



Integrate

old-school virtual analog synthesizer

User Manual

Introduction

As technology has improved, it has become possible for manufacturers to produce increasingly complex synthesizers. This has led to the creation of synths that trade off ease-of-use for maximum sound design potential, spreading hundreds upon hundreds of options and parameters throughout dizzying interfaces, full of tabs and menus. They provide oscillators and filters with such immense algorithmic precision that any soul or character is stripped away before it has a chance to flourish in the sound. **Integrate** is not like that.

Integrate has been designed with the goal of creating an old-school virtual analog that has just as much personality as any of the classics, paired with a fantastic user interface that makes exploring the sonic possibilities an effortless pleasure. It cannot do everything, but this is intentional; limitations breed creativity. No longer will you spend hours scrolling through lists of wavetables, paralyzed by the impossible task of deciding which is the best to use – a question that has no right answer. Instead, **Integrate** has a simple two-page layout where every parameter has been selected for its unique axis of inspiration. Every dial, slider and button is waiting to delight you as you delve into a world of aural possibilities.

Using dials and other controls

Dials and sliders

Most parameters in the user interface take the form of dials and sliders. To increase the value of the corresponding parameter, **left click and drag** your mouse upwards; to decrease the value, drag downwards.



While dragging the control, a tooltip will appear to display detailed information about the parameter's current value, and possibly a short description of what the parameter does.

To reset a value of a parameter to its initial position, **middle click** on the control, **double left click**, or **alt left click**.

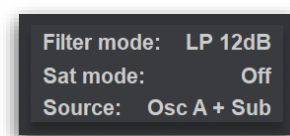
For some controls, holding **shift** while dragging will automatically snap to important values. For example, the Semitone parameter on the oscillators will snap to octave intervals.

The colored ring around dials shows the current modulated value of the parameter. This value is calculated by adding the main value (the one that you set) to any modulation sources (such as LFOs and envelopes) that are sent to the parameter.

You can also use your mouse's scroll wheel to change the value of sliders and dials. **Scroll up** to increase the value and **scroll down** to decrease it.

Menu dropdowns

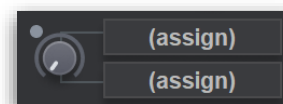
Left click a menu dropdown to open a menu showing all the possible choices for that parameter. **Middle click** or **alt left click** to switch back to the initial choice.



Modulation destination slots

Left click a modulation destination slot to select a target for the modulation source. All the parameters that can be modulated will be highlighted, and the parameters that cannot be modulated will be dimmed; some buttons will appear for additional modulation destinations that were not originally dials. **Left click** the parameter you want to modulate. **Left click** elsewhere to cancel picking a destination.

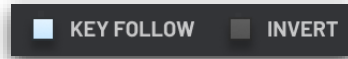
If a modulation destination has previously been selected, a button will appear on the destination selection screen, allowing you to remove the modulation assignment. You can also **middle click** or **alt left click** on the slot to remove the assignment.



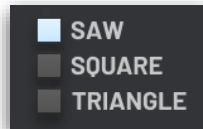
Important – To assign modulation to many of the parameters on the “Effects” page, the effect unit needs to be enabled first. Click the enable switch in the top-right corner of the unit.

Switches

Switches come in two main types. The first type is toggled on-off when you **left click** on it, and reset to its initial state when you **middle click** on it.



The second type is a group of switches where only one switch can be enabled at once. **Left click** a switch to select that specific options. **Middle click** any of the switches in the group to revert to the initially selected option.



Buttons

The remaining controls in the interface are buttons. **Left click** a button to perform its associated action. All buttons feature a label indicating their purpose.

Throughout the interface, there are various buttons marked with “Copy” or “C”. These buttons display a menu allowing you to choose to copy all the settings from another part of the patch. For example, if you select “LFO 2” from the menu on the “C” button in LFO 1, the settings of LFO 2 will be copied into LFO 1.

Interface layout

The interface consists of two pages, the “Main” pages and the “Effects” page. You can switch between these pages by clicking the corresponding buttons at the top-right of the interface. By clicking the preset display at the top-left of the interface, you will open the “Patch Browser” popup; click anywhere outside the popup to close it.



Main page



The main page contains the oscillators, filters, amplifiers and pitch controls. It also contains the modulation sources and matrix, but these are shown on the effects page as well.

On the left side are the two main oscillators, **Oscillator A** and **Oscillator B**. Each has options for FM/RM, Sync and a section for unison configuration. These oscillators are split into two parts, giving **A1**, **A2**, **B1** and **B2**.

In the right column are the two filters and amplifiers. The output of each filter is hard-wired to its adjacent amplifier. Each filter unit contains controls for the type of filter, the saturation mode, and a selection for its input source. Each amplifier unit contains an AHDSR envelope with slope tension parameters, a gain dial, and a waveform preview.

