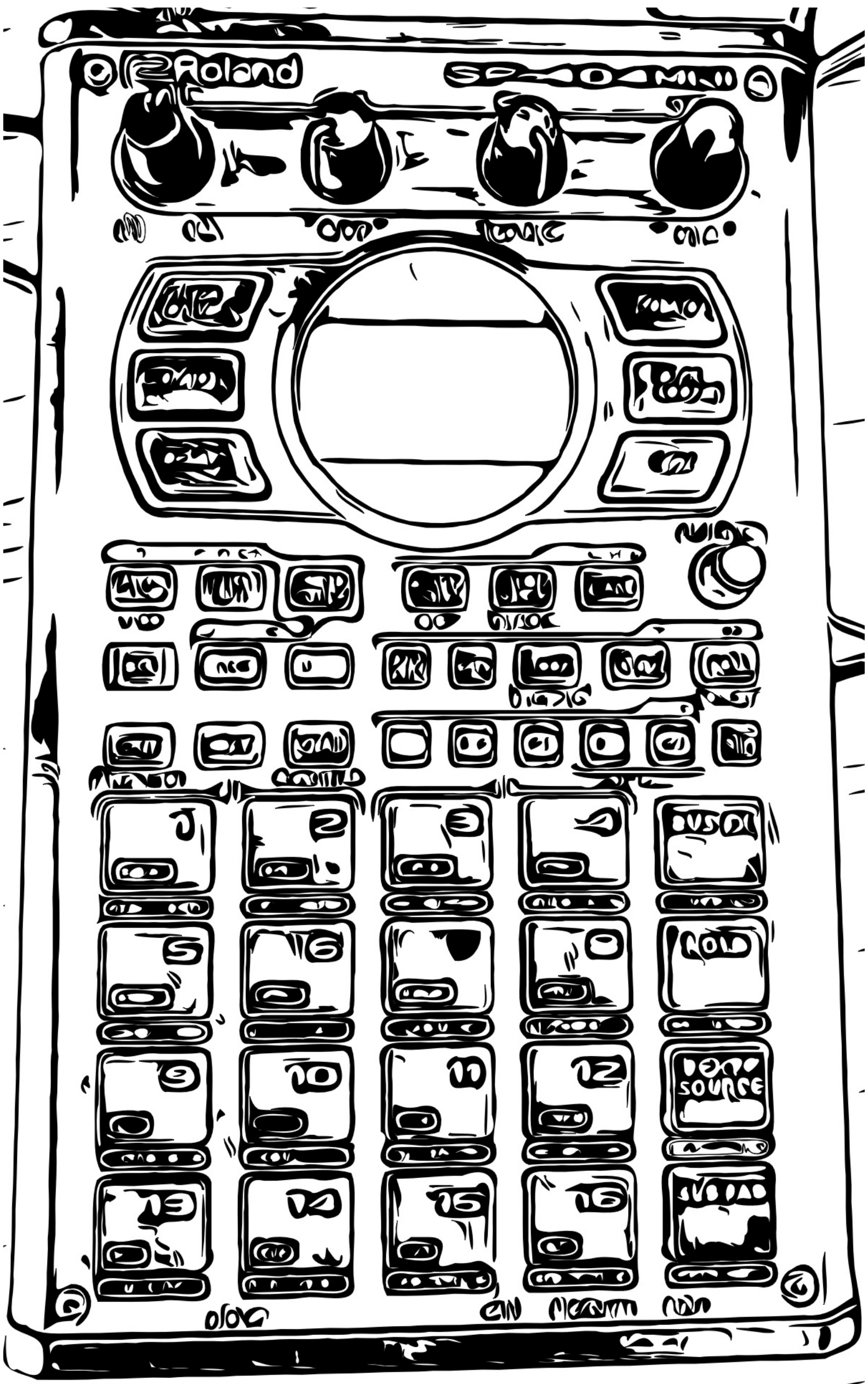
NearTao’s Guide to SP-404 mk2

An Unofficial Reference for Firmware 1.14



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Guide Versions

Guide Version	Mk2 Firmware Version	Change Date	Notes
0.17	1.14	Feb 18, 2022	Massive Effects section update, needs more work, but tried to give as much info as I could manage
0.16	1.14	Feb 10, 2022	Birthday update… fixed tables floating… Apple Pages you wonky… finish Skip Back section, a few various updates, Rename Mixer to Audio Signal Flow, finish Audio Signal Flow section.
0.15	1.14	Feb 9, 2022	End Snap update, added copy/delete into Sampling section, finished Sample Mode section
0.14	1.14	Feb 2, 2022	Added some notes on donations, fixed bump map to bitmap (hah), finished Sample Edit section
0.13	1.14	Jan 26, 2022	Filled out the Sample section
0.12	1.14	Jan 23, 2022	Fill out Data section, filled in a few extra questions.
0.11	1.14	Jan 18, 2022	General Edits, remove Company/History sections (for now), added Guitar Pedals section under MIC/GUITAR Input section, Added an Audio Output section, new FAQs, More content in the Audio Input section
0.10	1.14	Jan 17, 2022	Filled out and formatted the Frequently Asked Questions section.
0.9	1.14	Jan 13, 2022	Update about USB-C to lightning, update to Polyphony information (its 32 voice polyphony for mono), Added Polyphony experiment.
0.8	1.13	Jan 11, 2022	Mic/Guitar Experiment, added phono subsection, made Inputs a top level section separate from Sampling, Power section USB-C info/bugs, and battery bugs, added References, fixing nits where possible
0.7	1.13	Jan 10, 2022	Gain Staging to a point, attempt at some guitar info, experiment for digital clipping,
0.6	1.13	Jan 9, 2022	Sampling stubbed Gain Staging, filled out Controls, started MIC/GUITAR
0.5b	1.13	Jan 8, 2022	Reworking sections (Specificaitions/Sampling)
0.5a	1.13	Jan 8, 2022	Added links/contact info into the Overview section
0.5	1.13	Jan 8, 2022	Finish Spec section, add some contact info, decided to put this out there as a free guide
0.4	1.13	Jan 7, 2022	Move Getting Started to the top, tweak formatting
0.3	1.13	Jan 6, 2022	Flesh out Getting Started/Conventions, and add icons, fill out Specifications section.

Guide Version	Mk2 Firmware Version	Change Date	Notes
0.2	1.13	Jan 6, 2022	Add title page, start Overview/Getting Started
0.1	1.13	Jan 5, 2022	Initial Document Structure and Outline saved

Foreward

Not sure what to put here yet (or remove the section), but Firmware 1.14 just dropped, and that's going to cause me to make some changes to this guide. First big impact, is that there is a pdf manual now, which is going to make it a lot easier to write this document, without having to re-write every detail. This is a great change, and should let me focus more on workflows.

There are also a lot of great changes with the 1.14 firmware, and it is going to take some time for me to absorb them and figure out the best way to discuss using them in various ways.
* Pad Mutes are a big ++ from me
* The End Snap feature was a bit of a head scratcher, but I got if sorted out :D

A little bit more than a month in (Feb 2022), and I have to say… the guide is both further along and not as far along as I'd like. I keep marching towards getting some workflows down, but there are so many little things to make sure I understand how they work that it's a bit of a task to run everything down. And yet… I know I am also missing plenty of things.

Contents

Getting Started

You've got this book in your hand, a tablet, or loaded up on your computer, and are wondering how in the world you can get around. Well, the order of sections is somewhat arbitrary, so while I tried to put it together in a way that made the most sense to me, you should absolutely feel comfortable going through it in whatever way makes the most sense for you. Skip through sections, consult what you're interested in now, and come back for the rest later.

To keep things simple, I am creating a dedicated URL on my website, starting a thread on SP-Forums, and using my discord server.

NearTao Blog: https://neartao.com
NearTao Guidebook URL: https://neartao.wordpress.com/neartaos-guide-to-the-sp404-mk2/
SP-Forums URL: https://sp-forums.com/viewtopic.php?f=24&t=27232
Discord Server URL: https://discord.gg/qMuSpuxC4n

On the subject of donations, it's really not necessary. Say hi on my discord server, post a friendly comment on one of my songs, surprise me. I'd be much more interested in seeing people paying it forward, for example by donate to something local, call somebody you haven't talked to in a long time, or do something nice for some one just because you can. Reach out, I'd be super excited to hear what you've done!

If you read through all that and still want to donate something, use the link below. I may also start a cassette tape collection as I've started sampling off of them again… but who knows.

Donations: https://www.buymeacoffee.com/neartao

One other thing that comes to mind is that I've started collecting tapes again. Reach out if you're looking to get rid of some :D

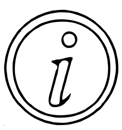
Conventions

The SP-404 mk2 has a lot to keep track of, so I wanted to come up with some consistent conventions for giving directions to operate it. This should help improve the clarity to navigate this device, which while it isn't terribly complicated, does have a decent enough set of controls that I want to have a quick short hand for us to be able to work together. I'll provide pictures where it feels appropriate, but I don't want to over burden this book, and the editing of it going forward with lots of images.

Icons

Throughout this book I will try to remember to use standardized icons to help present important information and warnings as they make sense to note and identify.

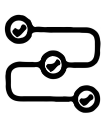
Warning: I will use this icon to indicate a brief section as a warning or heads up. This section will contain details that may not be well documented or are not well understood.   This will typically indicate something to be aware of on the mk2.



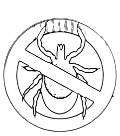
Information: I will use this icon to indicate a brief informative section. This will be something that is nice to know, but may not be essential for the operation of your SP-404 mk2.

Under Construction: this indicates that a section is still being considered, researched, or in the process of being rewritten. Don't be surprised if this information changes, moves, or is just removed.

Experiment: This is indicating that I have (or possibly intend) to run an experiment to get more details on how something works, or more specific information about the operations of the mk2.



Workflow: Indicates a short list of buttons presses or other things necessary to get a desired result



Bug: Indicates that there's something that probably isn't working quite right, not as documented, or not as expected. This is something I would expect Roland to fix on the mk2 eventually.

Controls

(VOLUME|CTRL1-3 CW|CTRL1-3 CCW) will all be referenced within parenthesis ( )
* Where CW is turn the knob clockwise
* Where CCW is turn the knob counter clockwise

/FX/ will all be referenced within forward slashes / /

|BUTTON| will all be referenced within pipes | |

{VALUE CW|VALUE CCW|ENTER} will all be referenced within curly braces { }
* Where CW is turn VALUE clockwise
* Where CCW is turn VALUE counter clockwise
* Where ENTER is push VALUE

[Pad 1-16|SUB PAD] will all be referenced within brackets [ ]
* These are velocity sensitive pads that can be effect volume based on how hard they are hit

<AUDIO OUTPUT> will all be referenced within < >
* Headphone Jacks
* L/Mono and R

>AUDIO INPUT< will all be referenced within > <
* L/Mono and R
* Mic/Guitar

Mic/Guitar & Power Switch?
Mic/Guitar Gain knob?
USB-C port?

`SUB FUNCTION` will all be referenced within back ticks ` `
* This will be used to typically denote functions accessed by hitting a pad/button twice, or pressing pads/buttons at the same time.

-> will denote pressing one pad or button followed by pressing another pad or button

+ will denote holding a pad or button while holding another pad or button

Examples

`CHROMATIC` |Shift|+[Pad 4] - Hold the Shift button and then press Pad 4 to enter Chromatic Mode

`F` |A|->|A| - Press the A Bank button twice to enter the F bank (honestly, don't do this one)

More examples like REC or… ???

Overview

I don't know about you, but I like guides. The *Hitch Hiker's Guide to the Galaxy* and *Zen and the Art of Motorcycle Maintenance* meant a lot to me growing up and I have read both books multiple times. Although I highly doubt that this book will have the level of social influence either of those books have had, I do hope that this book can help drive people to getting the most out of their Roland SP-404 mk2.

Super excited that Roland has launched a pdf manual for the mk2. Hopefully this will make things much easier for me to reference in the future, and be a good way for me to not have to replicate all of the information Roland has put together.

I am writing up this guide with the following goals.
* Give a solid overview of what the mk2 is and is not
* Show users how to use the mk2 and point out common pitfalls and mistakes
* Provide guidance and workflows to help make mk2 users get the most out of their instrument
* Reference external information where possible for users to do their own further research

Similarly there are a few things I am not trying to do with this book.
* Replace the existing Roland manual

Specifications

Get ready, this is all of the information from the data sheet, with notes that I have sprinkled around, as well as additional information for context. There are some things I am not entirely sure about, especially for specific language that Roland has chosen to use, or how ranges for some things are managed when driven externally from the mk2. As I get more documented I'll work out some experiments, tests, or do further research and reference back into the manual, community, or anywhere else to try and nail down this information.

Sampling

Polyphony

The mk2 is specified as having a maximum polyphony of 32 voices, but doesn't state whether each voice is monophonic or stereo in the manual. I have confirmed that the 32 note polyphony is for a monophonic sample, so if you use only stereo samples then you're going to have only 16 notes of polyphony since a stereo sample requires two mono voices.

Document an experiment and save off a project for readers to follow along with how we figure out if this is mono or poly.

Guess is that a voice is stereo… and 32 note stereo polyphony is pretty good.

Internal Data

Internally the mk2 appears to handle all samples at a 48kHz sample frequency at a 16-bit linear depth. It is unclear what the file format is for other metadata other than the boot screen(s) which are saved as Microsoft Windows bitmap files.

Exporting a project creates many files, but samples are saved as .SMP, and patterns are saved as .BIN. Pictures are saved as .BMP (bitmap files), and last but not least there is a PADCONF.BIN file, that presumably contains pad settings for the project.

Could be worth digging into these file formats and seeing what can changed/ modified. Seems like a future set of experiments to run.

Sample Import Format

The mk2 natively supports importing .WAV, .AIFF, and .MP3 files. It is unclear if there are sample rate, bit depth, or other factors that are important to know when importing samples into the mk2.

Seems like some good experiments out there to try different .WAV formats, stereo vs mono (or surround sound), different bit depths and sample rates… could be cool, or might be super messy outside of “standard” values.

Using the Roland Cloud SP-404 mk2 App allows you to import WAV, AIFF, MP3, FLAC, M4A files. Maybe anything else? Could be worth researching…

Sample Export Format

The mk2 only exports files as stereo, 16-bit, 48khz .WAV files.

Warning: Plenty of older devices only support 16-bit at 44.1khz. The older devices may happily play a 48khz file, but it will be out of tune because it will be playing at a different speed than the intended 48khz that the mk2 exports.

Skip Back

The skip back features allows the mk2 to record the last 25 seconds of audio from the final output mix (??? What does Roland refer to this as) into a buffer that the user can access to go back to.

The skip back certainly makes the mk2 an interesting end of chain device as it can let you capture snippets of a performance that you may not have been intentionally recording, but if you’re fast enough and want to revisit again to try to recreate or to sample for future use. Over time I think we’ll see this as a formative mk2 feature.

Storage

Data

Internally, the mk2 can store up to 16 projects, each project is made up of 10 banks, and each bank can have both 16 samples and 16 patterns. Some basic math lets us know that internally the mk2 can store up to 2,650 samples and 2,650 patterns (16 samples/patterns * 10 banks x 16 projects).

Internal Storage

The mk2 has 16GB of internal storage. From a factory install, only 14.21GB of internal storage is available, and the lost storage is due to preloaded samples and patterns, settings, metadata, factory restore data, and drive format.

From a factory reset, if you delete project 1 which contains the factory samples and patterns, you will have a maximum available amount of free storage of 14.38GB. You will not be able to address more available storage than this as there is no way to remove any of the hidden internal content necessary for a factory restore.

External Storage

The mk2 supports up to a 32GB SDHC card. While it might support the older SD card format that went up to 4GB, it definitely does not support SDXC cards or sizes over 32GB.

Unlike previous SP devices (202/303/404), the mk2 does not support streaming audio off of the SD Card. It can only be used for backups/restoring data, and importing/export projects and samples.

Maximum Sample Time

The mk2 can have a single sample that is up to 16 minutes long, documented as approximately 185MB per sample.

Pattern Sequencer

The Pattern Sequencer has a resolution of 480 parts per quarter note, which is quite good. Some devices do go to 960 parts per quarter note or possibly higher, but there was plenty of gear manufactured that only manages 96 parts per quarter note.

To increase the resolution for parts per quarter note, you can look into doubling the BPM that you record at. If you were originally going to record a piece at 100 BPM, recording at 200 BPM will effectively get you 960 parts per quarter note.

The Pattern Sequencer can record patterns of 1, 2, 4, 8, 16, 32, or 64 bars.

A technique for older gear to extend the pattern length (at the expense of parts per quarter note) is to halve your BPM, which will allow you to double the number of bars you record. By doing this you can go from 64 bars to 128 bars.

The Pattern Sequencer has a BPM range from 40 to 200.

Not entirely sure how it handles ranges outside of this from an external sync source. Worth investigating.

Finally the Pattern Sequencer supports Quantization and you can apply Strength.

Got quite a few questions here… but going to need some time to experiment, read docs, and better understand how this works.

Effects

The information on the tin states that there are 37 Multi Effects that can be used on the Bus FX or EFX (what the heck does this stand for???). There are a further 16 Input Effects, which mostly seem to overlap with the Multi-effects (will need to verify parameters, but there are 3 unique input effects at this time of Auto Pitch, Vocoder, Gt Amp Sim).

This reads like the mk2 actually has 40 effects at this time, but the Effect Type section shows that there are currently 42 unique documented effects. I suspect that the 3 unique Input Effects will eventually make it into the MFX section, and Roland has hinted that there are likely more MFX coming, so maybe we'll get an extra page or two in the MFX section. Only time will tell.

Interface

Pads

There are two firsts for the mk2 compared to the SP 202/303/404 product line. First and most noticeably there are now 16 pads instead of 8 (as the 202/303) or 12 (as the 404/404sx/404a). Second, the 16 Pads and Sub Pad are now velocity sensitive, which means that you'll be able to vary how loud a sample plays based on how hard you hit the pads.

Control Knobs

The mk2 has continued the 4 knob tradition since the 303, and has retained it's volume knob, and three CTRL knobs. These knobs are rotary knobs and have a clear start point and end point. This means that the knobs all have a definitive position and value based on where you have the knob set. You can somewhat adjust this behavior in your settings to allow for a more relative feel until the knob value catches up with the parameter value that you are controlling.

Value Knob

Finally the mk2 has added an endless/continuous knob that has a push toggle, to allow for relative input as well as an additional commit/enter command. This means that you can endlessly turn the knob clockwise or counter clockwise and it will adjust the specified parameter based on what the software settings the value knob is currently associated with.

My biggest concern for the mk2 currently is the rotary knob. It gets used very frequently, and is something I expect have issues in the future. If you're going to travel make sure that you don't jam the mk2 into a crowded backpack, or bring a separate custom purpose case to help protect your device.

Display

The mk2 has an OLED graphical display now, which although it is not high resolution by any means, is more than sufficient to display audio data, settings, options, and plenty of other information that makes than mk2 easier to work with than its predecessors.

External Connections

There are a number of connections for the mk2, and I'll do my best to break everything down.

Headphone Jacks

Yes you read that right, the mk2 has multiple (well okay 2) headphone jacks. It has both 1/4" and 1/8" headphone jacks, which is pretty handy for using different headphone types, and might be a good way to collaborate with other people with a single device.

The experiment **Do the headphone jack outputs share an amplifier?** Indicates that there is a single headphone amplifier, which shouldn't be a direct concern, other than there will be an overall lower listening volume if there are two connections attached the headphone jacks that is worth being aware of.

A word of warning on the headphones, the **(VOLUME)** knob controls both the main outputs and the headphones, so you cannot control this independently. What makes this problematic is that the mk2's USB-C output is also tied to the **(VOLUME)** knob, and is lower than the output for the main/headphone output.

Stereo Output

Interestingly the mk2 has impedance balanced 1/4" TRS L/Mono & R jacks. This gives the mk2 some advantages to reduce the noise floor and eliminate ground loop noises when using the right cables within an impedance balanced studio environment. ??? VERIFY TRS ???

The above said, I'm not sure how much this will help the average producer, as they are unlikely to be in a balanced environment, but I guess it is to say that it is a nice feature to have when you can take advantage of it.

