guitar, each having their concentration of energy in a different frequency band. The reason that instruments such as the Fender-Rhodes and Hammond organs became so popular in rock bands was because their spectrum of sound didn't step over these other common instruments too much.

With the plethora of synthesized sounds today, occupying larger and larger frequency bands, one has to be more careful when introducing instruments to a

mix. With lots of synthesized sounds playing together, certain regions of frequencies can get very busy, such as the 200-300Hz range.

One thing to note is that with additive synthesis, you can remove or reduce certain frequency ranges by dialling back particular harmonics, if you know where that sound is going to be played on the keyboard. This can be done with other forms of synthesis, such as subtractive, but it's more surgical with additive, and the adjustment occurs at the point of creation, not after the mix.

So the sound design doesn't have to stop at the composition stage and can be revisited at the mixing stage. One advantage of reducing or removing unwanted frequencies from the the sound at the source is that you don't have to use filters (EQ plugins are essentially multiple filters) that can add phase shift and other undesired distortion to the sound. It's much better if the frequency content you don't want isn't there in the first place, and this is something to bear in mind when using additive synthesizers such as Synclavier Regen.

Additive synthesis will be covered in detail in the Partials & Sound Design chapter.

4.2. Studio

If you 're working with sessions a lot and blending different sounds, you might find it laborious to tap a track, then tap levels, then use the **Swiper** to set something, then go to another track and do the same, *ad nauseam*. It's not as easy as working with a mixing desk. But it could be!

Volume and **Pan** settings can be linked to MIDI CC input. So if you have a MIDI keyboard with lots of CC knobs and/or faders, you can put them to use controlling the mix. And there are products that are devoted to CC controls, laid out more like a mixing desk, such as the Novation Zero, Studio Logic SL Mixface, Nektar Panorama, Korg NANOKONTROL, AKAI Professional MIDImix, iCON iControls, Faderfox EC4, Electra One, Livi Alias 8, and so on.

Example of MIDI keyboard with CC controls

Example of MIDI *mixer / DAW controller* interface

While on the Track **Levels** page, press the **MIDI-link** button and you'll be presented with the MIDI Controller page. If **Track Volume** is not already highlighted, tap to select it. Tap it again to *arm* it, then twiddle a knob. The **MIDI Channel** and **CC** values will be linked to the **Volume** parameter for the currently selected Regen track. You can set these manually too, using the **Swiper** and **Nudge** buttons, arrowing left and right to change selection from **Channel**, **CC** and **Abs/Rel**.

The **Abs/Rel** represents the mode your MIDI controller is set for sending data. Abs is absolute: this is typical for faders and rotary encoders with end stops. Continuous rotary encoders are typically set to relative (Rel) and when adjusted, the MIDI value they send corresponds to the change up or down of that value, rather than its absolute value. To further complicate matters, the MIDI protocol supports several relative standards, using different bits of data to communicate a positive direction and negative direction. When you *arm* and link a MIDI CC, Regen will guess which mode the encoder is set to for you, but this can be overridden manually if the parameter doesn't seem to respond as expected to your encoder adjustments.

Once you've linked a control to **Track Volume**, press the **MIDI-link** button again to return to the track Levels page. Now adjust the encoder and you'll see the VU for that track change as well as the value in the VK Display. Now you can play your keyboard and manually adjust the amplitude without recourse to the Regen at all. The Synclavier Regen, once set-up, could be on the other side of the studio to you, and you could be controlling the **Pan** and **Volume** of all 12 tracks remotely over USB.
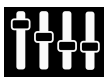
NB: When you save your session, it will save your current **Volume** and **Pan** levels regardless of how you last modified them (by Regen **Swiper** or by CC encoder). If you load a new session, provided your controllers are relative mode, you will recall all the starting **Volume** and **Pan** levels from the session and the MIDI links you set-up will now work on the newly loaded session. You'll get a little fly in the ointment with absolute encoders, because as soon as you adjust one it will send it's absolute value to Regen, overwriting the value, and it's not likely to be set to the position of the value if you've just loaded a fresh session, so you'll get a jump in level (the first time you adjust your encoders) which is only somewhat smoothed by Regen's smart synth engine.

All the parameters of the TIMBRE panel can be **MIDI-linked** to CC controllers. More on these parameters in the next chapter of the manual, suffice to say, a lot of these parameters have the ability to vastly affect the sound of a timbre, so mapping some of these parameters to hardware knobs can be of great benefit to the sound designer. Also, the parameters of the MASTER panel can be **MIDI-linked** with the same technique. And you only need do this once for your controller and then save the Studio settings.

There's no correlation between the MIDI filter of the **Track Crossfader** and the MIDI channel of the encoders linked to parameters, so you can have a MIDI mixer interface devoted to controls set to, say MIDI channel 15, and another MIDI keyboard set to, say MIDI channel 1, and your track 1 could listen to notes from the keyboard and CC information from the mixer interface. Alternatively, you could link the parameters of your track to be the same number as your track and the same channel could be set on the crossfader: in this way, you could have one MIDI keyboard with encoders to play and change values, and simply change its MIDI channel when you want to play a different track.

Saving MIDI Controller Settings

If you've set up all your tracks' volume and pan settings to your MIDI keyboard or controller, then you'll likely want to save that, so you don't have to do it again. The collection of information for MIDI Controller Settings is called a Studio and Synclavier Regen has the facility to load and save **Studio** settings.

If you tap **Home** then **Setup** then **Studio**, you'll be presented with a MIDI Controller Summary listing all the currently linked encoders. You can scroll through this list, and if you press **Enter**, it will take you to the **MIDI-link** page of the highlighted line. If you press **Save / Load**, you'll be presented with a bank and entry style list where you can save various studio settings to your SD card. There's a title and description field with ample room to detail what gear you were using and the purpose of the set up. NB: If you don't have an SD card, you can't save or load studios, but the current studio settings will be stored

when you tap the **Save State** option (on the **Home** page)so they can be recalled when you next use Regen.

Standard Mapping

A "standard" mapping is available to download for your use – included in the example User Card contents at www.synclavier.com/regen-quickstart. This can be used as a template and edited with a text editor to speed-up your own creations.

The mappings, shown in the table below, are per track, i.e., if you send a cc MIDI message on channel 1, you will alter the relevant parameter of track 1; channel 2, you will alter track 2; and so on.

| Track Parameter	MIDI message
Volume	cc7
Panning	cc8
FM %	cc2
FM Ratio	cc3
Chorus	cc12
Reverb	cc13
Filter Resonance	cc71
Filter Cut-off	cc74

4.3. Custom Tuning

On the Synclavier II, you were able to alter the master tuning and alter each pitch class individually. Similarly, Synclavier Regen brings you some overall tuning options, and takes it a few steps further.

This section covers the various ways that you can change the tuning of Regen. This is an advanced topic that many users can safely skip. All custom tuning

modifications affect every Regen track and the settings will be saved within a session. When a session is recalled, you can elect whether or not to recall the tuning information.

Overall Tuning

The simplest way to adjust Regen tuning is to use the overall settings parameter. Tap **Custom Tuning** from the **Session** page and Arrow Left fully to see this page:
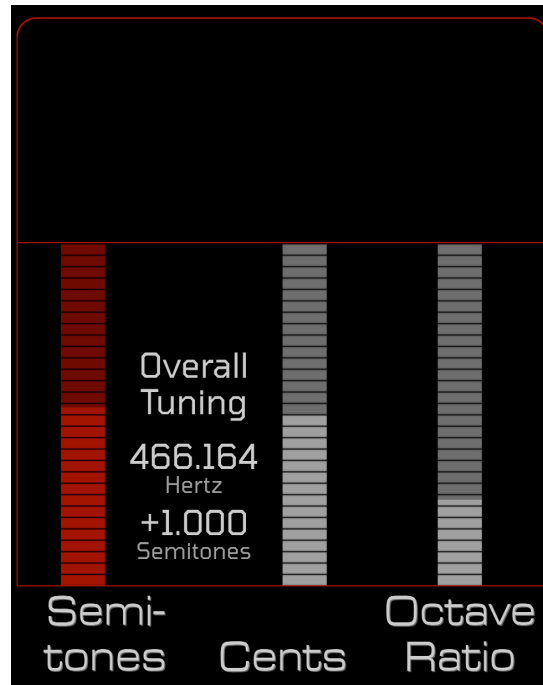
You can adjust the overall tuning of Regen from the A440 standard — https:// en.wikipedia.org/wiki/A440_(pitch_standard) in semitone and cent increments. Double tap **Semitones** to return to the default.

The **Semitones** adjustment was a useful feature of Synclavier II when used as part of a musical ensemble tuned to an alternative tuning. In large auditoriums that would warm up over the duration of the performance, the **Cents** control provided a useful feature to keep in tune with the other players.

Octave Ratio can change the relationship between an octave difference of input MIDI note and resulting pitch output; from, for example, the default double frequency per octave, to quadruple, with a value of 2.

NB: **Octave Ratio** can be modified per timbre and is covered in more depth in the Timbre Effects section of the following chapter. Of course, the tuning of

individual timbres and partials can be freely adjusted on Regen, which will be covered in later sections.

MPE & Pitch Bend method

Synclavier Regen fully supports the MPE protocol. You can connect an MPE controller to the 5-pin MIDI TRS port, one of the USB-A ports (with your Regen in Controller mode) or through a DAW on a computer or iPad and into the USB-B port (with your Regen in DAW mode).

With an MPE keyboard or other MPE interface, you can get "between" the notes. Pitch-bend messages are tied to individual notes and not global. Controllers like the Haken Continuum use this to send exact pitch information from the start of the note. Of course, MPE achieves this over MIDI by using all 16 MIDI channels, so you cannot add another interface on the same port. You'd have to tie up one of the MIDI interfaces on Regen just for one MPE input.

NB: Some sequencers support microtonal output by pitch-bend messages sent immediately after note on messages. Regen would, of course, support this method, but you'd be limiting yourself to 1 note of polyphony.

Once you've connected your MPE controller, tap **Home** and **Setup** then the keyboard icon, which will take you to the MIDI inputs page. Here you will have to ensure the MPE checkbox next to your interface is on (green), and you have the opportunity to select which "MIDI" channel it will be mapped to inside of Regen.
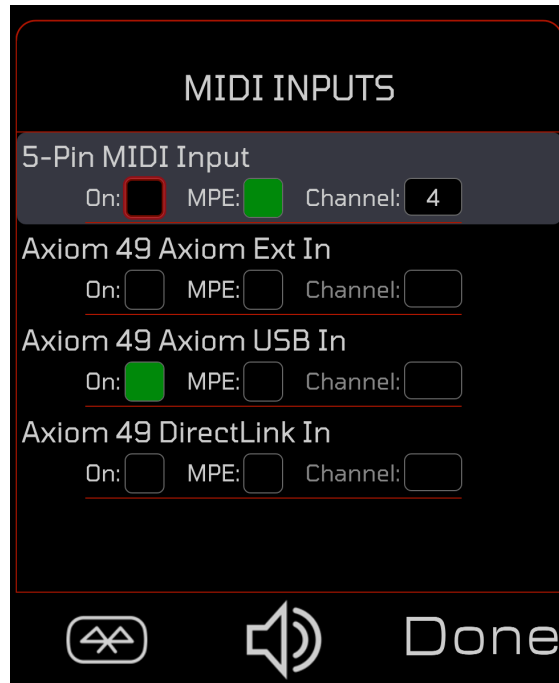
Tuning Models

When you tap the **Custom Tuning** button you will be presented with a page like this:
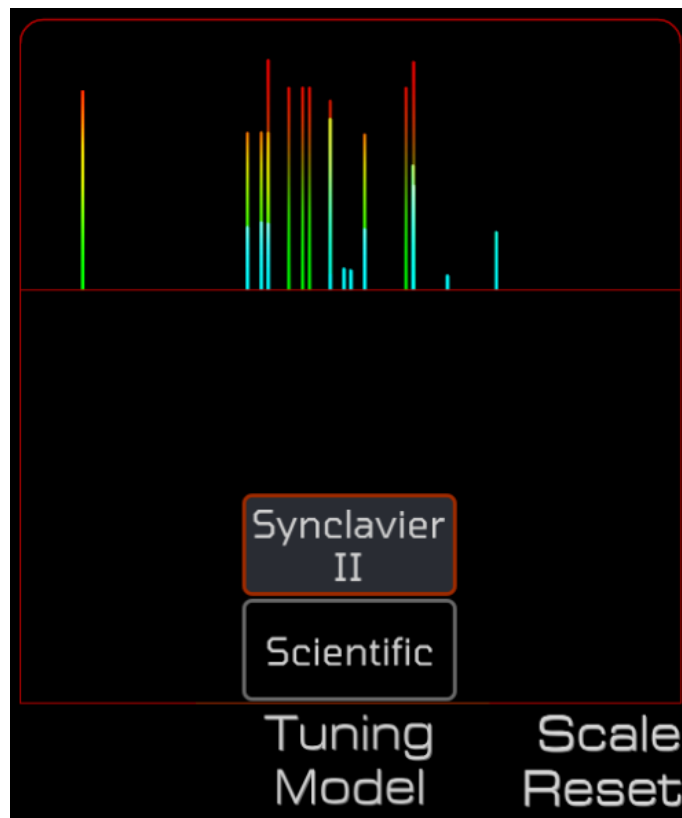
The first digital oscillators, like those used in the original Synclavier systems, can only produce a finite number of discrete output frequencies. For example, with Synclavier II, A-440 was synthesized at 439.990 Herz instead of the scientifically-correct 440.000 Herz. The octaves were also slightly out of tune, for example the A below Middle C was synthesized at 219.995 Herz while the octave below that was synthesized at 110.004 Herz. These small intonation errors were not audible on their own, but when playing a chord the result was a fuller and more dynamic sound, since the different pitches changed phase over time.

As with many acts of necessity, this and other Synclavier II nuances are responsible for the "vintage" sound of the machine. This tuning has been set as the default in Regen, but, since we're using a modern computing architecture for Regen, where more precise frequencies are possible, it can be turned off via the **Tuning Model** parameter, and indeed should be if you want to dive deeper into alternate tunings or it will get very confusing.

Setting the **Tuning Model** to Scientific generates frequencies with a resolution of about 1/1000 of a Herz at all frequencies and should be used to achieve precise microtonalities. The Synclavier II Tuning Model uses the nearest frequency that was available in the original instrument and should be used when you want to recreate the fuller less-electronic sound of the original instrument.

Pitch Class & MIDI remapping

This feature will be most interesting to those engaged in microtonal music production with interfaces such as the Lumatone. **Arrow Right** from the Custom Tuning Model page and you'll see this page:

```
Custom Pitch Class Tuning
1.    C           C            +0.0
2.    C#          C#           +0.0
3.    D           D            +0.0
4.    D#          D#           +0.0
5.    E           E            -13.0
6.    F           F            +0.0
7.    F#          F#           +0.0
8.    G           G            +0.0
9.    G#          G#           +0.0
10.   A           A            +0.0
11.   A#          A#           +0.0
12.   B           B            +0.0
      Pitch       Semi-
      Class       Tone      Cents
```

Use the **Pitch Class** control to select a particular class of pitch. You can then tune these up or down from their position on the equal temperament scale, in **Cents**. You can also transpose pitch classes to other notes up or down using the **Semitone** control.

To quickly select a pitch class (note) you can press whichever option button is lit a second time. This will toggle "arm" mode. Regen will then wait for the next key from your MIDI controller and the selection will change to that pitch class. A vertical green bar will show next to the **Pitch Class** to denote it's "armed" status.

For example, tap **Cents** then tap **Cents** again and the green bar will show. Now strike a key on your MIDI controller, and the selection will change to that key, and the green bar will vanish. The **Cents** option is still selected, so you can use the **Swiper** or **Nudge** buttons to adjust the cents value. Using this method, you can quickly change values for a number of pitch classes.

To reset a value to its default, double tap the option button. For example, tapping **Semi-tone** twice within 50ms will reset the semitone value to its default, i.e., the corresponding pitch class.

Playing with pitch classes provides a quick and fun way to explore other 12-note tunings. You can test out different temperaments, such as Just Intonation, Pythagorean , and Quarter-comma. You can send Regen some MIDI from a well-known tune and then play with these settings and hear the tune through a different lens.

MTS Tuning

If you **Arrow Right** again, you will see a similar page with these option buttons:

    MIDI Key | **Semitone** | **Cents**

Here you can go to the next level of control, changing any or all of the 128 individual MIDI keys. This has the potential to make a standard MIDI keyboard into quite a honky tonk piano, but these controls are more useful when you have a keyboard that can understand other scales.

The Lumatone, for example, has an isomorphic keyboard, whose keys can be set to different colours. By judicious use of those colours you can arrange a scale with more than 12 notes per octave without getting lost (this is dubbed as microtonal, since some notes are between the usual notes). That pattern of note progression can be repeated many times over the layout.

All of Lumatone's keys can be set to any MIDI key from 1 to 128, so you would use this Regen page to set pitches that correspond to your configured keyboard layout for each MIDI key. Since you are now using more MIDI keys per octave, you would, of course, have fewer overall octaves to play with within the 128 key limit.

> Toggling option buttons to set Regen "armed" to listen to incoming MIDI notes works on this page too, but different to the Custom Pitch Class page, when you arm for Semi-tone, you will set the semitone value rather than the MIDI Key selection.

NB: modifying the cents of pitches is additive. I.e., if I set each **Pitch Class** for C to be +100 cents and then I set C-4 to also be +100 cents, when I play the middle C on my keyboard I will hear D.

So this is the page where you would experiment with different tuning schemes, even if they don't fit into the 12 note per octave rule (are not chromatic),

since each MIDI key can be mapped. So much music can be produced and enjoyed in other scales; Turkish, Arabic, Indian, Hungarian, Byzantine, Persian and many more — https://en.wikipedia.org/wiki/Scale_(music)

> Tuning maps represent one part of the MTS specification that was added to MIDI in 1992. Realtime and non-realtime SysEx messages that can dynamically change tuning schemes when sent to devices over MIDI are not currently supported by Regen.

Scala Maps

Scala has become the de facto standard to represent tuning maps in file format. Synclavier Regen supports the import of both Scala Keyboard Maps (.kbm) and Scala Definition Files (.scl). Both are required as a pair to define a complete tuning scheme. To import them, add the files to a library folder on your SD Card. They will appear at the bottom of the **Timbre** list in an eggshell coloured font. To load them press **Enter** and the custom tunings will populate the Custom MTS Tuning page. You can further modify these tunings if they do not suit your exact needs, but you cannot save them out as scala files.

The scala format hails from a freeware computer program called Scala, which you can use to view and develop tuning schemes. Lots of information is to be found online regarding Scala and microtonal music production. To learn more, start here:

https://www.mizzan.de/scala-format

https://www.huygens-fokker.org/scala/

https://en.xen.wiki/

PLAYING WITH TIMBRES

5.1. Timbres

Synclavier II VPK model

From its inception, The Synclavier has always been a many-buttons one-knob interface. Great efforts were made to get the feel and accuracy of the knob as good as possible, and since there was only one per unit, no expense was spared.

Having many low-profile buttons meant you could select multiple at a time (multiple parameters or multiple partials) and change them in concert.

That ethos has carried through to this product. Regen doesn't have a knob, but a touch-sensitive value **Swiper**, and rest assured we've invested great energies to get the **Swiper** as responsive and easy to use as the knob.

Of course, every design choice closes doors on other options, and some synthesists prefer having a knob-per-function approach or variations on this. Rather than debate the many pros and cons, we will present the operation of this unit for sound design, and the advantages and limitations to the Synclavier approach will become evident. We hope once users get their feet wet on Synclavier Regen many of our design choices will seem vindicated for this product.

So with that out of the way, let's look at the timbre panel:

NB: We've already covered the **Levels** which pertains to tracks rather than timbres — denoted by a white colour.

FM

As per the convention of this manual, FM, or frequency modulation, will be covered in some depth later on. For now, know that adding FM to a timbre is a modulation (controlled distortion) of the sound that adds upper harmonics and inharmonics.

Load Subtle FM Example [SDT 3-3] to one of your tracks and play a few notes. Now select **FM** and you'll be presented with these options:

Timbre FM ± | **Timbre FM %** | **FM Ratio**

Tap **Timbre FM ±** and you'll see a value of 0% pop up on the VK Display. Play a few notes and turn up the value. By about 10% you should notice a change in the sound. It could be described as having more bite or grunge or body or metallic flavour, all adjectives used to describe frequency modulation.

This clavinet timbre already has some FM on its two partials. This is just adding more. A little goes a long way, and you'll soon realise that tweaking the timbre-level controls on this panel can dramatically affect the character of a timbre. When you get proficient using them, you'll be able to quickly morph preset timbres in many different directions.

Now double-tap the button. The value returns to 0%. This is the default value for this parameter. Many parameters have sensible default values that can be recalled with a double-tap, which is a great time-saver to remember.

Now reduce the parameter value, as you play some notes. You'll hear the sound become cleaner as those upper harmonics are reduced. By about -30%, the timbre will have no FM applied to it. So this button, as

the legend implies, can add FM to a timbre that doesn't have any or remove some from one that does. We call its action additive.

Now tap on the **FM** % and crank it up. Since you've dialled down the FM, you'll hear no difference. This is because **FM** % is multiplicative: it acts on the current level of FM on the timbre and multiplies it by its percentage value. If you double tap on **FM** ± to return it to 0% and go back to adjusting **FM** % you will hear its effect.

You may wonder why we need two controls for the same DSP parameter and it is a fair question. FM is one of those spices that can vastly change the flavour of the whole soup. Indeed, the keen listener will tell the difference of 100% and 101% on this timbre. It's also a key element to Synclavier sound design — see Pro Tip box below. Whereas the **FM** ± button should be considered a character changer, the idea of the **FM** % button is to provide a parameter to fine-tune the level of FM on a timbre without recourse to setting partial levels or envelope settings.

FM Ratio describes the relationship of the frequency between the carrier wave and the modulator wave. Again, more on that later in this manual, but just know now that it's another parameter on which small adjustments can make substantial changes to the character of a sound.

FM % **Ratio** is another multiplicative parameter. The FM Ratio defined on Subtle FM [SDT 3-3] partials is 3. So the modulator wave is 3 times the frequency of the carrier wave. If you set the FM Ratio to be 112% you'll get a ratio of 3.36, and the sound takes on a warbly consistency. FM Ratio can also reduce this ratio if set less than 0%.

The FM parameters can be MIDI-linked to CC controls, as described in the previous chapter, for on-hand instant timbre morphing.

> As has been noted above, a little FM goes a long way, and Synclavier gives you precise control over these types of parameters with the Swiper's fine mode (this level of control is hard to find on synths that rely on rotary controls, because they have to employ course and fine adjustments separately). But another thing that characterizes the Synclavier synthesizer (and some would say its users) is its ability to go to extremes: the maximum **FM%** setting produces outrageous distorting fold-over artifacts; **FM Ratio** can go to 16 for screaming intensity, a large filter resonance can induce wild self-oscillation; and so on. Most kinds of synth architectures can't go to these extremes or chose to preclude them because they are not "safe" — most useable sounds being found in the moderate parameter ranges — but experimentation at the extremes can be fruitful and Regen's presets are full of crazy but satisfying sounds that have been harvested from the nether regions.

Tuning

From **FM** you can **Arrow Right** to get to **Tuning**, or just tap the button, which will reveal the following options:

Octave | **Semi-tones** | **Cents**

These 3 parameters work at different resolutions to change the tuning of the timbre. They're additive. The partials (components) of a timbre contain a specified tuning. This will bring all partials up or down by the dialled in octaves, semitones and cents.

