

Moebius Lab by Amazing Noises



Moebius Lab is a multimorphic sound processor; it contains a synth, a varispeed stereo sample player, 9 time domain effects, 6 spectral effects, 7 slots for external audio application, and nearly 200 LFOs that can modulate each parameter.

The screen is divided in 3 horizontal areas: the top one contains the sound source (Sample player/External Input/Internal Synth), the middle one contains the Time Domain Effects Chain with four slots for external apps, the bottom one contains the Spectral Effects Chain.

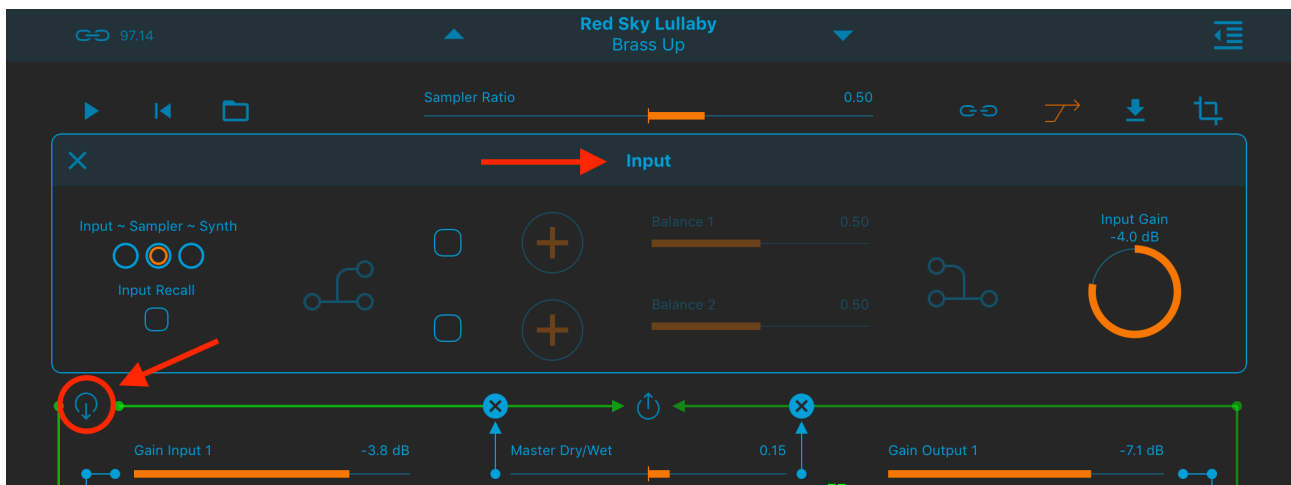
The two effect chains are connected in a sort of Moebius Ring: you can route the signal from the Time Domain Chain to the Spectral Chain and vice versa.

To select an input source, tap the circular icon on the upper left of the time domain chain: the Input Selection Window will appear at the top of the screen.

Important Changes Since Ver. 2

Moebius 2 is a major release which enables Plugins support: you can connect up to eleven RAU (Remote Audio Units) for both IAA (i.e. Inter-App Audio) and AUv3 (i.e. Audio Unit Extension). With this new feature Moebius works both as a Client and as a Host. For example, you can route the Audio Signals to external Effects and/or Instruments that you can play through the embedded virtual keyboard or via MIDI. Now there are no more limits to creativity!

The Input Selection Window



In this window you can select the Input source; you can also add two external audio apps by tapping on the “plus” icons in the middle. If you add an effect it will process the input before routing it to the Moebius Ring. If you add an instrument you can mix it with the internal input.

Input - Sampler - Synth Switch

You can choose between the internal sample player and a live input via Microphone, Audiobus or Inter-App-Audio or a Built-In Synth.

Input Recall

When this toggle is on, the source Input, the Sampler and the Tempo value are recalled when a new preset is loaded.

Please note that even if this switch is disabled, all data (Source Input, Sampler and Tempo) are still saved in the presets.

If the Recall Input toggle is switched on and the Sampler is subscribed for SYNC (i.e. the sync icon is switched on), the MASTER tempo will not be affected.

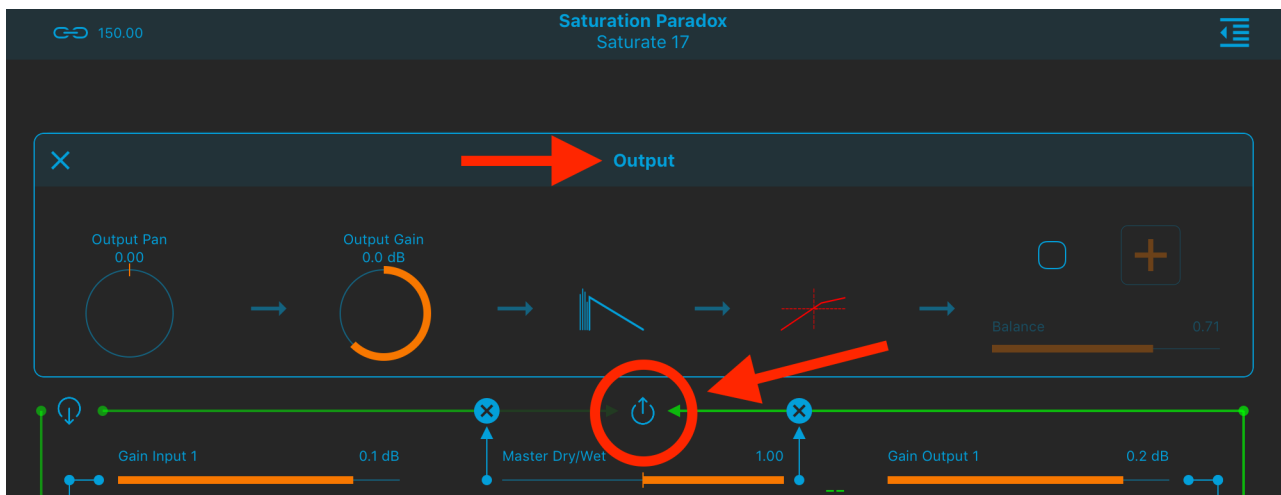
IMPORTANT NOTE:

When you are connected in Audiobus or Inter-App Audio as FX, Mic-AB-IAA will be selected automatically. Thus, every time you will disconnect from AB or IAA, you will turn back in Sampler. A warning message will be displayed in order to prevent dangerous Speaker audio feedback. The same behaviour will occur if you have not attached, in your iPad/iPhone/iPod Touch, an external device as headset or Mic. etc.