Stereo Input

Sadly the 1/4" L/Mono & R jacks are the TS (no ring to provide impedance balance) type and appear to not be balanced.

Personally this doesn't bother me too much, but it is a little bit strange to see balanced outputs and unbalanced inputs. It's probably fine, but if you find running off of batteries or USB adds some noise, try using DC power.

Dual Microphone/Guitar Input

On the front there is dual microphone and guitar jack that is a 1/4” TRS. For a guitar the ring isn’t doing anything, so just use a standard TS cable to hook up your guitar and/or pedals. For microphones you’ll need a TRS cable, as this is how power is provided to the microphone.

I’m not sure what types of microphones the mk2 supports, going to need to do some research to figure this out.

MIDI Input and Output

The mk2 supports MIDI in and out, using 1/8” TRS jacks. You can buy these cables from Roland as accessories. For supported devices you should be able to connect TRS MIDI out to TRS MIDI in across devices.

Roland lists some optional (not included in the box) accessory cables that you can buy, and I’ll dig more into the specifications of them later.

BMIDI-5-35
BMIDI-1-35
BMIDI-2-35
BCC-1-3535
BCC-2-3535

There are two TRS cable formats for TRS to MIDI, Type A and Type B. Roland uses the Type A format. Make sure you get the right format documented here.

USB-C Connection

The mk2 has a USB-C port. It can be used for powering the device, sending and receiving MIDI information, and sending and receiving audio information. It has a lot of utility and I’ll be writing about this extensively in sections to come.

As far as I can tell, this is class compliant audio and class compliant midi. This means that you shouldn’t need any special drivers to use the mk2 with devices that support class compliant audio and midi.

One oddity with the USB-C connection is that audio seems to be quieter over this connection for some reason. It seems like Roland put in a -12 or -24db (??? Measure) audio level adjustment, making you have to normalize audio to get it back up to a higher level with other content sampled on the mk2.

Power

You can power your mk2 using several options, including the provided AC adaptor, USB-C, or with batteries.

AC Power

The mk2 takes DC power in at the terminal. The power adaptor for the mk2 is an AC (wall wart) adaptor and is specified as a 5.7 volt and 2 amp device, and is center negative. It is noted that the mk2 draws 1.1 amps of power, so the 2 amps from the power adaptor is more than sufficient to handle the power requirements of the mk2.

Older SP models were 9v, and you could swap out power supplies with other SPs or even guitar pedals. This is no longer the case, so make sure when power your mk2 that you use the provided power or something with the same specifications.

Even if you have batteries or a powered USB-C connection, don't disconnect from Power as it will turn off the mk2. If you want to disconnect from power, turn off your mk2 first and then turn it back on once you have switched to your new power supply.

USB-C Power

The mk2 can be powered from another USB-C device or power adaptor. This can be helpful if you have a lot of USB-C devices to power or charge, and don't want to carry multiple different power adaptors around with you. One thing to note however, if you are powering over USB-C, the device or adaptor must provide 1.5 amps of power, or the mk2 will default to a different type of power. The older USB standards only provided 500ma of power, and would certainly be insufficient to power the mk2.

My iPad mini has a USB-C port, and it has been able to power my mk2 without any problems. This has been helpful in conserving battery power on the mk2, but does put a bit more strain on the iPad's power supply.

Although you can hook up your iPhone to the mk2, it is unable to supply the necessary 1.5 amps of power over the port. This is fine if the mk2 is plugged in, but if you are battery powered on the mk2, then you are going to get consistent messages about the mk2 using battery power. ??? Fixed in 1.14 ???

I'll get into it more later, but I have found you need a USB-C to USB-C cable to power the mk2 over USB. A USB-C to USB-A only supports 500ma of power which is not enough, and the USB-C to lightning does not provide power or class compliant audio/midi at all, for an iPhone you will need the camera kit.

Battery Power

To power the mk2 off of battery power, you will need 6 Alkaline or Ni-MH batteries. There doesn't seem to be a way to tell the mk2 what type of batteries you have inserted like some other devices do. Roland specifies that for Alkaline batteries you will get approximately 2.5 hours of use and for Ni-MH batteries you will get about 3.5 hours of usage. If you use rechargeable batteries you can likely expect the amount of time you get between recharges to be lower, but you'll go through less batteries.

If you are using rechargeable batteries, it can be a good idea to use the batteries in sets. I frequently will put some washi tape of the same color/pattern on a set of batteries for a device so that I will wear all of the batteries out evenly.

When you are plugged into an outlet, do not unplug from the outlet even if you have batteries in the mk2. The mk2 will not switch over to the battery power in time, and you will have to restart the mk2.

I just wanted to note, that it's not a great habit to get into leaving batteries in the mk2 for months or years without checking on them or replacing them. If you're not using the mk2 for extended periods of time consider taking the batteries out in case they leak so that they don't ruin your device.

The mk2 doesn't give any information to let you know about remaining battery power while in the interface. Although a power loss from no longer having remaining battery life may not be the worst, it can still lead to a loss of work or other issues. The mk2 really should have some sub menu to indicated battery life.

SP-404SX/A Project Import

The mk2 is able to import projects from the 404SX and 404A.

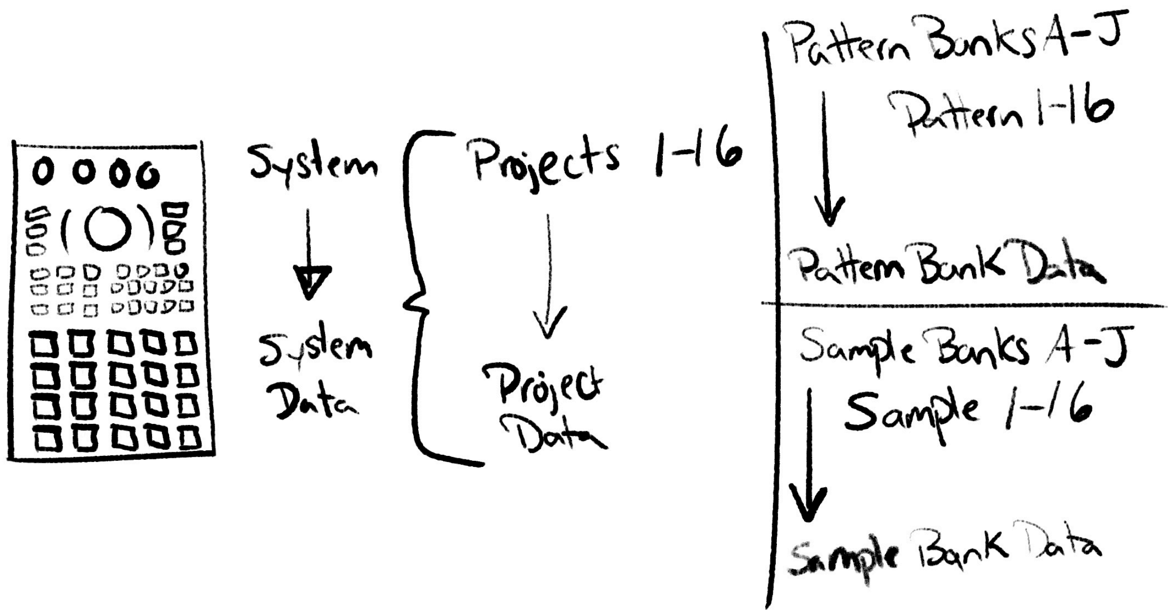
I don't have either of these units, so not able to test or try this out at this time. I believe that this needs the Roland cloud app, and not able to be done over the sd card on the mk2.
NOTE: I now have both… need to set aside some time to test.

Data

The SP-404 mk2 has several types of concepts and data that you will want to make sure you understand while you are working with your mk2. This will help ensure that you are managing your mk2 data appropriately. Below is a sketch to try and help show how I internally think about the structure of the mk2.

Going from left to right, the mk2 as a System has System Data and 16 Projects. Each Project has Project Data, 10 Pattern Banks, and 10 Sample Banks. Each Pattern Bank has Pattern Bank Data and 16 Patterns, and each Sample Bank has Sample Bank Data and 16 Samples.

Although the architecture of the mk2 is fundamentally quite simple to understand, the sheer amount of differing patterns, samples, and effects permutations give a vast array of options to create songs with.



System

At the highest level, the mk2 is made up of your System Data and your Projects. The System Data contains all of your System Settings such as how you configure your pad sensitivity, audio settings for +/- db, master effects bus settings, as well as other global settings that you will use across your mk2.

The only way that I am aware of to save your System Data is to make a backup of your entire mk2.

??? Does the App give you any way to adjust or backup/save System Data ???

??? Run an experiment to see which System Data is and is not backed up and restored upon a system restoration.

??? How does a factory reset impact this ???

Backing up your mk2 will create a directory on the SD Card named “BKUP”, and will then create a folder “BKUP<x>” where <x> is the backup number you choose from 1 to 99. The artifact that is created for the backup is a single file named “BKUP01.bin”.

Restoring a system backup on your will load from a directory named “BKUP<x>” where <x> is your restore number you choose from 1 to 99, and it will load erase the contents of the “BKUP01.bin” file appropriately.

If you want to restore a backup from another mk2 or move the backup files to another SD Card, make sure that you copy the “BKUP<x>” directory into the “BKUP” directory on another SD Card. You can rename the directory to anything from 1-99, just make sure that you select the appropriately numbered directory when you load up from backup.

??? Does it create more files based on the 1gb(‘ish) file size limit ???

??? Can you extract files from this file format ???

Projects

The mk2 gives you 16 projects that you can switch between, and each project has a unique set of 10 Pattern Banks (A-J) and 10 Samples Banks (A-J).

Exporting a Project will create a directory named “EXPORT” and a sub directory named “PROJECT”. You can export projects 1-16, and the exported directory will be named “PROJECT_<x>” where <x> is the number of the project that you are exporting.

Contained within the “PROJECT_<x>” folder is a file named PADCONF.BIN, and three directories named PICTURE, PTN, and SMPL.

The file PADCONF.BIN presumably contains information about how your pads are configured for the project, including information about Pad Mutes, and other settings that may impact how you setup the pads on your mk2. ??? need to spend some time making changes to see how tweaks impact this file ???

The directory PICTURE contains six .bmp files that are bitmaps for your system startup as well as images that can be cycled for a screensaver. ??? need more information on how to edit these files ???

The directory PTN contains a file named PTN00001.BIN, which is likely what has all of the information for patterns you have created on your mk2.

The directory SMPL contains multiple files named “BANK<x>-<y>.SMP” where <x> is the bank number (where A=1, B=2, C=3…J=10), and <y> is the 1-16 based on which pad the sample is associated with.

??? Beyond running some unix commands to try and parse what the file formats may be, I do not know what most of these files are at this time, and will likely run some future experiments to see if I can figure anything out about them in the future ???

Pattern Banks

Each Pattern Bank (A-J) contains 16 Patterns. Each Pattern Bank has a unique BPM, so if you change the BPM for one pattern in a bank, it will change the BPM for all other patterns in the bank.

Each Pattern Bank has a unique BPM, so if you change the BPM for one pattern in a bank, it will change the BPM for all other patterns in the bank. If you chain a pattern into a different Pattern Bank, the BPM will change to the BPM of the new Pattern Bank once the new Pattern starts playing.

??? Does strength, quantization, and other settings follow patterns as well ???

Patterns

A pattern can be anywhere from 1 to 64 bars, but currently it only allows for doubling increments, meaning 1, 2, 4, 8, 16, 32, or 64 bars. Pattern mode has the ability to modify the pattern Loop Start and Length, but Loop Start currently does not work, and Length can only be modified to match the same increments of the number of bars.

Patterns on the mk2 are restricted to a single track of pattern information, meaning that you can only play one pattern at a time. This means that you will need to make sure that each pattern only triggers the pads that you want to play, and may need to chain patterns together in order to play a full song.

When copying patterns between Banks, it will retain all of the settings listed above when you press **|COPY|** and select a source **[PAD]** and destination **[PAD]**. When copying a pattern to another bank, it will take on the BPM of the Pattern Bank it is copied to.

Sample Banks

Each Sample Bank (A-J) contains 16 Samples. Each Sample Bank has a unique Volume that you can change by pressing **|SHIFT|+|BANK|** which will allow you to modify the volume of all Samples within the modified Sample Bank.

If you keep similar sounds within a Sample Bank, modifying the Sample Bank Volume setting can be a quick way to adjust sounds together to keep them all relative to each other.

Samples

Each sample on the mk2 must be associated with a unique project, bank, and pad. The mk2 does not allow samples to be linked or referenced to each other, so there is no concept of non-destructive chopping like there is on other devices.

Every sample gets it’s Start/Loop/End point, Pitch/Speed/Volume adjustment, Envelope, and has unique settings liked to **|BPM SYNC|**, **|GATE|**, **|LOOP|**, and **|REVERSE|**. You can further adjust a sample’s FX Bus to Bus 1, Bus 2, or Bypass, by pressing **|REMAIN|+[PAD]** to cycle through the FX Bus options.

When copying samples between Banks, it will retain all of the settings listed above when you press **|COPY|** and select a source **[PAD]** and destination **[PAD]**. When copying a sample to another bank, it will take on the Sample Bank Volume of the Sample Bank it is copied to.

If you want to copy samples between projects, you will either need to export your sample to SD and then reload it into the new project, or you will need to play the **[PAD]** so that it is recorded to Skip Back, switch to the target project, and then press **|MARK|** to load Skip Back to pull the sample from the buffer.

Audio Input

Of the many ways that you may choose to utilize your mk2, probably the most consistent thing that you're likely to do will be to sample from an external source. This section will cover the different types of input, and how you may go about utilizing them within the mk2 and your other devices.

One thing to note about the inputs on the mk2, if you have EXT SOURCE on, or are in REC mode, it will pass through anything on the inputs from the 1/4" ins, MIC/GUITAR, and USB-C. This is good if you want this, but could be problematic as there is no way to switch between the 3 input sources individually if you wanted to.

MIC/GUITAR Level

On the front of the SP-404 mk2 there is a 1/4" TRS jack that can be used to (power/increase signal strength) of a guitar or a microphone. There is a switch labelled MIC/GUITAR, and a GAIN knob that can be turned to adjust the incoming signal boost.

I ran some experiments on over driving a line level signal on the mk2, and there doesn't appear to be anything special about running line level audio into the MIC/GUITAR input, if EXT SOURCE is turning red, then you are definitely digitally clipping your audio.

Microphone

When the MIC/GUITAR switch on the front of the mk2 is set to MIC, you can connect an unpowered microphone to the front INPUT jack.

Just discussing microphones would require a guide on its own, and overall is far too deep of a topic for this section. In this section I will just hit the most important notes.

The mk2 doesn't have a phantom power switch that would enable it to provide power to microphones that require it. As such, this means that you won't be able to directly use a condenser microphone without having a DI box or amplifier that you can plug the microphone into, and then hook that up to the mk2.

If you want to use a microphone with your mk2 then, you are going to want to be looking for dynamic microphones as they do not require power. Dynamic microphones tend to be lower cost, but the transient response and high frequency response tends to be lower than that of a condenser microphone.

Shure makes the SM57 and SM58 which are both solid dynamic microphone choices found on stage and in studios, and I have been using an AKG D880 for years that has worked just fine for capturing a multitude of audio options.

If you really need a condenser microphone, I might suggest looking into the Zoom product line. I have the Zoom H6, it is a small field recorder, supports up to six inputs, and can provide phantom power to microphones, as well as allowing you to record endless amounts of content on its own.

The manual does not make it clear what types of microphones are supported on the mk2, but since there is no phantom power switch, and the manual does not state that it can provide +12/+24/+48v phantom power I must assume that it cannot. Condenser mics will need power from batteries or another device.

On some older sampling gear a trick was to hook up your line outputs to the microphone inputs to add some distortion. This worked due to the the nature of the AD used. Unfortunately this old school trick doesn't seem to work as you just get the traditional digital clipping instead of getting any interesting distortions.

Guitar

When the MIC/GUITAR switch on the front of the mk2 is set to GUITAR, you can connect high impedance devices like a guitar or bass, as well as guitar pedals if you wanted to add one or more pedal effects to your guitar or bass.

Instrument level brings it's own challenges with noise. I don't have a lot of guitars to choose from (just the 3 string my dad made me)… may need some help beyond doing pedal stuff for thoughts/suggestions here. Hooked it up, and holy smokes is my 3 string noisy… woof. Guess will need to come back here later.

Tried a few things… noise floor straight from a single pedal into the mk2 is *high*, and I'm suspecting it might be a noisy power supply. I switched over to my pedal board and it so far sounds much cleaner, if noisy. I suppose a hold off on this section for the time being is in order, but even for just testing, got some fun results.

The mk2 doesn't have an inbuilt tuner, so if you're trying to tune your guitar or bass you'll need to look elsewhere. There are plenty of good phone apps and battery powered tuners. This is a feature I'm hoping we get in the mk2 at some point though, so we can tune instruments, samples, or whatever else needs tuning.

Guitar Pedals

One thing that is great about the guitar input on the mk2 is that it opens up a wide world of variety for FX. There are so many pedals that range from boutique manufacturers to well established brand names, and the can add so much unique character and sound.

I am not going to go into any deep discussion on impedance or signal levels, but there are a few things that are often worth reading up on with your pedals, or trying on your own. Some pedals will only work at instrument level, and will have to be connected the MIC/GUITAR input port on the front, but there are some that are also happy to function with a line level signal source and then output a line level source. Heck, even some pedals will provide stereo output and occasionally stereo input. It is definitely worth experimentation.

Either way, you can frequently turn the volume way down on a line level output, hook it up to a guitar pedal input, and then hook the pedal output to the mk2 input. You'll monitor your input with EXT SOURCE lit, and slowly increase the gain of your line level device. You'll probably notice clipping quickly if you go to far, but just pay attention to your levels and you might just find some fun new ways to record material for your mk2.

Line Level

For a lot of people, I expect that the 1/4” inputs on the back are going to be the primary way to get sound into their mk2. Plenty of instruments, radios, phones, tablets, and more output audio at line level.

There are a number of adaptors to get phono, 1/8” or 1/4” TRS, lightning, USB-C, and other connectors to output to to line level on 1/4” TS jacks for the L/Mono and R signal. The important part is to make sure that they output at line level.

If you only connect a device to the mk2 L/MONO input, the mk2 will still record a stereo sample. There doesn’t appear to be any way to record a mono sample to the mk2, and I’m just guessing that the only way to get a mono sample onto the mk2 is to import ??? or use the companion app ??? Need to test to confirm.

Phono Level

Before I dig into this section it must be noted, that the mk2 does not by itself support phono level on its inputs. While you can technically connect phono level up to the line level input jacks on the mk2, it is going to be incredibly quiet, and the audio will almost certainly have some distorted qualities to it.

Also, just because you have a device with RCA/Phono jacks doesn’t mean that your device outputs at phono level, it may be outputting line level. You’ll want to make sure you know what the specifications are for the outputs.

That said, most turn tables will output at phono level, and need to have their signal boosted up to line level. This is traditionally done through amplifiers, and is why you will see many people who sample off of vinyl will have a DJ style mixer as part of their setup, which does the job of signal amplification, volume control through the track fader and cross fader, signal equalization, and may even have some other effects.

USB

Possibly the most surprising feature for me out of the box for the mk2 was just how well integrated the USB-C class compliant audio is. Connecting it to a phone, tablet, or computer, with the right cables just works, with minimal setup or fuss.

On the iPhone and iPad, the mk2 becomes your primary input and output device, and it can even see the two inputs and four distinct outputs. This makes it very easy to just connect to one of these devices, load up an audio application, and quickly get to sampling and making music.

There is a system setting to adjust USB output, but there is no system setting to adjust USB input. This is a problem for iPad/iPhones as you won’t have any way to adjust the volume for applications that you cannot route through AUM or AudioBus 3.

For the iPhone/iPad at least (don't have Android to test), one issue is that the volume buttons won't adjust your audio levels. For music software you can use AUM or AudioBus 3 as a mixer, but for applications like YouTube, there is no way to adjust your output, so you may be better off using the line inputs instead.

Using a computer will require a bit more work, but from what I have seen it isn't much more. You'll likely need to adjust your audio outputs to direct them to the mk2, and if you are using a DAW you may instead want to configure its audio setup within your preferences. That said, this is still pretty easy, and I'll try and document it some more in a future section.