There's various reasons that you may want to do this: you might want to turn a lead organ sound into a gnarly bass instrument and drop it an octave; or you may know a piece well, but your vocalist sings in a another key to your usual vocalist, in which case you can minus or plus a semitone or two in much the same way a guitarist would use a capo; or imagine you're playing with an orchestra and the performance hall warms up throughout the night. Many of the instruments will be slightly sharp by the final opus, so you many want to add a few cents to your sound without resorting to changing partial settings.

You'll notice if you jump from page to page, e.g., FM to Tuning, that the selected parameter is remembered on each page. This is a useful feature to reduce finger athletics. A sound designer will often be going between 2 or 3 parameters to hone the sound she desires. We've striven to put related parameters together on the same page, to cut down on finger athletics, but where that's not possible, your prior selection being remembered will cut down on button taps.

Chorus

Chorus is an effect on many synthesizers. It creates a chorus voice, tuned slightly different to the base voice. Sometimes chorus is referred to as detune or unison on analog synths. If you tap on Chorus, you'll see the options:

Smart Chorus | **Hyper Chorus** | **Fine Chorus**

Like **Tuning**, the three options are different resolutions on the same thing. Smart Chorus is what you'll want to use most of the time, as it follows a logarithmic scale, so the amount relates to how us humans will perceive the effect. **Hyper Chorus** runs rampant and **Fine Chorus** is for when you're making your final tweaks.

Call up Additive Percussion [SDT 2-8] timbre to a track. This is a tuned percussion sound, which means that different keyboard keys will trigger different pitches, so the composer can find the right tuning of Clave for their composition. Play a few notes to witness this.

Then dial up the **Smart Chorus** to around 5%. You'll notice the sound has thickened up a bit. Keep increasing it and you find when you get to about 20% it starts to sound like 2 distinct sounds instead of one, and by about 35%, this distinction is obvious.

You'll also see the two vertical note lines, and the chorus voice will stretch out further to the right as you increase the chorus amount. Double tap **Smart Chorus** to return to 0%. You can also reduce chorus to a minus value and the chorus voice stretches out to the left. NB: When using negative chorus you change the fundamental frequency of the timbre, therefore the perceived tuning, in a way that positive chorus does not.

On the Chorus page, the main display provides a Combined Values pane, which shows the chorus as a ratio, semitone difference, and cents difference. You may be used to valuing chorus by one of these method, so all are provided. NB: A chorus ratio of 1 means no chorus, because the chorussed voice would be the same frequency as the base voice.

Unlike the FM controls on this panel, which under-the-hood simply modify the amount of partial FM, **Timbre Chorus** is an additional effect on top of any partial choruses that may be present on the timbre. You can see this by clicking on the **Chorus** button on the OSCILLATOR panel and dialling in some partial chorus on partial 1 (there is only one partial on this clave timbre). When you press a keyboard key, you'll see 4 vertical note bars appear (1 x 2 x 2).

NB: Chorus has no effect on partials with subtractive waveforms.

5.2. Timbre Effects

Many effects can be applied at the timbre level; Reverb, Detune, Grunge, and Stereo Spread, which we will discuss in detail.

There are a lot of features on this button, so when you first press **Effects**, you're presented with a menu page:

Bit Crush | Effects Filter | Timbre Reverb

Tapping an option button will take you to the selected sub-page. Or, if you **Arrow Right** once, you'll get the **Spread, Detune & Octave Ratio** page, and if you **Arrow Right** again, you're presented with the **Timbre Volume** page.

Bit Crush

**Bit Depth** | **Grunge** | **Alias Filter**

- option buttons are presented on the Bit Crush page.

Bit Depth sets the bit depth for each additive partial in a timbre. The default bit depth for Regen, as per earlier Synclaviers, is 12-bits for additive generators and 16-bit for samples. Many wave shapes change dramatically as the bit depth is changed. The **Alias Filter**, when on, will interpolate linearly for every sample point.

> The best way to explain this is with a sine wave. Cut all the tracks, tap **Levels** from the ENVELOPE panel and with **Carrier** Selected, double-tap the **Partial Volume** option button. The default additive sine-wave can now be heard when you play a key. Alternatively, you could load the Sine Wave [SDT 1-1] preset.

> Now go to the **Bit Crush** page. There you will see a scope graphic with a sine wave when you press a keyboard key.

Below is a graphic of one period of a sine wave. The internal DAC of Regen is set to 50kHz, just like Synclavier II. This means, every 20μs, come rain or come shine, Regen needs to output a sample. Time is plotted on the x-axis, and the black vertical lines are 20μs apart.

sine wave 6-bit   time ⟶
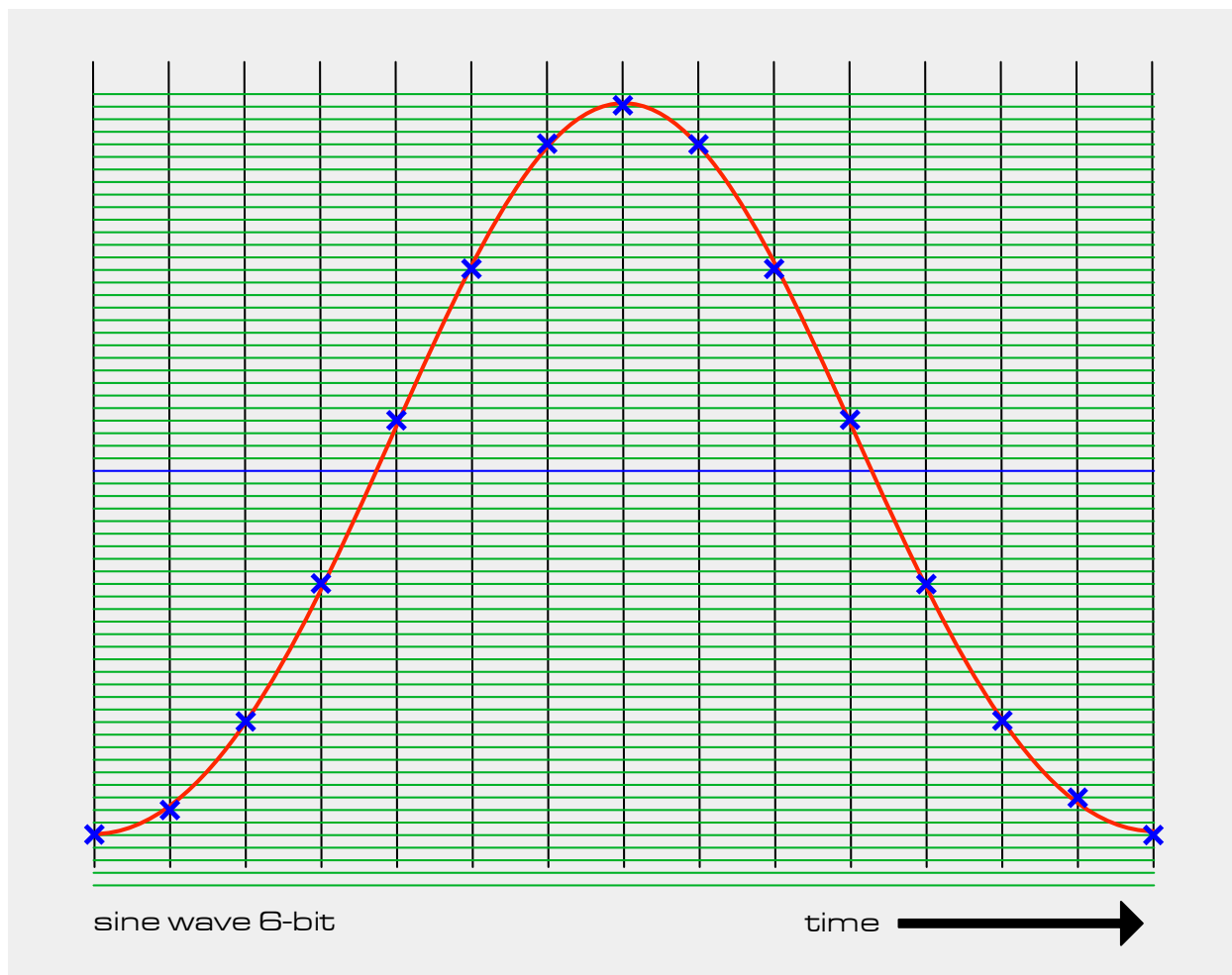
With 6-bit resolution there are $2^6 = 64$ levels. For each sample time there is a blue cross at the nearest level to the sine curve. If you look closely, you'll see some error. The blue crosses are a compromise and so some are under and some are over the red curve at the sample points. If you join up the blue crosses, it won't be an exact sine wave at 6-bits.

> Set the **Bit Depth** to 6-bit and press a key on your MIDI keyboard. You'll see some fuzziness on the scope, which is a fine-grained castellation of the sine wave. This manifests itself as quantization distortion in the sound. A 12-bit facsimile of a sine wave has much less error

and the quantization distortion is inaudible, having been pushed out of the audible range.

If you change **Bit Depth** to 2-bit, you'll see extreme castellation, and the sound is distorted into something unlike the original.

sine wave 2-bit   time ⟶
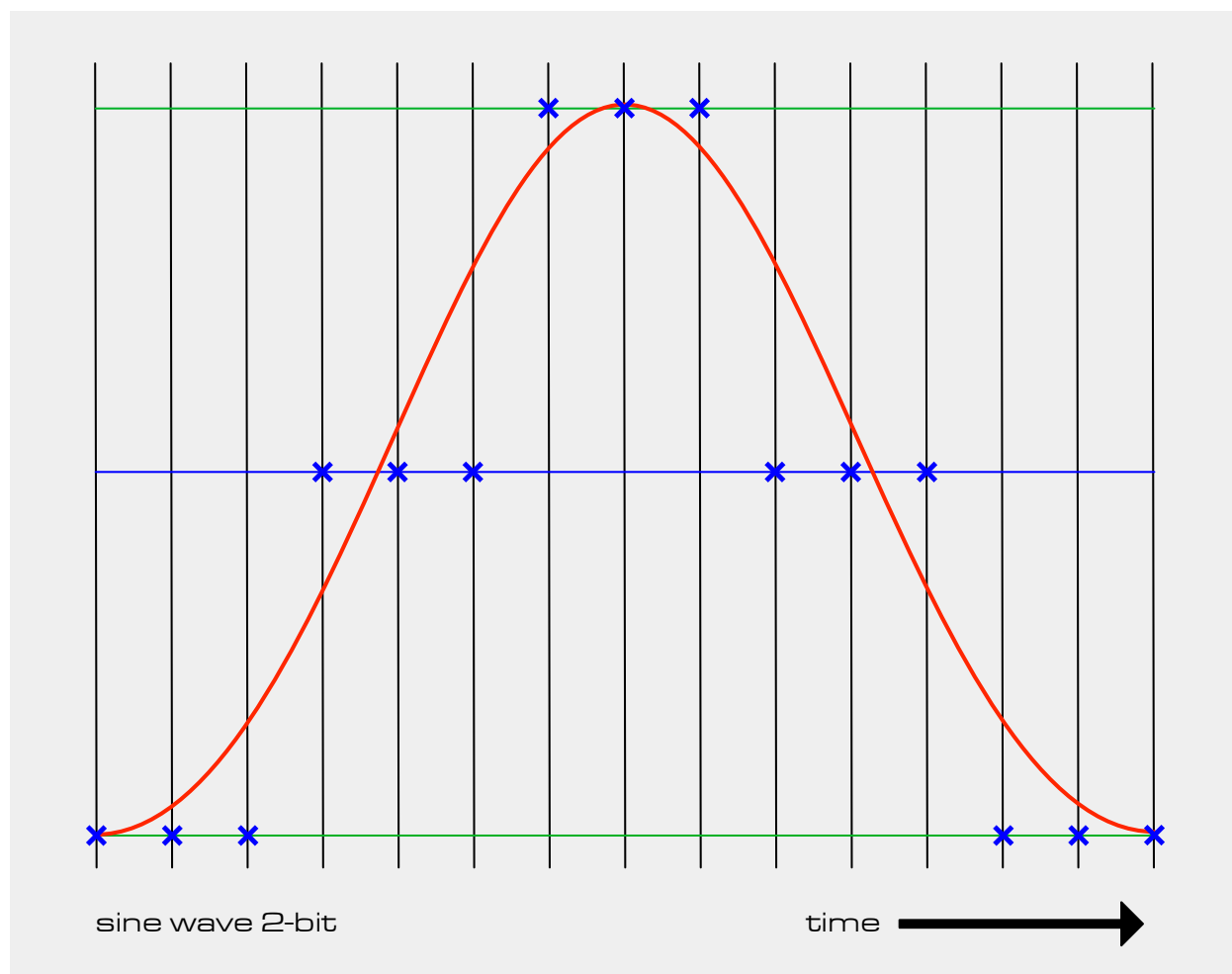
The image above shows a 2-bit representation. 2-bits gives 2^2=4 levels, but since we want one level to correspond to the zero-crossing point (middle) of waveforms, 1 of these levels is unusable. It's not difficult to tell that if you trace the blue dots, you don't get a very good approximation of a sine wave. In fact, triangle and square waves are identical to sine waves at 2-bit resolution.

Regards the frequency domain, you can consider the **Alias Filter** as removing the quantization distortion by running the bit-crushed waveform through a low-pass filter. Regards the time domain, it's actually linearly interpolating the points that don't fit. So in the sine-wave example above at 2-bit resolution, the only points that fit (an exact level for the current bit depth) are the peak and troughs of the wave. The other points, rather than go to the nearest level, are interpolated linearly (along a straight line) — and you'll get a triangle wave. You'll see this effect on the scope graphic, if you turn the **Alias Filter** On and play key on your MIDI keyboard.

For additive waveforms, **Grunge** is a fine level of bit depth. It's unconcerned with the 2-bit architecture of computers and instead sweeps a finite max number of levels linearly, presented as a percentage. The control is linear as opposed to the exponential control of **Bit Depth**. Both can be combined together and will mathematically sum.

For samples, **Grunge** instead decimates the sample, from 1/1 to 1/1024 on a exponential curve, which messes with the sample rate rather than the bit depth of the waveform: similar to the effect of the **Alias Filter** on the sample playback, covered in the Soundfiles section of the Partials & Sound Design chapter, but not limiting the effect to playback below the base key. The error introduced is akin to the stepping of a sample-and-hold DAC, but from Regen's sigma delta DAC!

The **Grunge** control has no effect on partials using the Subtractive waveform generator.

When playing with these controls, on many types of sounds, you'll likely find there's a sweet spot where you welcome some distortion, but not too much.

The three controls on this page have no effect on subtractive waveforms. A system limitation is that the **Bit Depth** and **Grunge** controls cannot be MIDI-linked unlike most timbre parameters.

Effects Filter

New to Synclavier with Regen, a timbre can have filter and reverb settings. Use the following options to configure your filter:

Filter Type | Cut Off | Resonance

You can select **Filter Types** of Low Pass, High Pass, and Band Pass, with 12dB (shallow) or 24db (steep) roll off. You'll see a graphical representation of the frequency response when you select something other than **Off**.

The **Cut Off** point is the frequency at which the filter attenuates the input amplitude by 3dB (amplitude is halved).

Resonance is controllable on the **Effect Filter**, adding an emphasis at the **Cut Off** frequency of the filter. You can MIDI-link the **Cut Off** and **Resonance** parameters to incoming MIDI CC controls and save those settings in your Studio.

Timbre Reverb

The **Timbre Reverb** consists of 3 subpages that can be navigated with the **Right** and **Left** arrows. The options presented are as follows:

Wet/Dry Mix | Choose Reverb | Bypass Master
Wet/Dry Mix | Hall Size | Room Echo
Stereo Image | Pre Delay

The keen-eyed among you will have noticed that **Wet/Dry** is repeated. This is a technique found in a few spots to cut down on finger athletics. As mentioned previously, the Regen designers have endeavoured to place commonly-related controls close to one another, so users aren't going back and forth all around the houses.

People that have used reverb effects will be familiar with the controls, and there is much about these ubiquitous controls online, but here's a quick summary:

Reverb Parameters

Wet/Dry	As with many effects, its the mix of unprocessed vs processed audio that's passed throughto the output.
Hall Size	Sets the emulated size of the space in which the reverberations take place; changing this control affects the delays between the original and reverb artifacts as well as the attack and release times of those reverb artifacts.

Room Echo	Sets the echo factor of the emulated space; changing this control will vary the strength of the reverberation, as if going from a heavily furnished space, to a shiny reflective capsule. This parameter's control has a exponential curve, since a little goes a long way.
Stereo Image	Controls how wide the stereo image can get; imagine listening to a HiFi in a highly reflective room. You will hear the source stereo image and also reflections, and in this space, there's nothing to stop reflections from the left speaker reaching your right ear and vice-versa, so this effect is mitigated or magnified with use of the Stereo Image parameter (and use of this parameter will turn a mono timbre into a stereo sound).
Pre Delay	This controls how much time elapses between the start of the dry sound and start of the wet sound; adding a pre delay makes your brain perceive the reverb as a more pronounced echo effect.

You don't need to dial in the above parameters individually. If you prefer, you can tap **Choose Reverb** and select from a library of Preset reverbs or User reverbs (if you have some saved on your SD card).

NB: The **Timbre Reverb** is an insert effect, not a send effect: each timbre can have a different reverb algorithm, which is applied to the track audio before mixing.

Regen comes with a MASTER **Reverb** that will affect every track, after mixing. Many Synclavier timbres come alive with a modicum of reverb, so it's typical to leave the master reverb algorithm set to a mild or moderate amount as you're browsing and playing timbres — preset Studio A [1-1] has been provided as a kind of "default" for this purpose.

If you want to save reverb into a timbre, but feel its reverb plus the master reverb would be overkill, you can tell the synth engine to route the timbre track audio to bypass the master reverb, by switching **Bypass Master** ON.

Here's a neat trick with timbre reverb that's worth trying. First start with an blank slate (cut all the tracks and set the MASTER reverb mix to 0%.) Then load

Tongue Drum [RW 2-1] to track 1. Play some notes and gradually increase the timbre reverb **Wet/Dry Mix** to 50%. You'll notice, as you approach 50%, the reverb effect becomes less pleasing; it's just a mess of sound, especially when playing lower notes.

Copy track 1 and paste it to track 2. Set the reverb mix of track 2 to 100% and track one to 0%. With both tracks at the same volume level, this is the same as having one track with a mix of 50%, but now you can use the track volume to adjust the level of your dry sound (track 1) and wet sound (track 2) individually.

Now we have done this, we can alter the nature of the wet reverb channel without affecting the dry sound. Select track 2 and go into the **Effects Filter** page. Select a **Filter Type** of Band Pass 12. Play some notes and you'll hear that the reverb effect is no longer overwhelming. Only the middle frequencies from the Tongue Drum timbre are passed to the reverb algorithm.

Using the **Effects Filter** is a great way to tame a reverb for certain timbres that need a more tailored reverb sound. But we can go further: try adding 2% of timbre FM to track 2 and you'll hear the reverb effect brighten up, especially in the sustain and release phases; adding some chorus gives the reverb a fuller dimension; and you can, of course, experiment with changing all kinds of parameters of the timbre on track 2 until you get the type of reverb you want. Then you can save your work as a session, which will save both tracks.

Spread, Detune & Octave Ratio

From the main Effects page you can **Arrow Right** to the Spread, Detune & Octave Ratio page, where you'll get presented with these options:

Stereo Spread | **Timbre Detune** | **Octave Ratio**

These controls aren't related to each other, so it's a bit of a miscellaneous holding page.*ƒ*

Stereo Spread is used to spread the partials across the stereo field. It's a quick way to add some stereo life to a timbre. The same can be achieved (statically) by setting the pan controls of each partial individually, but this doesn't mean **Stereo Spread** is redundant: it can be used as a modulator destination — more on modulators towards the end of the Partials & Sound Design chapter — but, as an example:

> Call up String Ensemble [RW 3-2] and play a chord,
> then move the mod wheel. Since the ModW is routed to
> **Stereo Spread**, you'll hear the stereo image widen as
> you increase the mod wheel position.

Stereo Spread works additively with partial **Pan** and the function takes account of how many partials you have. The lowest active partial is pushed out to the left by the dialled-in magnitude, the highest partial is pushed by the dialled-in magnitude to the right. The partials between are then evenly spaced betwixt the two ends (in terms of pushed magnitude, the absolute placement will also depend on partial **Pan** settings). A negative **Stereo Spread** will reverse this, i.e., lowest partial goes right, highest goes left. It can be tricky to hear what's going on when your partials sound similar, so wear headphones and make use of the soloing of partials to get familiar with how this feature works.

Timbre Detune is a curious control that the Synclavier inventors had the foresight to include on the original to make it *less digital*. It's interesting to note that the first crop of digital synthesizers focused on recreating sounds of orchestral instruments, so we have many Synclavier timbres of pseudo violins, flutes, cellos, tuba, and so on. Since then, a growing trend in digital synthesis is to model analog synthesizers.

One of the ways analog differs is that the tolerance of the components constituent in the sound-producing circuits affects the pitch and timbre of the sound. If I strike middle C on a certain preset of my digital synth, by and large, I'm going to get the same exact sound every time. With an analog synth, this isn't necessarily true: as electronic components, such as transistors, warm up, their behaviour changes, adding subtle nuances to the sound. Indeed some famous analog synthesizers and drum machines are well known for having variations from unit to unit. And what's more, on some architectures, different keys can be off-pitch by different amounts.

Musicians have been known to sample their analog gear note-by-note into their digital workstation keyboards to capture the uniqueness of those inconsistencies. The digital workstation can produce a sawtooth better than the analog device, but they'd much rather have the raw imperfection.

The Synclavier designers knew this, so incorporated a very small detune to the original that added a tiny random pitch offset to each keyboard key. Too much, and it would sound bad, too little and it would sound lifeless (too "digital"). The original level of "detune" is the default Synclavier II setting. But with Regen, we've also enabled you to increase this effect from +1 to +10, and also switch it off, if desired when you *want* something to sound "digital".

By definition, it's a subtle effect to hear, most noticeable (and useful) when playing chords.

Octave Ratio is another interesting control. It sets the relationship between the MIDI key input and the sounding pitch of a timbre's partials. For the vast majority of timbres, this is left at the default setting of 1, which translates to a doubling of pitch for every keyboard octave. If you wanted a keyboard octave to have 2 octaves of pitch range, you'd set this to 2; half the pitch range, set it to 0.5. NB: This stretching or compressing is centred around A4, which always stays as A4 (440Hz).

This might seem like a strange thing to do, but if you're designing sound effects or tuned percussion instruments, this might be exactly what you want. An example with an **Octave Ratio** less than 1 is Raindrops [RW 7-7]. To maintain the

effect of raindrops falling, the ordinary pitch range has been limited by reducing the ratio to approximately 0.3. An **Octave Ratio** of 0 would lock it completely, but having a small fraction enables some pitch variation, thus simulates varying raindrop intensities with different keys.

Timbre Volume

Simply put, the Timbre Volume is saved within a timbre, whereas the track volume (covered earlier) is saved within the session. Most of the time, you won't need to concern yourself with this setting. The vast majority of timbres have **Timbre Volume** set to the default, -10dB. A timbre cannot be heard unless its assigned to a track, so modifying the track level is how you would manage a timbre's volume. Also, the amplitude of a timbre is the summation of the amplitude of the constituent partials (and soundfiles if present), so the overall volume can be managed without recourse to **Timbre Volume**. So when might you need this control?