You can disable this message through the app Settings, see later for more explanations.

To open the Output Window tap on the circular icon above the time domain chain.

The Output Window



From left to right you can set the signal Pan, the Output Gain, add a Reverb, a Dynamic Processor, and an external audio app before sending the signal out.

Global Routing



You have 7 faders to route the signal globally.

1 – Master Dry/Wet: with this fader you mix the input and the output signal. When it is at the center the two signals are both at maximum level. Moving the fader to the right lowers the dry signal, moving it to the left lowers the wet signal.

2 – Gain Input 1: this is the gain of the signal sent to the Time Domain Chain.

3 – Gain Output 1: this is the gain of the signal coming from the Time Domain Chain.

4 – Gain Input 2: this is the gain of the signal sent to the Spectral Chain.

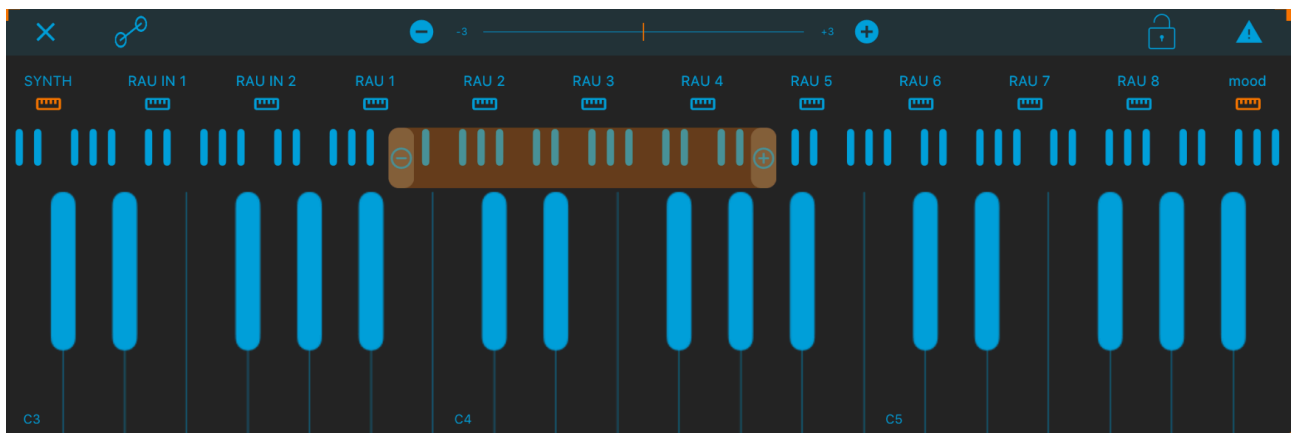
5 – Gain Output 2: this is the gain of the signal coming from the Spectral Chain.

6 – Feedback Chain 1: with this fader you can send the signal from the Time Domain Chain to the Spectral Chain.

7 – Feedback Chain 2: with this fader you can send the signal from the Spectral Chain to the Time Domain Chain.

Each node presents a vertical dry/wet fader.

MIDI Keyboard



Tap on the little keyboard icon on the bottom to show the keyboard: you can use it to play the internal Synth or the RAU (i.e. Remote Audio Units) instrument you load in the nodes.

Use the handle area to the top of the keyboard window to drag it up or down.

Tap the keyboard icon checkbox on the top to enable/disable MIDI communication with the RAU (i.e. Remote Audio Units) instruments.

Tap on the padlock to activate the “sustain mode”. In “sustain mode” when you tap a key it will remain on until you tap it again.

The “+” and “-” circular icons in the tiny keyboard are used to increase/decrease the number of keys visible in the keyboard.

The “panic” triangle generates all notes off.

The amplitude of each note i.e. its key velocity ($0 \div 127$), changes in relation to the vertical position of the tapping.

The **Aftertouch Y Scrub**, can be assigned to one or more parameters, see the Control Manager. Tap and hold a note, then scrub along the vertical axis (Y) to control the assigned parameters.

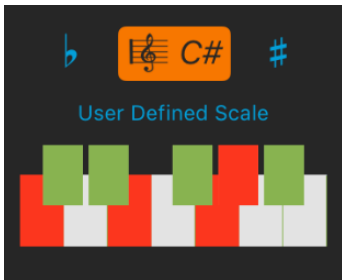


The Ribbon performs a continuous theremin-like pitch/amp glissando. The Ribbon will not affect the RAU instruments which remain quantized over the semitone grid.

You can play through a remote MIDI keyboard and use the Virtual Keyboard at the same time, for instance, to play the Ribbon.



Open the lateral slide view to configure a custom Musical Scale used by the Keyboard.



The \flat and \sharp transposition keys allow to change the base note of the keyboard (C) and the Scale button opens a menu where you can choose among several scales. The first item on the list (User Defined Scale) is a special feature which allows you to setup your own scale through the Musical Keyboard.

N.B. The transposition keys don't affect incoming MIDI notes.

Nodes

Internal Nodes Gestures:

Double tap a node to switch it on and off.

Single tap a node to open its edit window (more later).

Tap and hold a node to drag it in another position.

Inter App Audio Nodes Gestures:

Tap on the “+” sign to open an external app (IAA) or AUv3. The “+” sign is replaced by the app icon.

Double tap a node to mute/unmute the app.

Single tap a node to switch to the app window if the node is an IAA.

The user interface of an Audio Unit Extensions appears directly inside the Moebius app in a floating resizable window (tap the node to bring up the plugin user interface).

Tap and hold a node to drag it in another position.

Swipe up/down the app icon to remove the app.

All the nodes have a Dry/Wet slider on the upper left of the edit window. Here is an explanation of the processing of each node, with a list of parameters.

Time Domain Nodes

EXCITER

The Exciter Node distorts the signal below (Saturation) or above (Excitation) a given frequency.

Sat/Exc: When this parameter is 0, only the signal below the split frequency is distorted (see below), when it is 1, only the signal above the split frequency is distorted. Other values cause different mixes of distortion above and below the split frequency.

Split Freq: this is the parameter used to separate the two distortion bands (Saturation and Excitation).

Amount: the amount of saturation/excitation.

Feedback: the feedback reintroduces the distorted signal back into the circuit, for even more distortion.

RINGER

A ring modulator with distortion and feedback.

Frequency: Frequency of the modulator.

Freq/Pitch: you can set the modulator frequency in hertz or in MIDI note values.