Gain Staging

When people talk about gain staging, what is generally meant is to have all of your inputs at an equal input signal to each other. This is particularly important in studios with large mixers, because you want to use your mixing console as the master volume control for each audio channel, otherwise you quickly find that each instrument has a varying amount of volume and noise floor, and this leads to making it harder to mix.

Traditionally, for many audio sources, you would run a single sin wave as output for each source, at or near the maximum instrument/device volume without the signal distorting, and then adjust the input gain on the mixer until you hit a target volume such as 0db or -3db for each and every device/instrument. By doing this, you will be maximizing the amount of volume you have the instrument, and in general this will also help to reduce your noise floor. This allows each fader on the mixing desk to be able to control the volume of any sound relative to any other sound.

Now there are a couple things to consider here, first, the mk2 doesn't have any mixer faders, and second most of the volume gain you can do on the mk2 is done in software.

Going to have to come back to this section… I really need to figure out what the threshold is that is triggering the EXT SOURCE pad to light up red.

Controls

Monitor External Sources

When sampling, it is not always necessary to monitor your external sources, but if you don't want to enter record mode, and just want to hear how things will sound, if the sound is too loud or quiet, or if you are getting some kind of clipping or distortion this can be a quick way to hear how your source is sounding. To monitor your mk2 inputs, all you need to do is press **|EXT SOURCE|** and play your audio from your external device.

The experiment **Does EXT SOURCE turning red mean that you are clipping or not?** shows that if you see the EXT SOURCE pad turn red that you are digitally clipping. Unless you are looking for a specific or intentional effect, you should dial your audio source back down so that the EXT SOURCE pad stays amber.

Input Settings

You can adjust your input settings with **`INPUT SETTINGS` |SHIFT|+|EXT SOURCE|** which will bring up additional information on your display.

Modifier	CTRL1	CTRL2	CTRL3
None	REC BPM	ROUTING	LEVEL
SHIFT	FINE REC BPM	N/A	PAN

REC BPM (CTRL1) - This sets the BPM that you believe you are recording at in whole number increments. If you are just recording one shot samples then you probably don’t need to worry about this too much, but if you are planning on using LOOP or PING-PONG you may be more interested in getting this set properly.

FINE REC BPM |SHIFT|+(CTRL1) - This allows you to adjust the REC BPM value by one tenth (or .1) value at a time instead of adjusting the BPM value by whole numbers.

ROUTING (CTRL2) - This allows you to set either Mix or ExtIn. It does not directly impact what is played, but does impact what will be recorded. If you select Mix, then when sampling all content that is playing, whether it is from pads or from an external input will be recorded. When set to ExtIn, then only audio coming in from the Line In, Microphone, Guitar, and USB-C connection will be recorded.

Although this is undoubtedly useful, I think it would be helpful to have a few extra settings so that you could have a loop playing on a pad to hold a beat or rhythm, and then for example use Chromatic mode to just record a new bass line or piano section. However, this would require additional routing that just does not exist. Pre/Post Bus?

LEVEL (CTRL3) - This adjusts the volume of the sample that will be recorded. 0 is the lowest and will be so quiet that the sound will be imperceptible, and 127 is the loudest volume you can record. If you are approaching or going beyond clipping you may see the |EXT SOURCE| light up red.

PAN |SHIFT|+(CTRL3) - Adjust the stereo pan of the incoming signal. C indicates that the signal is centered, so whatever is connected to your inputs will be reflected the same on your speakers. For either an L or R setting a value of 1 is the lowest amount of panning and a value of 50 is the highest amount. At a value of L50 you should only hear the inputs in your left ear, and a value of R50 you should only hear the inputs in your right ear.

Input FX Settings

From the INPUT SETTINGS menu, you can get to the Input FX settings through the value knob by pressing **{ENTER}** to open up the menu. This will break the display down into three sections EFX Type, EFX Settings Row 1, EFX Settings Row 2.

To modify EFX Type when the row is underlined press **{ENTER}** and then rotate **{VALUE}**. To exit modify the EFX Type just press **{ENTER}** again once you are happy with your selection.

To modify EFX Settings Row 1 and EFX Settings Row 2, you only need to rotate **{VALUE}** to underline the row that you want to modify and then use the corresponding **(CTRL)** knob to change the EFX setting's value on that row.

By default the EFX Type will be set to Bypass, which indicates that the incoming signal is clean. There are sixteen EFX that you can select from here, three of which only appear in the Input FX menu.

Unique FX to Input FX - Auto Pitch, Vocoder, Gt Amp Sim.

Common FX to Input FX - Chorus, JUNO Chorus, Reverb, TimeCtrlDly, Chromatic PS, Downer, WrmSaturator, 303 VinylSim, 404 VinylSim, Cassette Sim, Lo-fi, Equalizer, Compressor.

For more details on the individual effects, please see the Effects section.

Audio Output

If you're going to do much of anything with your mk2, you're going to need to hook it up to some outputs. I guess you could technically do some field recordings with a microphone and not hook up the outputs, but this generally seems like an edge case to me.

Line Level

Most gear that has inputs is going to support line level. This is a quite common standard, though you might need adaptors to adapt the cable type from 1/4" to 1/8", phono, or other connection standards.

The L/Mono and R output jacks both provide a panned signal to the L or R channel, neither provides a stereo signal by itself. So it is possible to only connect to outputs to either the L/Mono or the R jack, but you will only be getting a mono output.

It is worth checking your gear to make sure it supports line level inputs. For example, if you want to connect to a DJ style mixer, it will likely have phono jacks which may mean it takes phono level (and not supported by the mk2), but it may have a switch to adapt the connection from phono to line level.

You can adjust your line level output relative to gain by going to **\`UTILITY\` |SHIFT|+ [PAD13]** then **\`GAIN\` (CTRL3)** and select **\`Line Out\` {VALUE}** to adjust your line level output gain from 0db to 12db in 6db increments.

The manual states that the mk2 has TRS outputs, which would suggest that the outputs are balanced. I have not confirmed this yet, and will need to do some research to determine if they are in fact balanced, or turn out to just be regular TS outputs.

Headphone

The mk2 has both a 1/4" and 1/8" set of headphone jacks. When you want to use your mk2 with headphones, you don't have to figure out which jack or adaptor you need to use, just plug it into the appropriate hardwired jack on your mk2.

If you have headphones with a high impedance, you may want to increase the gain on your headphones by following the instructions in the info icon below.

Even better, the pair of headphone jacks means that you can have two people connected the mk2 at the same time, which may open up some interesting audio path workflows for both a single person as well as a pair of people to get an audio setup off of a single device.

The experiment **Do the headphone jack outputs share an amplifier?** Indicates that there is a single headphone amplifier, which shouldn't be a direct concern, other than there will be an overall lower listening volume if there are two connections attached the headphone jacks that is worth being aware of.

Although the headphone and line level volume output are linked, as of the 1.14 firmware release you can adjust the headphone gain relative to line level gain by -18db to +12db in 6db increments by going to **\`UTILITY\` |SHIFT|+[PAD13]** then **\`GAIN\` (CTRL3)** and select **\`Phones Out\` {VALUE}** to adjust your headphone gain.

USB

The USB-C audio output on the mk2 is really handy, and is a quick way to get audio from your SP out to your iPhone, iPad, or computer. Any audio that is going out to your main outs is automatically also going out the USB-C port. If you are having problems with the USB-C outputs running too hot, you can lower (but not increase) the relative USB-C output level by going to **\`UTILITY\` |SHIFT|+[PAD13]** then **\`GAIN\` (CTRL3)** and select **\`USB Out\` {VALUE}** to adjust your USB output gain from -24db to 0db in 6db increments.

??? I do need to test if outputs 1/2 are main outs and 3/4 are cue outs ???

Controls

* Attenuator
* Noise Gate
* …

Sampling

If you've got a mk2 and you're not sampling on it, I guess you're using it as an FX box, but that's missing out on sampling which is arguably the most interesting and fun part of the SP line. For this section I really just want to break down how to sample, things I've learned while sampling on the mk2, and probably add some notes on stuff that could just work better.

Before you kick off your sampling journey, you'll want to be aware that the mk2 has two primary modes, Sample mode and Pattern mode. You change between Sample mode and Pattern mode by pressing **|PATTERN SELECT|** to switch between the two modes, and the button **|PATTERN SELECT|** led will be dim if you are in Sample mode, and bright if you are in Pattern mode.

Recording Behavior

Before we do a deep dive into sampling, there are some important things to understand about the behavior of the mk2 when sampling, otherwise you might be confused when things don't work quite like you expect.

Input Setting Routing

The routing setting under `Input Setting` **|SHIFT|+[EXT SOURCE]** allows you to adjust between Mix and ExtIn.

Mix	Record all internal (samples/pads) and external (line/mic/guitar/usb) audio
ExtIn	Record only external (line/mic/guitar/usb) audio

While it may be obvious from the options listed above, there is no setting to record only internal sounds (as of 1.14). If you want only record internal sounds on the mk2 then you may want to disconnect all line/mic/guitar/usb inputs.

Count In

The Count In behavior adjusts if the mk2 will add a one bar or two bar metronome count in, wait for some line level signal, or if it is just off and will start once you trigger recording.

To change the Count In setting you just press **|SHIFT|+[PAD10]** and it will cycle between COUNT-IN OFF, COUNT-IN 1MEAS, COUNT-IN 2MEAS, and COUNT-IN WAIT.

COUNT-IN OFF	Must press **|REC|** after selecting a pad to trigger the recording

COUNT-IN 1MEAS	Pressing **|REC|** after selecting a pad will count in one bar before recording
COUNT-IN 2MEAS	Pressing **|REC|** after selecting a pad will count in two bars before recording
COUNT-IN WAIT	Will begin recording once audio plays over the trigger threshold or **|REC|** pressed

Much of the behavior listed above is really only important if you are recording external samples as resampling will begin recording as soon as you trigger a sample or pattern.

Record Threshold

If you have set your count in value to COUNT-IN WAIT, you can adjust the threshold that it will record in at by adjusting Utility > System > General “Auto Trig Level”

??? Provide more details ???

Metronome

You can adjust whether the metronome is on or off by pressing **|SHIFT|+[PAD9]** to toggle the metronome on and off. There are further options to adjust the metronome in **`UTILITY` |SHIFT|+[PAD13]** and then turning **(CTRL3)** to select CLICK.

Regardless of whether you are sampling external audio or resampling samples/patterns, if the metronome is turned on then the metronome will play at the BPM of the currently selected Pattern Bank. Further, the metronome clicks won’t be recorded as any sampling or resampling activity, so you can always safely turn the metronome on or off when recording content.

Mono/Stereo Sampling

As far as I have been able to determine, the mk2 can only sample in stereo, regardless of how you attach your inputs to the mk2.

??? Can you import mono samples, or are those converted to stereo as well ???

File Naming

To see the file name of a sample, you can press the **[PAD]** and then hold down **|REMAIN|**. This will bring up a lot of the metadata information about the sample, including the file name of the sample.

File naming so far has been a bit of a mystery to me. When you record onto the mk2 it will frequently give a name of “New Sample”, but I have seen it pick other names from time to time which aren’t entirely clear to me, and this may have been adjusted in firmware 1.14, as I have not been able to reproduce the behavior since firmware 1.13.

You can adjust a sample’s filename in the Companion Application. If you have access to a Windows or Mac this may be the best way to organize and name your files.

Effects

While sampling you have the option of using Input FX, Bus FX, and Master FX into your samples. Doing this is considered destructive, as once you have recorded a sample with FX, it will always have those FX as part of the sound. For this reason, it may be desirable to record clean versions of your sounds, and then separate effected versions of the sounds if you feel you may need to go back to your cleaner audio.

Once you sample or resample a sound, the mk2 will disable your Bus FX, but your Master FX and Input FX will both remain on. This can make things a bit confusing as if for example you have a compressor on your Master FX, record a sample, and then play the sample, you will hear a loud whooshing from doubled compression.

End Snap

This feature was tricky for me to figure out at first, so if it is giving you a hard time, take a deep breath and read on.

The first thing to note is that End Snap is an option that is either on or off, and is not a trigger that actually causes a sample to snap to the end of a bar. So once you are in record mode whether you are setting things up or actually recording, you can press START/END to turn on or turn off End Snap mode.

What causes the snap to the end of the last bar is when you press **|REC|**. The record event is what will then look backwards and place the End marker to the last previous full bar. I have had great success waiting until the next bar that I do not want to record starts playing, and then press **|REC|** to end the recording.

Recording

External Audio

For sampling external audio, you will want to be in Sample mode (**|PATTERN SELECT|** led is dim). Whether you have enabled **[EXT SOURCE]** or not is irrelevant, as once you start the sampling process **[EXT SOURCE]** will be disabled, and monitoring will be turned on.

Once in Sample mode, Sampling in external audio is as simple as pressing **|REC|**, pressing an unlit/empty **[PAD]**, and then pressing **|REC|** to begin recording. This will allow you to record audio coming in from the line inputs, mic/guitar input, or the USB-C connection.

As noted above, but just a reminder, the mk2 will mix all audio sources together, so if you are playing content on the line inputs, mic/guitar input, or the USB-C connection simultaneously it will record everything. If this is what you want then it should be cool, but if not, make sure to disconnect or mute the other input sources.

Resampling Audio

For sampling internal audio (samples that are stored on your pads), make sure that you are in Sample mode (**|PATTERN SELECT|** led is dim). The main difference here is that instead of starting the process with **|REC|** you will need to start it with **|RESAMPLE|**.

To resample the content stored on a pad, press **|RESAMPLE|**, press an unlit/empty **[PAD]**, and then either press a **[PAD]** or **|REC|** to begin recording. Either way will start capturing sound immediately, but pressing a **[PAD]** has the advantage of ensuring that the sampling starts lined up with what is already recorded onto the **[PAD]**.

One thing to note, you should investigate Pad Mute behavior if you'd like to hear some pads play, but not have them resampled to the new pad. Pad Mutes can really open up a lot of interesting resampling techniques that were not available prior to firmware 1.14.

Resampling for some reason does not follow the COUNT-IN options set, which makes it difficult to capture some times of content using resampling.

Resampling Patterns

For sampling patterns (Patterns that are stored in your Pattern Bank), make sure that you are in Pattern mode (**|PATTERN SELECT|** led is lit). The process is similar to resampling samples assigned to pads and starts with **|RESAMPLE|**.

To resample the Pattern stored on a pad, press **|RESAMPLE|**, press an unlit/empty **[PAD]** to select where to sample, and then press a **[PAD]** that has a pattern on it to begin recording. This will record whatever is playing for the pattern into the sample pad, and you can end recording by pressing **|REC|** when you have the content that you want.

One annoyance with the mk2 when recording a pattern is that when you are resampling it does not display the bar/measure counter on the display. You'll need to remember the pattern to know when you should stop recording.

End Mark is clearly an important thing to use here to sample perfect loops, but my technique is off, I haven't been able to determine if you press **|START/END|** before the end of the loop, just after, or something else… the timing always seems OFF to me.

Managing Samples

Copying

To copy a sample press **|COPY|**, press the source **[PAD]** to copy from, the target **[PAD]** to copy to, and finish by pressing **|COPY|**, and this will make an exact copy of the sample and all of the settings associated with it

Deleting

When you make a mistake, or no longer need some recorded content, you can delete it by pressing **|DEL|->[PAD]->|DEL|**. This will remove the sample associated with the pad from your mk2.

Protect

In some cases you may have a sample that you want to make sure you do not accidentally delete or overwrite in some way.

??? **|SHIFT|+|COPY|** protects the current bank, any way to protect a single sample ???

Sample Edit

A lot of what is under the SAMPLE EDIT banner on the mk2 is geared toward what I would refer to as sample playback. To me, editing is something you do that destructively changes the characteristics of the sample, and not just changing how the device plays the sample back. Mostly a nit, but I find that it confuses the terminology for people coming from or going to a lot of other devices. At any rate, you'll find the handful of sample editing features that are destructive in the Start/End menu, everything else that you want to do to manipulate or change the actual sample itself you will have to do through Skip Back, Resampling, or offloading to other devices, which I'll go over in the appropriate sections.

To use the Sample Edit features you will need to be in Sample mode. You change between Sample mode and Pattern mode by pressing **|PATTERN SELECT|** to switch between the two modes, and the button **|PATTERN SELECT|** led will be dim if you are in Sample mode, and bright if you are in Pattern mode.

To edit a sample, you will also need to select it, either by pressing the **[PAD]** which trigger the pad, or pressing **|EXIT|+[PAD]** which will select the pad but also stop the pad from playing if it was already playing.

??? Do some more research to see if there is a way to select a pad without triggering it or stopping it ???

Start/End

On previous SPs, the **|MARK|** button was important for setting your start and end points, and really force you into using your ears to listen to samples. You can still use **|MARK|**, but the display certainly helps out a lot. The advantages of having a display to adjust start, end, and loop points visually cannot be denied, but it can make it harder to dig around for happy surprises. All of that is to say "don't forget to close your eyes and use your ears once in a while".

On the mk2 the **|START/END|** button is where you go when you want to adjust your start, end, and loop points of whichever **[PAD]** you have selected. You'll be treated to the display changing to a visual representation of the sample. Depending on the length of the sample will depend on how much time it takes to finish the display update for drawing the sample, but you can adjust your start, end, and loop points while it is rendering out the waveform.

(CTRL1)	Move the chop point.
(CTRL2)	Move the currently selected Mark (M with a line above and below it)
(CTRL3)	Cycle through Mark points
|MARK|	Create a Mark at the chop point

The section in the SP-404mk2 Reference manual "Editing a Sample (SAMPLE EDIT)" outlines controls much further, but bellow are a few of them captured

(CTRL1)	Adjust the start point of the sample
(CTRL2)	Adjusts the loop point of the sample
(CTRL3)	Adjusts the end of the sample
|SHIFT|+{VALUE}	Enter numeric value of the selected point (start/loop/end)

Hopefully the start and end points of the sample are straight forward, but START determines where a sample will start playing when you press a pad, and END determines where the sample will stop playing when you press a pad.

If you have **|LOOP|** enabled (as documented in the Sample Mode section), then an additional option for LOOP can be controlled, which determines where a sample will start playing from once it has reached the END point to continue playing. This allows for the entire sample to repeat if you set the START and LOOP point to be the same and is important for beats or other BPM timed samples, or you can set the LOOP point to some point after START which can be useful for sustaining sounds such as pianos or strings that you may want to play longer than what you have for sampled content.

The display will only render the left channel of audio. So if you have a sample that is panned hard right, the display will show a flat line as it does not render any of the waveform for sample information in the right channel. Just because a sample looks empty does not mean that it is empty.

Start/End Sub Menu

Once you have entered the **|START/END|** sample screen, you have the option to make a handful of destructive edits to the sample by pressing **{ENTER}** to get access to these options. These will allow you to modify your sample in various ways, but the changes are permanent. If you want to try them out, you will likely want to **|COPY|** the sample first to another pad before making any of these adjustments.

Normalize

Selecting Normalize will increase the maximum volume of the entire sample (not just what is between the start/end points) to 0db. This is an excellent way to get your audio content to be louder if it seems very quiet compared to the rest of the samples you are playing.

Personally I try not to normalize every single sample, as this can cause some future work reducing sample volume later. For example, if you have a drum kit, you may want to increase the volume of all sounds in relation to each other. Low pitch sounds such as a kick drum tend to need more volume and so will need to be louder to be audible, while high pitch sounds such as a high hat tend to need less volume. In other words, normalizing a high hat by itself may have the unintentional consequence of making it overwhelmingly loud, but normalizing a kick drum and high hat recorded on the same sample may keep them in a proper volume level in relationship to one another.

You may want to truncate your sample before you normalize it if you are trying to increase the overall volume of your sample, as the content that is outside of the start/end points may be louder than what is in between your start/end points, thus preventing normalize from increasing the sample to maximum loudness.

Truncate

Selecting Truncate will remove all sample content that is outside of the start/end points, and is an excellent way to save storage space and keep projects clean. Given the sheer amount of storage space on offer, this may not immediately seem relevant, but for those who want to maximize the 16GB of internal storage it will likely be necessary.