Certain types of timbres such as complex resynthesized sounds appear quieter than the average timbre. Similarly timbres containing many simple partials appear louder than the average timbre. In these cases, you might wish to apply a timbre volume to balance the perceived volume. Another reason for modifying the timbre volume would be if you're playing a timbre designed for a velocity sensitive keyboard, but you're playing a keyboard that doesn't support that keyboard. Those keyboards typically send a fixed velocity for every key strike, and it may *throw off* a timbre designed for a velocity curve mapped to volume, so rather than remove the velocity relationship, you can easily adjust the **Timbre Volume**.

So, **Timbre Volume** is provided for convenience, and there are a few times where it will be useful.

The **Partials Levelling** setting, when On, automatically reduces the timbre volume based on the number of active partials (sometimes referred to as normalizing). So if you add a partial it doesn't get *louder*. That behaviour is very useful if there is overlap in the frequency content of each partial. However, if the

partials don't sound together – like a drum kit on mutually-exclusive keys, or partials that are gated by a crossfade envelope, it's easier to design the sound with **Auto Partials** Off. If, while developing a complex timbre, it is sounding quieter than other timbres in your library, you may have to switch this setting Off. If you are getting crowded with sound as you add and manipulate partials, you can try switching this On.

TIMBRE Summary

So, in summary, the TIMBRE panel offers controls that have the potential to wildly affect the sound of a preset, and its parameters can be MIDI-linked to CC controls, so you can effectively have a row of shape-shifting sliders/knobs on your MIDI controller as part of your sound creation arsenal.

If you've mastered these controls, and yearn to go deeper, it's time to check out some partial-level stuff.

5.3. Note FX

Arp

Regen contains the classic Synclavier arpeggiator feature. The two **Arpeggiator** buttons are related, as evidenced by the little horizontal bar between the buttons. Under the left button, you have the options:

Repeat | Arpeggiate | Rate

and under the second, you have:

Pattern | Arp Spread | Note Chance

When **Repeat** is On, a note or chord is repeated at the selected rate as long as you hold down the key(s).

When **Arpeggiate** is On, a chord is arpeggiated once at the selected rate.

When the **Repeat** and **Arpeggiate** functions are both On, a chord is arpeggiated and repeated as long as the keys are held.

The **Rate** (sometimes called tempo) at which notes or chords are repeated, can range from 0.010hz to 50hz. Faster rates can change the character of the sound, giving your timbre a futuristic effect.

Try it out: select a timbre with uniform sound, such as an organ, press the **Repeat** button to turn on this function, and when you press and hold a key on the keyboard, it will sound repeatedly. Try changing the **Rate** and try the **Arpeggiate** function when playing a chord.

Tap **Pattern** and swipe or nudge to select a pattern from Performance, Ramp Up, Ramp Down, Up-Down, Up-Down no-repeat, and Random, and note the difference.

Arp Spread is used to spread successive notes across the stereo field, from left to right with a positive value, or right to left with a negative value.

Note Chance defaults to 100% so that all the notes in the arpeggiated chord sound all the time. You can alter the chance that individual notes will sound by reducing this value, and add random variation to your arpeggiations.

A good example of a timbre designed with arpeggiation in mind is Candy Apples [LFS 2-7]. Play one note and its rather pedestrian, play a chord and it comes alive.

NB: For timbres with long release times, when using repeat, it's likely you'll want to set Poly Retrig mode in preference of Polyphonic to save on voices. More on the various modes next.

Mode

Under the mode button, you'll find the Portamento and polyphony mode features:

Portamento | Port Rate | Poly Mode

Portamento makes the pitch "glide" between the notes you play, rather than changing the pitch instantly as you hit another key. Using **Portamento**, you can more closely simulate playing an instrument such as a trombone or slide guitar, or this might just be the effect you're after on a wholly synthesized timbre.

The **Portamento Rate** sets the speed with which the sound glides from one pitch to another in milliseconds. This setting might require some trial and error to get to the effect you desire.

Tap the **Portamento** option and nudge to select it. You can chose between Linear and Logarithmic: Linear changes the pitch from one note to the next linearly; Logarithmic follows an s-curve, so the initial pitch shift is fast, then it slows down during the bulk of the transition, and finally speeds up to land on the note. This transition profile is closer to how some acoustic instruments exhibit portamento — picture a trombone player moving the slide, rapidly at first and then smoothly and finally an abrupt stop.

Here's a table describing the polyphony modes supported by Regen. There's some inter-related behaviour between polyphony mode and soundfile loop mode, so keep this table in mind when we discuss soundfile modes later in the manual.

	Mode
Polyphonic	Provides for full polyphony. When a note on event is received (either from a keyboard or sequencer), a new voice is activated for the desired sound in all cases. Earlier notes of the same pitch that are in the final decay state are not affected; their decay continues without interruption. With very long final decay settings this can lead to undesirable phase cancellations or, in extreme cases, overloading the audio output.
Poly Retrig	Allows for full polyphony, however if a note on event is received that matches a note that is currently in the final decay state, the envelope generators for the matching note are re-triggered (the decaying note is swiftly muted with a 2ms ramp down, followed 3ms later with the start of the new note). The Poly Retrig mode is very natural for sounds with a long final decay such as bells or gongs. E.g. Regen Vibes [RW 3-5].

Mono Retrig	Same as Poly Retrig except that any note on will cause the retrigger, not just a matching key number, and of course the retrigger takes place at the new pitch. For a timbre with a lot going on, this can be a good choice, for example the famous Galactic Cymbal [RW 1-7]. NB: With a patchlist, the retrigger might be a different soundfile of course; or no soundfile, if there is nothing in range for that key.
Monophonic	Same as Mono Retrig, but with no retriggering of the attack part of the volume Envelope. Together with some portamento, this emulates early analogue synthesizers that had one voice and no volume envelope.

NB: The **Poly Mode** is used in conjunction with **Portamento** to get the desired fingering style.

5.4. Note Filter

There's a lot of power hidden under the **Filter** button on the NOTE FX panel. This note filter is new to Synclavier for the Regen product. Like the Effects Filter it pertains to the timbre you're currently working on and is saved with it. Unlike the Effects Filter, which applies to all sounds on the track, it applies to each note individually and contains a re-triggerable envelope.

Regen supports 1 Roll Off filter per partial, 1 Effect Filter per track, and 1 Note Filter per note. That can add up to a lot of active filters. We have made the filter algorithm as efficient as we can to *make the theoretical possible*, but there's a limit. If you're using a tonne of filters, it's always wise to check the CPU meter found on the MASTER **Volume** page. If, while playing some chords, you go into the red line, there's a possibility of Regen dropping notes. Lots of filters are certainly possible, and lots of Reverbs too, but what we've found is a combination of many filters and reverbs is what can crush the CPU.

On the Note Filter page, you have the following options:

Filter Type | **Filter Setup** | **Filter Env**

As with the Effects Filter, you can select **Filter Types** of Low Pass, High Pass, and Band Pass, with 12dB (shallow) or 24db (steep) roll off

Filter Setup and **Filter ADR** will take you to subpages. The Filter Setup page has options:

Pitch Track | **Cut Off** | **Resonance**

As described previously, the **Cut Off** point is the frequency at which the filter attenuates the input amplitude by 3dB (amplitude is halved).

And the **Resonance** adds an emphasis at the **Cut Off** frequency of the filter. As with FM Amount, a little goes a long way. Check out Starry Eyes [AM 3-2] to see that a filter with 4% resonance is enough to add the right punch to saw wave to make a useful popular music timbre.

Pitch Track moves the **Cut Off** frequency depending on what MIDI key is received. Recall that each note has its own individual filter.

Pitch Track can be turned Off by setting the value to zero.

With **Pitch Track** On, the **centre key**, nominally Middle C, is the only MIDI key that won't change the **Cut Off** from the dialled-in value

If **Pitch Track** is set to 1, then an octave delta of played key (from middle C) will cause an octave shift in **Cut Off** frequency in the same direction; 0.5 will cause half an octave shift; 2.0 will cause a shift of two octaves **Cut Off** for every octave delta of MIDI key; -1.0 will cause an an equal shift of **Cut Off** in inverse direction to the key played (from Middle C); and hopefully by now you get the idea, but if you need the concept cemented, the graphic on this page should help.

The graphic shows a line emanating from the **centre key** of the keyboard graphic to the dialled in **Cut Off** point of the drawn frequency response curve of the filter. Another line is plotted from a key an octave lower and a further line from an octave higher; those lines leading to the **Cut Off** points of curves plotted on the same x-axis to show the filter response if those keys were played.

With the **Pitch Track** option selected the **Swiper** will adjust its value. If you tap **Pitch Track** a second time, the **centre key** will show green and that value can be adjusted with the **Swiper**. While green, the parameter is armed to listen to incoming MIDI, so striking a keyboard key will also set the **centre key**. Double-

tapping the **Pitch Track** option button while the centre key is being adjusted will return it to Middle C; whereas double-tapping Pitch Track while adjusting Pitch Track value will alternatively set it to one of two defaults, 0.0 or 1.0.

NB: The Filter Envelope feature is discussed in the following chapter, in the Envelopes section.

NOTE FX Summary

Where the features of the TIMBRE panel add expression to the sound, the features of the NOTE FX panel add expression to your playing, and therefore the sound. And then there's the **Note Filter** at the end of the chain, affecting the whole timbre.

There's a symbiotic relationship between NOTE FX and the TIMBRE panels. All the parameters on these buttons are saved and recalled with the timbre.

PARTIALS & SOUND DESIGN

6.1. Taking the Red Pill

Partials have always been essential to Synclavier sound design. Indeed, a patent was filed in 1984 on the "Partial Timbre Sound Synthesis Method and Instrument" by the inventors.

When designing a timbre, very complex and dynamic sounds can be created by choosing different sounds for two or more partials. For example, to strengthen the attack of a brass instrument, you might put a percussive sound on one of the partials. Each of these partial can be modified in a variety of ways, including changing the volume envelope, adding vibrato, or real-time effects.

And as we've learned, the whole timbre with its multiple layers can be further modified by adding effects such as chorus or vibrato. Depending on whether you're working in music, sound effects, or vocal effects, timbre design can be used in many ways.

Working with Partials

Tapping the **Switcher** button will change the view. A timbre can have any number of partials from 1 to 12. When you press the **Switcher** button to change to PARTIALS view it will turn red and the 12 selector buttons will display the

partials for the last-selected track. Also, pressing one of the buttons on the partial panels (OSCILLATOR, LFO, or ENVELOPE) will switch to PARTIALS view.

An active partial is one with a non-zero volume and will be lit red. (NB: If a partial's LFO is being used as a modulator source, but the partial itself makes no sound, it will be lit white.) To change a setting on a partial (or several partials at once), you would tap or swipe the partial or partials and they would become selected partials, showing magenta.

As per tracks, all currently selected partials will change their values when you adjust a chosen parameter. And in solo mode, all solo'd partials will change. The way this works, is that the last selected partial* can increase or decrease to its extents, but it drags the other selected partials along with it — each other selected track will increase or decrease until either, they hit their extent or the selected track hits its extent. The ENVELOPE **Levels** Volume page is a good page to experiment with, because observation of the VU meters will demonstrate how it works.

When working in PARTIALS view, don't forget the cyan blob in the top-left of the main display, which shows the currently selected track. There will also be a light magenta blob showing the *last selected partial.

> Load timbre Combining Sounds #1 [SDT 5-5] to a track. Play a few notes to get a feel for the character of this timbre. It has a marimba come xylophone likeness of the kind we could imagine being played in Tibetan monasteries. FYI: If you have a velocity sensitive keyboard, you'll notice this timbre has been designed so that medium velocity strikes ring out, but harder strikes mute, just as if you were using a mallet to play the instrument and the harder strikes ended in the mallet being left on the playing surface. This effectis achieved by having a long **Release** time and zero **Sustain** volume.

In PARTIALS view, press the **Solo** button. One or more partials will show green to show that they are soloed. Pressing **Selector** buttons now will toggle the partial's solo state. Solo just partial 1 and strike a key softly and you'll hear a tone that's close to what the designer wanted, the *proto* xylophone example, if you will. There's a lot of body to the tone but not much character. Now solo partial 2 and you'll hear a more percussive sound that emulates the strike on the playing surface — notice this partial doesn't vary with velocity. Finally solo partial 3 and you'll hear a fuller sound that rings out. This is the bell like sound related to many of the *keyboard percussion* instruments, and achieved with applying an FM effect (more on that soon).

So in the case of these 3 partials, we can see that the whole is greater than the sum of the parts; and we have 12 partials to play with! Let's dissect one more timbre.

Load Combining Sounds #2 [SDT 5-6] to a track, solo both partials, one at at time, and listen to them. Partial 1 has a very slow Tremolo that undulates between left and right of centre, but other than that, the two partials sound the same: a basic tone sound. Then tap **Solo** to come out of solo mode and listen to both partials together: now it sounds like an organ with a distinct Leslie effect. This is because of a phasing effect between the two partials, caused by partial 2 being tuned slightly sharp of partial 1. With partial 2 selected, tap **Wave** and you will see the tuning on the main display, showing

441.4 Hz. Partial 1 is the default middle A tuning of 440 Hz.

 Many Hammond organs from 1936 onwards had two independent tone-wheels with one set slightly out of tune to the other to add richness to the sound, so this timbre is an example of modelling a real-world instrument to simulate it, using partial synthesis.

Partial Recall

Another useful feature adopted from the original hardware is Partial Recall. Its use was covered in the Library chapter, but here's a quick reminder:

Normal load of a timbre recalls all partials of that timbre to the currently selected timbre, obliterating anything in its path. Partial Recall allows you to select certain partials to be recalled to the current timbre, leaving the unselected partial/s intact. This is a great fun way to quickly mix and match partials to create a new sound.

When browsing a timbre library, if you tap the **Choose Partials** option, you'll see the selector buttons that represent active partials in the candidate timbre blinking. On the off blink cycle, currently active partials will remain solid. Simply select the partials you want to recall from the candidate timbre and those selector buttons stay lit longer. Then press **Enter** to recall those partials.

Partial Clipboard

To copy a partial, you can press the **Copy** button, and the VK display will indicate that the currently selected partial/s were copied to the clipboard. Press the destination selector button/s, then **Paste** to complete the copy. If the destination has a partial or partials, they will be overwritten.

The cut button will copy the partial to the clipboard, and erase the original. A useful function for organizing partials, just remember to save your timbre when you're done.

To erase a partial, typically it's enough to set the **Partial Volume** level to zero, but you can use cut to clear everything (and put it in the clipboard).

6.2. Additive Synthesis

There are many different synthesis techniques employed by modern synthesizers. The Synclavier I was designed to use the additive technique, and the Synclavier is still cited today as a prime example of this technique.

The combination of additive synthesis with FM modulation (to add inharmonic components) and the "partial timbre method," is the bedrock of Synclavier sound design, and partly responsible for its signature sound.

Additive synthesis begins with a simple sine wave. Other sine waves of different frequencies, volumes, and phases, are combined with the original to create a complex steady-state waveform. Each sine wave can be modified individually. You can create complex periodic waveforms by simply "adding" sine waves.
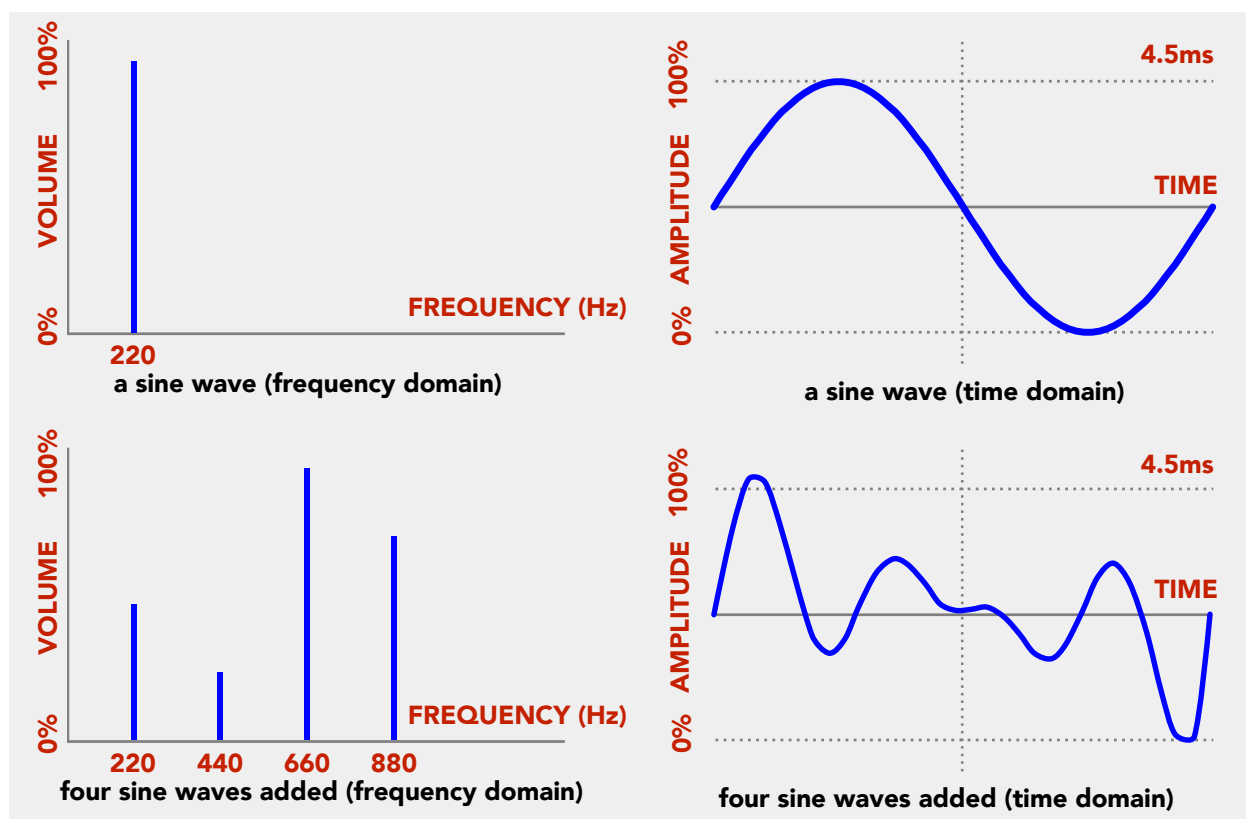
NB: There are many technical articles on Additive Synthesis and comparisons to other methods of sound synthesis, so this is not the place to describe everything going on under the hood. This section will focus on how to design a sound using the Synclavier Regen interface. If you want to learn more theory than is presented here, search for "additive synthesis" and "harmonic theory" on your favourite search engine.

Sine waves

The fundamental unit of any synthesized sound is the sine wave, a periodic waveform that can be represented graphically as amplitude changes over time.

Complex waveforms are created when sine waves are superimposed on each other. Over 200 years ago, Joseph Fourier, a brilliant French mathematician, observed that all complex periodic waves are in fact combinations of sine waves with different frequencies, amplitudes and phases. The sine waves of a complex

waveform can be represented on a frequency spectrum, which shows the relative volume of each harmonic.

a sine wave (frequency domain)

a sine wave (time domain)

four sine waves added (frequency domain)

four sine waves added (time domain)

In a steady-state waveform, the 1st harmonic is the fundamental frequency of the waveform. The 2nd harmonic is a sine wave whose frequency is twice the fundamental frequency; the 3rd harmonic is three times the fundamental frequency, and so on.

In musical interval terms, the 2nd harmonic is an octave higher than the fundamental. The 3rd harmonic is a perfect fifth higher than the 2nd, the 4th harmonic is a perfect fourth higher than the 3rd (and 2 octaves higher than the fundamental), and so on.

Note that, for equal temperament tuning, some intervals are nudged slightly for convenience (lots of information on the various compromises of tuning systems online if you want to venture down that rabbit hole), but in the case of the harmonics of additive synthesis, the intervals are exact. So, for example, if you play a well-tempered chord, the 5th of the chord is out of tune with the 3rd harmonic of the fundamental.

Harmonics

Okay, let's put this in the context of sound design on Synclavier Regen. Harmonic coefficients determine the relative volume of each harmonic. A coefficient of 0.0% means the harmonic is not present in the waveform. A coefficient of 100.0% means the harmonic is present at full magnitude. Harmonic phases determine the relative times at which the sine waves reach their peaks and troughs.

The analysis in the previous section was a bit left-side-brain. What you'll typically do, in practise, is alter each harmonic slowly until you get a sound you like. So let's explore how:

Select and **Cut** all your tracks and then partials, so we start with a blank slate. Then select partial 1. Tap **Levels** on the ENVELOPE panel and you'll see:

Partial Volume | **Partial Pan**

Double-tap the **Partial Volume** option to set the default of 0dB.

Tap **Wave** on the OSCILLATOR panel. Tap **Carrier** on this panel, if not already selected. You'll see these options:

Synth Mode | Semitones | Cents

Confirm that the default Additive **Synth Mode** is highlighted. Tap the **Define** button (or **Arrow Right** twice to get there). You'll notice that the display shows the coefficients of all 24 harmonics. Harmonic 1 should be set to 100% and the others to 0%. Only the fundamental harmonic will be present in the sound, so this is a sine wave. Play a key and you'll hear the sine wave in all its purest beauty.

Tap **Edit 1-8** and you'll see the **Selector** buttons 1 through 8 blink. Swipe selectors 2 to 4 to select them (they show magenta when selected). Then use the **Swiper** to add a value to those harmonics. On the graphical display, you'll see vertical bars appear next to the fundamental. Play the sound again and you'll notice it's *timbre* has changed. In fact, it only takes 1% to notice that the sound is more complex than a sine wave. Sound design is a fine art.

Settle on around 12% for harmonics 2, 3, and 4, then **Arrow Right** to the next screen where you can adjust phases. Again tap **Edit 1-8** and increase the phases of these harmonics. You'll also notice this is a subtle effect. Certain precise phase imbalances between the fundamental and other harmonics are sweet spots for dissonance and will be pleasing or not depending on your goals for the timbre design.

Don't forget that you can see a visualization of the waveforms as you're designing them by pressing **Volume** on the MASTER panel. The waveform from the last struck keyboard key will be shown in all its time-domain glory. And you can **Arrow Right** from there to also view the frequency content on the multiscope page.

You can create evermore complex waveforms by adding further harmonics, and tweaking the coefficients and phases, but the fun has just begun, because you can combine many of these waveforms (partials) together and apply effects to partials, such as FM modulation, tremolo, vibrato, chorus. And, with Regen, you can use synthesis techniques other than additive to generate your waveforms.

Bear in mind, sounds produced by acoustic musical instruments are typically very complex; each instrument has its own harmonic structure, which can vary from note to note, and depending on volume (e.g. a key struck harder on a piano will induce more resonances in the strings and sound box). Synclavier Regen gives you the tools to make some very complex harmonic waveforms (and to vary their behaviour over the keyboard range).

Pro Tip: From the **Edit** button, there is a **Quick Wave** function. If you tap this and tap **Nudge Up** to select a desired waveform, the harmonic coefficients and phases of the selected partial will be set for you. You can then observe and modify the coefficients and phases individually. So the quick waves can be used as building blocks or a jumping-off point for your own crazy waveforms.

FM Modulation

Thus far, we have been modifying the carrier waveform. Things can get really interesting (or weird) when you modulate the carrier wave with another wave.

Tap **Mod** on the OSCILLATOR panel, and you'll see the coefficients and phases of the modulation wave, which can be set in the same way; by **Quick Wave** or manipulation of individual harmonics: Synclavier

Regen lets you design complex harmonic behaviour into both carrier and modulator waveforms.

The modulation wave defaults to off (all the harmonic coefficients are zero). Select **Ramp** for the modulator waveform. You will not yet hear any difference in the sound, because you have to set a level for the waveform just like we did by setting the partial volume.