Shape: the shape of the modulator; 0 = sine, 1 = saturated sine.

Lowpass: a low-pass filter for the ring modulation.

Feedback: the amount of modulated signal sent back into the modulator.

KLAMPER

Clamps the signal with a few special features.

Klamp Level: when the signal amplitude is above this level, it is “klamped”; i.e. if the Klamp Level is 0.5, all signal values above 0.5, or below -0.5 are transformed according to the Mode parameter (see below).

Mode: you can choose among 6 modes of “klamping” for the signal exceeding the Klamp Level.

- Klamp: the signal is clipped.

- Fold: the signal is folded back into the limits.

- Wrap: the signal is wrapped in the opposite direction.

- Annihil: the signal is annihilated (reduced to 0).

- Sine: the signal is replaced with a sinusoidal oscillation.

- Square: the signal is replaced with a square wave oscillation.

Oscil Freq: frequency parameter for the “Sine” and “Square” modes above.

Split Freq: you can klamp the signal above or below the split frequency (see the Spec Balance below).

Spec Balance: when this parameter is 0, only the signal below the split frequency is klamped, when it is 1, only the signal above the split frequency is klamped. Other values cause different mixes of klamping above and below the split frequency.

Normalize: the klamped signal can be normalized, i.e. raised to the level specified by the Norm Level parameter below.

Norm Level: when the Normalize switch is on, this is the level for the klamped signal.

SWARMER

A “superchorus”, excellent to create super fat synth sounds (and more). It is composed by a chain of modulated delay lines.

Activity: how fast the delay lines are modulated.

Depth: sets the amount of delay in the delay lines.

Tank: this is the feedback parameter, sets the amplitude of the signal sent back to the circuit.

Lowpass: **a low-pass filter for the feedback signal.**

FILTER

The Filter node has two configurations:

Dub: dual low-pass high-pass state variable filter with resonance and overlap factor.

Morph: a triple low-pass/band-pass/high-pass state variable filter with resonance and continuous morphing between the filter types.

You can switch between the two configurations using the radio buttons on the right of the edit window.

Dub Parameters:

Cutoff: this parameter sets the cut off frequency and the filter type. From -100 to 0 it is a low pass filter with cutoff frequency gradually increasing from

0 Hz to 15000 Hz, from 0 to 100 it becomes an high pass filter with cutoff frequency gradually increasing from 0 Hz to 15000 Hz.

Overlap: the overlap factor influences the Filter behaviour. When it is 0 the Filter parameter has the effect described above. Increasing the overlap factor makes the two filter types (low pass and high pass) to gradually overlap (i.e. the high pass effect begins before the low pass section has completed its path), and when the overlap is at its maximum, the filter behaves like a band pass (because the low-pass and the high-pass sections have nearly the same cut off frequency)

Morph Parameters:

Frequency: from 10 to 11.000 Hz.

Type: the filter type can be morphed gradually from low-pass (0) to band-pass (0.5) to high-pass (1).

Other Parameters:

Resonance: this is the amount of resonance the filter has at its cut off frequency.

Order: this parameter gradually changes the filter order from 12 dB to 24 dB.

Env Depth: this parameter sets the amount of variation the internal envelope follower has on the filter cutoff frequency.

Env Speed: this parameter sets the speed of the envelope follower, i.e. how fast it follows the input signal amplitude.

SPLITDEL

A dual band delay with feedback.

The delay time can be expressed in milliseconds or in note values (1/4, 1/8, etc.). To switch between the two, tap the chain on the upper right of the edit window.

Split Freq: signal components below the split frequency are sent to Delay 1, the components above are sent to Delay 2.

Delay 1: the delay time of the first delay line.

Feedback 1: the feedback factor of the first delay line.

Delay 2: the delay time of the second delay line.

Feedback 2: the feedback factor of the second delay line.

REDUKTOR

This node can be used to decimate and do other “bad things” to your signal.

Bit Depth: you can change the bit depth of the signal

Resample: this parameter is a factor of the actual sampling rate: i.e. if the SR is 44100 Hz, a Resample of 0.5 would give a 22050 Hz sampling rate, 0.1 would give 4410 Hz and so on.

Int Freq: you can randomly interrupt your signal. This parameter sets how many interruptions statistically you have per second.

Int Ratio: this is the ratio between the interruption and the sound. If set to 0.5 the sound has the same duration of the interruption, if set to 0.1 the sound has 1/10 of the duration and the interruption 9/10 and so on.

Int Amount: this is the depth of the interruption. If it is 1 the sound is completely interrupted (silenced). If it is 0.5 the sound amplitude is halved during the interruption and so on.

Single/Dual/Alternate Switch: Single = both channels are interrupted at the same time; Dual = the two channels are interrupted independently; Alternate = the channels are interrupted alternatively.

None/White/Pink Switch: None = during the interruption the sound is silenced; White = the sound is replaced with white noise; Pink = the sound is replaced with pink noise.

Split Freq: you can decimate the signal above or below the split frequency (see the Spec Balance below).

Spec Balance: when this parameter is 0, only the signal below the split frequency is decimated, when it is 1, only the signal above the split frequency is decimated. Other values cause different mixes of decimation above and below the split frequency.

SHIFTER

A dual band frequency shifter.

Split Freq: the signal components below the split frequency are shifted by Shift 1, the signal components above the split frequency are shifted by Shift 2.

Shift 1: frequency shift parameter for the signal components below the split frequency.

Shift 2: frequency shift parameter for the signal components above the split frequency.

Shifter Stereo Flip: when this switch is on, the right channel shift parameters are inverted (Shift 2 for the lower frequencies and Shift 1 for the higher).

REVERSO

Plays the signal in reverse at regular rate.

Switch Rate L/R: it is expressed in note values ($1/4$, $1/2$ and so on). For each given value (e.g. $1/4$), the first half ($1/8$) is played normally and the second half ($1/8$) is played in reverse.

Reverse Prob L/R: the probability that a segment is played in reverse.

Inertia: the time it takes for the internal delay line to change the length of the reverse signal when the Switch Rate parameter is changed. It affects the pitch of the sound.

Stereo Flip: when switched on, the reverse processing is applied independently (and with different parameters) to the two channels.

Spectrum Domain Nodes

BRICKWALL

A spectral brickwall filter.

Low Freq: the components below this parameter are filtered out.

High Freq: the components above this parameter are filtered out.

Bandpass/Reject Switch: when in reject mode the brickwall filter filters out the frequencies inside the two frequency parameters.