Just remember that truncating a sample will permanently delete this sample content, making it unaccessible to you unless you either re-record the content, or are able to restore from a backup.

Truncation isn't necessary for a sample to play properly, and it can be useful to keep the extra information around just in case you want to go back to it. For example, if you have a drum loop and only need one hit off of it, it could be useful to keep the rest in case you wanted to explore other sounds in the sample later.

Emphasis

The manual says it increases high-frequency sound, but I hardly hear any difference in the audio. I guess I am too old to hear the sound enhancement it is talking about.

??? Is this worth an experiment to test what it may be doing ???

Cancel

Exits out of the menu.

Pitch/Speed

Being able to change the pitch or speed of a sample can help change the overall tone, help get two sounds to work with each other, or otherwise explore sounds and hear something new. One of the reasons I have (too many) samplers is because the approaches and algorithms for changing the pitch of sound are so varied and it can be really fun to explore the boundaries of technology.

You can switch vinyl mode by **|SHIFT|+{VALUE}** to turn vinyl mode on and off. With vinyl mode on the speed and pitch knobs influence each other so as you increase speed you also increase pitch, and as you decrease speed you also decrease pitch, similar to how adjusting the speed on a turntable would change the pitch of the record you were playing. With vinyl mode off the speed and pitch knobs can be changed independently so that you can play the sample slower, but at a higher pitch for instance.

Without resampling, the mk2 has a range of -12 to +7 semitones with vinyl mode on, and a range of -12 to +12 semitones with vinyl mode off. If you choose to resample whether you have vinyl mode on or off you can really stretch and change the samples that you load into the mk2.

Personally I find the audio quality with vinyl mode on to be very good, but with vinyl mode off the engine gets into more of a heavy artifact and granular sound especially as you get into the extremes of having either pitch or speed go up while you turn the other down. I am glad to have the option because it is absolutely great when you need it, but I'll stick with vinyl mode on for most of my use cases.

Some people have been disappointed with how Chromatic mode works because it is not polyphonic. You do still have the option of copying a sample and adjusting individual pads by changing the pitch to create your own sounds. It does take a bit more effort, but it does let you setup your own scales and it is polyphonic.

Pitch/Speed Sub Menu

This sub menu is hardly worth spending much time on, but it does allow you to detect the BPM of a sample by giving the mk2 a rough idea of the BPM of the content at hand, by selecting 100-199, 80-159, 70-139, or 50-99. I am going to guess that the mk2 is just using the length of the sample instead of doing any deep sample introspection, but it could be helpful if you are intending to do BPM SYNC to get various samples playing at the same speed together.

??? There is an experiment here to load up samples of different lengths and styles to see if it is just using length to determine BPM or something else ???

Chop

Chopping samples is a pretty big new feature on the mk2, but is something that has been available for a lot of other samples for some time. On previous SPs you'd have to copy pads and make adjustments, or in the case of the 202 offload a sample to another device so you could then sample it back in.

Chopping allows you to take one long sample, and split the sample up across multiple pads to play as distinct parts of a sample as individual pad hits. This can be anything from a single drum hit, a piano chord, a phrase or two of a song, really anything that you want.

The first thing to know about the Chop mode is that it doesn't respect your start/end/loop points. You are free to chop content of the entire sample, not just what you set for parameters in **START/END**. If you want to only chop on what is between your sample's start and end, then don't forget to **TRUNCATE**.

I don't want to do a deep dive on chopping at the moment, because there is so much to it and it is pretty well explained in the manual. Instead of duplicating what the manual says (for the time being), I wanted to focus more on what a few workflows of chopping samples looks like.

The short overview of the controls are as follows:

Each Mark will be assigned a new pad Starting with [PAD1] through [PAD16]. You cannot have more than sixteen chops on the mk2.

Lazy Chopping

This is what you get as soon as you load up the **|SHIFT|+|START/END|** Chop mode.

Okay, this is not what the SP calls it, but it's what I've always seen it referred to as on MPCs. Essentially you play a pad to start a sample, and then each subsequent pad you hit will set a marker for where the previous pad ends and the next pad starts. If you're halfway decent at this it can be a really quick way to just knock out an eight or sixteen bar sample into distinct pads. I also like this style because it can give you some happy accidents where things are chopped up perfectly. This is a great way to break down drums, pianos, pads, or anything that you want to get close to how you feel the flow of the sample. With the display, you've also got the option for getting far more accurate with the chop point positions if you want, it just depends on how much time you want to invest on your chops.

This method tends to work best on samples that have a shuffle, don't adhere to strong BPM, or other kinds of live content that you cannot rely on using Auto Mark to select Time Divisions. Creating Marks by playing sloppy, or intentionally with the volume off can be a good way to find interesting chops. Sometimes you'll get a chop that is between note changes, creates a gated stuttering drum, or other interesting sounds. As I mentioned, you can absolutely make these perfect, but don't forget to look for the beauty in weird chops as well.

Time Division

You'll find this under the Auto Mark **{ENTER}** sub menu, and it will allow you to chop a sample into precise equal sizes. This can be great if you are working with electronic content, or don't mind chopping on major beats to avoid/maintain issues with beats shuffling. Similar to Lazy Chopping, I'd suggest also giving it a try to use odd or wrong Time Division parameters, as it can expose some new ways to play sounds and build patterns.

Use this in particular for drums, or anything that maintains the beat/rhythm to the BPM. And don't be afraid to get creative. Instead of taking a one or two bar loop and chopping it up, it can be just as effective to take an eight bar loop, and break it down to eight one bar loops. It all comes down to how you want to approach the content and what you are trying to do with the samples.

With practice you may be able to get this to work with time signatures other than 4/4. For example, with 3/4 content, instead of selecting a time division of 4, 8, or 16, you may find that a Jazz loop may work with 3, 6, 9, or 12. Just give it a try and see what does and doesn't work for you. If you get stuck, don't forget to go back and try using the Lazy Chop method to tap out the pads by hand.

Level

Level is also found under the Auto Mark **{ENTER}** sub menu, and adds a marker at each location where the volume of a sample exceeds a certain level.

Personally I'm not as much a fan of this method, only because for some types of content it is going to put the marker after the attack has started. For example, pads that have a long swell or reverse cymbal crashes will put a marker at the loudest part of the sound. Sure it gets you in the ballpark, but you're likely to spend a lot of time fiddling with the marker points so that you get the right start and end of a sound. I'm also not convinced that 1-10 leaves enough of a range for catching sounds, but you may find this works well for you.

Complaints aside, if you are struggling to get decent chops, this is a method that can give some meaningful results by just trying a few different values. I've found a setting of 3 or 4 tends to give the best results.

A setting of 10 is going to be the most aggressive, and will almost certainly not have enough pads to capture everything, but can lead to some tiny snippets of content that might be interesting to loop or otherwise break down and turn into tight rhythms or tones for a song.

Transients

Transients will add markers where there is a significant volume change, and in particular can be good for content like drums where kicks, snares, and high hats tend to play on their own, or have distinct enough events that you can have meaningful separation of the sounds.

Where Level is chopping on the loudest part, transients is more aligned with the start of the attack of the sound. The chopping parameters for Transients is Hard, Mid, and Soft, and corresponds to how quick of an attack each sound in the sample is. Hard tends to work well on drum loops and pianos, while soft can be better for more subtle transient changes in pads or other instruments that tend to slowly build up into a sound.

Chop Value Sub Menu

Auto Mark

Auto Mark gives you three different options for chopping up a sample: Time Division, Level, and Transient. These map pretty well to other samplers, so if you are familiar with chopping elsewhere then you've probably used something comparable to one of these modes.

Overall, given the sixteen pad limitation I'm okay with the limits at work here, but people who are coming from more complex samplers like an MPC are going to be underwhelmed at how limited the SP chopping methods are in comparison. The SP will force you to work with smaller samples to chop, or longer chops.

Assign To Pad

Assuming you've gotten your sample chopped, this is where you need to go in order to get those chops laid out onto pads. The method is pretty straight forward, but I found it a little wonky at first.

Before you enter Assign to Pad mode, make note of the pads you want to export. If you only want a single kick and snare out of a drum loop, remember which pads they are on. **[SUB PAD]** would be a perfect way to play each marked chop here, but you *cannot* listen to the pads once you are assigning them.

With the warning (and request for this behavior) out of the way, we can move forward. **(CTRL2)** allows you to select which chop you want. If the chop is left with a "- - -" then it will not be assigned to a pad, and to assign a chop to a pad just use **(CTRL2)** to select the chop and then press **[PAD]** to assign it to an available pad. Once you are done you just press **{ENTER}** to commit.

Using the drum loop example, if you chopped a drum loop to eight chops, and you only wanted the kick that was on chop 1 and the snare that was on chop 3, you just use **(CTRL2)** to select chop 1, and assign the kick to **[PAD]**, use **(CTRL2)** to scroll passed chop 2 and go straight to chop 3, then press the **[PAD]** to assign the snare.

Once I spent some time with Assign To Pad it wasn't so bad, but coming from the MPC I was definitely over thinking how capable the Chop mode was. Although it may be fairly bare bones, it is still a great feature addition from previous SPs.

Delete All Marks

This option will delete all of the Marks that you placed in chop mode, so if you're not happy with where the marks are and how they are chopped you may want to just delete all of them and start over.

Cancel

This option will cancel you out of the menu as per usual.

Envelope

The envelope on the SP is pretty basic, and it can only be used to adjust the volume of the sample. If you have dreams of modifying pitch, frequency cutoff/resonance, speed, or other features, you're not going to find it here (at least as of 1.14).

As bare bones as the mk2 envelope is, it has an AHR (attack, hold, release) instead of an ADSR (attack, decay, sustain, release). The AHR setup does map well to the 3 control knob scheme, but I cannot help but wonder what may have been here. It is useful enough for some basic sound sculpting, and for my usage has been what I go to for eliminating clicks and pops from the start and end of samples.

For in depth details on the Envelope, check the manual.

My biggest complaint about the mk2 is that the Envelope release doesn't respect the Gate setting. On just about any other sampler, when you release a pad with Gate, it will play out the release. Instead, the mk2 just stops playback immediately with Gate on. Roland, please fix this!!!!!

Eliminate Clicks and Pops

If you're finding that you're samples have clicks and pops, and no matter how much you adjust the start and end points they continue, you may want to go into envelope to fix this.

Frequently by applying an Attack of 1 and a Release of 1, I have found that it will quickly reduce or completely eliminate the clicks and pops without much of a noticable change to the audio.

Adding Expression

The options with the AHR are a bit limited, but if you have a sample with Gate on, leveraging Attack or Release can be a way to have a sound change a bit more depending on how long you hold it.

As an example, I have recorded some high hat loops that play three or four high hats. With Gate turned on I can then have some control over the volume of the hi hats based on how long I hold down the pad, and can get some interesting sounds by cutting the sample playback short.

Copy

To copy a sample on a pad, press |COPY|, then press [PAD] to select the source pad, press [PAD] to select the destination pad, and finally press |COPY| to execute the copy operation. This will copy the sample and all of the associated settings to the new pad.

??? Anything that is not copied ???

Delete

To delete a sample and the associated samples on a pad, press |DEL|, then press [PAD] to select the pad you want to clear, and finally press |DEL| to execute the delete operation.

Sample Mode

Sample Mode is mostly settings that control how a sample is played back, and in some cases can have an immediate change to a sample that is currently playing. None of these sample settings are destructive, and with the buttons on the mk2 allow you to quickly change these settings with the press of a button.

BPM Sync

BPM Sync allows you to toggle if a sample plays based on the **PITCH/SPEED** setting, or instead if you are going to play the sample in relationship to the speed of the BPM that is set. Typically you are going to want to turn on **BPM Sync** to get one or more loops to play in time to a BPM. Clearly, if you're not concerned with timing or a BPM, then **BPM Sync** is probably going to be a less interesting feature for you.

A few things to consider though, a sample such as a One Shot that is a distinct sound, such as a Kick Drum, doesn't inherently have a BPM by itself, so setting a BPM for a single hit or single note doesn't necessarily make sense. For content that is rhythmic or looping in nature you are going to want to properly set the BPM of a sample before you enable **BPM Sync**, because otherwise it will not play in time properly to your patterns or other BPM Sync'd material. Generally speaking **BPM Sync** only really makes sense for a sample that is a loop, so that it can be repeated many times.

If you have drum loops, sustaining pads, or instrument progressions BPM Sync can help you keep everything playing in time. For instance, many modern loop libraries will tell you what BPM a sample is played at in the file name, and if you import these samples you can then go to **|PITCH/SPEED|** and press **{ENTER}** to set BPM.

Gate

The Gate setting effects how playing a pad will influence playing a sample. This can be a good way to give an additional amount of expression to a sound so that you could adjust how long the sample plays for.

With **|GATE|** lit, the sample will play as long as you hold the pad down, until the end of the sample. When **|GATE|** is unlit, the sample will play out until the end of the sample once you hit the pad.

For short samples like closed hi hats, it may be hard to notice any difference between **|GATE|** being lit or unlit, but it can be possible to take an opened hi hat sample, and play it quickly to simulate a closed hi hat, and then hold then pad down longer to let the full open hi hat sound play out as a one shot.

Although I will set Gate on and off based on what I am trying to accomplish with a sound, I find it to be an incredibly expressive way to get more range out of a sample wether it is a single sound, several sounds played one after another, or a loop. In all of these cases you can use Gate to determine how long the sample will play out for, and it can open up a world of options. Besides the opened hi hat, I've used Gate on spoken words, bass patterns, swung drums to let additional notes through from time to time, and creating interesting rhythms with noise samples.

Loop
The Loop setting determines if the loop will sustain indefinitely when on, or just play through a single time when off. At the surface, if you load up multiple loops all at the same BPM (or setup with BPM Sync), you could theoretically trigger them on and off and they will all stay in rhythm forever. Another option to explore with loop is to sample sustaining sounds, find loop points, and let the sound play for as long as you would like.

Another option for loop is to record content that doesn't have a distinct BPM, and just different independent loops to create evolving sounds. This can be as simple as recording off several distinct sounds, enabling loop, and then triggering the loops as you wish.

With Gate on, the loop will play repeatedly as long as you hold the pad down.

With Gate off, the loop will play repeatedly until you trigger the pad again, which will stop the loop from playing.

Ping-Pong
Ping-Pong is an alternative behavior to Loop, that is accessed by pressing **|SHIFT|+|LOOP|**, and the mode is indicated by the |**LOOP|** blinking instead of staying lit. With Ping-Pong enabled, the loop will play forward, then play in reverse, and will continue this back and forth pattern until the Gate conditions to end are fulfilled similar to Loop.

Ping-Pong has a number of creative uses, and can frequently be used for smoother volume and pitch transitions for content that isn't inherently rhythmic. Pads for example can often be extended in playtime by enabling Ping-Pong strategically to get a sound to seamlessly go from playing forward to reverse without the listener noticing.

I have also found Ping-Pong to be a life saver when having chopped up a sample into distinct parts that live played content in particular may have subtle variations in a chopped segments length. While you can get some mileage out of time stretching to length a note, sometimes a Ping-Pong deployed strategically can blend a chop much more nicely.

Reverse
Reverse is pretty straight forward, and just determines if a sample will play forwards or backwards. It may not sound particularly fancy, but it can create some interesting effects to a sample quickly, and who doesn't like a reverse cymbal crash to lead into a drop?

Roll
When you hold down Roll and press a pad, it will play the note in 1/2, 1/4, 1/8, 1/16, 1/32, or 1/64th notes adjusted by pressing **|SHIFT|+|ROLL|** to cycle through the roll setting.

Oddly, Rolls are not recorded into pattern mode, so are only really useful for either resampling or adding a Roll to a pattern that is not permanent. This feels like an oversight, and I do hope that this is something that is able to be recorded into pattern mode in the future.

Mute Groups

To enter Mute Group mode, press **|SHIFT|+[PAD8]**. Mute Groups allows you to group pads together so that if any pad in the group is played, it will end the other pad in the group from playing. There are ten mute groups, allowing you to setup several different Mute Groups for different purposes in your project.

For me purposes, I will frequently use a Mute Group to take a sample that has a lot of content in it, that if I played multiple pads at the same time, would trigger too much sound.

An example of this would be taking a house drum loop, that plays a kick on ever beat, a snare on the 2 and 4, and a hat on every eighth note. This will typically leave you with chops that play a kick/hi hat at the same time, a kick/snare/hi hat at the same time, or a hi hat at the same time. Given the combinations, it is very unlikely that you will want to play and of these three sample types at the same time, so you may want to place all of these chops into the same Mute Group.

I'll also reach for Mute Groups when I am working on an instrumental for a song, which may have a bass, piano, and horns, playing at the same time. There is a lot of harmonic content playing, so frequently when you chop this up you will only want one pad playing at a time so that you do not have all of the sounds playing together.

Pad Link

To enter Pad Link mode, press **|SHIFT|+[PAD7]**. Pad Link allows you to assign multiple pads to a group, so that when you trigger one pad, it will trigger multiple pads at the same time. This can be useful for stacking sounds together to create a bigger sound, such as playing two or more kick drums at the same time, or using this to play chords if you setup your Pad Links to trigger multiple different note assignments for a sound.

Personally I will frequently reach for Pad Link when I find a sample that is okay, but needs to be punched up a bit. For example, I may have a Kick drum that I like, but it is lacking in the low end. A quick solution can be to load up an 808 bass onto a pad, assign both the Kick and 808 to the Pad Link so that they both trigger at the same time, and then I can tweak and adjust both pads individually until I get them to work well together.

Another usage of Pad Link is with loops, as it allows you to trigger sets of loops together at the same time, so that they will all start and end in time with each other. This can be useful when for example you have several layers to a song, and it is important that they are all in time. You'll find that some loops may be okay out of time and effectively give some swing to the song, while other more rhythmic portions may be important to stay in tighter time with each other.

Pad Mute

To mute a pad, press **|SHIFT|+|REVERSE|+[PAD]** for the pad that you want to mute. This will mute the pad until you press **|SHIFT|+|REVERSE|+[PAD]** again, and can be a great way to create different sections and versus of a song without having to create multiple new patterns or make multiple Pad Links to create the variations. You will know that you have enabled Pad Mute because the pad will light up red.

Typically when I make a pattern, I will make multiple copies of the pattern and make small adjustments or add in different sounds. Pad Mute can help take some of the burden off of having to copy so many variations because if you want to drop off the hi hats for example, it is as simple as pressing the button combination to mute the specific sound or sounds so that they no longer play, until you choose to unmute them.

I have found that Pad Mute also works well for loops and long running samples, allowing you to mute a sound from playing by muting the pad while you don't want it to play. It doesn't matter so much if the pads are in sync or not, but you will want to remember that this is a hard volume off for the audio and may sound harsh if not used appropriately.

Timing can be important if you choose to use Pad Mute, at the end of a bar or as part of a drop it can be an effective tool, but if you're not able to time it well within the context of the song it may be confusing to listeners as they hear a sound start to play and then just drop out.

A neat trick that was also added into firmware 1.14 is that you can have Pad-muted samples monitored to your headphones by going to Utility -> General -> Pad MUTE -> Master (instead of MasterPhn). This can make it much easier to record variations of loops from patterns or other sources, or play to existing tracks.

Pad Stop

To stop a pad from playing, the quickest way to end it is to press **|EXIT|+[PAD]**. This is particularly useful when you have turned Loop on and Gate off, and the pad keeps triggering when you press it.

Chromatic

Chromatic mode allows you take a sample and play it chromatically across all of the 16 pads by pressing **|SHIFT|+[PAD4]**. This can allow you to play out some melodies or tap out some scales from a single sample.

Unfortunately Chromatic mode has a few caveats (as of 1.14). The sample is monophonic, so you cannot play chords. You cannot play the sample into the pattern sequencer. There is no way to make your own custom scales. There's no arpeggiator to play with the Chromatic mode.

Personally I'm happy that Chromatic mode is here, despite the flaws, and I hope that it learns some new tricks in future firmware releases. It's a nice to have, even if you are pretty much stuck playing something and needing to use **Skip Back** to get at whatever you just played.

My advice is that instead of using Chromatic mode, you may find it better to **|COPY|** a sample out to multiple pads. This has the benefit of allowing for playing chords, you can create your own scale, and you can record this into Pattern Mode.