Tap **Mod** on the ENVELOPE panel, then tap **Levels** and you'll see the options:

Partial FM % | FM Ratio | (Fine)

Notice that we have **FM** % in the first position whereas the carrier waveform presents **Volume**. It's not really important what units are used for the magnitude of the waveform, but specifying a % reminds us that the modulator waveform is applied to the carrier waveform.

You'll remember from our discussion on the TIMBRE features, that a little goes a long way when it comes to FM. Tap **Partial FM** %, sustain a keyboard key, and slowly swipe up and you'll not only hear the sound change dramatically, but you'll see the waveform shape change on the graphical display.

The frequency of each harmonic in the carrier waveform is being modified in real-time by the changing amplitude of the modulator waveform. In

practice, what Synclavier Regen is doing under the hood to achieve this effect is adding "sideband" components to each harmonic.

When you get a level of FM you like, go back to adjusting the modulator coefficients, and you'll see that changing the modulator waveform affects the character of the modulation. Finally, modify the **FM Ratio** to suit, from the ENVELOPE **Levels** page. This is the ratio of $\dfrac{\text{frequency of modulator}}{\text{frequency of carrier}}$.

So the modulator waveform's frequency isn't defined directly, rather, its ratio to the carrier frequency. The default is 1 which means the modulator has the same frequency as the carrier. This has a particular characteristic. In fact, every whole number ratio imbues a distinctive *character* to the resultant waveform.

To help understand how **FM Ratio** acts, it may help to visualize the resultant waveform. For just 1 partial, set the carrier and modulator waveforms to Sine using the **Quick Wave** function (or call up Sine [SDT 1-1]). Then return to the **Mod**ulator **Levels** page. Set the **Partial FM%** to 10%.

While sustaining a keyboard key, gradually dial up the FM Ratio from 1.0 until 2.0. As you do, the modulator increases in frequency, so it undulates the carrier more times in a cycle. This will be evident on the waveform in the VK Display. At 2, it will be undulating the carrier exactly 2 times in a cycle. Since the carrier returns to zero at the same time the modulator does (albeit after

two excursions) you'll see the waveform stays static, and the same is true of every whole number ratio.

Real numbers, in between the whole numbers, are *phasier*. Halfways are interesting, i.e., 1.5, 2.5, 3.5, 4.5. So too are fractional values, wherein the modulator is lower in frequency than the carrier, as they add a different flavour. And don't forget to use the (Fine) control to reach some lesser-used ratios. There are so many interesting possibilities. Plus, with Regen, you can dial down the tuning of the carrier to a subsonic frequency, and then when you apply FM, you're just hearing the FM sidebands in the audible range, as in the case of FM Gallop [PSA 7-3].

The sheer breadth of character that can be had by judicious use of FM is staggering, or maybe not, if you consider that a whole generation of synthesizers were built on this technology since its development by John Chowning, and swathes of genres of music as well as film and TV soundtracks are reliant on *FM* sounds.

Lots of factory preset timbres employ frequency modulation, and you can learn much by studying the parameters set on these timbres. Take a look at Bank 3 of Sound Design Tutorial library as a starting point, but there are dozens of factory timbres you could draw inspiration from.

It should be mentioned that although you can create complex additive waveforms for use as the modulator, often times, a simple waveform will suffice. Complex modulator waves leads to very complex output waveforms and that comes with aliasing distortion. Maybe you want something filthy and distorting, but if not, when using a complex modulator, you'll probably want to also use a LP filter on the timbre to tame some of the top-end.

The following diagram shows the signal flow of a partial through Regen's synth engine and should help to clarify how this key element of Synclavier sound design operates:
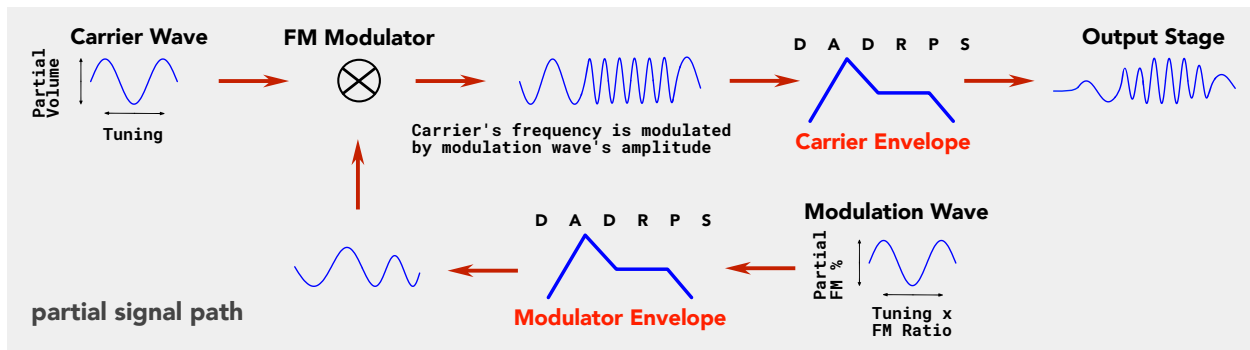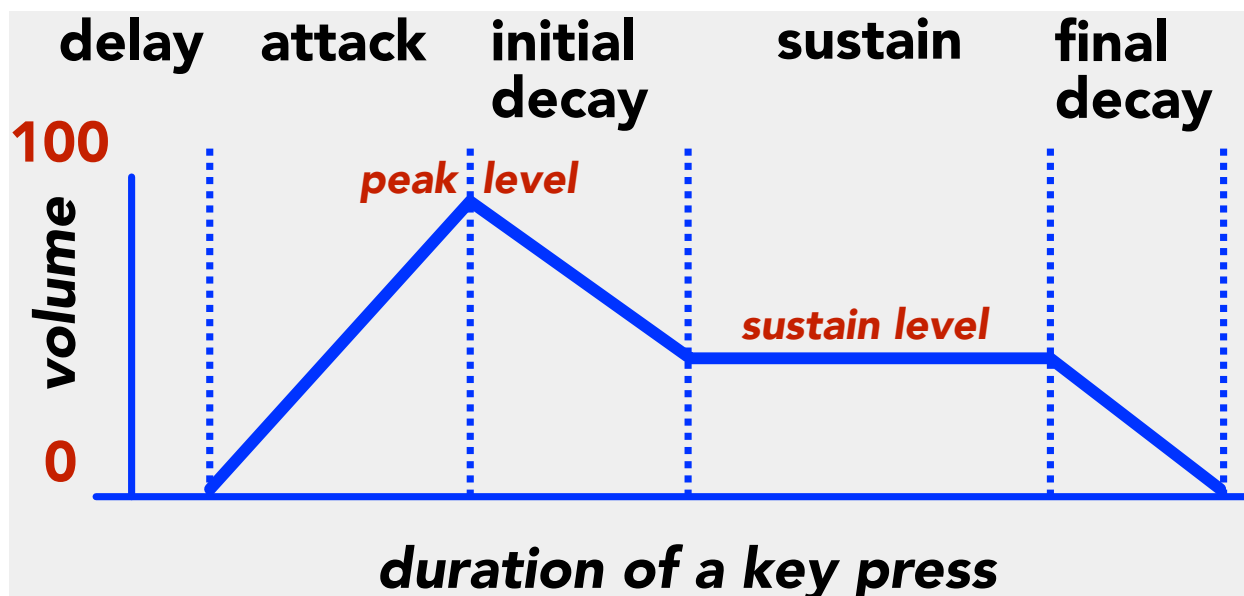

6.3. Envelopes

It's time to cover envelopes – a building block so fundamental to sound design, they're found on almost every synthesizer. There's lots of background theory on the internet on this subject, so we will just summarize it here.

An envelope dictates how the partial will behave over time. In the case of the carrier volume envelope (sometimes simply called volume envelope, sometimes simply called carrier envelope), the envelope is applied to the carrier's amplitude. So, as can be seen in the diagram below, attack stage governs how quickly or slowly a sound builds. Percussion sounds will have a near-zero attack whereas string sounds may have a second or two of attack time.

Synclavier also supports an initial decay stage and sustain stage, with settable levels. Sustain occurs when you hold your finger on a keyboard key or use a sustain pedal. The final decay or release stage is prompted by the release of that finger. Regen also supports the shape of the decay. In the diagram below, a linear shape is shown, but you can also set curves decays which affect the character of the sound.

The diagram shows a volume envelope with labeled regions: **delay**, **attack**, **initial decay**, **sustain**, and **final decay**. The vertical axis is labeled **volume** ranging from **0** to **100**. Key points labeled: **peak level**, **sustain level**. The horizontal axis is labeled *duration of a key press*.

Carrier Envelope

Below, the adjustable parameters of the volume envelope are listed. They're found on the ENVELOPE panel's two **ADSR** buttons, as well as the Delay & Decay page "hidden" between **Levels** and **ADSR**. You can navigate to this page using the **Left & Right Arrow** buttons, or you can tap **Levels** and the left **ADSR** button (**ADSR 1**) simultaneously and you'll go directly there. Both **Levels** and **ADSR 1** will light brightly when this page is shown.

As you adjust these parameters, you'll see a helpful graphical representation of the envelope, much like the diagram above, change.

Volume Envelope Parameters

Vol Env Delay	Time from key depression to beginning of sound. If you want the sound to start after the key is struck, swipe a time above zero. Using different delays on multiple partials to offset them means the sound evolves as the different partials come in and out.
Decay Adjust	Lengthens the decay times (initial and final) as you play down the keyboard below middle C. The units are percentage of decay per octave. Applying some additional Decay Adjust to #chime and #percussion timbres can make the results more natural sounding. Applies to all decays (initial and final) for Volume and FM envelopes.

Decay Shape	The original shape of the Synclavier II decay was an approximate parabolic curve, since it was efficient to compute on the hardware of the time. This is one of the many small characteristics of the original that contributed to its signature sound and is available for selection on Regen.
	Also available, are logarithmic curves, that are the predominant decay shape used in modern synthesizers, and are useful in creating certain types of sounds, such as the plinky sounds we associate with modular synthesis. Various slopes are offered from -10dB to -80db. A linear slope is also provided, useful for certain applications, such as percussive or non-musical sounds.
	Applies to all decays (initial and final) for Volume and FM envelopes.
Vol Env Attack	Time from beginning of sound to peak volume. Swipe to select an interval between 0-15,000 milliseconds.
Vol Env Decay	Time from peak volume to beginning of sustain level. Swipe to enter a time interval between 0-30,000 milliseconds. Also sometimes called initial decay.
Vol Env Release	Time from key release to end of sound. Swipe to enter a time interval between 0-30,000 milliseconds. NB: Release, when it occurs, will replace whatever stage the note is currently in, be it attack, initial decay, or sustain. Also sometimes called final decay.
Vol Env Peak	Relative volume at the change from attack to decay. Tap this button and swipe to set a level between 0-100.0%. Zero is no volume and 100.0% is maximum level.
Vol Env Sustain	Relative volume from end of initial decay to key release. Tap this button and swipe to set a volume level between 0-100.0%. Zero is no volume and 100.0 is maximum level.

For a demonstration of how dramatically the carrier envelope can change the sound, try this: load Multiple String Percussive [SDT 5-4] to a track and play a few

notes. Select all the active partials and tap **Carrier,** then **ADSR 1**. Select the **Vol Env Attack** option and swipe up to around 4000ms. You'll see the graphical representation of the selected partial's envelopes change and rather than individual envelopes, they blend into one another. Now if you play a note or chord, you'll hear that you've turned a plucky string instrument into a pad. You can increase the **Vol Env Release** time and set the **Vol Env Sustain** to >90% to complete the transformation.

Modulator Envelope

As you might discern from the diagram in the previous section, the modulator envelope – sometimes called the harmonic envelope, since it adds harmonic complexity – dictates when and how much FM modulation is applied to the carrier waveform per partial. There's no point repeating ourselves, but if we listed the table for volume envelope again, for the modulator wave, the **Decay Shape** would be absent, because it is one parameter that controls all decays. Also, the **Decay Adjust** would be absent for the same reason. In its place is the **FM Adjust**. This parameter, if set to something other than 100%, increases (or decreases) the FM amount as you play up and down the keyboard from Middle C. It's a simple mechanism to colour the sound depending on what notes you're playing.

For a demonstration of how dramatically the modulator envelope can change the sound, try this: load Rock Keyboard [RW 2-7] to a track and play a few notes. Select all the active partials and tap **Mod** then **ADSR 1**. Select the **FM Env Release** option and swipe down to 0ms. Also tap **FM Env Decay** and reduce this to around 12ms. Now if you play a note or chord, you'll hear that you've quickly turned a grungy lead sound into an

ethereal accompaniment instrument. The satisfying ringing sound of the original is more pronounced in the decay stage because there's no FM on the partials. If you really want this to ring out, you could further modify this timbre by increasing the **Decay Adjust** (under the **Carrier** mode). So, not what the sound designer of Rock Keyboard had intended, but something distinct and with a different application.

Filter Envelope

Having introduced the concept of envelopes and examined the volume and harmonic envelope features, we now return to the Note Filter and examine its envelope feature.

From the Note Filter page, tapping **Filter Env** will take you to the Filter Envelope page, where you have these options:

Env Attack | **Env Decay** | **Env Release**

These option buttons take you to further subpages. **Env Attack** has the following options:

Start Level | **Attack Time** | **Peak Level**

The **Start Level** sets the starting level for the envelope. Unlike the volume and harmonic envelopes, whose output (think y-axis) is magnitude, the filter envelope's output is cut off frequency. And also, unlike the other envelopes, the starting position doesn't have to be zero.

The **Start Level** is additive, so a note can start and the cut off frequency of the Note Filter could be an octave lower than the **Sustain Level** of the cut off frequency by setting the **Start Level** to -1. In the case of a low pass filter, this means the note will take some time to "open up." The time taken is the **Attack Time** — the input (x-axis) is of course time, as with the other envelopes, and the units used are milliseconds.

The **Peak Level** occurs at the end of this attack time, and can also be set above or below the **Sustain Level** of the cut off frequency for the Note Filter.

From the **Peak Level**, there's a decay stage to the **Sustain Level**, whose duration is set by the **Decay Time**. These parameters can be found on the Env Decay page. You can **Arrow Right** to get to this page, or press the back button and tap **Env Decay**.

Arrow Right again, or tap **Env Release** to get access to these options:

Sustain Level | **Release Time** | **End Level**

The **Sustain Level** is the same parameter as the top-level **Cut Off** parameter, just with a different name in this context. It's the level at which the filter cut off frequency is sustained after the attack and decay stages complete.

As per the other envelopes, we have a release stage, which begins when a note off event is received / finger is lifted. The **Sustain Level** will head towards the **End Level** and the time this takes is set by the **Release Time** parameter.

6.4. Subtractive Synthesis

Subtractive synthesis takes the opposite approach to additive synthesis. Rather than start with a sine wave and add harmonics to it, we start with a more complex wave and filter out stuff we don't want.

We can't talk about subtractive synthesis without mentioning filtering, so if you haven't read those sections yet, go back and check them out. Additionally, there is a simple multi-mode filter provided on all subtractive generators, whose settings are per-partial. This is very useful for crafting sounds for different areas of the keyboard, or processing the same oscillator differently based on the filtered output, so for instance, sending only the higher frequency content to a reverb algorithm.

Noise Generator

With the **Carrier** wave selected on the OSCILLATOR panel, tap the **Wave** button and change the **Synth Mode** to Subtractive. Now if you **Arrow Right** or tap **Edit**, you'll see:

Wave | **Roll Off** | **Pulse Width**

The oscillator generator for the base wave can be set to Stereo Noise, Mono Noise, or Super Saw. Noise has an ultra-wide bandwidth, meaning it contains *all* frequencies. You can *shape* the noise through use of a volume envelope and filter.

The multi-mode roll-off filter is provided to change the noise colour; turning the **Roll Off** up, will change it from white noise to pink noise, as it cuts out some of the higher frequencies. The label will read Low Pass. Turning the parameter down makes the filter a High Pass, reducing some of the lower frequency content; this is useful for building timbres for lead sounds where you want to reduce some of the bass energy so it doesn't crash over other bass instruments. The frequency where the filter starts to "roll off," is sometimes referred to as the 3dB point, because its the frequency where the output of the filter is 3db lower than the input.

Using a low pass filter and volume envelope, you can turn noise into a range of sounds. Shaped Noise [SDT 2-1] is an example where the end product bears little resemblance to the source generator. It contains one simple partial with a Stereo Noise wave generator. Coupled with a high pass filter and snappy volume envelope, it satisfyingly simulates the sound of a hand clap.

Super Saw

The noise generators don't require many parameters, as they're simply noise, so the **Define** button is inoperable. If you select Super Saw as your generator, you'll see these options:

Unison Voices | **Unison Detune** | **Stereo Spread**

Firstly, you can select the amount of voices, or how many voices lay in *unison*. This is how many saws are placed in one partial, just like the harmonics in an additive generator. If you select single and play a note, you'll see the waveform of a saw on the main display. Adding more voices will super-impose additional saw waves, hence the name "super" saw. Each voice quantity has a distinctive style to it.

You can create a saw waveform with additive synthesis techniques (start with a sine wave and add harmonics of diminishing magnitude using the formula coefficient of n = 1/n), but what's going on under the hood will be different.

An additive generator creates a wavetable, the playback of which varies depending on the note pressed. Playback interpolation of samples uses a linear algorithm. There's a brick-wall filter above 24 harmonics which prevents so-called fold-back distortion.

The wave of a super saw generator is created in real-time, based on the note pressed, so each sample is correct to the saw shape without the need of interpolation — it's continuous (or as continuous as a digital synth will allow) — as if you sampled an analogue saw wave with no anti-aliasing filter (well, actually a special antialiasing filter, as Regen does something clever with the samples to prevent any nasty artifacts from this synthesis technique).

You can see this difference too: if you play a note with a single saw generator and watch the oscilloscope display, and compare this to an additive saw, you'll notice the subtle "ringing" on the additive waveform symptomatic of anti-aliasing (brick-wall) filtering.

Each method of saw creation has its advantages, disadvantages, and subtle sound signature. Being continuous and sample accurate, the super saw generator is the best choice for filtering and, by extension, subtractive synthesis techniques.

Being a wavetable, the additive generator can have **Bit Crush** effects applied. As discussed in the Playing with Timbres chapter, this is the classic Synclavier sound that is so sought after.

Unison Detune sets the amount of tuning spread between the individual saw waves (not applicable to single saw). This is shown as a ratio, and in semitones and cents. That massive thick super saw sound prevalent in house and trance

music and horror and thriller movie soundtracks is created by detuning several saws.

The **Stereo Spread** places the different saws in the stereo field. With the middle saw staying centred, a positive value will push the lower saws to the left and higher saws to the right. A negative value will do the opposite.

Subtractive synthesis is a deep area, and since this is new to Regen, the factory presets that make use of it are sparse, but here are a few of timbres using saw generators, illustrating the breadth of sound possible:

Phat Bass [RW 5-5] has 1 super saw voice with a bit of FM added. Without much voodoo a phat bass sound is created. Played through the DC-coupled outputs of Synclavier Regen you can shake the room!

Regen Pad #1 [RW 3-6] and Regen Pad #2 [RW 3-7] is a simple #pad sound that is created with several blended saw waveforms, some additive and some subtractive.

Raindrops [RW 7-7] employs one partial with a 3-voice super saw. This is coupled to a low pass filter that's driven by an LFO to create a simulation of the sound of raindrops.

Hour Glass Transmission [RW 3-8] has lots of super saw partials and a noise generator partial for good measure, driven through a resonant filter, to create a #pad sound rich in harmonic content.

6.5. Soundfiles

The ability to play samples (soundfiles in Synclavier parlance) wasn't on the original Synclavier II machines, but was added as an extra in the early 80s. This opened up a world of possibilities to Synclavier owners, such as polyphony when playing sampled monophonic synths, and playing life-like orchestral instruments traditionally difficult to synthesize. A single soundfile could be transposed in pitch by the synth engine, depending on the key struck, or groups of keys could be assigned to different soundfiles for more accurate modelling. Special effects could be easily uploaded and used in film scores.

Regen has pulled out all the stops for sample playback: each partial contains a patchlist of up to 128 soundfiles; different sample playback modes are implemented, such as loop and one-shot; and over 555 varied sample files have been included factory-installed for royalty-free use in your productions.

Factory-installed Samples

When browsing the libraries, you can press the **Sample** button and you'll be presented with a list of all the samples in the current library. All factory-installed samples are royalty-free and lossless format.

Selected Legacy Samples contain hand-picked original Synclavier samples, many from the NED days, the majority of which are 50 kHz sample rate.

Since Regen's internal DAC runs at 50kHz, these NED soundfiles playback without any transformation at their native frequency, but will have interpolation applied when played back in a frequency other than their base key. Synclavier uses a variation of cubic spline interpolation for sample playback, which we have found to be both efficient to calculate and musical sounding.

Synclavier Digital has also collaborated with leading sound designers to bring a range of artist samples exclusive to Regen. These modern, sometimes eclectic, electronically created samples complement the more traditional samples of instruments found in the Selected Legacy Samples library.

While browsing samples, you can press and hold the **Audition** option to hear the sample playback at its native sample rate. When you've found a sample you like the sound of, you can tap the **Enter** button to load it to the current track. What this actually does is create an empty timbre on the selected track, then places the sample at the top of partial 1's patchlist. This isn't always what you want to do.

Recalling Soundfiles to Partials

Often, you want to keep the other partials intact and load a sample to a specified partial. If, rather than tapping **Enter**, you tap the **Choose Partials** option, you can then select a partial on which to load your sample from the

selector buttons. When selected, press **Enter**, and your soundfile will be loaded to the chosen partial, and all other partials will be preserved.

> Find a sample you like and load it to a track by pressing **Enter**. Then navigate to the OSCILLATOR panel to see what has taken place: tap **Wave**, and ensure that just the first selector button is selected, corresponding to partial 1. You'll see that Samples has been selected as the **Synth Mode**.

Sample Patchlists

> Tap **Edit** and you'll be presented with this partial's patchlist with your chosen sample in the first position. From here you can:

Choose File | Insert Row | Delete Row

Existing soundfiles can be replaced and additional ones added, up to 128. A partial is only capable of playing one sound file at any one time, so why add more than one? Because soundfiles can be arranged to trigger in keyboard ranges. This allows for multi-sampled (sometimes called poly-sampled) instruments, where each key played triggers the sampled instrument in that same note or note group.

Ranges cannot overlap — if you have more than one soundfile patched to the same key or key range, you will see the "**Conflicting Samples***" warning. Directly underneath this warning is the graphical representation of the keyboard with coloured blocks depicting the patches, so you can quickly identify which ranges conflict. With conflicting samples, Regen will sound only the first one on the patch list.

Each MIDI note triggered creates a new *instance* of a timbre with all its partials, so unless you select a monophonic playing mode, the patched samples do not *choke* each other when played; you can play a chord with more than one sample triggered from the same partial's patchlist.

One advantage of having separate samples for multiple ranges is to capture the nuance of a real instrument. Certain instruments, piano being a prime example, behave differently depending on the key struck. It's not sufficient to transpose one sample in pitch to recreate the nuance of such instruments. Additionally, a piano will sound different depending on how hard you strike a key. Not only will the sound be louder if you strike a key harder, but it will cause more reverberations in the sound box and will somewhat excite the unstruck strings, thus producing a thicker sound. Check out the Anthony Marinelli — Pianos and Keys library for examples.

Since we have 12 partials, we could sample up to twelve different velocities and use the Partial Crossfader (discussed in a later section) to map these samples to MIDI velocity ranges.