RUNNER

A spectral delay with feedback.

This node has three “spectral LFOs” to modulate the amplitude, the delay and the feedback of the spectral components.

The edit window shows three oscilloscopes, one for each spectral LFO. A little circular icon on the bottom right of the oscilloscopes open the parameter window.

Amplitude Parameters:

Min: sets the minimum amplitude (the minimum LFO value).

Max: sets the maximum amplitude (the maximum LFO value).

Cycles: how many cycles of the spectral LFO are generated in the spectrum space.

Shift: when this parameter is not equal to 0 the LFO scrolls in the spectrum space, generating dynamic variation in the components.

Shape: the waveform of the LFO: Sine, Square, Triangle, Ramp Up, Ramp Down, Pulse.

Delay Parameters:

Same as the Amplitude Parameters, except **Min** and **Max** which are expressed in seconds.

Feedback Parameters:

Same as the Amplitude Parameters.

GRIP

This node samples and holds spectral components of the incoming signal, based on their amplitude.

Amp Center: This is the center of the held spectral components amplitude range; all components whose amplitude falls in this range are held.

Amp Range: The amplitude range width.

You can edit graphically the two parameters above by tapping the small circular icon between the two knobs.

Amp Decay: Amplitude decay for the held components: it ranges from 0 (shortest) to 1 (longest).

Sample & Hold: the percentage of components held (when they are included in the Amp Range).

Phase Hold: this switch freezes the components phase: the result is a drone resonance, whose frequency depends on the sample rate.

Phase Mult: multiplies the freezed phases value.

BLUR

A (noisy) “spectral reverb”. The duration of the spectral components are prolonged (blurred).

Amount: length of the blurring. When this parameter is 1 the spectrum is frozen.

Blur Min Freq: the minimum frequency to be blurred.

Blur Max Freq: the maximum frequency to be blurred.

Unblurred: the amplitude of the non-blurred components (those outside the Min-Max range above).

In/Out: when toggled the blurred components are those outside the Min-Max range.

Stereo Flip: Blurs the entire spectrum; the components inside the Min-Max range are on the left channel, those outside are on the right channel. The behaviour is reversed if the In/Out switch is on.

DEGRADER

A spectral decimator.

Reduction: a division factor for the spectral components. If, for instance, the components are 512 and the Reduction parameter is 4, the components become $512/4 = 128$.

Rounding: generally the amplitude of a single component goes from 0 to number-of-components (i.e. if we have 512 components, the amplitude of a single component can range from 0 to 512). The rounding factor sets the value of a single step in amplitude. If it is 0 you have all possible values, if it is 1 you only have integer values, if it is 2 you only have even values, if it is 3 you only have multiple of 3 and so on.

Min Freq: minimum frequency affected.

Max Freq: maximum frequency affected.

Gain: sets the gain of the decimated signal.

FLUTTER

Flutter is a sort of animated brickwall. Two spectral sinusoidal LFOs define the frequency boundaries. The spectral components which are inside or outside (depending on the Spectral Balance) these boundaries are suppressed.

Upper Freq Limit and **Lower Freq Limit:** the effect takes place only inside these limits.

LFO 1 Bands: in how many cycles the first LFO covers the signal spectrum.

LFO 1 Freq: at what the rate the first LFO slides along the signal spectrum.

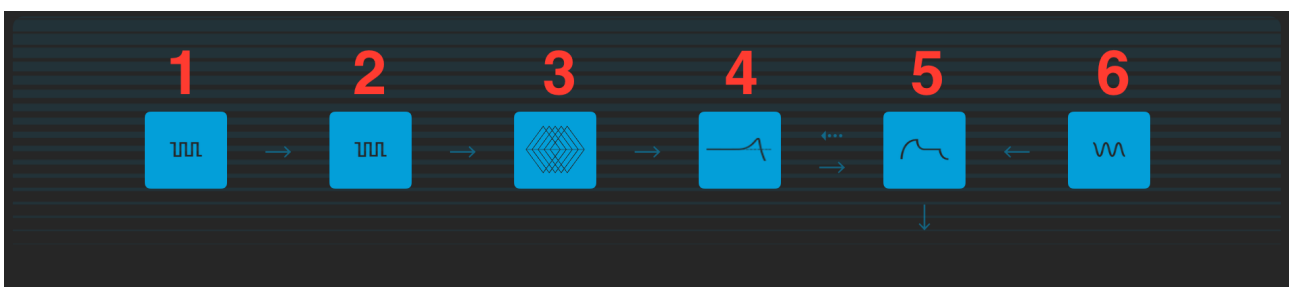
LFO 2 Bands: in how many cycles the second LFO covers the signal spectrum.

LFO 2 Freq: at what the rate the second LFO slides along the signal spectrum.

Spectral Balance: when this parameter is 0 the spectral components which are inside these boundaries are suppressed; when it is 1 the spectral components which are outside these boundaries are suppressed.

Residual Gain: this is the gain of the frequencies not processed, i.e. the frequencies not included in the band between the Upper Freq Limit and Lower Freq Limit.

The Synth



You can use the internal monophonic synth as a source by choosing it in the Input Selection Window.

The synth contains two **band limited oscillators** (1 and 2), a **complex oscillator** (3), a **ladder filter** (4), an **envelope generator** (5) and a **sub-bass oscillator**.

Tap on one of the six icons to open the relative edit window. Inside an edit window you can navigate to the other windows using 6 little icons on the top. Each component of the synth, apart from the envelope generator, can be switched on and off by tapping the “power” icon in the upper left of each edit window.

BAND LIMITED OSCILLATORS

The band limiter oscillators have a generator capable of morphing between the four classic waveforms: sawtooth, square, triangle and sine.

They also contain a noise modulator.

Waveform: use this parameter to morph between the different waveforms. Tap on the small icons above the fader to precisely select one of the four waveforms.

Harmonics: this parameter sets the quantity of harmonics of the waveform, i.e. its brilliance.

Amp: the amplitude of the oscillator.

Duty Cycle: ratio between the first and the second half cycle in the square and triangular waveforms.

Detune: pitch transposition in semitones.

Freq Shift: adds or subtracts a fixed amount of hertz to the pitch.

Ladder Send: the amount of signal sent to the Ladder Filter.

Glide: pitch portamento between notes in milliseconds.

Rand Freq Fact: multiplication factor for the frequency of the random modulator.

Rand Mod Depth: depth of the random modulation.