Fixed/Velocity

I'm lumping both the Fixed mode accessed through **|SHIFT|+[PAD1]** and 16 Velocity mode accessed through **|SHIFT|+[PAD2]** together, as they are both about adjusting the volume of a sample.

Fixed Velocity will make it so that any pad hit will be played at the maximum volume. This can be helpful when you are just trying to playing something out and you want to ensure that you have consistent levels on it. This can be important when you are playing back a chopped up loop and you don't want fluctuations in the volume of the sound that doesn't come from the source material.

16 Velocity mode will take the currently selected pad and spread it across all of the pads, played at increasing velocity levels. This can allow you to play a single sound at a consistent level based on the pad you hit. For example, you can play a pattern with full level notes and ghost notes by playing the appropriate pads to create variations in the velocity.

Cue

There are two ways to send samples to Cue (headphones only).

Cue mode allows you to send a single pad (no more than one), by pressing **|SHIFT|+[PAD3]**, this enters Cue mode, and then any pad you press will be output to Cue as long as you hold it. This feature can be used to quickly listen to different samples if you want to see what content is on a pad without it going to your main speaker outputs.

When in Cue mode, you can only play samples that are currently not playing. Samples that are not available to audition will not be lit up. This can limit the usefulness of Cue mode to some extent, but if you are in the middle of a performance can be a life saver to figure out what sounds are on your pads.

As of firmware 1.14, you can also send multiple muted pads to Cue. First you need confirm you have the appropriate setting by going to Utility -> General -> Pad MUTE -> Master (instead of MasterPhn). This will tell the mk2 to send any pads that are muted to the headphone outputs instead of the master output. Then, you can toggle pad mutes on and off by pressing the **|SHIFT|+|REVERSE|+[PAD]**. This will both mute the pad from the monitors, but also send the output to your headphones.

Regardless of how you send the pad to Cue, just be aware that it will skip the effects, so if you want to send audio to Cue that is effected, you will need to resample the audio so that it has the sounds printed so that you will be able to hear the sample with the effects.

Bus

The mk2 has two bus effects that you can configure to be either in serial (Bus1 into Bus2) or parallel (Bus1 is separate from Bus2). The configuration if the busses are in serial or parallel mode is adjusted by pressing **|SHIFT|+[PAD15]** to adjust EFX settings. Pressing **{ENTER}** will switch between Type A (serial) and Type B (parallel) bus modes.

Regardless of the configuration, you can select if a pad is set to send output to Bus1, Bus2, or skips the busses be pressing **|REMAIN|+[PAD]**. Pressing **|REMAIN|+[PAD]** will cycle between the currently selected bus and skipping the bus. If you want to send the pad to a different bus then just make sure that you press **[BUS FX]** to switch to the other bus.

Skip Back

The Skip Back feature is pretty much always listening to whatever audio is playing through your mk2. Any audio that is destined for the Skip Back will end up in a 25 second continuously recording buffer, which means that in general, the mk2 can be a decent end of chain device, especially if you like experimenting with sounds and what to have a fail safe to try and capture some happy accidents. Although it might not meet every need, due to the nature of how it works it definitely enables some interesting workflows that were not available on previous SPs.

Once the mk2 has started “hearing” audio, it will start recording that audio to the Skip Back buffer, and you will know that there is something in the buffer when you see that the **|MARK|** button begins slowly pulsing on and off. To enter Skip Back mode, simply press **|MARK|** and you will be brought to a view of all of the contents in the Skip Back buffer that will be up to 25 seconds of audio.

There is setting to control **[SUB PAD]** behavior under UTILITY -> SYSTEM -> GENERAL called “Sub Pad Mode” that can be set between Retrig or Skip Back. Retrig will retrigger the last pad played. When set to Skip Back, pressing **[SUB PAD]** will open up the Skip Back interface just like **|MARK|**. Why is this better?

The way that a buffer works is that once it is full, it will start overwriting the oldest data with the newest data. If you try and record 50 seconds of audio into the buffer, you will end up with only the last 25 seconds in the Skip Back buffer.

You can adjust the Skip Back and how it works in a few fundamental ways. The two primary changes you make are related to how content does or does not get recorded to the buffer, and can be adjusted based on your Input Settings and whether you are sending a sample out to monitor or cue.

One thing I’d like to point out, that prior to firmware 1.14, Skip Back was practically necessary for a lot of workflows on the mk2, but with Pad Mutes added, there are better ways to record some kinds of content. This leaves Skip Back an interesting feature, but I’m going to outline what I see as valid use cases at this point, instead of covering everything you could possibly do with it.

Audition Buffer

When you are in the Skip Back buffer, the way that you listen to the buffer is by pressing **[SUB PAD]** to trigger the Skip Back to play the buffer contents.

Save Buffer

Once you are in the Skip Back mode, you save the Skip Back buffer to a pad by pressing **|REC|** and then selecting a free pad to save the Skip Back buffer. This is quite fast, and pretty efficient.

Routing Behavior

By adjusting routing behavior, you have some control over what content does and does not hit the Skip Back buffer. This can be important in determining at a high level if you are grabbing everything coming into and playing on the mk2, or just external instruments.

As discussed in the Sampling "Recording Behavior" section, there is a routing setting under the **|SHIFT|+[EXT SOURCE]** called Routing. When routing is set to Mix, all audio that is sent to monitor and all input that comes from your Line Inputs, MIC/GUITAR Inputs, and USB-C Inputs will be mixed together. When routing is set to ExtIn, only audio that comes from your Line Inputs, MIC/GUITAR Inputs, and USB-C Inputs will be mixed together.

There isn't any "Internal" only setting, so if you need to make sure you are recording only internal sounds from the mk2 to your Skip Back buffer, you will want to make sure to either lower the gain or mute external devices at the source, or just disconnect them.

Pad Mute/Cue

Don't overlook Pad Mutes for the Skip Back recording, this is a great way to make sure that you can still hear some samples play out of the headphones, but that won't be recorded to the Skip Back buffer.

If you want to only record some of the pads to your Skip Back, you'll need to make sure your routing is set to Mix, and use Pad Mutes as outlined in the Sample Mode section. To adjust Pad Mutes, press **|SHIFT|+|REVERSE|+[PAD]** to change muting behavior for each pad. When pads are muted, the audio will not play to the Skip Back, though it may play out to headphones depending on your settings.

By using Pad Mutes, you can create some variety to your loops or patterns by muting and unmuting content, and the Skip Back buffer is a good way to make several attempts at it until you get the timing right so that you can go back and extract the best play through.

Edit Buffer

When you enter Skip Back mode by pressing **|Mark|** you will have a very similar view to what you would see when sampling editing in the Start/End menu. You're presented with the option to set start, end, and loop points with the **(CTRL)** knobs, you can set gate, loop, and reverse to adjust playing style, and you can even truncate, normalize, and emphasis a the buffer just like you would any other sample.

This can be a great way to audition something content that is in the buffer, and if you know you want it then you can work with it to get it sounded correctly before you save the buffer to a pad.

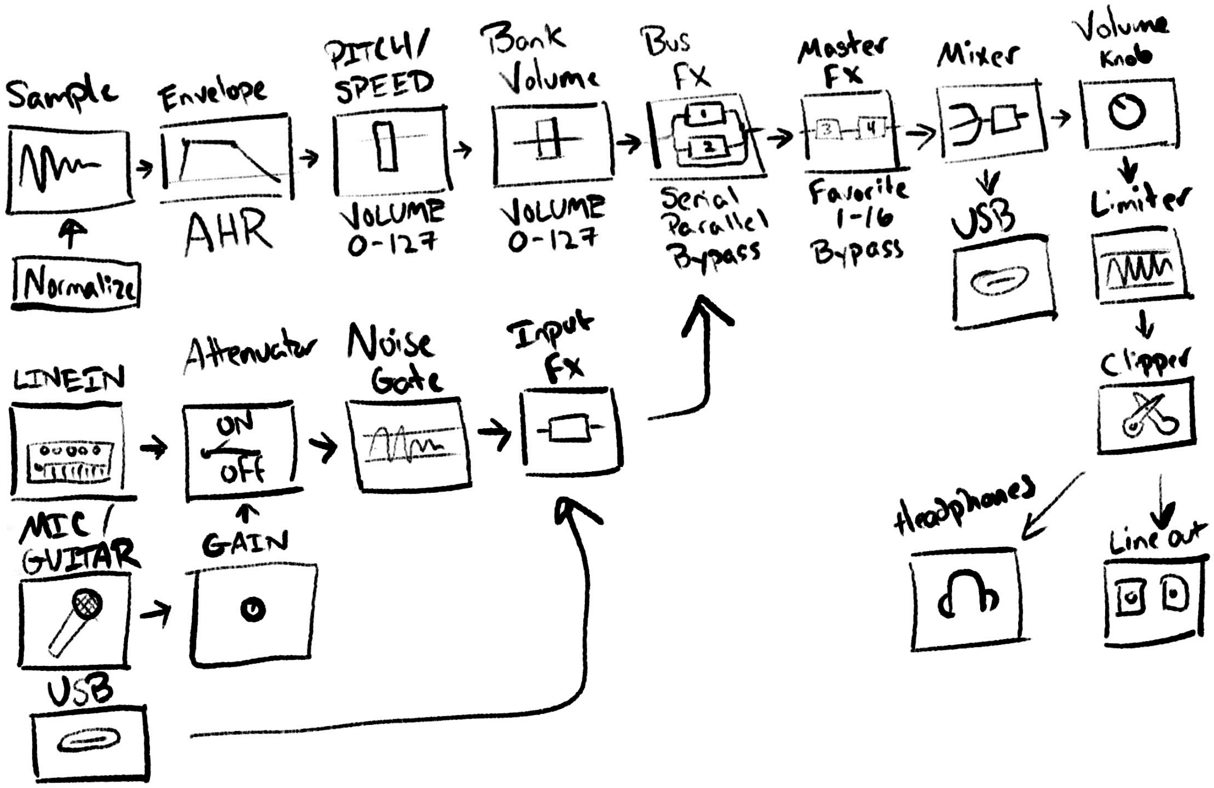
I do want to caution people against truncating the buffer. Only because it is a destructive function, and you are probably better served to just save the buffer first, and then go about your normal sample editing business because then you can make copies of the pad in case you ever wanted to go back to the original sample.

Audio Signal Flow

Alright, first thing is first. Let's be honest, the mk2 doesn't really have a mixer. That said, there's a lot of places to adjust the volume of your audio, and I wanted to capture it all in one place to try and make it clear what and where you may want to adjust volume for a sample.

Diagram

The mk2 manual has a nice diagram in the Appendix named "Audio Diagram". This shows a view that is important to understand, but I want to break this down a bit different and address each part in sections. I think I captured most, if not all of what is going on within the mk2.



Even for the rather simple architectural design of the mk2, there is a *lot* going on here to take in, and many things that can impact the volume from sample or input, all the way to your outputs on USB, Headphones, or Line Outputs.

Getting the right volume of a sample can be hard on the mk2, because there are no db meters, or anything else within the box to help you. As with many things on the SP line (and hey, sampling in general), trust your ears. If it sounds good, it probably is good. Just keep this diagram in mind if something gets hard to trace.

Sample
As usually, to modify a Sample, you'll want to press **|START/END|** to enter the Sample Edit mode.

Honestly, this is pretty straight forward, but a sample does have a volume to itself. Personally I try to sample everything in, within relation to other sounds, but you don't always have this luxury. I guess I just want to help make it clear that you normally want a Kick Drum to be louder than a Hi Hat. In general, lower frequency content should be louder, and higher frequency content should be quieter.

I will try and get the Kick and Bass drum to about the same level and I will usually target around -9db for both. I will then balance other percussive content around this, but I'll usually start at around -20db. For other instruments or sounds, I'll target -20db to -30db. This is my starting point, but do what works for you.

Just be warned, if you're struggling to get a sound loud enough in relationship to other content, you may want to run it through a normalization process. This is destructive and almost always will require decreasing the volume someplace else in the process though. This is usually my last resort.

Alright, so maybe it's not a bug… but Roland, please give us more than just Normalize, being able to adjust a sample in increments is a basic feature just about anywhere else, and would help keep making volume changes to single spaces as necessary.

If you're getting clicks and pops on your start and end of a sample, I would start here, and do my best to try and find as close to a 0 cross point as possible. I haven't really been able to tell how sticky the mk2 is to 0 point crosses, but if you zoom in far enough you can frequently get close enough. This is definitely important to find when you get into looping a sample, as it may be the only sane way to make sure that the loop point and end point match up at a transition point.

Envelope
To get into Envelope mode, press **|SHIFT|+|PITCH/SPEED|** as normal.

The envelope isn't really a place that I would go to for adjusting the volume of a sound, as it is really more for changing the character of the sound with the AHR. However, it can impact the volume of a sound, and if you are having volume issues with a sample, I'd definitely take a look here, as a high Attack, Hold, or Release value can have a significant impact on the sample's volume.

Envelope has been a consistent place for me to go to as well if I am getting clicks and pops on a sample, and Sample Edit just isn't cleaning it up well enough. I know I mentioned this elsewhere, but setting Attack to 1 and Release to 1 has been a very fast way to adjust issues at least on the Start and End point of a sample when trying to find a 0 cross point in Sample Edit hasn't gotten the job done.

PITCH/SPEED
To get into the sample's Volume, you'll need to press **|PITCH/SPEED|**. I cannot be the only one that finds this a weird place to adjust the volume of a sample, but here it is.

I'll be honest, I don't know if this volume change comes before or after the envelope, but frankly the nature of how the two work it shouldn't matter. If I ever get information that specifies which comes first and it is wrong on my diagram I'll fix it, but for now the order should be sufficient for discussion.

To adjust the Volume of a sample, you can change it in this mode from 0-127. A setting of 0 effectively mutes the sample, while a setting of 127 will be the highest the sample can play. If you've Normalized a sample, this is most likely going to be the setting that you want to come to in order to adjust the volume of the sample in relation to the other samples you are playing.

The Volume change here does not provide gain to a sample. Setting this to 127 will not make the sample louder than what it was recorded in at. This setting functions as a pretty straightforward attenuator that will only be able to make the sample play at a lower level.

Bank Volume

You can get to the Bank Volume by pressing **|SHIFT|+|BANK|**.

The behavior as of firmware 1.14 is a little weird. No matter which bank you select, it will open up a menu to edit the currently opened bank. Just make sure that you pay attention to which Bank Volume you are adjusting when making the change.

Similar to the Sample Volume under the **|PITCH/SPEED|** menu, this setting is from 0-127, and impacts all of the samples within the selected Bank. Also similar to the Sample Volume, this can only further reduce the sound, as a setting of 127 will play the samples in the Bank at their loudest volume, and a setting of 0 will effectively mute all samples in the bank.

With a little organization, or adjusting your workflow, Bank Volume can be an effective way to adjust many sounds in relationship to each other. Consider having a bank of drums, then a bank of bass, and a bank of piano chords. With this in mind you can adjust the Volume of types of sounds in relationship to each other.

Line In

The signal level for your Line Input is best controlled by adjusting the volume on the device that is outputting. In general, you should set the Volume on your device to be as loud as possible without clipping (**[EXT SOURCE]** turns red).

Mic/Guitar

The signal level for your microphone and guitar is unlikely to have a gain control, though some might. The important thing is to adjust the switch to MIC/GUITAR on the front as appropriate for the device that you are connecting.

Gain Knob

There is a gain knob on the front of the mk2, beside the MIC/GUITAR switch, and this is used to amplify the incoming signal. Adjust it to taste, but note that you typically want to do this without clipping (**[EXT SOURCE]** turns red).

Attenuator
I suspect that the attenuator is referring to some of the settings that are available to adjust how loud incoming signals are, but I am not positive.

Noise Gate
The noise gate is in place on the mk2 to ensure that when there is a sufficiently low signal level, it will drop the incoming signal to effectively mute it until it hears a sufficiently high level signal again.

??? Test what this volume threshold his ???

USB In
The most important thing to note about the USB input is that it skips the Gain Knob, Attenuator, and Noise Gate. This is most likely a design decision that was made because only digital audio is passed between the mk2 and the USB host, and as such it won't have to manage noise generated by analog signals.

There's not a great way to adjust the volume of incoming USB input. Personally I kind of feel like this is in bug territory, and it is frustrating that allow there is a USB Out gain, there is no USB In gain. I'll continue shaking my fist angrily at the sky.

Input FX
The Input FX section can be opened by pressing **|SHIFT|+[EXT SOURCE]**.

All of the inputs (Line, Mic/Guitar, and USB) pass through the Input FX section. This can be set to Bypass so that no effects are used, but it can also be used to inject one of sixteen different effects.

If you are finding that your inputs seem to have an unexpected volume or character to them, definitely check here to see, and set to Bypass while you try and track down any possible issues.

Bus FX
There's a number of things to know about this section, but I'll start by letting you know that this is the first place where you can adjust both the character of a sound, and depending on the effect, also increase the Volume of a sound over a sample's original level.

Let's recap on a few things for Bus FX before we move on.
* You adjust BUS FX as Serial or Parallel in EFX Settings **|SHIFT|+[PAD15]** (Type A or B)
* You select which Bus you are modifying by pressing **[BUS FX]**
* You adjust which Bus a sample is assigned to by pressing **|REMAIN|+[PAD]**

The big points to remember is that you can decide if you want to send a pad to Bus 1, Bus 2, or bypass the Bus FX all together and go straight to the Master FX section.

I won't be able to cover everything here, but there are effects here that can actually increase the volume above what a sample was set to. Isolator is able to adjust Low/Mid/High individually to up to +12db, which can lead to clipping. The Compressor and the 303 Sim are both able to use compression to increase the overall volume up to the maximum volume, making them good candidates to get a louder sound out of your samples, but at the potential expense of reducing your dynamic range (the difference between how loud and how soft you can play a sound).

I'm reluctant to dig into too much more in the Bus FX section at the moment, because I'd rather spend most of the time digging into the details in the Effects section, but depending on feedback I may add some more content here in the future.

Master FX

The Master FX are useful, but I also find it a bit of an odd duck. You can get into the Master FX section by pressing **|SHIFT|+[PAD16]** and you will be brought to a menu where you can select to use Bypass, or one of the up to 16 favorites. The Master FX are labelled as Bus 3 and Bus 4, but as of firmware 1.14 this is the only way to access them, so you're not really able to do much other than set them and forget them, or torture yourself with menu diving.

At this time, I mostly just use the Master FX as something I turn on when I am trying to put the finishing touches on a track I have recorded, and usually turn it off otherwise. It is just too easy to forget that the Master FX are on, and adjusting your audio without realizing or thinking about. Double compression is easy "**whoosh**".

The Master FX section is very similar to the Bus FX section, and there are effects that you can use to boost the overall volume of your audio. Using Isolator or Compressor here is going to have the same effect on as a sound as it will in the Bus FX.

When something seems to be working a bit weird, or volumes aren't as you'd expect for your samples, the Master FX section is definitely an out of the way place to look, as it is easy to forget that it is on, and you have to specifically go here to see what the settings are.

Mixer

The mixer is where all of the audio sections come together and are mixed. As far as I can ell everything is mixed equally based on each section's settings. It is unfortunate that you cannot adjust anything here, I would say that overall there is enough control in previous sections to manage this.

The only thing I'd really like to see made available would be some kind of level meters. A level meter on the Mixer, and strategically placed at the Bank Volume section would really go a long way towards knowing roughly what the levels were of your sounds.

Volume Knob

Once the audio gets past the Mixer section, the Volume Knob can be used to attenuate the output that goes to both your Line Output and your Headphones. Unfortunately they are both tied together, so to separate them you need to make adjustments in the UTILITY -> SYSTEM -> GAIN section.

The Volume knob does not effect the volume of audio that goes out to the USB connection.

Limiter

After the Volume Knob, the mk2 has a Limiter that will ensure that your audio does not go over a certain threshold. This is likely somewhere between 0db and -3db, but I haven't found anything to clarify this.

??? I have not tested to see what kind of limiter it is, and I haven't checked the manual to see if it specifies ???

Clipper

Generally you'd use a Clipper to help drive your audio output gain. I'm assuming that the mk2 is doing something like this, but I suspect an experiment is in order here.