And within those 12 partials, we could have up to 128 different samples assigned up to to 128 different keyboard keys or ranges. It's rarely necessary to sample every note; typically one sampled note would be applied to a range of MIDI keys, but even in the case of a grand piano, the 88 notes could all be assigned to the same partial, so you could, in theory, have 12 levels of velocity all on one timbre.

In the case of Anthony's piano, the top 16 notes do not have a damper, so they were placed in their own partial which was set to one-shot so that these notes sustained, even after the key was released, for maximum authenticity. This kind of sectioning by keyboard range limits the amount of partials you could use for velocity levels, but sometimes keyboard ranges do behave differently.

Your mileage may vary, but in arranging the Anthony Marinelli pianos, we found that any more than three velocity levels didn't provide much more authenticity and just produced more work to blend the levels; two or three velocity levels was enough in this case.

When you tap **Choose File**, you can use the LIBRARY buttons to navigate the sample files in the libraries, as usual. While browsing for a sample, the Edit button to blinks to remind you that you are looking for a sample to add to your patchlist. You'll see a screen such as this:
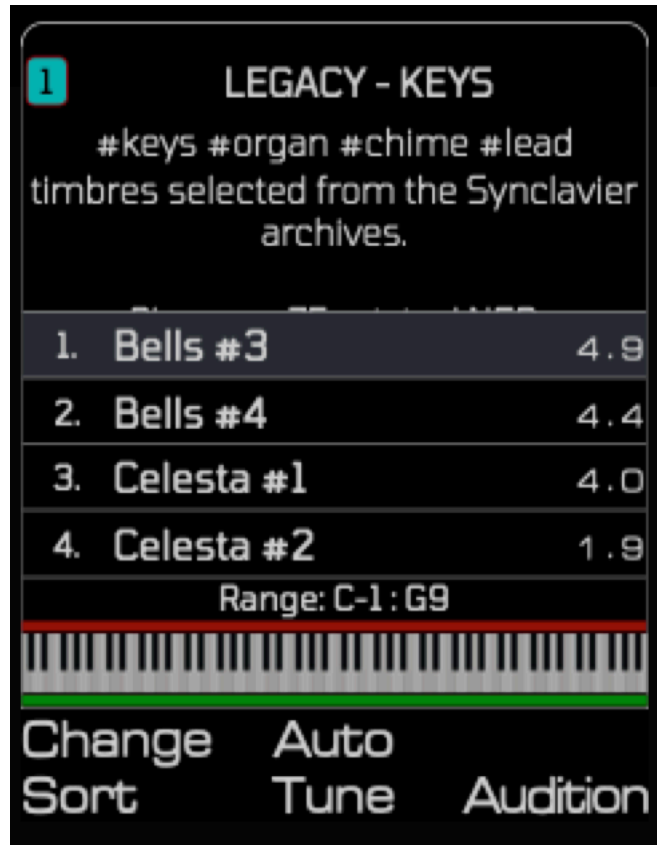
You can **Audition** samples, and when you find one you want press **Enter** and it will be placed on the patchlist. However, you'll also notice a graphic of the keyboard and a Range readout. When choosing soundfiles for a patchlist the Range setting is pre-armed: so, before tapping Enter, can you select a range for your sample to be sounded by doing one of the following:

1. Press and release one keyboard key, and this will set a range of one key, e.g. D3.

2. Press and hold one key, and then press one more key, then release: this will set a range, e.g. C3-B3.

Using this procedure, it's quick to enter a bunch of samples and map them to ranges, but admittedly a full grand piano may take more than one cup of coffee.

Another convenience feature that can be used before you tap **Enter** to select the soundfile is **Auto Tune**. Synclavier uses a fast Fourier transform technique to identify the **Base Key** of the sample. All the factory samples are pre-tuned, so this step is unnecessary, but if you're loading user samples from an SD card, you may

want to **Auto Tune** them or load them and then select the **Base Key** from the **Define Pitch** screen after loading.

If you change your mind, and wish to cancel the **Choose File** operation, you can tap another button, such as **Wave** and **Edit** will cease blinking.

NB: Other common uses of patchlists are "sound boards" of sound effects, each triggered by a different keyboard key, and, of course, drum kits.

> Call up Monster Percussion Pack [SS3 6-5] and click **Edit**. Select the 8 active partials individually to view the patchlists. This drum pack conforms to the General MIDI specification (https://en.wikipedia.org/wiki/General_MIDI#Percussive) and contains 67 drums plus 2 metronomes, each on their specified MIDI key.
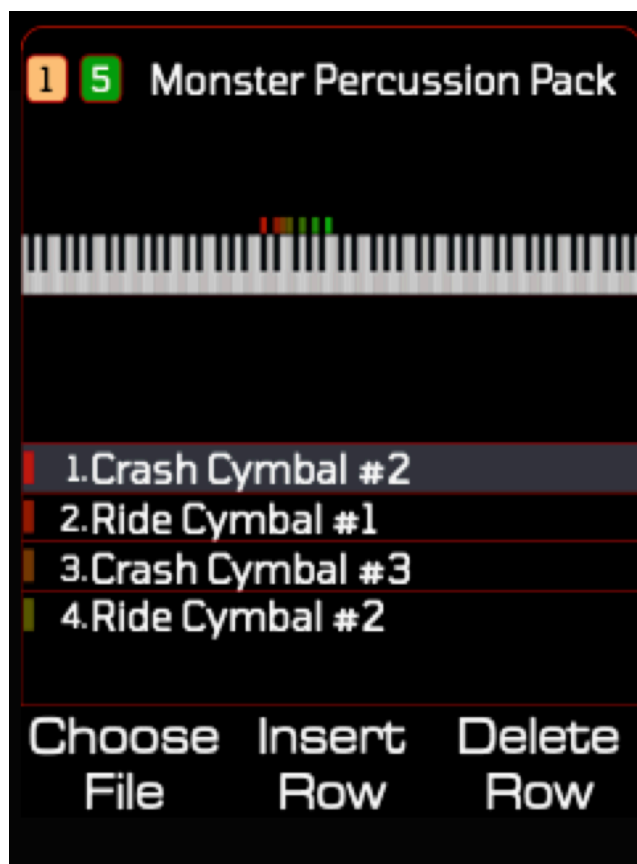

> Notice the information on the main display as you peruse these patchlists. On the top line, you have the

track blob, followed by the partial blob that you're viewing, then timbre name. There's a representation of the keyboard and small coloured rectangles above it that correspond to the coloured rectangle to the left of the soundfiles, signifying the keyboard ranges.

Since we used 8 partials and not 2, the patchlists are more manageable, the longest being 10, but more importantly, they're grouped by instrument category. So if, when using this timbre on a particular composition, you decide all the cymbals should be louder, you can adjust the volume of partial 5 without altering the other instrument categories.

You can use the **Load Up** and **Load Down** buttons to traverse the patchlist when on the soundfile pages. This will become more useful on other pages where the **Nudge Up** and **Nudge Dn** are used to modify values.

Tap the **Define** button (or **Arrow Right**) and you'll see a lot of information on the currently selected soundfile:
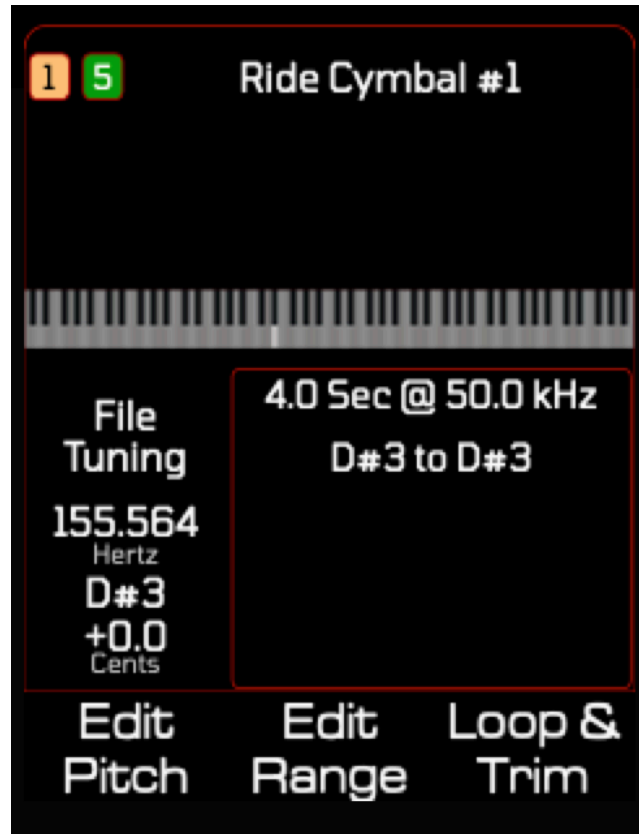
At the top is the sample name. The tuning information is on the left. In the right info block you'll find the sample length and sample rate followed by the range map. Press **Load Up** or **Load Down** to select a different soundfile in the partial's patchlist and view its information. This **Define** page can be considered a menu, since it leads to other subpages.

There are a few drum kits included in Regen that comprise samples, but here are some examples of non-percussive multi-sampled instruments for you to take a look at:

Good Disposition [RW 2-1]
Harpsichord Multi [RW 5-5]
Gravitas Organ [RW 4-6]
Full Piano [AM 1-1]
Ambience Clarity Piano [AM 1-5]

Dulcimer Multi #2 [SDT 5-2]

Sample Tuning

Tapping **Define Pitch** will take you to the **Tuning** page, where you'll have options:

Base Key | Tuning Offset | File Volume

The **Base Key** is the MIDI note that would trigger the sample at its native sample rate. If we had a **Base Key**, for instance, of G2, if you strike G2, the sample would play without any pitch transposition; if you played A2, the sample would play back faster to increase the pitch; F2 would cause the sample to play back slower for a lower pitch.

The playback rate and pitching is worked out for you by the synth engine, within the range that has been set for that soundfile. The preset samples provided have all been tuned, so when you load a factory sample, its **Base Key** and **Tuning Offset** will be configured for you (although you can override this after loading if you wish).

Real (acoustic) instruments require regular tuning and are subject to various environmental elements, such as humidity and temperature, which can affect their pitch. So many samples of real instruments may be a cent or two sharp or flat. The **Tuning Offset** is where you would correct such errors. Setting the **Tuning Offset** to +1 cent is stating the sample was 1 cent sharp, so when you play the base key, it will be played back 1 cent flatter to correct.

A small tuning error isn't a problem until it becomes one, when harmonizing with other instruments, for instance. Indeed, it's difficult to detect small errors without playing the sample against a perfectly in-tune instrument, or using a strobe tuner. And of course, sometimes these small mis-tunings are part of what makes a sample sound more "real."

You can use the **Swiper** and **Nudge** buttons to set the **Base Key** and **Tuning Offset,** or you can ask Synclavier Regen to **Auto Tune** the sample for you by tapping **Base Key** and **Tuning Offset** simultaneously. Regen will analyze the sample file and pick out the fundamental frequency using discrete Fourier analysis. It doesn't work for very complex samples or inharmonic samples, such

as many percussion instruments, but it works well enough for most uses. You can also access this **Auto Tune** feature when on the **Choose File** page.

NB: All the soundfile tuning information will be saved with your timbre when you've finished designing it, but if you load up the same soundfile on another timbre, Regen won't remember how you last had it tuned.

> If a soundfile is not accurately tuned, it may sound playable in one octave, but stretched in another, as the perceived tuning error increases across the keyboard range.
>
> When using FM, the modulator frequency is calculated as the product of the **FM Ratio** parameter and the carrier frequency. The carrier frequency is known absolutely when using Additive or Subtractive carriers, but for Samples its calculated from the **Base Key** with **Tuning Offset** (and key played). So if a sample is not well tuned, not only will it sound off, but it will affect the quality of results of any FM applied; an FM ratio that behaves in a certain satisfying way on some samples might *break down* on other, untuned samples.

Volume Matching

File Volume is a further convenience provided. It's very difficult to computationally analyze a sample file and come up with a useful assessment of its volume. You can easily figure out metrics such as maximum or minimum amplitude or a root-mean-square value, but due to the dynamic nature of many musical sounds, the volume is in the ear of the beholder.

So, if you're creating a large drum pack from recordings of different sets, or collating samples of grand pianos from different studios, for example, you will likely want to tweak individual soundfile volumes to blend well with each other.

Range & Alias Filter

From the **Tuning** page, you can press the **Back** key to return to the **Define** menu page, or you can **Arrow Right** to the **Range** page. Here you'll see:

Low Key | **High Key** | **Alias Filter**

After loading a sample, you have the opportunity on this page to change its range. You can tap **Low Key** or **High Key** and **Swipe** or **Nudge** to alter it, or you

can double-tap **Low Key** or **High Key** to arm it to listen for incoming MIDI notes. When armed, you'll see a thin green line.

You'll notice, on these **Define** pages, the range is highlighted on the keyboard graphic, and the green rectangle below a key denotes the currently selected soundfile's **Base Key**.

The soundfile **Alias Filter** isn't related to range, but space was getting tight on these OSCILLATOR pages, so that's where this wound up. Although similar to the **Alias Filter** under **Timbre Effects**, it works in a different way, and those that are interested in the nitty gritty should read on.

Flux Capacitor

In the good old days of Synclavier II, the voice cards used variable-sample rate DACs. The DACs worked very differently to the DACs of today — which we won't discuss here, as it's a huge topic — suffice to say the old DACs were of the sample-and-hold variety.

When you play a soundfile on a Synclavier II system, it works out what sample rate the DAC should run on based on a formula of the soundfile's recorded rate and played note. So if you play a run of keys, you'd be instructing the voice card to play at different sample rates.

This works well for playing notes higher than the key of the recorded sample; you simply playback the sample faster. But for lower keys, playing back the sample slower added distortion, because no interpolation was taking place.

sine wave sample-and-hold DAC   time ➡
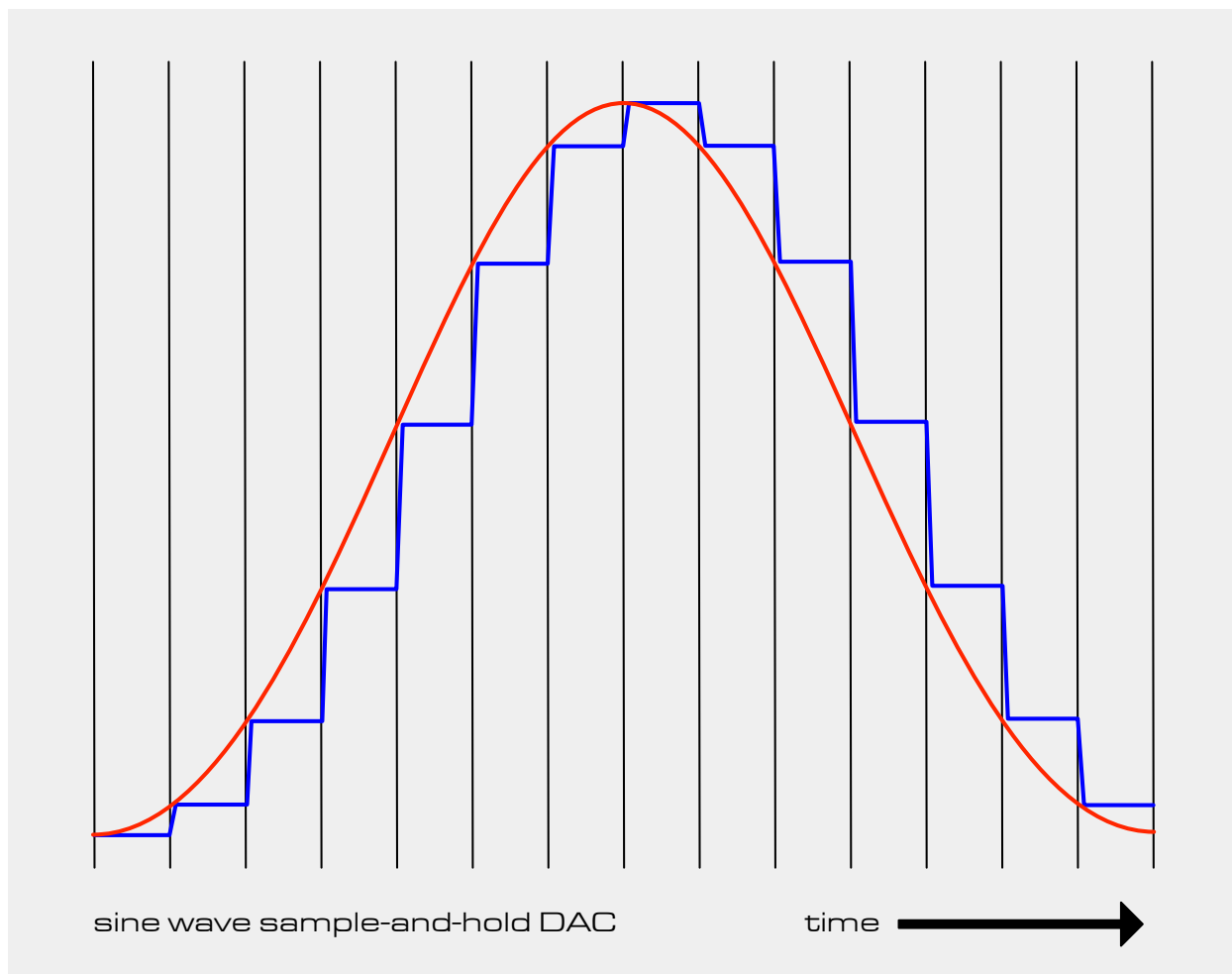
The image above shows the output of the DAC following a sine wave. At every sample point, the level is set and held until the next sample point. The transition takes a finite time (slew time) to occur.

Typically, a DAC output would be sent to a reconstruction filter which would smooth out the rough edges and re-create the original sine wave. As it happens, the Synclavier II voice cards didn't include any filtering. To a certain extent, the nominal capacitance, inductance and resistance present in the output wires and coupling would provide some filtering, but many of the higher frequency artifacts of the stepping would fall outside the audible frequency band in any case. That is, until you play the sample at a lower frequency than it was recorded. Since there's no interpolation on Synclavier II, at lower pitches, the samples are

simply held for longer and the resultant "decimation" distortion appears in the audio as a kind of grunge.

This distortion is often desirable. It's why many people talk fondly of the early samplers such as the MPC, E-mu, Ensoniq, Fairlight, and Synclavier. Still today, people will record into these devices and pitch down their samples to get the desired grunge prevalent on hip hop records and other sample-based genres from 1980 onwards.

Synclavier Regen interpolates sample playback using a modified cubic spline algorithm. If you turn the **Alias Filter** Off, it reduces the amount of interpolation performed to simulate the Synclavier II DAC output through the Regen sigma-delta DAC, so you can recreate the grunge of the original.

Our thinking is, if you go to the effort of loading your own samples into the Regen, then not only should you be able to add the Synclavier effects, but even without effects, they should still sound enhanced. Otherwise you're just playing back samples, and any computer sample program can do that. The Synclavier II added character to sample playback that was very musical, and the Regen faithfully recreates this.

> For an example, try loading Pad #10 from Selected Legacy Samples. The **Alias Filter** should default to Off. Play middle C, then an octave lower, then an octave lower. You'll get the idea. Then turn the **Alias Filter** On and play the same notes.

Also note that you can add some of the same type of distortion to all soundfiles on a timbre using the Grunge control from the TIMBRE Effects page. This adds some decimation distortion to samples regardless of the ratio of sampled key to played key.

Sampling Real Instruments

If you intend to sample an instrument of your own for use in Regen, it pays to be aware of a few things. Firstly, the more notes you can sample, the more realistic

the timbre created from those samples will be. Regen supports up to 128 samples per partial — (see the Sample Patchlists section above)

The more samples, the smaller range each one will occupy. If you are stretching one sample across the whole keyboard range it will start to "squeal" after a few octaves; there's interpolation, but ultimately information is dropped. And going down, it sounds "muted" after a few octaves because the higher harmonics "run out." For most realism, you'd sample in the native area of the instrument, e.g. A4 would be good for a violin, A2 would be better for a Tuba. But realism isn't always the aim if you're just using one sample. Maybe you want it to squeal, mute, or grunge.

Also note, if you want to stretch a sample a long way down, you could sample in a higher bit rate, but stretching up it doesn't matter what bit rate you chose; as long as the sample is high-fidelity enough when played back in its Base Key. For most purposes 48kHz is sufficient, with a bit depth of 24. And lossless compression formats, such as .flac, are the best method to stuff lots of quality samples on an SD card.

Looping & Trimming Samples

You can **Arrow Right** from the Range page to get to the Loop page, or go via the top-level **Define** page. The Loop page has these options:

Loop Mode | **Loop Start** | **Loop End**

There are 5 loop modes, described in this table:

Loop Mode Behaviour

No Loop	Final decay kicks in when key is released, à la piano damping mechanism.
One-Shot	Entire soundfile plays once; no damping mechanism. Think dulcimer. Most percussion sounds should be set to this mode.
Always Loop	Soundfile loops always; final decay from the envelope generator kicks in when key is released. Required for the NED #loop files (or any sound with a built-in loop crossfade).

Loop w/Tail	Soundfile loops when key is held; proceeds past loop point when key has been released; final decay is processed (assuming a release time on partial's volume envelope). Useful for sample files that contain a final decay stage.
One-Shot Loop	Soundfile loops when key is held; proceeds past loop point if key has been released; and always plays to the end of the soundfile. This is a useful mode for chunks of sound, such as breakbeats and the included #beatloops.

The **Loop Start** and **Loop End** points can be set on this page and looping information will be shown in the info pane when applicable. When looping, the *playhead* will jump from the **Loop End** time to the **Loop Start** time.

If you Arrow Right from the Loop page, you'll get to the Trim page, with options:

Start Time | End Time

These parameters are fairly self-explanatory. You can non-destructively trim the start and end points of samples. Often, there'll be a short pause at the start of a sample before the first transient occurs and at the end of many samples there'll be a period when the sampled instrument is inaudible, but sufficient sample noise remains that's worth trimming out. You'll see that the trimmed parts of the sample are shown on the graphical representation as greyed-out segments.

In some cases, you might have one sample file that contains several sounds, such as a drum breakbeat, and you can use the trim parameters to select just one of these sounds. If you have such a sample, you can have it loaded on your patchlist several times with different **Start** and **End Times** on each occurrence, but Regen will hold only one copy in memory.

As you adjust the markers, you'll notice that the unplayed region (outside of Start Time and End Time) will be shown on the sample waveform display in a grey colour, the played region will be green and the looped region will be yellow. NB: The screen has enough space to show 2 seconds of sample time with accurate resolution: if your sample is longer than 2 seconds then if you trim the start or

end, you will not see the grey regions, instead the waveform will scroll and zoom accordingly.

It's sometimes necessary to dial in a very precise **Start** or **End Time**, and this can be done by the **Nudge** buttons, which are sample accurate, or by holding your thumb on the **Down Nudge** button and using your index finger to swipe, wherein the **Swiper** will operate in fine mode providing 10 samples per increment rather than 100. There's also a fast scrubbing mode, which scrolls 1/200th of the sample per increment — this is activated by using the **Swiper** while holding down the parameter button (i.e., **Start Time**, **End Time**, **Loop Start** or **Loop End**).

The loop region needs to reside inside the trim region, so if you try to set the **Loop Start** lower than the **Start Time**, it will push the **Start Time** down for you. Likewise, if you try to set a **Loop End** higher than the **End Time**, it will push the **End Time** up for you.