COMPLEX OSCILLATOR

The complex oscillator uses a new synthesis technique we called “Split Synthesis”: it is based on different processing of the positive and negative parts of the waveform.

Overtones: this parameter emphasizes the overtones of the oscillator using a frequency shifter. Negative numbers are not “undertones” but overtones obtained using negative frequency shifting (try it to hear the difference).

Overtones Divisor: the frequency shifting set with the first parameter is divided by this one. This creates undertones and/or inharmonic components.

Amp: the amplitude of the oscillator.

Simple Shifting: this is an additional frequency shifting which increases the overtones frequency shifting above, creating beatings and inharmonicities.

Recirculation: a delay line with feedback is included in this oscillator. The Recirculation parameter sets the number of delay repetitions per cycle.

Recirculation Feedback: this is the feedback parameter for the delay line. If it is set to 0, the recirculation has no effect. Values above 1 generates heavy distortion.

Ladder Send: the amount of signal sent to the Ladder Filter.

Simple/Complex Mix: there are two split synthesis modules in cascade: the output of the first is the “simple” one, the second is “complex”. With this parameter you can mix the two outputs.

Complex Overtones: with this parameter you can shift the frequency of the second module. If the Simple/Complex Mix parameter above is 0, this parameter has no effect.

Complex Shifting: this is an additional frequency shifting which increases the Complex Overtones frequency shifting above, creating beatings and inharmonicities.

LADDER FILTER

A high quality anti-aliased analog emulated Ladder filter.

Cutoff Ratio: the cut-off frequency as a ratio of the note frequency; f.i. if the note frequency is 220 Hz and the Cutoff Ratio is 3, the actual cut-off frequency will be 660 Hz (220×3).

Reson: resonance factor. When it is 1 or more the filter resonates by itself, even with no signal in input.

Env Sens: envelope sensitivity: the amount of cut-off frequency influenced by the envelope. For example, if the Cutoff Ratio is 3 and the Env Sens is 2, the actual cut-off ratio goes from 3 to 5 and back to 3 following the amplitude envelope shape.

ENVELOPE GENERATOR

The classic ADSR envelope.

Vel Sens: velocity sensitivity; the difference in dB between the softest (velocity 1) and the loudest (velocity 127) note.

Attack: first envelope segment, this is the time it takes the note amplitude to go from 0 to 1.

Decay: second envelope segment, this is the time it takes the note amplitude to go from 1 to the Sustain level (see below).

Sustain: amplitude of the note after the attack and release segments. This segment lasts as long as the note is held.

Release: last envelope segment, this is the time it takes the note amplitude to go from 1 to 0 when the note is released.

Curve: exponential factor for the ADSR segments. When the value is > 1 the ADSR segments are not straight lines but exponential curves.

SUB-BASS OSCILLATOR

A sinusoidal oscillator which can be used to reinforce the fundamental of the sound, or to add a sub-bass component one or two octaves below the fundamental frequency.

Amp: the amplitude of the oscillator.

0/-12/-24 Switch: the transposition of the oscillator frequency in semitones (no transposition, 1 octave below, 2 octaves below).

Glide: pitch portamento between notes in milliseconds.

The Output Window Post Processing Effects

In the Output Window you can activate two “post-Moebius” effects; a Reverb (“PlateVerb”) and a Dynamics Processor

PLATEVERB

As the name suggests this is a classic Plate Reverb. In the Output Window tap on the icon next to the “Output Gain” knob to show the PlateVerb Edit Window.

Balance: with this parameter you can mix the dry and wet (reverberated) signals.

Pre Delay: you can add a short delay to the reverb effect to separate it from the dry signal.

Low Cut and **High Cut:** with these parameters you can equalize the sound before sending it to the reverberation circuit. The equalizer doesn't affect the dry sound.

Damping: with this parameter you can attenuate the high frequency reflections faster than the low frequency ones. The higher the parameter is, the more high frequencies are attenuated.

Size: the size of the reverberation space. If you set this parameter to 1 and the Damping parameter to 0, you have an "almost infinite" reverb tail.

DYNAMICS PROCESSOR

After the PlateVerb, you can change the output sound using a compressor and expander effect.

This is used to either reduce or expand the dynamic range of an audio signal.

The Dynamics Processor controls set high (compression) and low (expansion) threshold volume levels. When the incoming audio is between these levels, it is left unchanged. Audio that rises above the upper threshold is cut in volume to bring it back down to that level (compression). Audio that falls below the lower threshold is cut in volume to make quiet sounds even quieter (downward expansion).

Additional controls adjust the attack and release time for the plugin's volume envelopes, the expansion ratio, the compression headroom, and a master gain to boost or cut volume on the Dynamics Processor's output.

Threshold: when the signal is above the threshold it is compressed.

Headroom: upper limit for the compressor, a limiter prevents the signal to go above this threshold.

Attack and **Release:** activation and de-activation times for the compression when it is above the threshold.

Exp Threshold: when the signal is below the Exp Threshold it is backward expanded.

Exp Ratio: expansion ratio.

Plugins Support

Moebius can host up to 11 Nodes (i.e. RAU, Remote Audio Units), which can contain Audio Unit Extensions (AUv3) and Inter-App Audio elements (instruments, generators, effects and music effect) on iOS 9 and later. The Audio Unit Extension is a new audio Apple standard (more on it below). Both types of plugins can receive the host transport state and a beat clock signal for sample accurate synchronization (provided the plugin implements it).

AUv3

Unlike the standard music apps, the AUv3 plugins can be launched multiple times. This way you are able to create multitimbral instruments with the same synth plugin or use multiple instances of an effect plugin (filter, delay, etc.) in multiple audio pipelines.

The user interface of an Audio Unit Extensions appears directly inside the Moebius app in a floating resizable window (tap the node to bring up the plugin user interface).

The state of current Audio Unit Extension is saved with the Moebius presets and, of course, you can create multiple instances of the same AUv3 plugin.

Inter-App Audio

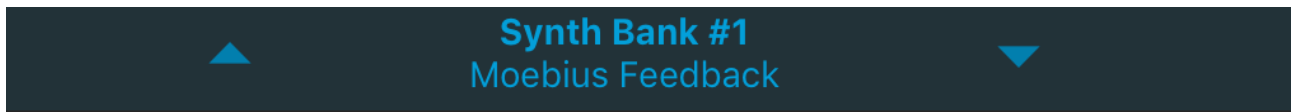
An IAA (Inter-App Audio) node is actually a separate app, so when you tap the node inside Moebius, the display switches to the IAA app.