??? Consider how best to get content off of the outputs to determine what the clipper is doing ???

USB Out

The USB Out skips the Volume knob and just comes off of the Mixer. As a result, there aren't many good ways to adjust the output from the mk2 over USB. However there is one setting under UTILITY -> SYSTEM -> GAIN called USB Out that allows you to adjust the volume of the overall output to USB by 6db increments.

Line Out

This is your main/monitor line output. It will play anything that is specified to go out to your monitors (ex. is not Pad Muted).

Headphone Out

This is your cue output. It is capable of playing cued content and monitored content.

Effects

The mk2 has a total of five effects available. There is one input effect that can be used on any audio that comes in through the Line Input, Mic/Guitar Input, or the USB Input. There are then two Bus effects that the input and samples can be routed to or bypass, and the Bus effects can be Serial (one after the other) or Parallel (both are separate). Finally there are two Master effects that all audio will pass through before going to the outputs.

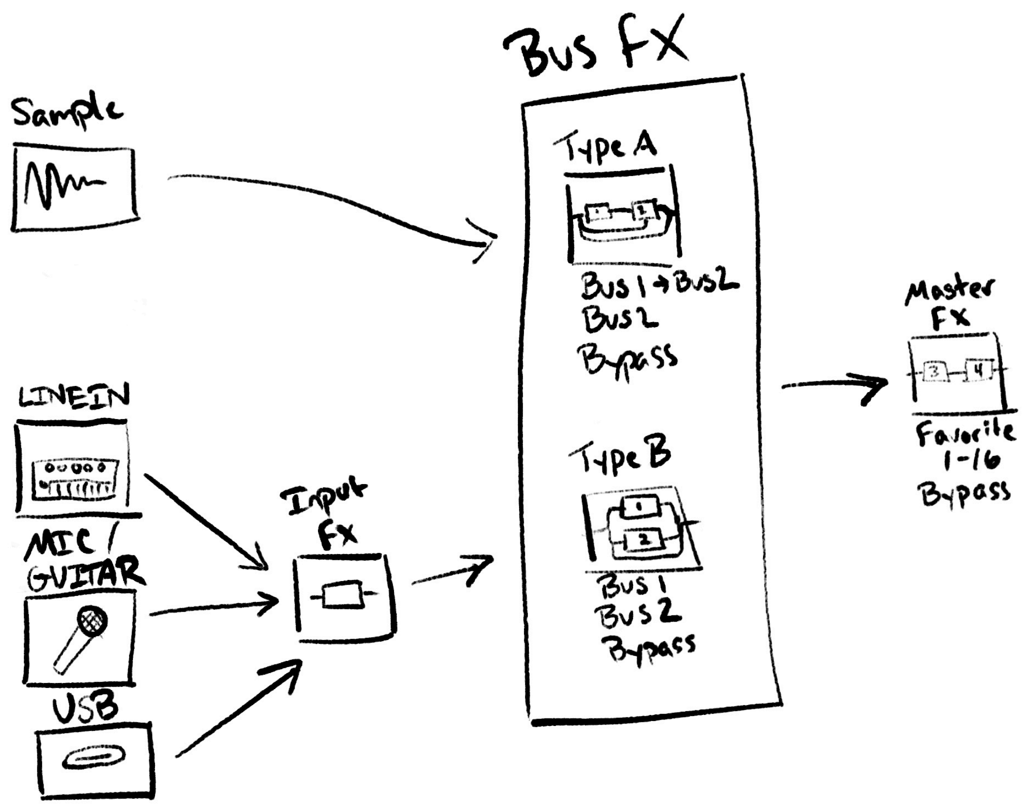
With this information, if you were using the mk2 as just an effects bus then, you could technically get up to five effects running at the same time for any content that is being input. Given the cost of a lot of guitar effect pedals, this might put the mk2 into a reasonable price category just as an FX box.

Unfortunately, there's no clean way to play your samples through the input effects, but given that there are only three input only effects, I wouldn't see this as a big loss.

??? I am quite certain you can use the Monitor/Cue to technically output Cue'd content back into Line In, but need to test ???

Diagram

The Mixer diagram is more detailed, but this view is intending to show at a high level how you can route audio through your Bus FX (1 & 2) and Master FX (3 & 4) for audio inputs and samples.



??? Leaving this as a quick note to come back to, but when using Resonator, and bypassing Bus 1/2, audio was also bypassing the Master Bus 3/4… ??? Is there something special here or are the diagrams wrong?

Controls

Settings

Effect Types

This table is intended to just show which effects are available for which effect slots of Input FX, Bus FX, or Master FX. I thought I had seen this table someplace, but the way it is formatted in the reference manual didn't quite work for me.

Effect	Input FX	Bus FX (1 & 2)	Master FX (3 & 4)
303 VinylSim	X	X	X
404 VinylSim	X	X	X
Auto Pitch	X		
Bypass	X	X	X
Cassette Sim	X	X	X
Chorus	X	X	X
Chromatic PS	X	X	X
Compressor	X	X	X
Crusher		X	X
Distortion		X	X
DJFX Looper		X	X
Downer	X	X	X
Equalizer	X	X	X
Filter+Drive		X	X
Flanger		X	X
Gt Amp Sim	X		
Ha-Dou		X	X

Effect	Input FX	Bus FX (1 & 2)	Master FX (3 & 4)
Hyper-Reso		X	X
Isolator		X	X
JUNO Chorus	X	X	X
Ko-Da-Ma		X	X
Lo-fi	X	X	X
Overdrive		X	X
Phaser		X	X
Resonator			X
Reverb	X	X	X
Ring Mod		X	X
SBF		X	X
Scatter		X	X
Slicer		X	X
Stopper		X	
Super Filter		X	X
Sync Delay			X
Tape Echo		X	X
TimeCtrlDly	X	X	X
To-Gu-Ro		X	X
Tremolo/Pan		X	X
Vocoder	X		
Wah		X	X
WrmSaturator	X	X	X
Zan-Zou		X	X

A few interesting things to note:
3 Unique Input FX: Auto Pitch, Gt Amp Sim, & Vocoder
1 Unique Bus FX: Stopper
2 Unique Master FX: Resonator, & Sync Delay

The 1.14 mk2 reference guide calls out two “VinylSim” effects. One references the 303 VinylSim and the second references the 404 VinylSim. There are no additional VinylSim vinyl simulation effects.

A few weird things here for sure, but can probably be cleaned up.

Descriptions

Instead of going through and describing the effects the same as they are described in the manual, I’m going to go over each effect, outline what it might be useful for, and any other notes that might feel relevant at the time.

---

303 VinylSim

This is an emulation of the Vinyl Simulation from the SP-303. It sounds good to my ears, but I have not done a side by side comparison of the effect yet. The stand out feature here is the compression, as you can dial in how much or how little you would like on the track.

Overall I’ve been happy with the parameters and sound of this effect. ??? Is it worth doing an A/B comparison of this effect ???

---

404 VinylSim

This is an emulation of the Vinyl Simulation from the SP-404. Some people prefer the noise of this simulation over the 303 and vice versa. A reason you may reach for this effect is because you can adjust the frequency characteristics of the playback, which you cannot do on the 303 simulation effect.

Honestly, you’ve got both, try them both out and see what you like.

Auto Pitch

The manual says that this processes the human voice to create a variety of characters, which is true, but it can also be used on instruments.

I’ve personally used the effect a tiny bit on the mk2, and found it to be suitable for the task. It compares well to similar boxes with similar auto pitch effects, and if you want to sound like a robot it does have some room for changing the formant and robot parameters.

Cassette Sim

This effect simulates the sound of playing the audio through a cassette tape. All the fine effects are available including adjustments to the tone, tape hiss, the age of the tape, distortion, wow/flutter, and tape catches. There is a lot going on, and a lot to like about this simulation.

Even though I really like the 303 VinylSim effect, lately I've found myself reaching for this effect more when I want to get a bit more of a dirty sound. Maybe it is just because it is new, maybe because there is more control over the effect, or maybe it is just because it is so good. Compared to some other tape sims, it's missing a few features, but because it was kept to six parameters, I have to say, the probably hit the right ones. Other effects in the chain can pick up the slack for whatever else you may need.

Chorus

The chorus effect is frequently a good choice when you want to fill out a sound and use a sample or sound source to harmonically fill itself out. With the depth and rate parameters you can go from pleasing harmonic characteristics to spaced out alien noise.

I put this effect in the bread and butter section of effects. It's always good to have around, but at this point it is not an exciting effect that people will reach for. Definitely use it to fill out your melodic, vocal, or other content that could use some voice doubling to make it a bit more present.

Chromatic PS

This is a two voice pitch shifter, that allows you to change the pitch for each voice in range of -24 to +12 semitones. This provides a large range with which to work with, allowing you to create chords out of a single note.

I do like this effect for the ability to create some interesting progressions from a single note, but even if you stick to semitone groups that work, it feels like it frequently creates some inharmonic or otherwise sour notes. This does feel like an effect that is worth a deep dive though, and with particular uses may provide some interesting ideas on how to work towards building out a song.

Compressor

The compressor is like most any compressor, it makes everything have a smoother volume level, and reduces volume variations. Or, in other words, it reduces the dynamic nature of your audio. This will tend to be a tool you will reach for to help keep a more consistent playing feel for instruments, but can also be used to make drums or other sounds snap just a little bit more to increase their presence.

I am a strong believer that a little compression goes a long way. Used sparingly this effect can give your tracks a bit more oomph, or help an instrument pull up into a piece. Similarly though, used too much, compression can make everything feel the same in your track, and kill any subtle emotion from dynamic changes in playing styles. Just be careful with it, and you'll do fine.

Crusher

The crusher is pretty clearly a sample rate reduction effect, and it sounds like it adds a bit of ring mod to it as well. The filter parameter seems like it is a pretty standard low pass filter, and the rate adjust how low the sample rate will go. There is even a balance parameter for adjusting how much the original sound is mixed with the crushed effect.

I usually like this kind of sound, but I feel like the resonance or ring mod parameter needs to be exposed, because for a lot of content it just feels like it rings a bit too much. It's not necessarily an unpleasant sound, but sometimes it just need to be dialed back.

Distortion

The distortion effect is about as aggressive as you'd want to go in the family of saturation, overdrive, and distortion effects.

I'm not sure what it is about this compared to the overdrive, but I feel like I like the tone a lot more on this effect. Glad that there are options, but this is the one I find myself grabbing more frequently when I want that distorted sound.

DJFX Looper

The DJFX Looper effect will loop over a small amount of audio based on the selected length parameter. The effect can make for some impactful transitions, but personally I feel like this effect gets overused on SP songs. Take care people, to only use this effect where appropriate.

It is good for adding some stutters and glitch effects though, definitely worth practicing with it to see what you can get.

Downer

This effect slows down the playback speed of the audio, and has a resonant high pass filter to further sculpt the sound and to help keep the audio from getting too bass heavy.

I find this to be a good effect for creating a darker segment of your track, and depending on the rate, can add some additional rhythmic and ringing tones to your audio.

Equalizer

This is a pretty standard three band equalizer, though it doesn't have quite the granularity of modern three band equalizers. It operates very similar to Isolator, but instead of having set low, mid, and high frequency bands, you instead have a bit more control over the bands to be a little bit more prescriptive on how you modify a sound.

Personally, I'm not reaching for this much. Yes it is more configurable than the Isolator effect, but if I really need to do heavy EQ work, I'm probably going to want to reach for a dedicated device to manage the EQ adjustments, and not a six parameter equalizer. Don't get me wrong, I am glad that this is here, but unless Roland ever adds the ability to save and name settings so I can cycle through different favorite settings, this is one that I'm not sure I'll have the patience to reach for frequently.

Filter+Drive

This effect is simply a low pass or high pass filter with overdrive. It can be put to good use to overdrive a sample, and then pull back either the high end or low end depending on which filter mode you chose.

It would be an effect that would traditionally be seen in a guitar pedal chain such as an overdrive followed by a low pass filter, and as such is good for processing guitars, but I do find it can impart character to other samples such as drums if used sparingly or as a way to create layers of sounds through resampling.

I do think that this effect could use a bandpass filter as well, but that does tend to require more parameters than the mk2 has available. If you do want a bandpass, my advice would be to consider opening the filter all the way up here, and following up with the Super Filter and use it’s band pass filter.

Flanger

A flanger is an effect that plays a signal slightly out of phase and slightly delayed from each other. The flanger is quite similar to a phaser, but it tends to have more of a comb filter sound that can produce a bit of a whopping noise. It can make some interesting stereo effects due to the phase shift.

I’m not sure what else to write about the flanger, it sounds fine, and is another bread and butter effect especially for a guitar. Check it out?

Gtr Amp Sim

This effectively simulates a guitar amplifier. The selection of options is certainly sufficient, and the parameters on offer are pretty much to the point to allow you to adjust between bass and guitar sounds.

I find this effect to be good, especially as I don’t have a guitar amplifier or any other hardware that can directly simulate one. This isn’t likely to blow anybody’s socks off or win any awards, but for those of you who just need a guitar amp, and don’t want to carry one around, I think you’ll be pleasantly impressed with what this offers, especially when you consider that you can stack another four effects on top.

Ha-Dou

The manual says that this effect generates a wave-like sound, and while this is true, I might describe it almost as similar to the Resonator effect in that it is making a sound based on your audio, but both your audio and this effect are mixed together. There is no way to make this effect 100% wet, so you will always have the audio from the track playing alongside this effect.

So far, I have not really found a solid or consistent use for this effect, but honestly I haven’t played with it very much at this point. For me it feels like a somewhat inconsistent reverb, making it a possible sound design tool, but I am not sure what else to do with it right now.

Hyper-Reso

This is a resonator with the manual describing it as creating melodies and bass lines easier. There is a fair amount of tuning options and ability to adjust the spread, scale and character, allowing you to really dial in a sound.

I'm a bit on the fence for this effect, as it does feel like it kind of makes everything through it feel a bit similar. On the other hand I do kind of like this effect, it can give a bit of a dark edge or bright sheen to select samples, and I suspect, like a lot of other effects, when used judiciously it can really help strike a tonal character in a few places that are worth investigating. Overall, I just don't think it is the effect that you want to push everything through.

Isolator

The Isolator effect is a three band boost/cutter. Although you can only adjust volume for the three bands based on low, mid, and high, it is a quick effect that can allow you to give some oomph to part of the track while taming other parts that might be a bit too much.

Although it doesn't allow you to adjust the frequency bands for the low, mid, and high bands, I do find this to be a pretty quick and useful effect to change up a sound while playing live.

JUNO Chorus

This effect models the Roland JUNO-106 and JX series chorus, and although it has limited parameters compared to the stock chorus effect, it also has a much different character. Definitely try this out, but due to the limited parameters you're either going to like what you hear or you won't.

I am happy to see the balance parameter here, and I find that this does a lot to help adjust the tonality of the effect in another dimension that you cannot quite do with the stock chorus effect.

Ko-Da-Ma

This effect is somewhere between a sync'd delay and a reverb, and can add some very interesting background texture to a track.

In my uses I have found it to be quite responsive to both bass lines and drum kits, and at larger values can create some very interesting melodic lines. It is not a go to effect for me, but it is something that can add some interesting variety and texture and is certainly worth checking out.

Lo-fi

The manual says that Lo-fi degrades the tonal character, and while this is true, it isn't exactly apparent what it is doing. It sounds like there may be some sample rate reduction, possibly some sample bit reduction, but the manual isn't specific on what this effect is doing.

To my ears it does sound good and adds some character to a sound, but you may end up finding you spend a lot of time adjusting the lofi type, tone, and cutoff without knowing exactly what it is doing. I place this in that sample and bit rate reduction category, but I do feel like the manual could have done a better job explaining what the effect is actually doing for at least the lofi type setting.

Overdrive

This effect overdrives the sound, and is most likely to be the effect you'll reach for and immediately think you made whatever you sampled turn into a guitar. Is this a good thing or bad thing, it might depend on how hard you push the overdrive.

For me, just like saturation and distortion, I find overdrive to generally be good in moderation. You can add a lot of sonic character to a sound with overdrive, and can help sounds punch through a mix, but it can also take away a lot of your available space to place sounds. In the end, I tend to find overdrive better when you have fewer elements, and worse when you have a lot of different elements in a track working together.

Phaser

The phaser is your general phase-shifted sound effect. It is related to the flanger, but it is a bit of a simpler effect.

The effect works as advertised, and is one of those bread and butter guitar effects. Try it on some guitars or melodic content, it is sure to sound quite 70's to you!

Resonator

The Resonator effect allows you to turn your mk2 into a physical modeling synthesizer of sorts. It uses the Karplus Strong method to simulate plucking strings by playing a sound through it. This method usually loops a short sound through a filtered delay to generate a string note. It is a fascinating method, and definitely worth looking into if you're interested in using your own effects to achieve similar results.

The first weird thing about this effect is that it is (as of firmware 1.14) only available in the master effects (bus 3 & 4).

Reverb

The reverb effect is your standard run of the mill reverb. It simulates an ambient, room, and two different hall reverbs. Being able to adjust both the low cut and high cut, as well as pre delay may make this a reverb that a guitarist might reach for.

What I don't like about this reverb is that there is no ability to adjust the mix or balance, your main signal is always on, and you can only adjust the level of the wet signal. This makes it hard to suggest this reverb as useful for creating large and lush pads because you always have the full signal of whatever you are running through the reverb. It is a serviceable reverb, I'd just like to see it able to do a bit more.

Ring Mod

The Ring Mod effect is typically described as adding a metallic characteristic, and this is no different here. If you want to make a sound feel more like a bell, brass, or metal plates, this is the effect for you.

I'll figure out something else to write here later, but this is just a completely familiar sound in the guitar space, and I'm not sure what else to write at the moment.

SBF

This is the sideband filter, and allows you to adjust to a few different filter types, and thankfully you can adjust the balance from 100% dry to 100% wet, which really opens up more interesting uses as a sound design tool. The width seems to let you adjust the combs for the filter, and it definitely has some resonance going on which can produce some subtle to in your face ringing.

I find this effect to be quite interesting from a sound design tool, and as an effect that you can mix in and out to really change the sound of a track.

Scatter

The scatter effect has several scatter types, that take an audio buffer and break it apart and play it back into different patterns based on the scatter type. It is similar to the Slicer effect, but it puts slices of the audio out in a more gated style and can alter the playback direction of the sound.

I find this effect leads to a more glitchy sound than Slicer, and the depth and speed parameter can create a variety of tones with the main sound.

Slicer

This effect slices up an audio source into small pieces, and can create an effect similar to a gated filter. By adjust the pattern and speed parameter the effect can cover a lot of different rhythms.

I find this effect to be fairly effective on rhythmic content like drums, but it can be incredibly effective on sustained sounds like a bass tone or a pad that plays for a long time.

Stopper

The Stopper effect is only available for Bus 1 and Bus 2, and allows you to get the sound and quality of stopping a record or a tape. I am sure that the reason this only shows up on Bus 1 and Bus 2 is because the menu diving to turn it on and off would be more hassle than the effect is worth.

My advice for this effect is to consider only using it on some elements of your track, so that you still have some kind of rhythm, melody, or other sound playing so that you have a reference to be able to bring the track back in. Otherwise, once stopper is engaged, you'll have no frame of reference for when to disable it and get your track playing again.

Super Filter

The Super Filter is documented as having an extremely high slope, and the rate of filtering can be adjust to creating a consistent pumping effect on the audio it is applied to.

With the high slope, I find the effect makes it quite good for use as a filter gate if pushed to the highest depth. It doesn't afford any directly interesting patterns, but the rate parameter can get you some of the way there by adjusting note frequency and changing to dotted or triplets.
With depth, you can get this filter to go from subtle to very extreme.

Sync Delay

This effect only appears in the Master FX Bus 3/4, and is a delay that will be in sync with the tempo of your currently selected pad bank.

??? Need to play with this one, as I didn't even know it existed until I put the table together ???

Tape Echo

The Tape Echo effect simulates having between one and three tape heads on a tape to produce an echo, and it does allow for wow and flutter to be added to the echo'd sound.

I do like the sound of this effect, but I wish that it allowed you to modify the dry/wet balance instead of just how much the level was for the effected audio itself. As it stands, it does allow you to bring the tape echo sound in, and honestly you can use the Cassette Sim effect if you want to have wow and flutter on your main audio. There are options obviously, but the Tape Echo and Cassette Sim while sounding similar don't produce quite the same result.