6.6. Partial-level Effects

Chorus

Chorus was discussed in the section on Timbres, since it can also be applied to timbres. The **Chorus** on the OSCILLATOR panel affects the selected partials. A ratio of 1 means no chorus, and turning it up or down will create a chorussed voice above or below the base tuning.

Chorus works for partials with Additive and Samples synth modes. For partials with Subtractive synth mode, the button will be inactive. Instead, you would use the **Detune Width** on the **Define** button to add chorus to your Super Saw partials. This is an equivalent effect and uses the same algorithm.

Pitch Track isn't related to chorus, and at first it seems a weird place to put it — and it is a bit — but it doesn't really belong under the **Define** button either, since it doesn't pertain to soundfiles in a patch list — pitch track is set per partial.

Pitch Track simply tells the synth engine to change the partial's pitch depending on what keyboard note is struck. Of course, this defaults to On. If you want to lock pitch regardless of keyboard key, for instance, in the case of sound effects or certain percussion sounds, you can set this to Off.

A few of the preset samples are coded to turn **Pitch Track** Off when they load, but bear in mind that this is a partial-level parameter, and not a soundfile level, so if you load a sample using the **Choose File** method it will leave the existing **Pitch Track** setting alone, unless you're changing the only file in the patchlist.

Vibrato

The **Vibrato** feature modifies the pitch of a sound as it's being played. It applies frequency modulation, but unlike the FM feature, the modulating wave is a low frequency oscillator (from 0 - 50hz).

There are 3 pages of parameters, accessed via the **Arrow** buttons:

Vib/LFO A Attack | **Vib/LFO A S-Curve** |
Vib/LFO A Wave | **Vib/LFO A Rate** | **Vib Depth**
Vib/LFO A Invert | **Vib Quantize** | **Vib Bias/LFO A**

Vibrato Parameter

Vib/LFO A Wave	The shape of the pitch changes. See table below.
Vib/LFO A Rate	The vibrato wave can repeat between 0.00 and 50.00 Hz.
Vib Depth	How much for the pitch to alter, between 0.00 and 25.00 semitones.
Vib/LFO A Attack	The vibrato increases gradually to its full depth from the beginning of the note to the end of the selected attack time, which can be set between 0 and 30 seconds.

Vib/LFO A S-Curve	When On, this changes the attack from linear to an s-curve: the change will be faster to start with, will slow down for the middle of the attack, and speed up towards the end. If you set **Vib Depth** to 100% and the **Vib Attack** to 12s or so, you should be able to hear and see (on the graphical display) the difference between s-curve and linear attack.
Vib/LFO A Invert	Inverts the direction of the vibrato wave. E.g., a sine wave starts at zero and increases in its first quadrant. With **Vib Invert** on, the sine wave would start at zero and decrease in its first quadrant.
Vib Quantize	The fluctuations produced by vibrato are normally smooth and gradual. However, you can quantize, or step, in semitone intervals the fluctuations in pitch. After you set the vibrato wave, depth and rate, turn on the quantize function by pressing **Quantize**. To witness this effect, try listening to a partial with vibrato set to a low **Vib Rate,** a high **Vib Depth,** and toggle **Quantize** On and Off.
Vib/LFO A Bias	Create a vibrato in which the pitch fluctuates entirely above the key pressed. Press Bias and play a note. As a result, the lowest point of pitch fluctuation is the pitch of the note played. The highest point is twice the number of semitones set for the vibrato depth.

Try modifying the vibrato of a partial timbre now. A sine wave vibrato of 6Hz is typically used to create the style of vibrato associated with vocalists and string or wind instruments, so this might be a good place to start your experimentation.

> Remember that you can put the **Swiper** into fine mode by holding your thumb on the **Down Nudge** and use your finger to operate fine mode.

Vib/LFO A Wave	Shape
sine	
triangle	

ramp (sawtooth)	
inverted ramp	
square	
random	

Tremolo

The **Tremolo** feature modifies the amplitude and/or stereo panning of a sound as it's being played.

There are 3 pages of parameters, accessed via the **Arrow** buttons:

Trm/LFO B Attack | **Trm/LFO B S-Curve** | **Trm Phase**
Trm/LFO B Wave | **Trm/LFO B Rate** | **Trm Depth**
Trm/LFO B Invert | **Trm /LFO B Alt** | **Trm/LFO B Sync**

The Tremolo wave can be used to amplitude modulate (AM) the carrier wave and/or is used to modulate the panning, controlled via the **Trm Phase** setting. A full description of the parameters is given in the table below:

Tremolo Parameter

Trm/LFO B Wave	The shape of the amplitude changes. Use **Swiper** or **Nudge** buttons to select one of three tremolo wave shapes: **sine**, **triangle**, or **square**.
Trm/LFO B Rate	To select a rate between 0.00 and 50.00 Hz for the tremolo wave.
Trm Depth	A depth of 100% is full tremolo, which would vary the waveform from its full amplitude to zero over the time period of the tremolo wave. If you set this lower, say 25%, then the carrier wave would be reduced by a maximum of 25% of its full amplitude.

Trm/LFO B Attack	The tremolo increases gradually to its full depth from the beginning of the note to the end of the selected attack time, which can be set between 0 and 30 seconds.
Trm/LFO B S-Curve	When On, this changes the attack from linear to an s-curve: the change will be faster to start with, will slow down for the middle of the attack, and speed up towards the end. If you set **Trem Depth** to 100% and the **Trem Attack** to 12s or so, you should be able to hear and see (on the graphical display) the difference between s-curve and linear attack.
Trm Phase	This defaults to 180° which allows full left/right stereo panning effects. Turning the knob full down to 0° or full up to 360° will allow tremolo (Amplitude Modulation) effects with no left/right stereo panning. Other settings between 0° and 360° will provide varying degrees of panning versus amplitude modulation depth.
Trm/LFO B Invert	This parameter changes the direction the tremolo wave starts in. When the parameter is Off, the stereo position moves to the left first, from its original (panned) stereo position. When the button is On, the stereo position moves to the right first, from its original stereo position.
Trem/LFO B Alt	The Alternate parameter turns on a function that alternates the starting Left/Right side of a Left/Right Pan for every new MIDI note on event. (when this is on, invert only affects the first note played.)
Trm/LFO B Sync	When On, all notes played will use the same LFO, so the phase will be synced. When Off, each new note on event will trigger a fresh tremolo LFO started for that note. Sync On is great for playing chords as now matter how tight the keys are struck the effect will sound homogenous. This is especially useful when you map the Tremolo LFO to destinations such as filter Cut-off. Sync Off is great for emulating real-life instruments or when using Tremolo for panning at a low Rate: each note can appear and disappear at a different spot on the stereo field.

Try playing with the tremolo settings now. Try changing the phase setting so that you experience stereo panning, as well as pure amplitude modulation.

6.7. Partial Crossfader

Now that you know how to create partials from scratch, it's high time we talked about the Partial Crossfader. Earlier, when discussing additive synthesis, we briefly visited the **Levels** button on the ENVELOPE panel, which gave us access to partial **Volume** and **Pan**, useful when blending several partials sounding at the same time. But what it you don't always want every partial to sound, or not at the same volume? In that case, you'll need to make use of the Partial Crossfader.

Recall our discussion on the Track Crossfader in the SESSION chapter. The Partial Crossfader is very similar, with some notable differences.

The Track Crossfader *filters* based on MIDI channel, and, optionally, keyboard range, and can apply fade-in and fade-out based on the keyboard range (to effectively crossfade). This is useful for playing Regen multi-timbrally, or for layering several timbres together (stacks).

The Partial Crossfader, cannot *filter* based on MIDI-channel — that's the job of the tracks — but *can* filter on keyboard range. This is useful for blending several sounds (partials) together for a split (or multi-sampled) instrument. Additionally, the input can be set to *filter* based on Pitch Bender, Mod Wheel, Keyboard, Velocity or Pressure, or any MIDI CC (of the track MIDI channel). These additional inputs make the Partial Crossfader the place to go for building complex dynamic timbres that respond organically to MIDI input.

The best way to explore the Partial Crossfader is to see it in action, so we'll walkthrough some examples:

> Call up Traditional [RW 4-4] and click Levels on the ENVELOPE panel. **Arrow Left** to get to the Partial

Crossfader and you'll see something like the image below on the main display:
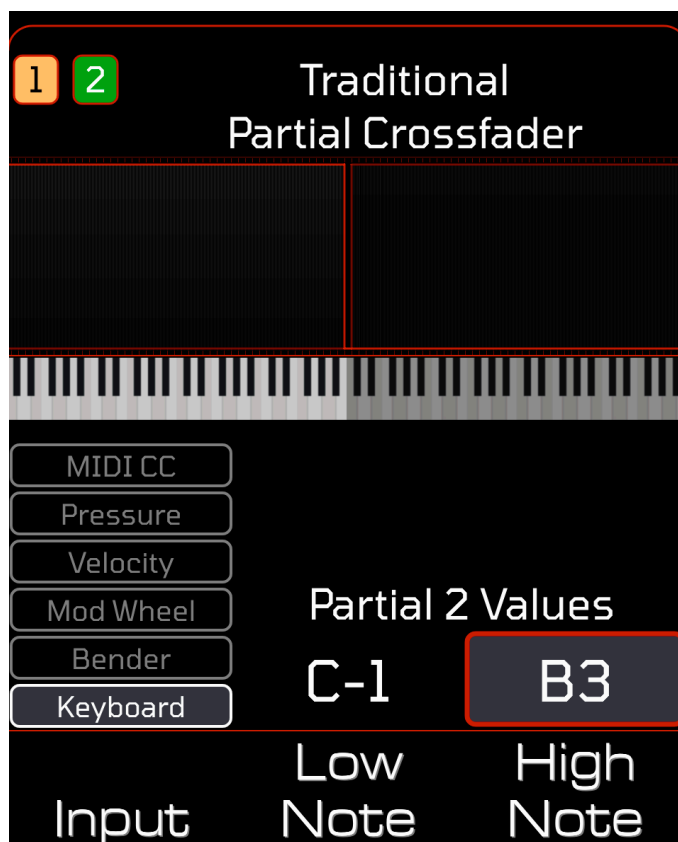
This is the simplest use of the Partial Crossfader — you'll notice the graphic above the keyboard shows two different shades of red, representing the two partials: partial 1 is an additive generator modelling a violin and its range is set for maximum volume from note C4 and up; partial 2 is a harpsichord sample set for maximum value from note B3 and down.

Splitting the keyboard for bass and treble is commonly done, and can easily be achieved with the use of the Timbre Crossfader, but this happens to have been done

inside the timbre in this case. **Low Note** and **High Note** function as they do on the Track Crossfader.

If you **Arrow Left** from the Partial Crossfader page, you'll see the same 3 fade parameters present on the Track Crossfader Fade page, which work in the same way.

Fade Shape | Fade In | Fade Out

Let's look at a more complicated example that actually *crossfades*:

Call up Dulcimer Multi #1 [SDT 5-1], and, on the Crossfader page, you'll see the main display as below:
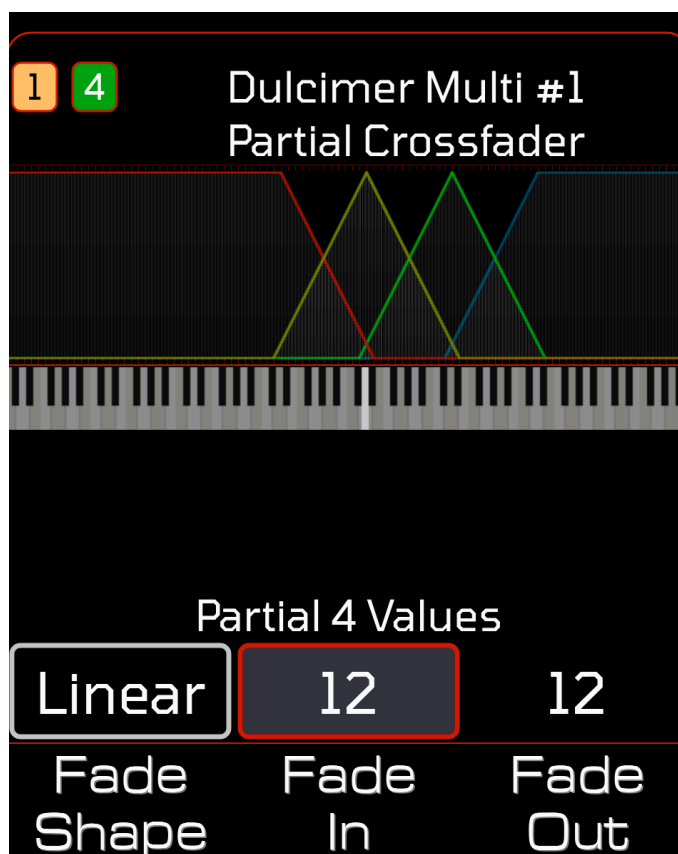
Here we have four groupings of transfer functions. This is a multi-sampled instrument for a dulcimer.

The Synclavier Regen Tutorial directory contains 12 samples of a hammered dulcimer. Strings D3, D4, D5, and D6 have been recorded for soft, medium, and hard strikes. It should be noted that these samples are by no means the final word on multi-sampled dulcimer instruments — for sure, many higher quality sample libraries of dulcimers could be procured — but they serve well for instruction and experimentation purposes.

The 4 groups of transfer function are split for each note, e.g., the red line is for D3 notes and is shared with partials 1-3, which contain each, soft, medium, and hard samples. Unlike the last example, we have used **Fade In** and **Fade Out**. Using a value of 12 (semitones in the case of keyboard range) means the fade takes an octave to fully transpire.

If you **Arrow Right** to the Levels page, you'll notice that many of the 12 partials (1 sample on each) have had their levels tweaked. You can take many precautions during the recording phase, such as careful microphone placement, even hammering technique, several takes of each note, etc., but you often find during the reconstruction stage that some samples are louder than others when they're played "as an instrument", and the levels invariably need some tweaking.

For example, Partials 4-6 have been set to a range of one note — D4 — and have a **Fade In** and **Fade Out** of 12. So, when you strike the D4 note, you'll hear that group of partials at their loudest possible volume, and you'll hear very little volume from partials 1-3 and 7-9. As you play further up, you'll hear more of 7-9 and less of 4-6,

until you hit the halfway point, G#4 ,where you'll hear both groups crossfaded equally. NB: If you play notes while viewing this page, you'll see handy little bars appear, showing where you are across the crossfader.

Crossfading multi-sampled instruments can help make a timbre sound more realistic by blending the edges, and there are many ways to do it. Dulcimer Multi #1 is a straightforward approach, setting the middle samples to sound on the key they're sampled in and setting fades from there, but you could employ larger ranges, or gentler crossfades; whatever sounds right for the instrument you're building. Some users may prefer that the samples only pitch up, or, less-commonly, only pitch down, so they get one type of interpolation (the flavour of interpolation Regen applies varies depending on whether it has to transpose the sample up or down). For this example, that would mean setting a range for the D4 sample of D4 to C#5, or D#3 to D4, respectively.

We will return to this example when we discuss Modulators in the next section, but for now, load Dulcimer Multi #2. This is using the same 12 samples, but arranged in a different way. Play both and discover which you prefer, or which sounds more natural to you.
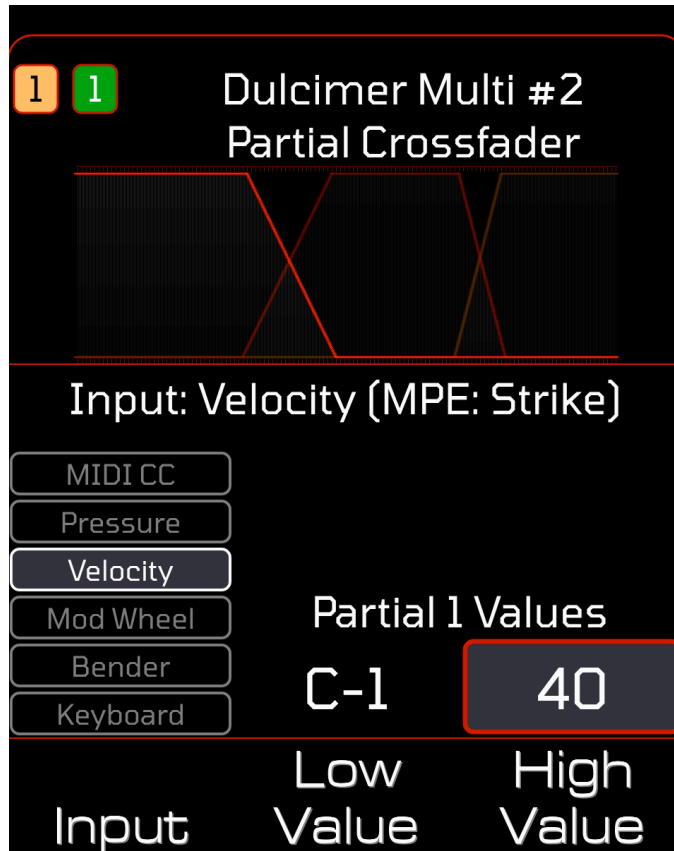
As you can see from the Partial Crossfader settings, we have switched the **Input** to Velocity. The x-axis of the transfer functions becomes velocity, soft to hard, left to right (the y-axis is always volume). Play a few notes at different velocities and you'll again see the vertical indicator bar, but it will be further right depending on how hard you hit a key. The x-axis units are 0-127 since that is the MIDI input Regen receives for Velocity messages.

Fades of 20 have been used for partial 1 to 2 (soft to medium), but for partial 2 to 3 (medium to hard), a steeper fade of 10 was employed. Since the Partial Crossfader has been tied up with crossfading for

velocity, the range controls on the patchlists have been used to filter for the note played.

And you don't necessarily have to crossfade for a multi-sampled instrument to sound authentic. Have a look at the Crossfader of Tongue Drum [RW 2-1] and you'll see the partials are either on or off for a given velocity — this works well for a set of samples whose character of sound varies substantially from one level of velocity to the next. — in fact, with more percussive samples, you will want to use discrete ranges: if you try to blend, for example, different velocity-level strikes of a snare drum, you're likely to get phase cancellations that will make the result sound muddy.

For greater realism, velocity is also mapped to volume with a Modulator (covered in the next section). NB: The partials of Tongue Drum all reference the same soundfile, but at different starting points, as the soundfile is a recording of the same note of a tongue drum being hit successively harder. Regen is smart enough to only keep one copy of the soundfile in memory when using this timbre and all the partials point to it.

Another example of using the Partial Crossfader for velocity that's worth some examination is Tympani Fest [RW 6-7]. In the image below, you can see how the harder you strike a key, the more partials are added (and the louder the sound), but none are removed. This breaks the rule of not blending percussive samples, but the aim of this timbre is not tight drumming; rather a sound effect with a cacophony of sound. Know the rules then pick when to break the rules!

Tympani Fest
Partial Crossfader

Input: Velocity (MPE: Strike)

MIDI CC
Pressure
Velocity
Mod Wheel
Bender
Keyboard

Partial 1 Values

0   127

Low   High
Input   Value   Value

Many Strings
Partial Crossfader

Input: Mod Wheel (MPE: Slide)

MIDI CC
Pressure
Velocity
Mod Wheel
Bender
Keyboard

Partial 1 Values

41   127

Low   High
Input   Value   Value

The same is true of Many Strings [RW 3-1], image shown above, but this time the mod wheel controls how many partials are added to the sound. If you load this timbre and navigate to the Partial Crossfader page, you'll see that if you move your mod wheel, the vertical indicator bar appears, and you don't have to play a note to observe that.
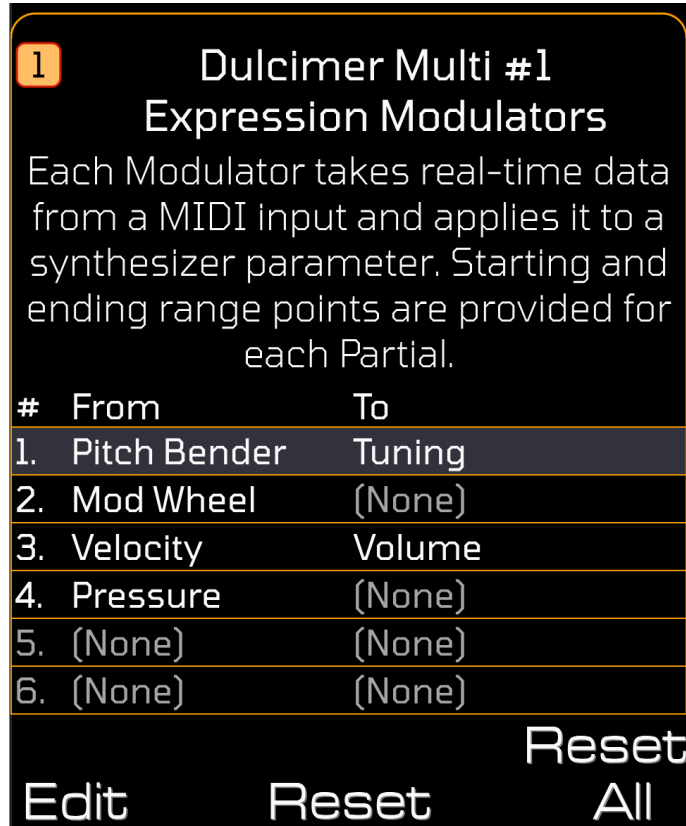
All MIDI messages are coded for channel, including the mod wheel, so, for example, if you're sending mod wheel messages on a channel that's filtered out by the track, you won't affect the sound of the timbre on that track, because those messages won't reach the Partial Crossfader.

6.8. Modulators

Welcome to the most complicated section of Synclavier Regen. As complicated and powerful as it is, our aim was to present Modulators to the user in as straightforward a way as possible. We spent a lot of time carefully thinking through the modulator interface and believe we've developed a system that's easy to work with.

The Basics

To get to the modulator page, tap any button whose parameter you may wish to "modulate", for example, **Filter**, then hit the MIDI button . A shortcut is to press Vibrato and Tremolo together. You'll see a page such as this:

```
┌──────────────────────────────────────┐
│ ┌─┐        Dulcimer Multi #1         │
│ │1│      Expression Modulators       │
│ └─┘   Each Modulator takes real-time data
│        from a MIDI input and applies it to a
│        synthesizer parameter. Starting and
│        ending range points are provided for
│                   each Partial.
│       #   From              To
│       1.  Pitch Bender      Tuning
│       2.  Mod Wheel         (None)
│       3.  Velocity          Volume
│       4.  Pressure          (None)
│       5.  (None)            (None)
│       6.  (None)            (None)
│                                 Reset
│        Edit        Reset          All
└──────────────────────────────────────┘
```

As stated on the page, modulators take data from a MIDI input and "modulate" a chosen parameter. You can have up to 12 modulators per timbre (saved with the timbre), each with a source and destination, which you alter with the **Swiper** or **Nudge** buttons.

The Pitch Bender (sometimes called Pitch Wheel) is often routed to Tuning, as in the example above. You can remove this routing by selecting the **Reset** option. Tapping **Reset All** will clear all routings.

If you press Edit, you'll be presented with a subpage with these options:

Edit From | Edit To | Define Range

- those buttons will lead to 3 further subpages where you can modify the source, destination, and range, of the highlighted modulator, respectively.