The IAA Instruments and Music Effects can receive MIDI messages directly from Moebius: there's no need to run a separate Virtual MIDI port.

Note that the state of the IAA nodes is NOT saved with the Moebius presets. Please be aware that all RAU nodes (Instruments or Music Effects) receive MIDI from the same port and channels.

The Preset Menu

To show the Preset Menu tap on the preset name on the upper side of the main window.



Here are the menu items:

Open Presets: opens the Preset Manager (see below).

Open Snapshot Pad: opens the Snapshot Pad window (see below).

Save Preset: saves the current parameters configuration as <unnamed> in the current Bank.

Save Preset As...: prompts for a new name for the preset and saves it.

Overwrite Preset: overwrites the last selected preset with the current parameters configuration.

Select Random Preset: selects randomly one of the stored presets.

Init: recall the default value for all the parameters.

Create Random Sound: opens a window where you can randomly alter the parameter values. The central slider sets the percentage of each random variation, and you can decide whether to include the gain parameters or not with the switch labeled "Assignations". A tap on the crossed arrows icon generates a new random configuration. The values can be higher or lower than the preceding values. When the upper or lower limit of a parameter is reached the new values are folded in the opposite direction. There are two additional icons, two zigzag arrows pointing upward or downward. A tap on the downward arrow generates random values which are lower than the previous ones, a tap on the upward arrow generates random values which are higher than the previous ones. When the upper or lower limit of a parameter is reached the new values are wrapped in the opposite limit. The curved arrow on the upper left is the global UNDO and resets the parameters to the values they had before the random window was opened.

Cancel: closes the Preset Menu with no action.

Tips & Tricks

Double tap on the **Knobs** to show the options. In the '**Edit**' field you can enter the value from the keyboard. Tap '**Random**' to generate a random value, '**Default**' to reset to the factory settings and configure the knobs dragging behavior. Step fine resolution sets how many points of movement result in a one degree rotation in the knob's position. Only used in the horizontal/vertical slider modes. You can set Knobs behavior from Settings

Long Press Tap on the **Toggles** to open Control Manager Assignments

Moebius can play audio in the background and/or mix its own audio with other iOS apps. Moebius also supports Apple IAA (i.e. Inter-App Audio for iOS), AU (i.e. Audio Unit) and Audiobus, so you can send and receive audio through applications.

Swipe gesture (right to left) on a preset or bank to rename/duplicate/delete it. See Presets for more details.

You can load Presets (current selected bank) by remotely sending a MIDI Program Change message. See MIDI section, below for further details. You can select the MIDI channel where you can receive PC messages. A PC value of zero will select the first preset in the list, a value of 1 for the second etc... The range is from 0 to 127.

When you are connected in Audiobus or Inter-App Audio as FX, Input Mode will be selected automatically. Every time you will disconnect from AB or IAA, you will turn back in Sampler. A warning message will displayed be in order to prevent dangerous Speaker audio feedback. The same behaviour will occur if you have not attached, in your iPad/iPhone/iPod Touch, an external device as headset or Mic. etc... You can disable this message through the app Settings, see later for more explanations.

We suggest you to disable the screen rotation, otherwise the accelerometer doesn't work properly.

Swipe up/down the Inter Audio App or Audio Unit V3 (i.e. RAU) icon to remove the node.



The main tools-bar contains the Transport buttons to play/stop/rewind the built-in Sampler and for the RAU (Remote Audio Units) Clients, see 'Plugins

Support'. Since Moebius acts as an Audio unit HOST, these buttons are indispensable to trigger on/off the RAU clients (if they require these informations), as well the Moebius Virtual Keyboard to play RAU instruments.



Captures Output stereo signal (post Effects) and creates an audio file *wav* format 44.1 kHz 16 bit in the Documents Folder. Files created by *Moebius* are saved with an id-name followed by a number, for example: **Output Rec 1.wav, Output Rec 2.wav etc.**

Each new file will be numbered with the next consecutive number.



Enable/Disable Accelerometer X/Y and LFO, see the Control Manager for more details.

SYNC



The goal of SYNC is to setup a global Metronome (MASTER BPM). This will affect the LFO (Low Frequency Oscillator) and/or the Sampler (or Sequencer etc...) when enabled to receive SYNC.

You can sets the BPM (Beats Per Minute) by scrolling up/down (or left/right) the label value (♩ 120 ←○→) or tap out the tempo.

Single Tap or Long Tap the minus/plus buttons to increase or decrease by 0.25. By default 120 BPM corresponds to 1 Hz frequency for the LFOs.

Ableton Link description

Ableton Link is a new technology that synchronizes the beat, phase and tempo of Ableton Live and Link-enabled iOS apps over a wireless network. It lets you play devices together with the freedom of a live band. Anyone can start and stop their part while others keep playing, and anyone can adjust the tempo and the rest will follow. You can use Link to play with several copies of Ableton Live, with Live and iOS apps, or with iOS apps together – even without Live in your setup.

MIDI Clock description

The clock signal is broadcasted via MIDI to ensure that several MIDI-enabled devices such as a synthesizer or music sequencer stay in synchronization. The iOS implementation is designed to provides precise, jitter-free MIDI clock transmission and robust, error-tolerant MIDI clock reception.

Quantization

When you are connected to an Ableton Live Session, the Play/Stop/Rewind transport will wait until the next sync quantum boundary before starting.

The 'Sync' button waits for the next beat (Tempo Forte) from the MIDI Clock or Ableton Link and resets all phases for LFOs, Sequencer etc.

Send MIDI Clock

Will create a virtual MIDI port for the MIDI Clock Output.

In Play/Stop mode the MIDI Start/Stop Clock messages are sent according to Transport Play/Stop (Sampler, Sequencer etc...). In 'Always' they will not be sent every time, but only at start.

Please note when you are connected to Ableton Link, the incoming MIDI Clock is disabled while you can send MIDI Clock out, converting the Ableton Link to MIDI Clock.

Receive Sync

- 1) Local-Host
- 2) MIDI Clock signal
- 3) Ableton Link.

When the Ableton Link its enabled the Local-Host and MIDI Clock will be disabled and viceversa.

In Local-Host when the app is connected through Inter-App Audio, will receive the Sync from the HOST (if available, i.e. tempo/start/stop) and the current BPM is overridden.

When MIDI Clock is activated, you cannot change the BPM (Slave) since is controlled by the MIDI Clock (Master). Through the 'CLOCK' button you can select a MIDI Clock Source.