TimeCtrlDly

This delay allows you to adjust the time of the delay, and although the pitch is going to change while you are adjusting it, it is a pretty smooth change. The low pass and high pass filter damping really get you some ability to adjust how the feedback and delay repeats sounds. Definitely a nice sounding effect to create some backing audio.

Personally I like the features of this delay, and think it is a good tool for creating ghost notes, alternative rhythms and a slew of other delay related sounds.

To-Gu-Ro

The manual says that this is an undulating effect, and it certainly has some rhythm to it. To me, it almost feels similar to the Downer effect or a kind of dropper effect where it is taking the audio that is being played and changing the rate that it is played back.

I'm not sold on the use of this effect much though, as the input is still playing, and you end up with your original signal mixed with this effect. I find it odd that you're not able to use this purely as an effect for interesting sound design applications.

Tremelo/Pan

This effect will vary the volume and panning of a sound, and is another staple of guitar pedal effects. When set to tremolo mode it can be used to make a gated filter kind of effect, and when set to pan mod, it will pan your signal between left and right.

I like this effect for adding some space to a sound, or adding some subtle dynamic character to a sound. This makes it good for content like pads and melodic content that you'd like to add a bit more movement.

Vocoder

This is a pretty standard vocoder effect, where you take one sound, and impart the quality of another sound onto it. This gets you a lot of those kinds of sounds where it a synthesizer seems like it is speaking.

To me, this is a decent effect, and definitely has application outside of just voice work to just experiment with sound design. It's also nice that you can control the note and pitch bend from an external keyboard as this opens up some additional expression for this effect.

Wah

This is your general wah-wah sound. Another staple of guitar pedals.

I'm not sure what else to say, give it a try. I promise I will come back and write up more when I am not mind dumbed by these general purpose effects.

WrmSaturator

The Warm Saturator effect is quite characteristic of a saturator effect, allowing you to drive the signal in similar way to what an analog tube preamp might sound like. The difference between saturation, overdrive, and distortion can be subtle, and you'll want to be careful not to drive your signal too hard or it may distort. That said, this does lean towards a warm sound, and can help add some much needed character to digital samples.

Personally I find saturation is best when used right on the edge of overdrive or distortion. Certainly the more sound, the subtler I would suggest being, though for a single source if it sounds good, then it be good to drive it harder. My suggestion is when you use this to try and drive your audio to distortion and then pull it down to where you like the sound. From here turning the effect on and off to see if you like the change can help hear subtle changes to the sound from saturation and see if they are appropriate for the sound or track.

Zan-Zou

The document says that this effect applies a delay to the negative phase of a sound, and definitely gives an interesting sonic afterimage. When using this, you'll probably need to tune the high pass frequency dampening to make sure you get the sound that you are looking for.

I need to try out this effect more, but I suspect it will be something interesting for introducing a bit of additional stereo presence for pads that could give them some more space. Certainly something to play with and see how resampling some audio can give some more tone to a sound.

Patterns

DJ Mode

System/Config

Settings

When you enter **“UTILITY" |SHIFT|+[PAD 13]** you can get to all of your settings for how your SP-404 mk2 will operate. The UTILITY MENU allows you to make SYSTEM changes, PAD SET changes, EFX SET changes, IMPORT (and export) Projects and Samples, BACKUP, and FACTORY reset.

Turning **{VALUE}** will allow you to select which UTILITY MENU you would like to access, and pressing **{ENTER}** will allow you to enter the specific UTILITY MENU that you have selected.

By turning **(CTRL3)** you will be able to scroll through settings for GENERAL, CLICK, MIDI, GAIN, and VERSION.

These settings will effect the overall operation of your SP-404 mk2.

Does this change… project to project… saved with a backup/restore? Experiment to find out!

Import/Export
* File Structure
* Projects
* Samples

Midi
* Each midi channel is to address each bank of a mk2 project.
* Supports velocity, but doesn’t appear to have any other CC automation available.

Boss (and Roland???) use Type A TRS to Midi cables.

External Connection

* Midi - channels, controlling external gear (probably needs it’s own section really)
* Phone
* Tablet
* Cassette/Turntable/FM Radio
* Computer

Integration

* Computer
  * Reason Studio
  * Ableton Live
* Guitar
  * Electric
  * Pedals
* Microphone
  * ???
* Hardware
  * Akai MPC
  * Mixer
  * Midi Interface
* Eurorack
  * Audio
  * Midi
* iPad/iPhone
*

Customization

Face Plate

Knobs

* You can change the knobs by pulling up on the knob to slide the current one off, and replace it with a similar knob. Size??? Dealer???

Chroma Caps are well known to work - https://store.djtechtools.com/collections/chroma-caps-cables/products/chroma-caps-knobs-and-faders

Display

* How to update the boot up screen on your display

Cases

Analog Cases has the Pulse case that fits both the SP-404 and the mk2.

* https://analogcases.com/products/sp404-pulse-case
* Notes: Has room for the power supply and a good amount of cables

Magma Bags has the CTRL CASE SP-404 that will fit an SP-404.

* https://www.magma-bags.de/ctrl-case-sp-404.html
* Notes: Mk2 is not listed as fits. Does not appear to have room for power or cables.

SP 404 mk2 App (roland cloud?)

Sample Packs

Appendix A - SP Use Cases

* SP as a buffer
* SP as a tape recorder
* SP as an effects box
* SP as a master bus

Appendix B - Beat/Drum Patterns

Appendix C - Workbook

* Foley Loops
* Steps to record a 16 bar loop and chop to pads
* Musiquie Concrete
* Other Tape-Music techniques with unsync’d timing
* Collaboration - how to use the mk2 to work with others

Appendix D - Finger Drumming

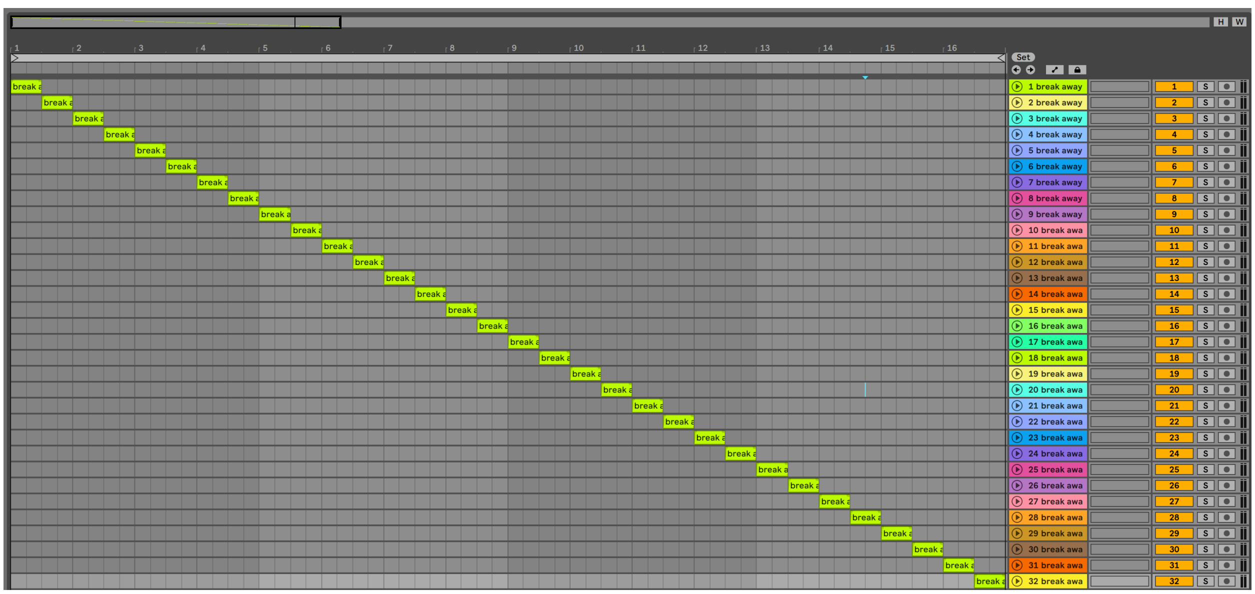
Appendix E - Experiments

Is the mk2 polyphony monophonic or stereo?

Most devices denote polyphony based on how many simultaneous sounds or differing notes that they can play at the same time. For samplers, this is traditionally how many monophonic samples can play at the same time. A stereo sample requires playing two monophonic samples so that you can get one sound for the left channel and one sound for the right channel.

Theory
I'm not terribly sure here, my hopes and dreams however are that the mk2 is able to play 32 stereo samples at the same time. Time to find out if my hopes and dreams will be crushed or not.

Method
To actually test this out, I have created a silly 32 track Ableton Live track, where each track plays half a bar of a Drum n Bass song I wrote years ago Break Away (linked if you actually want to hear the song). From here I just kept scrubbing from each track just one half bar section of the song, until I had 16 bars, or 32 half bars. In Ableton Live it looked like this:



With this done, I then exported each track to a separate stem, leaving me with 32 tracks, each with 16 bars in length, that only play one small portion of the song, and loaded up the tracks to my SD card so that I could load them up on the mk2.

Loading them into the mk2 was a little tedious, just because I had to load one stem at a time, and to keep my sanity I loaded track 1-16 to Bank A Pad 1-16, and track 17-32 to Bank B Pad 1-16. One last pass was to turn GATE off for all of the pads, because even though the samples were mostly silence, I wanted to make sure that they played the full length of the sequence.

Pattern Sequence #1

I set the BPM to 84 (which is the BPM of the stems), and created a new 8 bar pattern that would play all of the Bank A pads. 8 bars is only half of the stems, and requires 16 stereo polyphony (or 32 monophonic polyphony) to play.

It loops properly, and there are not dropped notes.

Pattern Sequence #2
Similar to Pattern Sequence #1, I set the BPM to 84, but this time I created a 16 bar pattern that would play all of the Bank A and Bank B pads. 16 bars is the full number of stems, and requires 32 stereo polyphony (or 64 monophonic polyphony) to play.

You can immediately notice dropped notes, as pads that had been lit up, are now randomly (or based on event entered order) are stopped to allow other notes to play.

Further Testing
To be really specific, I would consider running almost the same experiment, but output all of the samples as mono left and right channels. This would give further proof that the mk2 can handle mono samples, and that it is able to output 32 mono voices at the same time.

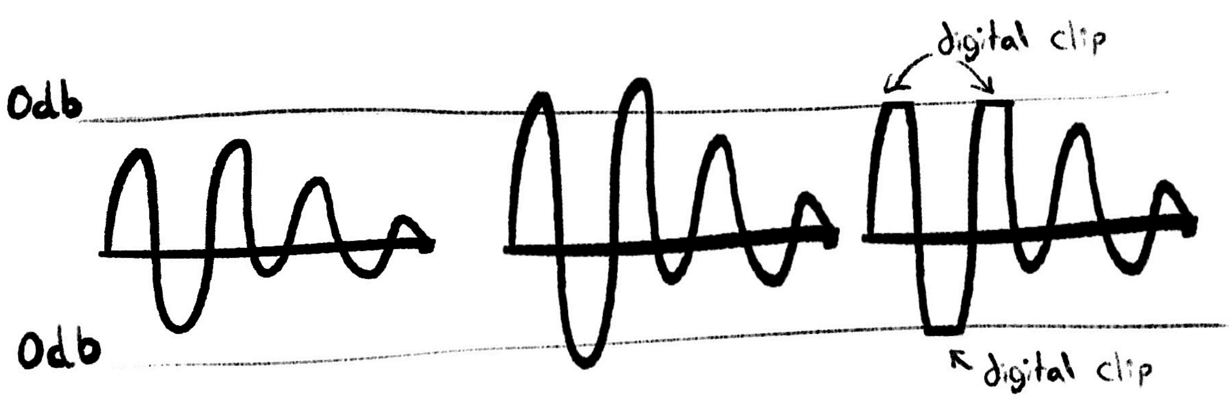
Conclusion
Sadly, the mk2 is 32 note polyphonic for mono samples, and 16 note polyphonic for stereo samples. I am actually fine with this, I was able to make tons of music with 32 note mono polyphony on the likes of the MPC 500, but my gripe is now that there aren't any direct ways to convert stereo samples to mono samples once you have them loaded into the mk2.

Probably more of a feature request, but I'm marking it as a bug for now (no icon for feature request), that to fully take advantage of your polyphony within the mk2, the mk2 needs to have an option to convert samples to mono… probably under the **|START/END|** menu that you can to with **{PUSH ENTER}**.

Does EXT SOURCE turning red mean that you are clipping or not?

Clipping occurs on audio devices when a signal goes over their maximum threshold. On analog devices, this can sometimes produce interesting results and new harmonics, but on most digital systems, 0db is the maximum for a sound.

Below is a sketch to illustrate what happens when a signal has been digitally clipped, and is a result of the input signal being louder than the 0db maximum. If you see flat tops to your sampled waveforms, unless you are working with square waves, then you're almost certainly experiencing some digital clipping.



Theory
I believe that the mk2 is going to digitally clip on the 1/4” inputs.

Method
I hooked up my trusty Moog Voyager to the 1/4” inputs on the mk2, and dialed the Moog in to a single oscillator with a triangle waveform. This allowed me to pretty clearly visually see if the tips of the triangle were still in tact or not when I was recording samples.

Recording #1
I dialed in the output volume from the Moog Voyager until the EXT SOURCE lit red, and then dialed it back slowly until it went back to the amber color. I then went into the sample and visually inspected the points of the triangles, and as far as I could tell everything we nice and pointy.

Recording #2
For the second recording I adjusted the Moog Voyager output volume until it was just barely triggering the EXT SOURCE to light red and stay red. This took a little bit of slight adjustments because it takes a few seconds of not playing a note for the EXT SOURCE pad to change back to amber, but I was able to dial in the sound where it was staying right on the edge. When I looked at the second sample, I could see that the tips of the triangle waveform were starting to flatten.

As a note, the audible difference is certainly there, but it is barely imperceptible. Also, there are reasons you may want to intentionally do this to a clip as it can add some additional intentional artifacts. I wouldn’t do this for a final mix myself, but as part of sound design and creative process why not have some fun?

Recording #3
The third recording I just pegged the Moog Voyager to maximum audio output. This caused some very heavy digital distortion of the audio that was instantly recognizable, and to my ears not terribly appealing unless you want everything to sound like a square waveform. Although I did not have to zoom in to hear this, once I did the damage was incredibly perceptible and it was easy to see just how much of the sound was actually left.

Further Testing
Some more work could be done to export and load the samples into a DAW or audio editor, this would just further confirm if the signals are clipping, or if there are other headroom considerations with the mk2 that are not well documented.

Conclusion
It wasn’t too much of a shock to me to see digital clipping on the 1/4” inputs of the mk2, most AD convertors take the data straight in and just clip away quite happily. It is confirmed that if you see red on your EXT SOURCE pad, then you are clipping your incoming signal. Whether this is desirable or not is up to you to decide.

Does the mic/guitar input produce analog or digital distortions?

On some old school gear such as the Akai s900/s950, a common way to distort a signal and add some character was to hook up whatever you wanted to sample into the microphone jack instead of the line input. This would create an analog clipping effect that some people found musically interesting.

Theory
I believe that the mk2 is going to digitally clip on the microphone/guitar input just as it did on the line level input. Still, I think it is worth the experiment just in case this yields something interesting.

Method
I hooked up the Line Level output of my Moog Voyager to the mic/guitar input of the mk2, and set the output level of the Voyager all the way down, and the gain on the mic/guitar was set to 0, since the output from the Voyager is definitely going to be hotter than a microphone or guitar. I then setup the patch on the Voyager to output a single triangle waveform so that it would be easy to determine if there was digital distortion.

Recording #1
My first recording was with the mk2 set to microphone input, and the gain was set to 0. I slowly increased the volume output of the Voyager until the **EXT SOURCE** pad just barely turned red, and then backed it down so that the pad would stay amber. Visual inspection of the waveform showed that it was being recorded as expected with no digital distortion.

Recording #2
The second recording had the same setup as Recording #1, but I decided to increase the volume of the Voyager so that the **EXT SOURCE** would remain lit while I played a note. With this change I could immediately hear the tone of the sound change, and when I inspected the waveform I could clearly see that the audio had been digitally clipped.

Recording #3
For the third recording I set the mic/guitar switch to guitar, and immediately noticed that I had to increase the output volume of the Voyager to get the **EXT SOURCE** pad to light up red, and then back it down again to amber so that I could get my recording. Loading up the waveform on the mk2 I didn't notice any distortion or digital clipping.

Recording #4
The final recording was also set to guitar, and I increased the output volume of the Voyager so that the **EXT SOURCE** pad would light up red. I could immediately hear the distortion as it started making the triangle audio sound more like a square wave. Visual inspection showed that the signal was digitally clipped just like Recording #2.

Further Testing
I suspect using an actual guitar or microphone would give the same results, but for completeness it would likely be good to verify the characteristics with other audio sources.

Conclusion
It appears that both the microphone and guitar inputs digitally clip just like the line level inputs. Other than as a specific type of distortion, I don't think that people will find it useful in general to drive their signal to the point that the **EXT SOURCE** pad lights up red, and causing digital distortion.

However, with all this said, this was just me listening by ear and visually inspecting the waveforms on the mk2. Somebody who wants to devote more time, and go use graphic equalizers or other techniques may find that there is some additional magic that I was unable to determine.

Do the headphone jack outputs share an amplifier?

The mk2 has two headphone jacks, which opens up the ability for two headphones to be connected at the same time. This may not be ideal though as both jacks are controlled by the same volume knob, and traditionally when using splitters for headphones, the overall volume tends to be lower when two headphones are attached due to the increased power draw.

Theory
My expectation is that Roland only put a single amplifier into the mk2, with the expectation that most people are only going to use a single pair of headphones.

Method
For this test, I connected multiple sources to the headphone jack outputs at the same time, and measured the volume output change on a Zoom H6 which was connected to the outputs.

Two Headphones
This was a pretty basic test, and just listening by ear. I connected a pair of DT 770 Pro 8ohm headphones to the 1/4” jack, and a pair of Apple headphones to the 1/8” jack. By ear, I could hear a small but noticeable increase in volume when I disconnected one of the pairs, and a small but noticeable decrease in volume when I connected a second pair of headphones.

DT 770 Pro 8ohm and Zoom H6
For this test, I plugged the Zoom H6 into the 1/4” headphone jack outputs using a TRS to left and right channel cable splitter. With the H6 and the DTs connected, I then adjusted the gain on the mk2 until I was reading a -6db output volume from a tone sound being played by the mk2. When I removed the headphones, the signal went to roughly -3db, meaning that the DT 770 Pro 8ohm was lowering the overall volume output by about 3db.

Further Testing
To be complete, it would probably be worth getting an assortment of headphones and testing for impedance differences and the impact to the overall output volume.

Conclusion
Personally I’m satisfied that there is likely only a single amplifier for the headphones on the mk2, and you’ll just want to be aware of volume changes based on how much of a load you are putting on the output.

Do mono samples import as mono or get converted to stereo?

Description

Theory

Method

Experiments/Results

Further testing

Conclusion

Is there compression on the MIC/GUITAR inputs?

Description

Theory

Method

Experiments/Results

Further testing

Conclusion

Experiment Template

Description

Theory

Method

Experiments/Results

Further testing

Conclusion

Appendix F - System Comparison

Not sure if this is interesting for folks or not… and this list could grow forever… if I get to it this section should probably move *way* down, maybe into an appendix?

Akai MPC Live/X/One
Boss SP-202
Boss SP-303
Boss SP-404
Boss SP-404(SX/A)
PO-33
Blackbox 1010

???
SP-404 (2005) - Sampling frequency 44.1 Signal processing 24 bits
SP-404A (2007) - Sampling frequency 44.1 Signal processing 16 bits
SP-404SX (2009) - Sampling frequency 44.1 Signal processing 16 bits
SP-404MKII (2021) - Sampling frequency 44.1 Signal processing 16 bits ??? 48khz?

Appendix G - Frequently Asked Questions

Hardware

Does the mk2 support velocity sensitivity on the pads?

Yes, the 16 sample pads, as well as the sub pad, are all velocity sensitive?