Continuing with the example above, you'll see that modulator 3 has been routed from Velocity to Volume. It's our old friend, Dulcimer Multi #1 [SDT 5-1]. Recall that this timbre used the **Partial Crossfader** to control

the keyboard ranges of the various samples, but that the samples were also categorized by fierceness of hammer strike. If you load this timbre, go to the modulator in question, and tap **Define Range**, you'll see this screen:

```
 1     Modulator 3
      Velocity - Volume
  1.       127.0         0.0
  2.        64.0        64.0
  3.         0.0       127.0
  4.       127.0         0.0
  5.        64.0        64.0
  6.         0.0       127.0
  7.       127.0         0.0
  8.        64.0        64.0
  9.         0.0       127.0
 10.       127.0         0.0
 11.        64.0        64.0
 12.         0.0       127.0
Change Change Change
 Shape  Range  Range
```
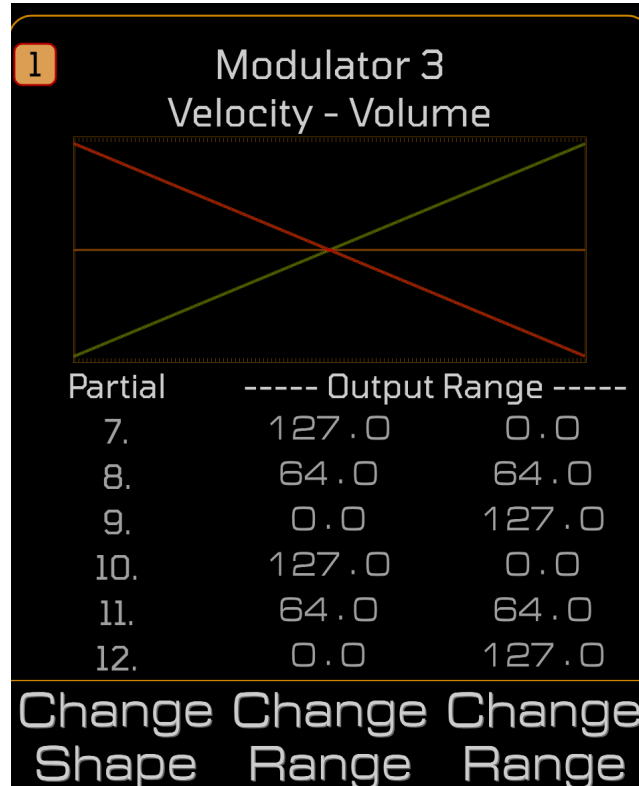
This lists each of the 12 partials and their minimum and maximum destination value, based on the smallest and largest input values of the source. The numbers go from 0 to 127.

To change a value or values, you would tap the required **Selector** button/s to toggle those partial's selected state, tap which **Range** extreme you want to modify, then **Swipe** or **Nudge** the value.

In this case, partial 1 would have a volume multiplier of 127 (full) applied when the Velocity was at its minimum setting, and the Volume would reduce all the way to a multiplier of 0 (off) when the Velocity was at the highest setting. So, we hear partial 1 at its loudest when a key is struck softly. This is the same for partials 4, 7 and 10. You will recall from the earlier synopsis of this timbre that these partials hold the soft samples.

The medium samples are on partials 2, 5, 8, and 11, and you can see from the images that the multiplier applied to volume is the same value regardless of the velocity source value. And the hard samples, housed in partials 3, 6, 9, and 12 start quiet and get louder with greater velocity.

You can use the **Arrow Left** and **Right** buttons to change the views: a graphical view with the transfer function of all partials is displayed à la Crossfader, and reduced screen real estate means you only get the values of half of the partials at once, e.g.:

```
┌─────────────────────────────────┐
│ ① │        Modulator 3           │
│         Velocity - Volume        │
│                                  │
│                                  │
│                                  │
│   Partial      ----- Output Range ----- │
│     7.         127.0         0.0  │
│     8.          64.0        64.0  │
│     9.           0.0       127.0  │
│    10.         127.0         0.0  │
│    11.          64.0        64.0  │
│    12.           0.0       127.0  │
│  Change  Change  Change          │
│  Shape   Range   Range           │
└─────────────────────────────────┘
```

So, the strategy used to design this multi-sampled timbre was to blend the three levels of sample fierceness by the strength of the input Velocity. This somewhat mimics the behaviour of a real dulcimer. Clearly, a skilled player can vary the amount of force she applies to the strings in more than 3 categories — its a continuous range of strike velocity — and the nature of an instrument like a dulcimer is that the volume of the struck course of strings vary in volume as they're hit harder, but that increased energy also excites other (unstruck) strings of the instrument in complex ways.

Some of this complexity is captured in the 3 levels of samples and makes its way to the finished timbre in execution, but it's a compromise. There are many ways to do this, as already eluded to, but each will be a compromise. Having said that, dulcimers are expensive, delicate, heavy, take up space, and require frequent time-consuming tuning, so when you consider all of that, maybe the compromise makes sense for you.

Okay, enough about Dulcimers. As beautiful and tranquil as they look and sound, sometimes you want something more menacing.

Getting more Advanced

Load Talking Trumpet [RW 8-5] and play a few notes, and then we'll try to dissect the craziness. There are three active partials, all set to Samples synth mode, each with one soundfile: respectively, they're samples of a didgeridoo, a tympani, and a trumpet.

Partial 1 has **Pitch Track** Off, and a 16Hz tremolo at 35% and is there as a kind of background flutter sound. Partial 2 is also pitch-locked, so won't change based on the note played, but it does change pitch as it has an extreme 25 semitone vibrato applied at 10Hz. If you **Solo** Partial 2, you'll hear the pitch jumping around as you play and sustain a note. Remove the vibrato effect and you can actually tell it's a tympani. Partial 3 is our trumpet, it has **Pitch Track** on, but the **Octave Ratio** for this timbre is set to 2.02, and this stretching of the pitch makes it much less like a trumpet. Rather than **Solo** Partial 3 to hear it, turn down the **Partial Volume** of 1 and 2 and play and sustain a key in the range C5 - B5. You'll notice a discontinuity in the sound. The sample is set to loop and the join is far from perfect, but that's not why you hear the discontinuities. To figure that out, we'll have to investigate the Note Filter settings.

Tap **Filter** and you'll see we have a low pass filter with a 24dB slope. Tap Filter Setup and you'll see a page like this:
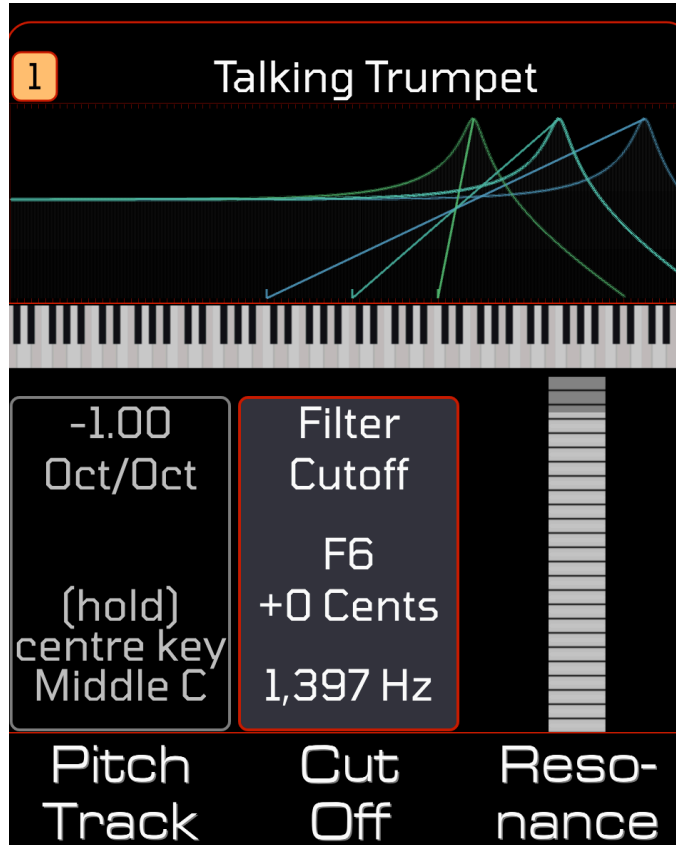
Resonance is cranked way up at 90% and, as such, this filter is able to self-oscillate.

The Filters on Regen are of the IIR variety, which use feedback loops to attenuate frequencies. With resonance, certain frequencies around the cut off point are reinforced rather than attenuated, and once these frequencies are bouncing around in the filter, they can linger, even after a note's final decay is complete.

Notice that the **Pitch Track** is set to -1.0. It's unusual to *pitch track* the keyboard in the negative direction, but there are a few musical use cases for this. In this timbre, the effect adds to the quirkiness of the overall sound, but it means there's a sweet spot: because as you play higher on the keyboard, the low-pass filter *closes* and

eventually, too much of the high-frequency information is filtered out for there to be anything useful left.

Now tap the MIDI button. On the Expression Modulators page, you'll see that Vibrato LFO 2 is routed to Filter Cut Off. This is the really interesting part of the timbre. The extreme random vibrato that was applied to partial 2 has also been "exported" as an LFO (low frequency oscillator) and routed to the note filter's cut-off. So every note played has a filter that opens and closes in a rapid random fashion. This is what's responsible for the "talking" effect.

This technique is used widely in subtractive synthesis, and is often referred to simply as "sample and hold filter effect" since the filter's cut off jumps to random frequencies and is held for a finite period, much like the sample-and-hold DACs we discussed in an earlier section. Check out the intros of Deep Purple's A' 200 and Emerson, Lake & Palmer's Karn Evil 9 - 1st Impression - Part 2 for examples of this technique in action.

NB: The filter has a 10 octave range, so when you map a modulator to filter **Cut Off**, you're almost certainly want to adjust the range values to *zoom into* a useful region. In the case of this timbre, we've restricted the LFO source to act from 54.0 to 127.0 on the destination. This is actually a large region for filter **Cut Off** and, by design, some of that region is outside the useable filter range, so the sound comes in and out.

With LFOs and Envelope sources, when you tap **Edit From**, you'll have an option to **Choose Partial**, where you can select which partial's Vibrato wave, Tremolo wave, Volume Envelope, or FM Envelope, respectively, to export.

If you're routing to a partial-level control, you'll also have the option "Each Partial" that would route each partial's LFO or Envelope to its own partial for the destination parameter.

And if you map "Each Partial" to a timbre-level or note destination, such as filter **Cut Off**, partial 1 is selected as the source for the modulator.

If you tap **Edit** then **Edit Range** you'll see a page like this:
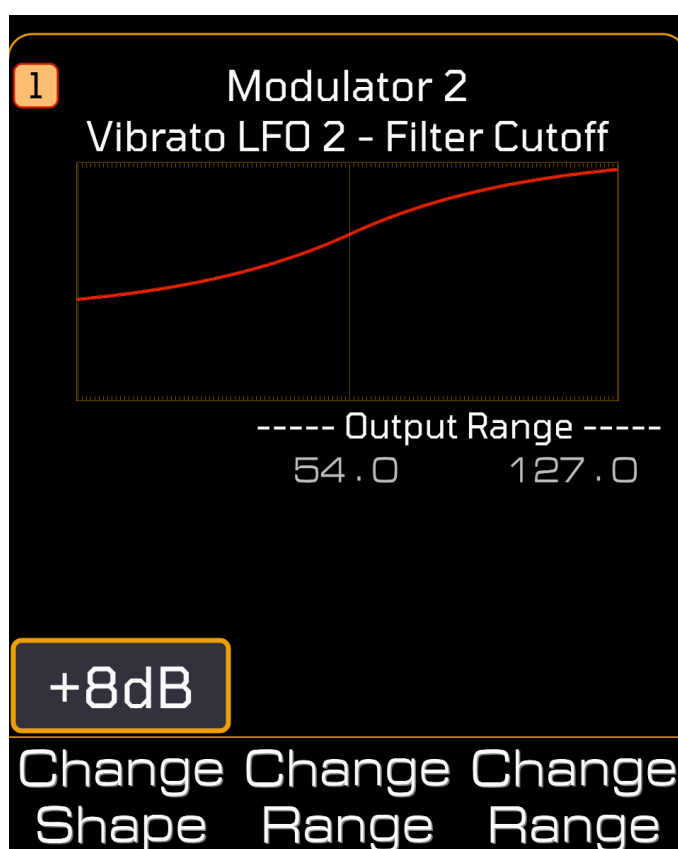
Note, there's only one line of Output Ranges. That's because some destinations, apply to the timbre rather than the partials, because they pertain to timbre-level

functions. Try making this effect more or less pronounced by decreasing or increasing the start **Range** from 54.0 (the range limits the travel of filter cut-off, but the swing of the LFO is always full). Note also, that the transfer function is a curve rather than a line.

When the **Shape** is set to non-linear, the LFOs and Pitch Bender sources follow an S-Curve, since they're resting value is in the middle: they go plus/minus, as opposed to Mod Wheels, Pressure, Envelopes, and so on, that go in one direction only. This curve has been set to a steepness of 8dB.

One neat trick used in FM Swell [SDT 3-8] is to match the curve steepness to the filter and route the same source to Cut Off **and** Volume (with an inverse relationship), so as the filter position is changed to remove some upper harmonics, the volume is increased to compensate. In this timbre, the Mod Wheel, acts not so much like a brightness control, as a character control: you can choose whatever character of FM Swell you want, and it should remain roughly at the same output level.

You can use a MIDI CC controller for a source, and if you select this, you'll have an option to **Choose CC** — tap that to arm and listen for the next MIDI CC message as usual, or simply dial in the CC code. NB: It's going to adhere to the Track Crossfader settings so will process CC messages from whichever MIDI tracks are *unfiltered*.

As an example, many experienced synthesists like to use a pedal for expression. As opposed to a sustain pedal, which is on/off, these have a variable level depending on how much you push down. Often these connect through a MIDI keyboard controller and will typically appear as CC11. In many of the factory timbres, the Mod Wheel mapping could be changed for an expression pedal CC and with some tweaking of the Shape, you will have a way to add lots of expression while keeping both hands free to play the keyboard.

> Just to be clear, a modulator routing can only operate within the range of zero to the parameter value set in the timbre. For instance, returning to the Talking Trumpet example, the filter Cut Off is set to 1397Hz. The extent of the LFO, 127 range value, will set the filter Cut Off to 1397Hz, it can't go higher. This is as opposed to the MIDI-link function described in the SESSION chapter, where the parameter's actual value is mapped to an input and can be changed within the full range of that parameter by adjusting the CC controller. The two functions act differently, because they're for distinctly different purposes.

Commonly mapped modulator sources have sensible default positions that are applied when a timbre is first loaded. If you don't touch the source control before playing keys, Regen will likely not have received a MIDI packet for that source; in which case, it will assume that the source is in it's default start position:

Mod wheel (CC 1) is set to 127.
Volume Pedal (CC 7) is set to 127.
Pan Pedal (CC 10) is set to 64.
Expression Pedal (CC 11) is set to 127.

You can save start positions in timbres if you wish to override the default. To do this, move the source controller to the required position before you tap the **Save** option.

It's sometimes useful to do this, for example, The Life Synth [AM 42] has both Mod Wheel (c1) and Expression Pedal (cc11) mapped to filter cut-off. This is so that keyboardists without an expression pedal can use the mod wheel instead. If you want to use the expression pedal to vary cut-off, you get the best useful range, if you set your mod wheel to off (zero), so that is obviously a better starting position for this timbre than the default 127 for mod wheel.

Modulator Miscellany

There are a few more characteristics of Modulators worth mentioning:

- If you set Tuning as a route destination, you'll have an option for **Bend Depth**, which corresponds to how many semitones the full range is.
- If you set Chorus/Detune as a destination, Regen will modulate **Chorus** on Additive and Samples partials and **Unison Detune** for Subtractive

Supersaw partials, and if your Supersaw is 1-voice, it will add a Chorus voice for you.

- If you set Stereo Spread as a destination, you have range settings for all 12 partials. As covered previously, TIMBRE **Stereo Spread** works by adding an **offset** amount of panning to each partial, determined by the partial's position in the timbre and what value you set this parameter to. With the use of a modulator, you can limit the range of the **offset** on a per partial basis. Additionally, for Supersaw partials, the **Stereo Spread** of the Subtractive controls will be modulated by this destination.

- If you use Crossfader as the source, this will commandeer the Partial Crossfader, effectively rerouting the output of the crossfader to that modulator rather than its default volume destination. This is particular useful if you wanted to make a parameter a function of keyboard note. With the Partial Crossfader's input set to Keyboard, you could select Crossfader as a modulator source and then Tremolo Depth could be the destination; the higher the note struck, the higher the tremolo. And of course, this effect could be tailored on a per partial basis by adjusting the modulator ranges. NB: if you want the Crossfader to still act on volume, but additionally use its input for tremolo, you would use two modulators set to Crossfader source, with one destination set to Volume and one to Tremolo Depth.

- You can route an LFO A to an LFO B parameter (Depth, Rate, Attack) or B to A, but you cannot route A to A or B to B to avoid circular dependencies.

- If you map, say, Mod Wheel, to Starting Frame and Ending Frame, you'll effectively solo a frame — which one is based on Mod Wheel position. While sustaining a note, Mod Wheel would scrub through frames, honouring the fade times.

Many more creative uses of Modulators can be found. Here's a few examples:

Dueling Harps [SDT 4-1] routes an LFO to Pan so that the two harp samples move in the stereo field and cross each other at the midpoint. It also uses a modulator to route a volume envelope to reverb, so reverb increases as the

timbre sustains NB: this modulator takes its source from the last played note of the track, since reverb is a track-level effect.

Leftfield Organ [RW 5-3] routes a Volume Envelope to Stereo Spread, so that the stereo field of the sound widens as each note is played;

Combining Samples [SS3 3-5] routes Mod Wheel to Chorus, allowing you to dial in your required chorus level (and resulting brightness it brings to the sound), or modify this level while playing, for an ethereal texture;

Pressure Chime [SDT 5-8] routes Pressure to Filter Cut Off **and** Filter Resonance, so when a little bit of aftertouch is applied, a fizziness is added to the bell decay sound;

FM Swell [SDT 3-8] routes Mod Wheel to Filter Cut Off. Since the note filter is low-pass, this lets more sound through. Mod Wheel is also mapped inversely to Volume, so the overall volume doesn't change as you vary the Mod Wheel position, only the character of the sound.

Noise Scorch #3 [RW 8-1] routes Mod Wheel to Filter Cut Off **and** Filter ADR (varying the duration of the envelope), and adjusting the mod wheel while playing the timbre sounds like tuning a short wave radio, because frequencies *jump out* from the noise; and so on.

So there's a tonne of useful stuff that can be done with Modulators. With all the possibilities for routings and ranges, it will take time to craft a timbre to do what you desire, but the work shouldn't stop there. It will also take time to become skilled at playing the timbre, especially if you have an expressive controller with aftertouch or an MPE keyboard (see below).

An acoustic (real) instrument takes years of practice to master. Think of a violinist applying vibrato. The fingers learn just how much pressure and movement is required on the strings to get the instrument to *sing*. Now imagine you have routed Pressure to Tuning (or even Vibrato Depth). The first time you play a piece likely won't be as good as the fifth time or fiftieth time, because you need to learn the feel of the aftertouch and hear its effect.

Even two weeks of practice on a timbre is a mere fraction of the time compared to an acoustic equivalent, and although the Regen doesn't pretend to do the same things an acoustic instrument can, the power of the modulators means there's a lot of expression to be coaxed out under an experienced pair of hands.

MPE Controllers and Keyboards

The MPE protocol defines 5 dimensions — Strike, Press, Glide, Slide, and Lift — and each note, as it is played, provides an individual value. So whereas a traditional keyboard would have a global value for pitch bend that would affect every note being played, MPE sends individual pitch bend messages for each note. Most standard MIDI keyboards do send velocity levels with each Note On message; and a few MIDI keyboard models have polyphonic aftertouch that send individual pressure values; but individual pitch bend and slide per note provides a whole new level of expression for the player.

The following table shows how these 5 dimensions are interpreted by the Modulators.

MPE parlance	Modulator From field (source)

Strike	Velocity
Press	Pressure
Glide (x-axis)	Pitch Bend
Slide (y-axis) cc74	Mod Wheel *
Lift (Release)	Note Off velocity is not supported

* Slide being interpreted as a Mod Wheel position is very effective for Synclavier timbres. Mod Wheel is often used to add volume, vibrato, or alter a filter cut-off; actions that open a sound up; and to be able to slide up the keys to achieve this, per note, provides immediate expression for MPE players.

6.9. Resynthesis

Developed by New England Digital in the 1980's, Resynthesis is a technique to create sounds with precisely-controlled harmonics that vary over time. Think of it as sampling before there was sampling. Back then, memory was expensive, so storing samples was impractical.

Now, memory is not in short supply, but resynthesis is still useful for a variety of reasons. Perhaps you're craving the characteristic of a resynthesized sound. Similar to early speech synthesis, the process imparts a low-fi artificial character to the sound. Or perhaps you wish to transpose the sample more than an octave without it breaking down. Maybe you want to elongate a sound by looping the sustain stage more easily. Or totally mess with the sound's tuning and FM over its duration. And maybe you just want to experiment to get something that sounds unique. Of course, once you've resynthesized your sample, you'll have saved vastly on memory and disk space, which will speed up loading and reduce CPU impact, so no bad thing. And each note played will have the same duration, as opposed to a sample that will complete quicker the more it's pitch-stretch up.

Resynthesis has been optimized for simple samples. It works by breaking down the sample into small frames that can be represented by a set of additive harmonic coefficients and phases. If the soundfile is long and complex, or lacks harmonic content, this cannot be done.

Rich sounds with lots of harmonic content, for example a piano, is not going to resynthesize well, if at all. There's too much going on. You'd have more success with something like an clarinet. And percussive sounds like cymbals have a great deal of non-harmonic content, so those types of sounds will typically yield bizarre results. And samples that are constantly changing with rhythmic or melodic content, are best left as samples.

On Synclavier Regen, resynthesis is easy to accomplish, as the following walkthrough illustrates:

How it works

A resynthesized sound is referred to as a Hybrid waveform. You can have one per partial. Start by loading a sample onto your partial, for example, Wine Glass #1 from Selected Legacy Samples. NB: If there is more than one soundfile on the patchlist, all but the first will be ignored. The **Start Time** and **End Time** of the soundfile will be honoured, so that a section of a sample can be resynthesized.

Select Analyze under **Synth Mode.** If your sample isn't tuned, now is the time to tap **Auto Tune** to tune it. Then tap **Next**. You'll be presented with:

Frame Count | **Hybrid Attack** | ⚡⚡

The **Frame Count** is how many frames the sound will be spilt into, so for longer or more complicated samples,

you would set a larger number. For Wine Glass #1, a relatively simple sample, we can stick with the default of 20 frames.

Hybrid Attack is how much attack of the original sample to play before the sound blends into generated additive frames. You can select 0ms for a completely resynthesized sound, but for more realism, you would keep a small slice of the original sample at the beginning. If you are going for realism, you should know that the brain stops paying so much attention after the attack of a sound — it's already decided, this is a harp, or this is a cello, for example, by the time the attack phase has completed — so by keeping some of the original sample at the beginning, you can sometimes, but not always, create a more convincing resynthesized sound, assuming that's what you're going for. For the purposes of this walkthrough, select 20ms.

Tap the ⚡⚡ button and Regen will analyze the sample and create a "Hybrid" waveform for you. Once the BUSY light shuts off, the partial is ready to be played. Try playing a few notes. You'll notice that the basic character of the sample has been preserved, but the attack is less percussive. Our wine glass sounds more flute-like.

NB: If you get the message "Sample Pitch Could Not be Analyzed," it means that the sample selected is too complex to be analyzed and resynthesis has failed.

Once analyzed, you can easily go back and forth and compare to the original sample, by changing Synth Mode back to Samples (or Analyze). Try experimenting with different resynthesis values — you'll have to click the ⚡ again to regenerate the frames. You'll find the higher the **Hybrid Attack**, the more of that percussive element is preserved, and the higher the **Frame Count**, the more authentic the result, especially in the higher harmonics.