SAMPLER

You can feed *Moebius* with a stereo sampler or stereo signal from microphone/line or Audiobus/Inter-App Audio when connected.


Touch and drag the waveform to scrub the audio file. The speed of the scrubbing is relative to the position of the finger inside the waveform: higher positions produce faster scrubbing.

The Sampler state saved in the presets, is composed by File Path, Loop Points (Zoom), Play/Pause and selected Input Source.



Tap the symbol to set the loop points. When in "loop edit mode" you can:

- tap and hold inside the selection for the fine regulation of the loop
- double tap inside the selection to select again the whole file

When a loop is set, deselect the (tap again ) to zoom-in into the loop selection.



Reset scanning to the start position.



Waveform Scrubbing Mode, when selected the pointer jumps to the **Absolute** position of the finger that touches the waveform. If unselected (**Relative**) it does not jump but the scrubbing occurs from Sampler position toward finger touch.



Load an audio sample, see Samples Manager for mode details.



Switches the Sampler Play function between Ratio and Time Signature mode. When in Ratio mode (blue icon) you can set the play ratio: 1 means play the sound file at the original speed, 0.5 means play at half speed, -0.5 means play at half speed and in reverse and so on. When in Time Signature mode (orange icon) you specify the number of Measures and Quarter per Measure the sound file contains, and the tempo is linked to the general Metronome Tempo; for instance if we specify 2 measures of 4 quarters each, and the general Tempo is 120 BPM, the sound file will be executed in 4 seconds.

← **BPM** The button appears only when the Sampler is in Time Signature mode (see above). If you tap the <-BPM button you overwrite the general Metronome Tempo with a tempo calculated using the sound file length and the Measures/Quartes setting. For instance, if the sound file length is 2 seconds, and it has 2 measures of 4 quartes each, the general Tempo is set to 240 bpm. This function is not available when the General Metronome is in SYNC mode (MIDI Clock or Ableton Link).

N.B. You can disable the Crossfade and change Waveform style from the Settings. See the Settings for more details.

PRESETS

The Preset Manager will collect all the **Banks** (i.e. *.moebius suffix*) and **Snapshots**. A *Snapshot* is a simple “photo” of the parameters in the UI (i.e. User Interface). Both the Matrix and Keyboard/Sequencer are included but they can also be excluded, see the Settings for more details.

So, once you find an interesting sound you can take a snapshot (i.e. save this new preset) by touching the ‘plus’ icon and this new preset will be added to the selected Bank. The top four elements of the Snapshots list are assigned for ‘HV Snapshot Pad’, you can then sort or delete the Snapshots in the List by tapping the ‘List Icon’. You can even share the selected Bank and you can restore the factory Built-In Banks from the ‘Cloud’ icon.

Backup



Factory preset-banks (Excluded from iCloud back up)



Library preset-banks (if available as In-App Purchasing) (Excluded from iCloud back up)



User preset-banks (Included in iCloud back up)

If you delete, overwrite, rename or save new presets in factory/library banks, the icon will change as User preset-banks , being backed up.

Snapshot banks are located in Documents folder, you can access files via iTunes Sharing, see ‘Samples’.



You can add new files by copying in the app's documents (through iTunes or from another app) and refresh the list from there.



Load a preset, randomly from the banks list.



Delete all Presets for a selected bank. A warning message will be displayed.

The Snapshots bank file is a kind of Property List format (*.moebius*), you can edit as text and modify the values just rename in *.plist*

Swipe (right to left) on a preset or bank to rename/duplicate/delete it.



The purpose of the unique **Snapshot Pad** is to obtain intermediate values between four snapshots (corners). You can change snapshots assigned in the corner through the Snapshots list (see above).

The Snapshots Pad performs complex interpolations between four snapshots on a two-dimensional space. By scrubbing your finger on the Snapshots Pad a temporary new snapshot will be created by mixing the four snapshots depending on distance. Thus you can modify hundreds of parameters with a single touch, this is also called *Hyper Vectorial Synthesis*.

Snapshots Pad can be resized on fullscreen for detailed controls (left bottom icon).

N.B. Snapshot Pad have no effect on Switches and Sampler.

SAMPLES

The Samples Manager can read/write in the Documents folder, you can also share files through AudioCopy, Document Picker (iCloud, Dropbox etc...) and AudioShare. Samples manager can also read the Built-in files, but you cannot rename or delete them. Touch on a file in the list to show the actions popup. Here it is possible to **Load** a file in one of two Samplers (i.e. Sampler Ch.1 and Sampler Ch.2), Open In... contains all system actions over the file, according to file type. For example you can send files as an e-mail, open file with a listed app etc...



You can rename files in the Documents Folder, if you omit the file extension then Samples Manager will fill the original audio file extension for you. N.B. Samples manager shows only audio files supported, the other files in the Documents folder will not be seen.



You can pre-listen to the samples before you load them.



Starts audio capture from the microphone immediately and saves new stereo audio file in wav format 44.1 kHz, 16 bit in the Documents Folder. Files created are composed by id-name followed by a number, for example:

Input Rec 4.wav, Input Rec 5.wav etc...

Every new file will be numbered with the next consecutive number.

N.B. You can record a mono or stereo file in wav format 16 bit 44.1 kHz according to the iDevice channels. Even if the Moebius is connected to

Audiobus or IAA, the Grab Audio will record always from iPad microphone/line



You can import files from AudioCopy, Music Library, Document Picker (iCloud, Dropbox etc...) and AudioShare.

Access to the iPod library, this allows the application to read user's songs, audio books, and audio podcasts. The file can be converted in .wav (PCM 16 bit 44.1 kHz) or .caf formats and stored in the local app Documents Folder. If the file exist at path then progressive number will be attached to the file name.

The document picker feature lets users select documents from outside your app's sandbox. These include documents stored in iCloud Drive and documents provided by a third-party extension. Users can open these documents directly, editing them in place. This access simplifies sharing documents between apps and easily enables more complex workflows.

Use iTunes to share files between your computer and your device

- 1) Open iTunes on your Mac or PC.
- 2) Connect your iPhone, iPad, or iPod touch to your computer using the USB cable that came with your device.
- 3) Click your device in iTunes.
- 4) In the left sidebar, click File Sharing.
- 5) Select an app to see which files are available for sharing in that app on your device.