Does the mk2 support pressure sensitivity?

For velocity of the initial hit yes, but you cannot adjust the pressure of your hit to bend or effect the velocity once the note starts playing.

Similarly, when you are triggering a roll, adjusting the pressure of the pad will not adjust the velocity of future roll events.

Hopefully this will be addressed in a future firmware update.

Does the mk2 have a D-Beam?

No, the mk2 does not have a D-Beam.

Samples

Does the mk2 autodetect zero point crossings?

Possibly to some extent when zoomed out, although it is hard to tell from the display.

Not when zoomed in, the mk2 will happily pick points that will clip and click, but you can typically zoom in far enough to avoid this from being an issue.

If you need to remove a click you can set the attack on the envelope to 1.

Can the mk2 do non destructive slicing?

No, each pad on the mk2 is a unique sample. You could copy a sample multiple times to different pads and adjust start/end points to taste so that the sample stays in tact, but the pads will not all reference the same sample.

Can the mk2 sync sample playback to MIDI Clock?

Yes, if BPM SYNC is enabled for the sample, it will sync the samples playback to a MIDI Clock. If you need time stretching set VINYL MODE to No.

How does the mk2 handle Pitch changes?

Pitch shifting on the mk2 is more like the SP202, where you had a CTRL knob to adjust the pitch speed as a somewhat continuous variable, similar to how variable speed controls work on a cassette player or record player would work. The mk2 also is able to make pitch adjustments in semitone increments similar to how a modern sampler would adjust the pitch of a sample to follow a chromatic keyboard.

Can you change the pitch of an entire bank of samples simultaneously?

??? Research ???

Does the mk2 have a mute group function?

Yes, there are 10 mute groups. You can assign a pad to any of the 10 mute groups, and when a pad on a mute group is triggered, it will/mute choke other pads playing in that mute group.

Can you sample while other samples are playing?

Yes, this is called resampling. The mk2 can resample while you play samples, whether FX are on or off.

Further, the Skip Back function is always recording the last 25 seconds of audio on the mk2, and is another way to get samples from content that has already been played.

Is there a way to assign only some of the chops to pads?

Yes, once you have chopped your sample to taste push ENTER and choose assign to pad. Use CTRL2 to cycle through the chops, and when you are on a chop you want hit a green pad to assign the selected chop to it. When complete, press ENTER again and choose execute.

Are there apps for creating sample chains?

OP-1 Drum Utility, Octachainer, and Sample Crate on the app store are worth investigating.

Sample Create outputs samples at consistent intervals which might be extra handy on the mk2.

Can you resample audio output and line/mic/guitar/usb input at the same time?

Yes, set INPUT SETTING and set ROUTING to MIX to record everything.

Can you start sampling when meeting a threshold?

Yes, press SHIFT+Pad 10 multiple times until you see COUNT-IN WAIT.

Should you normalize your samples?

Maybe. If a sample is not loud enough often normalize will get it to be much louder, but then you may need to reduce the VOLUME under PITCH/SPEED to taste because the sample may now be too loud.

Can you loop pads independently of each other?

Yes. Pads set to loop will play at through their start, end, and loop points as set, regardless of BPM that is set.

Where does end snap set the marker?

End Snap sets the end point of a sample to the previous beat of when the START/END was pressed. If you have a one measure loop, then you will want to press START/END just after the beat of the 2.1.1 and before 2.2.1.

How does reverse work for already playing samples?

Pressing reverse on a sample that is already playing will switch the playback direction and start it again from either the start or end point of the sample depending on whichever direction it is going.

Pads

How many pads can be sent to cue at a time?

Only one pad can be sent to cue at a time ??? Need to confirm ???

Can the mk2 trigger two pads from a single event?

Yes, pad linking can be setup on the mk2 to trigger one or more pads at the same time.

Are there suggested pad settings?

A reddit user posted to try: Curve type log, Threshold 1, Gain 10, Trig Span 5, others seem to be happy with this.

Another poster suggests: Curve Type: Fix; Threshold: 40; Gain: 0; Trig Span: 1

Is there a way to select a pad without triggering it?

Press the ENTER on the encoder and then press the pad.

???This stops the pad from playing???How to select a pad that is already playing???

How can you stop a pad from playing?

Hold EXIT and press the pad that you want to stop playing.

Skip Back

Does Skip Back record external inputs too?

Yes, it records based on your INPUT SETTINGS, whether it is MIX (Outputs, USB-C, MIC/GUITAR, and Line) or ExtIn (USB-C, MIC/GUITAR, and Line).

Can Skip Back be setup to bypass Master FX 3/4?

No.

Effects

Is it possible to have 5 FX at once?

Yes, if you use the Input FX, 2 Bus FX, and 2 Master FX you can use 5 FX at the same time.

Can the FX menu to stay on screen instead of going back to BPM?

Yes, you can get the FX menu to stay on screen by holding REMAIN and then pressing the FX that you want to stay on screen.

What is the delay buffer trick?

Essentially it is an effort to keep a delay buffer at or near infinite sustain to bring new elements into a set, but use the feedback level to adjust how long to keep in elements as well as fade them out.

DAKIM Boiler Room
Ways I Use Cheat Codes

??? Can the mk2 do it ???

Can you select an effect by holding MFX and pressing Pad 1-16?

Yes.

Do effects like reverb tail out when a sample is muted?

???test, but should be yes???

Do effects like reverb tail out when the effect is muted?

???test, but probably no???

Do the BUS FX maintain state being power cycles?

No, the BUS FX will always revert to off with default settings between power cycles.

Sequencer

Can you play triplets in a roll?

No, the mk2 currently does not support triplets in a roll, and you create bars that are odd bar lengths. This makes it much harder to do alternate roll patterns other than what will fit with a four on the floor style pattern.

Hopefully a future firmware update will enable dotted, triplet, and other roll patterns.

Does the mk2 have an arpeggiator?

No, although the mk2 has a chromatic mode, this is not currently compatible with pattern mode, and means that there is no arpeggiator.

Hopefully a future firmware update will add an arpeggiator.

Is BPM a global setting, or is it tied to something else?

??? This seems to be global, and not tied to a pattern ???

??? Seems adjustable per bank ???

Does the mk2 have pad mutes?

Yes, as of firmware 1.14 the mk2 has pad mutes.

Does the mk2 pattern sequencer support parameter locks?

No, the only information that the pattern sequencer records is pad number, pad velocity, pad start, and pad end for each event.

Does the mk2 have a step sequencer?

No, there is no step sequencer on the mk2.

Can the mk2 export stems from patterns?

No, the mk2 can only export samples or projects, but does not have a way to export stems to work in another DAW.

??? Is pad mute a workflow workaround ???

??? Bank velocity to zero method ???

Can you resample a pattern?

Yes, the mk2 can resample from a pattern now. While older SPs were unable to this, it is something that has been added to the mk2.

Can the mk2 make events random or have random parameters?

No, the mk2 does not have the ability to make a specific step or event random.

Does the pattern sequencer send MIDI notes?

No. Although pressing a pad will send a MIDI note, the pattern sequencer does not send MIDI notes.

How do you resample a perfect loop from a pattern?

Set pattern chain repeat to off, then when you resample a pattern sampling will stop at the end of the chain.

Does instant pattern change play the next pattern from the play head?

No, when instantly changing to a new pattern it will start playing the next pattern from 1.1.1.

Can you play two patterns at the same time?

No, the mk2 is only able to play one pattern at a time.

Can the mk2 work in other timings than 4/4?

No, the mk2 pattern sequencer and metronome only works in 4/4 timing, but you can record audio/samples in any timing that you would like.

Polyphony

Is the mk2 32 note mono polyphonic or 32 note stereo polyphonic?

The mk2 is 32 note mono polyphonic. You can see the Experiment on polyphony above.

Can an external keyboard play the mk2 polyphonically?

Yes, an external keyboard can trigger more than one pad at a time. This will be effective for drum kits, but will likely need a higher level of sampling and pad setup to set an instrument to work chromatically.

DJ Mode

Do tracks need to be analyzed to play at the correct BPM?

If the BPM of a track drifts, can you adjust the track?

Can you layer additional samples on top of DJ tracks?

No. You cannot play anything other than the content loaded to the two tracks while in DJ mode.

Can you sync tracks in DJ mode to an external Midi track?

Do you have access to the FX while in DJ mode?

Are there track markers in DJ mode?

No, there are no track markers in DJ mode.

Storage

Does the mk2 come with an SD card?

No, the mk2 currently does not ship with an SD card in the box. Retailers may make special offers, but Roland is not including an SD card at this time.

What can the mk2 do with an SD card?

The mk2 can backup and restore, and it can import/export both sample and

What SD Cards are recommended?

SDHC cards up to 32gb will work properly with the mk2.

The mk2 does not work with SDXC cards, which are above 32gb.

What format(s) are compatible with the mk2 and SD card?

Fat32 works.

exFAT and NTFS do not work.

Can the mk2 stream audio from SD like previous SPs?

No, the mk2 cannot stream audio from SD. It can only import and export samples and projects, or backup/restore the entire system from SD.

Can you have a loop playing while you audition samples from SD?

Yes, as of firmware 1.14, while you have a pad playing or looping on the mk2, you can go into the IMPORT/EXPORT menu and use SUB PAD to audition samples while the loop continues to play.

Can I mount the internal storage onto my computer?

No. The only way to get access to the internal storage through your computer is with the dedicated SP-404 mk2 application.

Audio

Can the mk2 be used as an audio interface for iOS or a computer?

As a class compliant audio device, the mk2 is able to be seen as a 4 output and 2 input device by iOS and computers that support class compliant audio.

When the EXT SOURCE lights up red, is my input clipping?

Yes, if you see the EXT SOURCE pad light up red, then the mk2 is digitally clipping anything that goes over the 0db threshold.

Can the mk2 be used as an amplifier for a microphone, bass, or guitar?

Yes, the mk2 can take the input signal from the MIC/GUITAR jack, amplify the signal, be routed through FX, and will then output the signal to the 1/4” outputs, headphone, and USB-C port based on your monitor and cue settings.

Is Thunderbolt compatible with the USB-C connection on the mk2?

Thunderbolt 3 adopted the USB-C standard, and is directly compatible with the USB-C connection on the mk2.

Older Thunderbolt connections require a special adaptor, but I don’t know which part of this will work.

Does the mk2 support bluetooth midi or audio?

No, the mk2 itself does not support midi or audio over bluetooth. You could invest in wireless technology that you could plug into the mk2 that would add this functionality though.

Can the mk2 be used as class compliant audio for the MPC Live/X/One?

??? Need to experiment with this ???

Is there a limiter on the mk2?

The manual details that from the Master Volume output, for Line/Phones output there is a limiter that then goes into a clipper. For USB-C audio output the signal goes straight out bypassing the limiter and clipper.

Can you tell if the mix of the mk2 audio output is peaking?

No, besides the EXT SOURCE turning red for external input, there does not appear to be any led or other indicator that the mk2 is peaking overall audio output.

How can you set the input gain for the line level inputs?

You either need to adjust the volume output of your external device, or go into the INPUT SETTING menu and adjust LEVEL.

Is it possible to trim a sample in bars/beats to make seamless loops?

Yes, with the START/END trigger on 1.14.

No on 1.13, but it's possible to trim to the exact length in samples. One quarter note = (60/BPM) * 48000 samples. In START/END mode, hold SHIFT and press ENTER to enter numerical values.

Alternative formula ((240 * bars) / bpm) * 48000 = length

Does the MIC/GUITAR input compress the incoming audio?

??? This came up over on the gear space forums… need to dig into this ???

MIDI

Does the mk2 transmit note and velocity data to the midi out?

Yes, it sends midi note and velocity data. Each bank sends a note on it's associated MIDI channel, meaning Bank A outputs to MIDI channel 1, Bank B outputs to MIDI channel 2, and so on.

Each bank outputs C2-Eb3.

Can the mk2 pass MIDI input/output between MIDI jacks and USB-C?

??? Need to determine this ???

Can an external sequencer play sounds on the mk2?

With proper midi connections an external device can sequence and play samples assigned to pads on the mk2.

Each bank on the mk2 is assigned a midi channel, so Bank A is MIDI channel 1, Bank B is MIDI channel 2, and so on.

??? Pad to MIDI Note mapping ???

Can the mk2 resample while receiving MIDI in?

Yes.

??? Need to test further???

Does the mk2 respond to MIDI CC data?

No.

Can you quantize MIDI after it has been recorded to a pattern?

No, the mk2 does not have a way to quantize notes in a pattern once it has been recorded. You will need to erase the notes you want to quantize and then re-record the parts of your pattern with quantize turned on.

External Sync

Can the mk2 send sync data to an external looper to keep it in sync?

??? This requires that the mk2 probably sends start, sync, and stop data over midi… ???

Does the SP-404 mk2 support AIRA link?

No, the mk2 does not support AIRA link. See the video Introducing the Roland SP-404 mk2 Creative Sampler and Effector.

Does the mk2 receive start, stop, and sync?

??? Need to research this ???

Can the mk2 sync loops to an external clock similar to Ableton Live?

Cables

Will a USB-C to Lightning cable work?

No, the USB-C to Lighting does not provide power, and it will not allow the mk2 to connect audio or midi to an iPhone or iPad.

To use a lightning port, you will need a USB-C to USB-A cable, and then the Apple Lightning Camera Kit to connect USB-C to Lightning. The Camera Kit will allow the mk2 to function as a class compliant audio and MIDI device, but it will not provide power to the mk2.

Why doesn't my USB-C to USB-A cable work?

Some USB-C to USB-A cables only provide power, and are not intended to transfer data. Confirm that your cable is able to provide both power and data.

Can you hook up two mk2s together over a USB-C cable?

No, the mk2 cannot function as a host, so they will not be able to communicate.

Power

What rechargeable alternatives are there to batteries?

MyVolts has created a solution called ReVolt, that powers a device using standard battery sizes through a USB connection. While similar to USB-C, this may be a good alternative for those who cannot power the mk2 over USB-C with a power bank because it does not have a USB-C output.

USB Rechargeable batteries

USB Powerbanks.

What USB Powerbanks work?

Anything that outputs over USB-C and can provide 1,500ma of power or more.

???Look into Birdcord for USB to DC power adaptor(s)???

How long do Eneloop Pro rechargeable batteries last?

Roughly four hours.

The standard Eneloop rechargeable battery provides 2000ma of power, versus the Eneloop Pro which provides 2550ma of power.

Can the mk2 be powered over USB-C from an Apple charger?

Yes, as long as the charger provides 1500ma of power over the USB-C cable.

Firmware Updates

Where do you copy the update files onto the SD card?

You copy the SP404MKII_APP0.bin and SP404MKII_APP1.bin files into the root folder of the SD card and NOT the unzipped folder containing the two BIN files.

Will the mk2 have paid firmware updates?

No, see BEATPPL Podcast 81

Companion App

Is the companion app available on iOS?

No, it is only available on Windows and OS X a this time.

Can the companion app run on an M1 Mac?

Yes, this is confirmed to work.

Do you need your SP connected for the Companion App to work?

No, the SP is not required to install or launch the Companion App.

Factory Content

Will restoring to factory content change my firmware version?

No, your firmware version will stay at whatever is installed.

How to restore factory content?

The factory content is hidden on the internal 16gb of storage.

To restore to factory defaults with project 1 populated with samples and patterns:

SHIFT+PAD13 (UTILITY) -> FACTORY -> ALL DATA

Please note, this will wipe out any other samples and projects you have on your mk2. So you may need to back that up first if you want to save anything.

Appendix H - Accessories
Batteries

Headphones

Device

Panel
Edit Section
- Volume -
- CTRL 1/Cutoff
- CTRL 2/Resonance
- CTRL 3/Drive

Control Section 1
- Filter+Drive FX
- Isolator FX
- Resonator FX
- DJFX Looper FX
- Delay FX
- MFX
- Display

Control Section 2
- (Pattern Sequencer)
- Pattern Select
- Pattern Edit
- (Sampling)
- Del
- Rec
- Resample
- Exit (Pattern Stop)
- Copy
- Remain (Current Pad)

Control Section 3
- (Sample Edit)
- Start/End (Chop)
- Pitch/Speed (Envelope)
- Mark
- (Push Enter)
- Value Knob
- (Sample Mode)
- BPM Sync
- Gate
- Loop
- Reverse
- Roll (Roll Set)

• (Bank)
• A/F
• B/G
• C/H
• D/I
• E/J
• Shift

Pad Section
• Pad 1 (Fixed Velocity)
• Pad 2 (16 Velocity)
• Pad 3 (Cue)
• Pad 4 (Chromatic)
• Bus FX (Mute Bus)
• Pad 5 (Exchange)
• Pad 6 (Init Param)
• Pad 7 (Pad Link)
• Pad 8 (Mute Groups)
• Hold (Pause)
• Pad 9 (Metronome)
• Pad 10 (Count-In)
• Pad 11 (Tap Tempo)
• Pad 12 (Gain)
• Ext Source (Input Settings)
• Pad 13 (Utility)
• Pad 14 (Import/Export)
• Pad 15 (Pad Settings)
• Pad 16 (EFX Settings)
• Sub Pad (Project)

|DJ Mode|
• |Ch1|
• Pad 1 (Bend+)
• Pad 2 (BPM +)
• Pad 5 (Bend-)
• Pad 6 (BPM-)
• Pad 9 (|<<)
• Pad 10 (Sync)
• Pad 13 (>/||)
• Pad 14 (Cue)
• |Ch2|
• Pad 3 (Bend+)
• Pad 4 (BPM +)
• Pad 7 (Bend-)
• Pad 8 (BPM-)
• Pad 11 (|<<)
• Pad 12 (Sync)
• Pad 15 (>/||)
• Pad 16 (Cue)

Rear
DC In
Power Switch

USB-C
1/8” TRS Midi In/Out
1/4” Line In R & L/Mono
1/4” Line Out R & L/Mono

Front
1/4” Headphone Jack
1/8” Headphone Jack
Mic/Guitar Gain Knob
Mic/Guitar Switch
1/4” Mic/Guitar Jack

Side
SD Card Slot

Bottom
Serial Number
Battery Compartment

Glossary

Polyphony - Specifies the total number of unique sounds that be played at the same time. Traditionally each note of polyphony is a monophonic (not stereo) sound, thus meaning that a stereo sound is normally considered to be two notes of polyphony. The Sp-404 mk2 is specified as having 32 notes of polyphony, which are mono, which means that the mk2 has 16 notes of stereo polyphony.

Volume

CTRL (1/2/3)

MIN/MAX

Pattern

Pattern Sequencer

Sequencer

MFX

Filter

Drive

Cutoff

Resonance

Drive

Polyphony

Index

References

Roland Links

Official SP-404 mk2 Manuals

Official SP-404 mk2 Downloads (firmware and skin template)

Ultimate Guide to Customizing the SP-404 mk2

Unofficial Links

Downloadable SP-404 mk2 Manual in PDF (firmware 1.10)

Technical

https://minimidi.world - Type A and Type B TRS to Midi resource

Specifications for TRS Adaptors Adopted and Released - Link to TRS Midi Specification

Articles

Roland Engineering: Designing the SP-404 mk2

Forums

Elektronauts Roland SP-404 Mk2 Thread

Elektronauts SP-404 Mk2 Production Tips/Tricks/Problem Solving/Bugs

MPC-Forums SP-404 mk2 Thread

SP-Forums SP-404 mk2 Sub Forum

Modwiggler Roland SP 404 mk2 Thread

Gearspace Roland SP-404 mk2 Thread

Reddit SP-404 Sub

Reddit SP-404 mk2 Sub - Seems dead

Videos

Official Roland SP-404 mk2 Launch Video

Introducing the Roland SP-404 mk2 Creative Sampler and Effector

Loopop SP-404 mk2 Review / 9 Tips & Ideas / Tutorial

Mike Acosta Roland SP-404 mk2 First Look

Bo Beats Roland SP-404 mk2 - the new king of creative samplers?!

Ave Mcree Roland SP-404 mk2 A Hip Hop Producer Review

Accurate Beats Roland SP-404 mk2 - Beatmaking and First Impressions

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LOFI Hip Hop jams on the SP 404 mk2 - spvidz

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Ave Mcree!! How to make your first beat

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