Once the Timbre Frames have been created, they can be edited with the **Frames** button — see the following Frames section to dive deeper.

6.10. Frames

NB: we introduced the concept of Frames in the previous section on resynthesis, since resynthesis relies on frames, but this section covers all the parameters and some possibilities of using frames for sound design.

Let's dive right in with a look at an example.

Call up the timbre, Looper [IN 6-8]. Then select Partial 2 and tap **Frame** and you'll see this display with the following options:
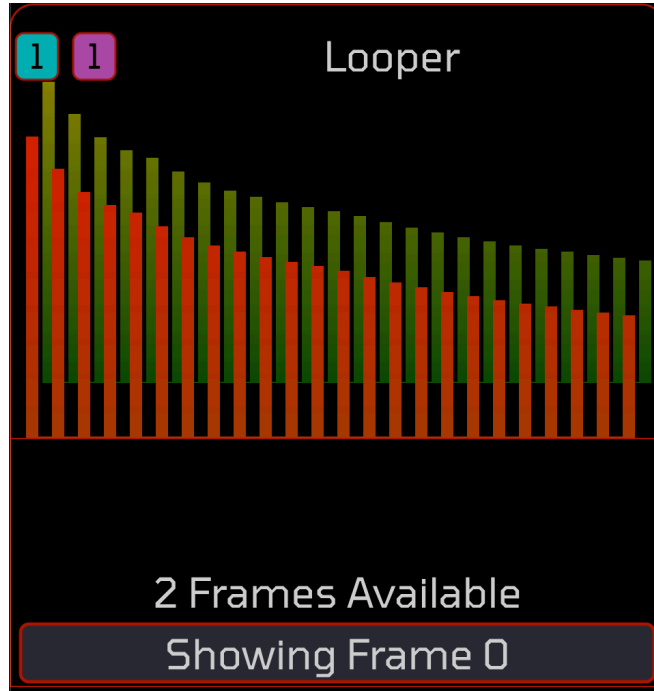
Clone / Delete | Change Splice | Change Vol/ Mod

This timbre has 4 partials, each with 2 frames.

The perspective display illustrates how the harmonics vary over time. The first Frame is called 0 and is highlighted in the above graphic. Tap the **Nudge Up** button and you'll see the set of harmonics "behind" are highlighted and the message now reads "Showing Frame 1."

In this example the harmonics are the same across the 2 frames, but frames can also have different volume, modulation and tuning. In this timbre, the sound designer has altered the pitch frame 1 for partials 2 and 4. Solo partial 2 and press and hold a key. You'll hear the the pitch shift up (and if you hold the note long enough,

you'll hear it shift back, since this partial's frame loop mode is set to a loop mode).

Now tap **Change Vol/Mod** and you'll be presented with the sub pages where frame modifications can be made, you'll have the options:

Frame Select | **Frame Volume** | **Frame FM%**

If you **Arrow Right** you'll have the options:

Frame Select | **Semitones** | **Cents**

On this second sub page use the **Swiper** or **Nudge** buttons to switch between the two frames and you'll see there's a pitch increase of just over an octave. Change **Semitones** to +36, play a note and you'll hear an even more intense shift. Now let's adjust the timing. Tap **Back** to go up to the main **Frames** page then tape **Change Splice**. Select Frame 1 and you'll see there's a 2.1 second delay before the pitch shift occurs then there's a crossfade of 960ms. Dial that down to 240ms to make the transition more abrupt and dial down the **Delay Time** to the same amount. Play a note and you'll hear the sound has changed quite a bit, it's more like a see-saw effect. But let's change things even more.

Still solo'ing partial 2, tap **Define**. You'll see the familiar harmonics page for the Additive Synthesis generator, but with a twist: the line below the harmonics shows which frame we're viewing. On this page, you can again

use the **Swiper** and **Nudge** buttons to select which frame before you tap an **Edit** button to adjust harmonics.

Okay, so let's select frame 1 then **Edit 1-8**. Strike a keyboard key and you'll notice that on this page, when you're in **Edit** mode, you're solo'ing the frame, so you're just hearing the pitched-up frame. Toggle **Edit** again to hear the whole partial. Let's make the second frame more harmonically rich, dial up all first 8 carriers to max. Recall that the harmonics are normalized, so we haven't increased the volume, just the harmonic content, brining the second frame closer to an impact wave than the original ramp shape. You can change the modulation waveform too. Why not do that now: tap **Mod** then edit harmonic 1 and reduce it to 0% for a cleaner sound. Tap **Frame** again and you'll see how the perspective view of the harmonics has changed.

There's near-infinite possibilities for what you can do with frames, and this example just used 2 frames. From the main **Frame** page if you tap **Clone / Delete** you'll get the options:

Frame Select | **Clone Frame** | **Delete Frame**

and if you Arrow Left from here, you'll get the options:

Loop Mode | **Loop Start** | **Loop End**

Delete is self-explanatory. **Clone** will create a new frame ahead of the currently selected frame, with the current frame's parameters cloned as a starting point. You can have a maximum of 100 frames. **Loop Start** and **Loop End** are also self-explanatory and the **Loop Mode** gives you the loop options of ping-pong, or bounce-back (which should be obvious from the labels), with or without tails. Without tail means that when a key is released, the partial would end at the

current loop cycle, but with a tail, it would continue looping throughout the key release phase (a la the previously-discussed sample loop mode).

Other examples of frames can be found throughout the preset libraries, or you can start creating your own (from scratch or from a resynthesized sample).

6.11. Layers

So far, when examining the sample patchlist we have indexed each sample by a keyboard range (or single key). In version 1.10 of Regen firmware, the dimension of layers was added to the patchlist. The index for the layers is nominally Velocity, but it can be changed to Mod Wheel or Random.

The layers page can be found between the **Wave** and **Edit** buttons (by navigating there or pressing both buttons simultaneously). If you call up a sample from the library then navigate to it, you'll see this:
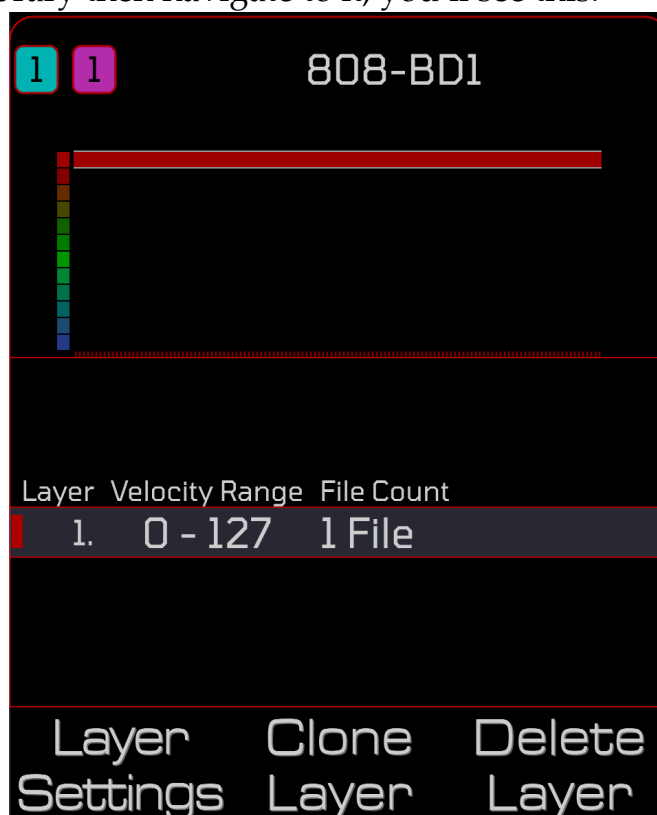

The default setting is to have one layer — shown in red — that is active across the whole velocity range (MIDI values 0-127). If you then tap on **Layer Settings**, you'll get a page similar to this:
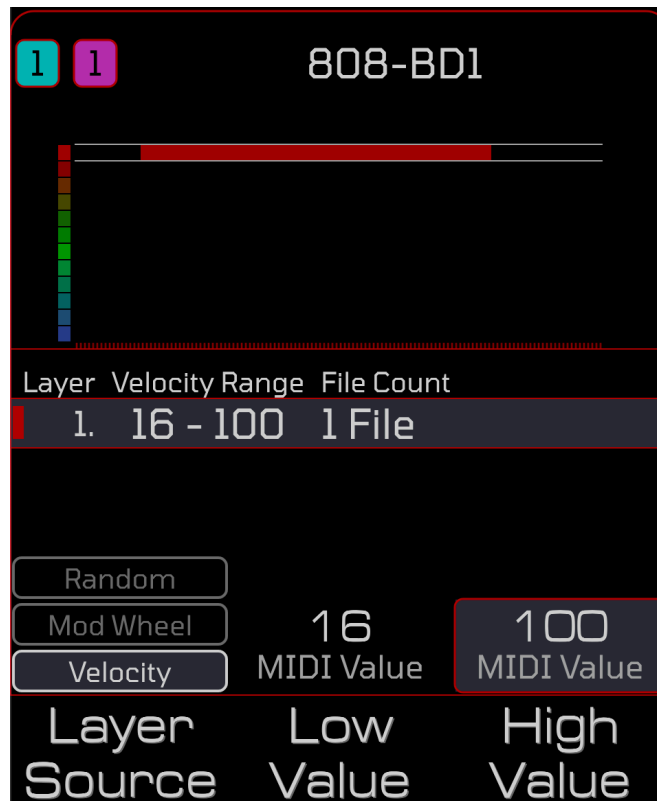
If you change the **Low Value** and **High Value** you'll see the active area of the red bar diminish. If you play a key, it will have to be struck in the chosen range of velocity to activate any of the soundfiles in that layer.

   To return to the main layers page tap the **Back** button. From there you can add additional layers, and as you do, you'll notice you initially get a "**Conflicting Layers**" warning message. Just like when setting the patchlist ranges, if two layers overlap then Regen doesn't know which one to activate, so after cloning a layer, its range will need to be set.

   Here's an example with 5 layers called Chromophone:
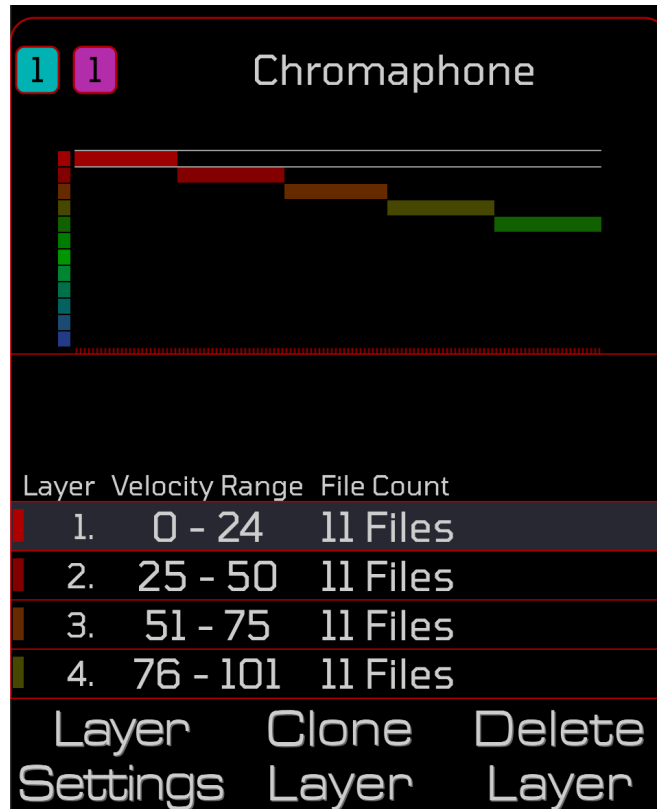
All 5 layers are shown on the graphical portion of the display and are seen to be contiguous with no overlap. The list shows the details of the first four layers (you can **Nudge Down** to view the fifth). The velocity ranges are given in the details and in this example, each layer contains 11 files. This can be confirmed by navigating (**Arrow Right**) to the Edit page, as shown below:
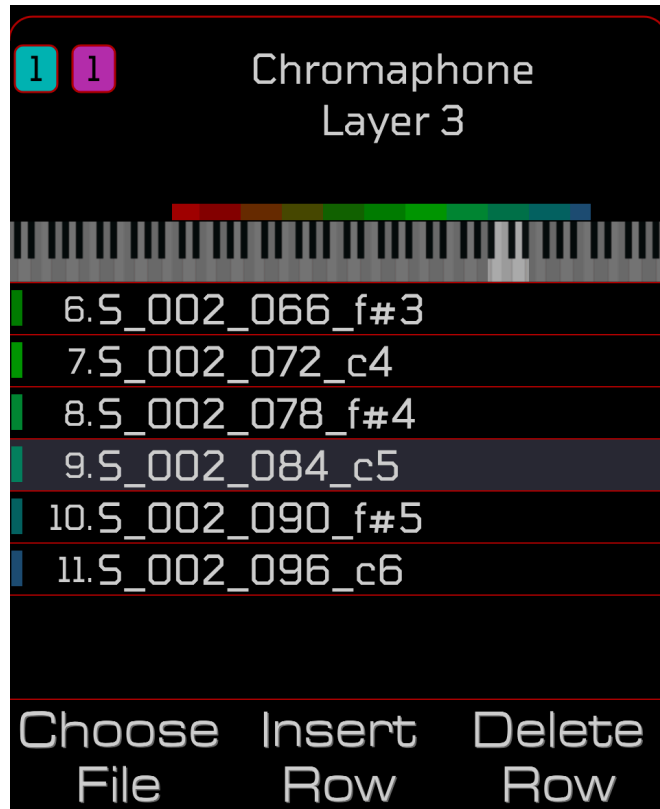
The Edit page shows the 11 soundfiles that can get triggered if the velocity is between of 51-75 (inclusive), and each one of these 11 files has a different keyboard range specified.

> While perusing the range information on the Edit page, if you have multiple layers, you can navigate them using the **Load Up Load Dn** buttons.

This extra dimension to patchlists is, of course, most useful for multi-sampled acoustic instruments. You'll recall from earlier discussion on sampling instruments that more realism can be gained by having sampled multiple velocity levels as well as multiple keys, especially for dynamic instruments such as a dulcimer or piano. The Dulcimer #2 [SDT 5-2] timbre was an example given where 3 partials are used to store samples at different velocities. With the addition of the layers dimension to the patchlist, only one partial would be necessary for this.

However, recall also, our discussion on crossfading. Many acoustic instruments are highly sensitive to velocity input and many different levels would need to be captured to recreate a realistic instrument. If you have

insufficient layers — such as the case with the dulcimer preset only containing 3 levels — then you'd be well advised to stick to the partial method, so that the velocity layers can be blended into each other with the Partial Crossfader.

Imagine playing a piano passage where you think you're striking the keys at roughly the same velocity, but halfway through a melody line, you go from 50 to 51. In the dulcimer example with 3 layers, it's going to trigger different samples for you. You'll likely have velocity mapped to volume, so the volume isn't going to jump too badly, but it may still sound odd to trigger a different sample with its own timbre. As a rule of thumb; 5 or more velocity levels is good, 4 an absolute minimum, 3 you're going to need to use the crossfader approach; but each instrument will be a little different.

A usage of layers that is not so sensitive to this effect is percussion. Drums are often sequenced over MIDI, so a pattern has a fixed set of velocity values for every bar — no surprises — but velocity can also be used to add variety, and in the case of percussion sounds, authenticity can be gained by having a variety of samples from the same drum or cymbal.

And velocity need not be used as the index for layers. The **Layer Source** could be changed to Mod Wheel, in which case the **Low** and **High** values correspond to the position of the Mod Wheel. Or layers could be triggered randomly, and Regen will pick one of the layers at random for each new note. NB: With **Layer Source** set to Random, **Low** and **High** values do not need to be set, each layer is weighted equally.

SoundFonts

So, if the procedures outlined above sounds tedious to you, do not fear. Synclavier Regen is capable of recognizing and loading Sound Font files in the SF2 format. This is a standard format for containing batches of samples indexed by keyboard range and velocity level. You can purchase packs of soundfonts from many sample vendors and many soundfonts are available for free on certain websites.

There are also some excellent computer based soundfont editors — again, some paid for and some free. The benefits of performing the intermediate step of

organizing your samples into a soundfont on a computer will be become obvious after use: not only will you be able to use a bigger screen, but you will be using software that was developed specifically for this task.

The Polyphone editor is an excellent place to start if you wish to explore soundfonts. The editor is free to use and cross compatible, and the polyphone website (https://www.polyphone-soundfonts.com) contains many good examples of soundfonts from the sampling community.

SampleRobot is also worth checking out – https://samplerobot.com/pages/samplerobot – this is a professional multi-sampling assistant with transient and loop editing, as well as a sample librarian with 1GB (pro version) of high quality sound font files included.

Once you have your samples in SF2 format, you can copy the `.sf2` files to a library folder on your SD card. The `.sf2` file contains within it all the necessary samples.

Select your user library on Regen and tap the **Samples** button. As you scroll your list, you'll see any soundfont files recognized shown in a peach colour. Press **Enter** to call one up and it will configure itself into the timbre, overwriting any previous patchlist information with the mappings from the soundfont. From then on, it will act like a Regen timbre and when saved will be stored in the Regen preset format (leaving the soundfont as reference for the samples or saving the samples out to your SD card if necessary).

TROUBLESHOOTING

Before troubleshooting, please check our website at www.synclavier.com/regen-downloads for software updates. Instructions on how to update your Regen to the latest firmware is given there.

#

No Sound

- Try switching on the **Audio Test** function.

  - This will narrow down the focus to the outgoing chain: Regen will output audio on the currently selected Audio interface regardless of incoming MIDI messages.

  - Continue to fault-find the issue with the suggestions below. When you have found the issue, you can switch the test off.

- Check cabling

  - Are all the cables in the correct sockets?

  - We have found that certain ¼" connectors, especially the TRS plugs used on headphones, can vary in tip shape. If wiggling the connector temporarily fixes the problem, you may have a connector that is out of the usual tolerance, so please try using better quality cabling.

- Are you using an external audio interface?

  - Check your interface has sufficient power.

  - Check Regen is in Host mode.

- Are you sending the correct MIDI information.

  - Check that the MIDI channel you're sending if going to be received by one of your tracks, but checking the Tracks Crossfader parameters.

- Did you check all Regen levels in your chain?

  - If any of MASTER Volume, track volume, timbre volume, partial volumes, or soundfile volumes are too low, you may not hear the sound.

Bad Sound

- Do you have a mains-frequency hum?

    - If you're not using balanced outputs, you should switch to using them if you can. If you can't, ensure the GND LIFT switches are both set to GND.

    - If you are using the balanced outputs, try changing the GND LIFT switches on the rear of the unit. Also check the receiving gear to see if it has similar switches.

- Are you experiencing a clipping distortion sound?

    - **Check the VU levels under the MASTER Volume** page. If, when playing a track, the meters are red-lining then you should dial back the MASTER Volume or that specific track volume.

    - You may be over-driving your receiving gear (e.g., mixer, preamp), so try dialling back the MASTER Volume.

- Do you hear nasty random noise distortion?

    - This could be cross-interference from an attached peripheral, such as an external MIDI keyboard or sequencer. Does the distortion go away when you unplug a certain device? In which case, you could try a USB isolator, or service this device.

Uneven Sound

- Is one channel, left or right, higher in volume than the other?

    - First check the pan setting on the track/s you are using. Double tap **Pan** to reset to centre.

    - Try a different factory timbre or use the test tone to check stereo image. Perhaps the sample or timbre you are playing is weighted more on one channel by design.

    - If this affecting the headphones output, try adjusting the headphone volume. If Regen was previously shut-down while sending information to the headphones volume control, then it could conceivably become unbalanced, and an adjustment to its volume will correct this.

- Is the sound out of tune?

    - If you're using a USB audio interface, check the sample rate matches what Regen reports on the Audio Settings page. If your audio interface as a control panel it may be possible to change the sample rate without Regen's knowledge. It is better to set the sample rate from the Regen Audio Settings page.

#

MIDI Device doesn't show up

- Are you in DAW mode?
    - Check on the Settings page from Home. You need to be in Controller mode to use the 4 USB A ports.
- Is your MIDI device USB "bus" powered?
    - If it is, you may be exceeding the 1 amp that Regen can supply (shared) to all its USB ports. You should try using a powered USB hub.
- Is your MIDI device self-powered?
    - Is the MIDI device's power supply plugged in, switched on, and working?
- Is your MIDI device class-compliant.
    - Most USB-2.0 or later devices are, but it's worth checking. If not, it may not be compatible with Regen.
- Are you drawing too much current?
    - Regen can handle 1 amp total on its 4 USB A ports. More than this, and all 4 USB ports are shutdown as a precaution against faulty USB hardware. Remove your USB devices one by one until you find the culprit, however you'll, have to restart Regen each time, because the over-current switch latches off. If there is a genuine reason for that accessory to draw a lot of power — for example, it has a lot of LEDs, it generates audio, etc. — then you will have to use a USB powered hub between Regen and the accessory. If you don't think the accessory should draw more than 1 amp of current, then it may be faulty and need replacing.

#

Notes played are not what is being heard

- Tap MASTER Volume then Arrow Right to the info page. MIDI notes should appear as they come in. Are you getting any notes?

  - If a keyboard is connected, but no notes appear, to deselecting and then reselecting your keyboard on the MIDI interface settings screen.

- Do you have MPE selected for the keyboard you're using and it's not an MPE keyboard, or vice-versa?

  - Change MIDI settings under home page.

- Are you playing a timbre that does something strange to the pitch?

  - Check settings such as **Pitch Track** and **Octave Ratio**.

  - Try a different timbre and see if you still have this issue.

- Have you filtered out the MIDI channels being used?

  - Check the Track Crossfader settings for your active tracks.

- Are all notes being transposed to the same MIDI channel?

  - If you have a non-MPE controller (keyboard), check you have the MPE setting switched deselected for that controller on the MIDI Inputs page under Settings. If not, then all notes will indeed be sent to the specified MIDI channel, regardless of the MIDI channel settings of your keyboard.

A useful page to track down MIDI issues is the display under MASTER **Volume**. Tap that button and then Arrow Right to view the multiscope display. All MIDI notes received will appear in that display.

#

A note doesn't end

- Do you have a note filter with high resonance?

  - If so, it could be self-oscillating.

- Do you larger reverb tails? Or long release times on a partial?

- Are you playing long-samples with one-shot mode?

- You may have a stuck MIDI note. This means a MIDI interface has supplied a Note On event for a note but not a following Note Off event.

  - In this case, press Home twice and all tracks will have any playing notes reset.

Display screens are blank or cycling different colours

- Wait 1 minute

  - If it doesn't resolve itself, power cycle the unit.

> **Error message on Regen Screen**
> - "Audio I/O Error: Missing Clock"
>   - If you are using a USB Audio Interface, it might not be fast enough to keep up with the Regen clock. Try increasing the latency value on the Regen settings screen.
> - "Audio I/O Error: Data underrun"
>   - This message means that Regen is struggling to keep up with the demand of notes in time for audio generation while abiding to the 1ms latency rule. Check the CPU meter under the MASTER **Volume** page. If you are *red-lining* the CPU you will have to reduce the load to avoid this error, either by reducing the number of voices (fewer simultaneous tracks and timbres or fewer incoming MIDI notes) or reduce the number of reverbs and filters applied.

#

If the above guide fails to help, you could try Google or Youtube to see if anyone has had a similar problem and solved it. Or, please get in touch with us. There is a messaging system on our website and messages sent to us are checked regularly.