CONTROL MANAGER



MIDI



Accelerometer X



Accelerometer Y



LFO

After selecting **MIDI**, *Moebius* enters into "learn mode" i.e. it "hears" incoming MIDI Control Change messages. It's then possible to send messages with MIDI hardware, Virtual MIDI and Network MIDI. Flashing screen means the association between the parameter and the MIDI message has been successful and displays the corresponding control Change and Channel.

Alternatively you can enter direct CC and MIDI Channel number through number keyboard. You will also receive an additional setting: 'Widget Range' that allows you to resize the range of action on the parameter. This is because you can use all 128 steps MIDI to control precisely particular part of the parameters.

Accelerometer X and **Y** are used to control the parameters through the horizontal or vertical inclination of the device.

LFO, Low Frequency Modulation is a table look-up oscillator with seven built-in shapes. The frequency of LFO is expressed in Hz and BPM (Beats per Minute). You can get the period time by dividing $1/\text{freq}$. The LFO Frequency value is converted in BPM (Beats Per Minute), you can adjust this value through the minus/plus buttons, these will replace current value with the integer BPM (previous/next) or you can TAP to set your own Tempo.

- φ Will reset to zero the phase of current LFO.
- Φ Will reset to zero the phase of ALL LFO actives.



Enabling the Link symbol, you will disable the 'Low Frequency Oscillator in Hz' Slider, so the value of the current LFO is sampled from the global 'MASTER BPM'. You can adjust the BPM value from SYNC view. When the MIDI Clock or Ableton Link is enabled (see MIDI Manager and SYNC) the BPM value will affect all LFO enabled for SYNC mode. The goal of the 'SYNC Subdivision' Slider is to rescale the 'MASTER BPM' from SYNC according to the relative symbol (abbreviate musical note time values):

Normal : 1/1, 1/2, 1/4, 1/8, 1/16, 1/32, 1/64

Triplets : 1/1T, 1/2T, 1/4T, 1/8T, 1/16T, 1/32T, 1/64T

Dotted : 1/1D, 1/2D, 1/4D, 1/8D, 1/16D, 1/32D, 1/64D

Where 'D' is a **dotted** note and 'T' is **triplet**.

With 1/4 (Default), the BPM follows the 'SYNC BPM' or 'MIDI Clock' BPM when enabled, i.e. 120 BPM corresponds to 1 Hz.

- 1/4 - each step is a quarter note (or BPM), e.g. 60 BPM is 0.5 Hz
- 1/4D - each step is a dotted quarter note (or 1.5 x BPM),
- 1/4T - each step is a triplet quarter note (or $0.5 / 3$) etc.

♪ Reset default value ($1/4 = 120$ BPM)

MIDI

You can enable/disable the incoming MIDI signal from all the physical, virtual and network ports available. The MIDI Clock is managed by SYNC, see the SYNC section for more details.



If the Core MIDI Network is connected a wifi symbol will appear in the Network port cell.

Channel sets the input channel for the selected port, the *apeFilter* DK (Keyboard) will receive the Note On/Off messages in OMNI (from all 16 channels) or, one of the 16 channels.

Configure Bluetooth LE MIDI

With the release of iOS 8 and OS X Yosemite, sending and receiving MIDI data is supported using Bluetooth Low Energy connections on any iOS device or Mac that has native Bluetooth Low Energy support. All established connections are secure which means that pairing is enforced and connections cannot be made to your devices without your explicit consent. After a connection is established, it simply appears as an ordinary MIDI device that any MIDI application can communicate with.

Enable NRPN

NRPN stands for "Non-Registered Parameter Number" and is part of the MIDI specification for the control of electronic musical instruments. NRPNS allow for manufacturer-specific or instrument-specific MIDI controllers that are not part of the basic MIDI standard.

Unlike other MIDI controllers (such as velocity, modulation, volume, etc.), NRPNS require more than one piece of controller data to be sent. First, controller 99 - NRPN Most Significant Byte (MSB) - followed by 98 - NRPN Least Significant Byte (LSB) sent as a pair specify the parameter to be changed. Controller 6 then sets the value of the parameter in question. Controller 38 may optionally then be sent as a fine adjustment to the value set by controller 6.

This fine adjustment is part of the conventional MIDI controller specification, where any of the first 32 controls can be optionally paired with a control offset 32 higher. This is the rare 14-bit Continuous Controller feature of the MIDI specification, and NRPNS simply take advantage of that existing option in the same way offering 16,384 possible values instead of only 128.

(<http://en.wikipedia.org/wiki/NRPN>)

When you enable NRPN, you should not employ CC 99 and 98 as the normal CC in order to avoid an undesired overlap. An NRPN message must be sent with this order: CC 99 98 6 38 where the first couple is the CC number and the second couple is the data. You can MIDI learn as for the normal CC, and NRPN works for all knobs and parameters. Although an NRPN CC number can be greater than 127, you will not be able to learn CC beyond 127.

Program Change for Presets Selection

You can select Presets (current selected bank) by remotely sending a MIDI Program Change. You can select the MIDI channel where you can receive PC messages. A PC value of zero will select the first preset in the list, a value of 1 for the second etc... The range is from 0 to 127.

SETTINGS

Buffer Size set the Audio Device latency. When you run *Moebius* from Audiobus, or Inter-App this value will be taken from Audiobus or Inter-App Host. Warning! When an app runs first and keeps running in the background, the Sampling Rate and Buffer Size will impose for the whole iOS while running.

FFT Size

This is the Fast Fourier Transform Size expressed in samples.

It must be a power of 2 such as 512, 1024 or 2048 (to give a few commonly used values).

For example, if your FFT size is 2048 samples long, the FFT analysis gives you 1024 equally-spaced frequency bins from 0 Hz to Nyquist.

FFT Overlaps

The basic algorithm is known as a Short Term Fourier Transform (STFT) which is a set of overlapping "windows" of input and output data. This kind of setup is necessary to reduce unwanted modulation on input to the FFT and allows us to modify the FFT data without unwanted overlap-add artefacts on output. You can experiment with NO overlap in order to hear the windowing modulation between FFT frames. A good value for Moebius is 2 (default).

Snapshot Pad Resolution Update sets the time interval for the interpolations, from about 15 to 350 milliseconds. Short times for higher resolution but higher consumption of CPU.

Save/Restore the Last App Setup, when you exit from the application, save the status and reload upon next opening of the app.

Others Settings

The switch in this section are self-explanatory. The first hide all warning messages and User Interface flashing when you store or load a preset etc... The others are designed for include or exclude Control Manager Assignments and Knob Color when you store a new Snapshot.