Below these, there is the sub oscillator and pitch controls. The sub oscillator only has a few options; it produces low-passed waveforms at a chosen octave below the main pitch. The pitch controls allow configuring portamento (pitch glide) and the master octave.

Effects page

The effects page is divided into different sections, one for each effect that can be applied to the patch. Effects can be enabled or disabled using the toggle switch in the top-right corner of that section.

The effects page also contains the **Oscillator Character** controls, which is not an effect per se, since it is always enabled and affects how the oscillators behave, rather than being an algorithm that is applied to the final sound.

All the effects can be bypassed using the “Bypass FX” button at the top the interface. When this button is blue (on), the effects will not be applied.

In this screenshot, you can see that only the **Delay** and **Reverb** effects have been enabled:



Patch management

Each instance of the synthesizer has a bank of 132 patches. You can save this bank to a file, or load patches in from an existing bank file. You can also save and load single presets to and from files.

When you modify a patch in the bank, those changes will be stored locally in that instance of the plugin. To use the modified patches in other instances, you will have to save the bank out and then load it in those instances.



Patch selection and management happens in the **Patch Browser**. Click the green box in the top-left corner of the interface that displays the current patch name to open the **Patch Browser**. You can click anywhere outside the popup to close it.

From here, you can click the name of any patch in the bank to switch to it. When you switch to a different patch in the bank, your changes to the patch you were previously working with will not be lost, and if you switch back to that patch, it will be as you left it.

If you hold **Shift** when selecting a new patch, the browser popup will close automatically after the new patch has been activated.

Double left click the name of a patch to rename it.

Along the bottom of the **Patch Browser** are buttons that let you manage the bank and individual patches.

Load bank: Load a new bank. This will overwrite the existing bank; make sure you have saved it first if you want to keep it. You can load an empty bank where every patch is initialized to a basic sawtooth, a factory ROM, or a bank from a file. By default when the synthesizer is created, Factory ROM A is the initially loaded bank.

Save bank: Save the current bank data to a file. This can be loaded with **Load bank**.

Initialize patch: Resets the currently selected patch in the bank to an initialized state, a basic sawtooth. The previous patch settings will be lost; make sure you have saved it first if you want to keep it. Hold shift while clicking this button to automatically close the browser afterwards.

Randomize: Randomize the currently selected patch in the bank using the smart-randomize algorithm. This is a useful feature for generating strange texture and effects, but also for exploring what the synthesizer is capable of. The previous patch settings will be lost; make sure you have saved it first if you want to keep it.

Rename: Rename the currently selected patch in the bank.

Save patch: Save the currently selected patch settings to a file. This can be loaded with **Load patch**.

Load patch: Load the patch settings from a file into the currently selected patch in the bank. The previous patch settings will be lost; make sure you have saved first if you want to keep it.

Settings menu

The settings menu is accessed by clicking on the button labelled “Settings” in the top-right corner of the interface. It contains settings global to all instances of the synthesizer, saved separately from your patches. None of the settings affect the DSP output. The menu also contains other entries, such as opening auxiliary popups.

100%-200% scale: Sets the size of the user interface.

Skins: Sets the appearance of the interface. This is purely an aesthetic choice; it does not have any effect on the output sound.

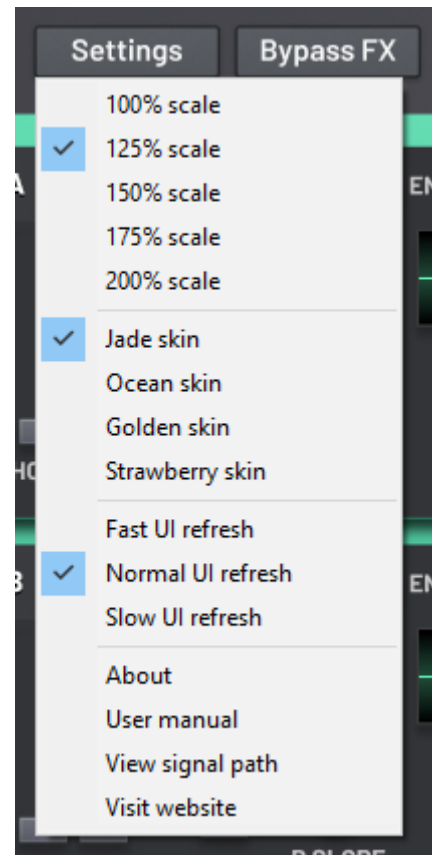
UI refresh speeds: Sets how fast the waveform previews, LEDs and dials animate. Note that when you are interacting with the interface (such as moving your mouse over its window), these items will refresh at a higher rate, regardless of this setting. Select “Slow UI refresh” to reduce CPU usage when the editor is open.

About: Shows the about popup. Click anywhere to close it.

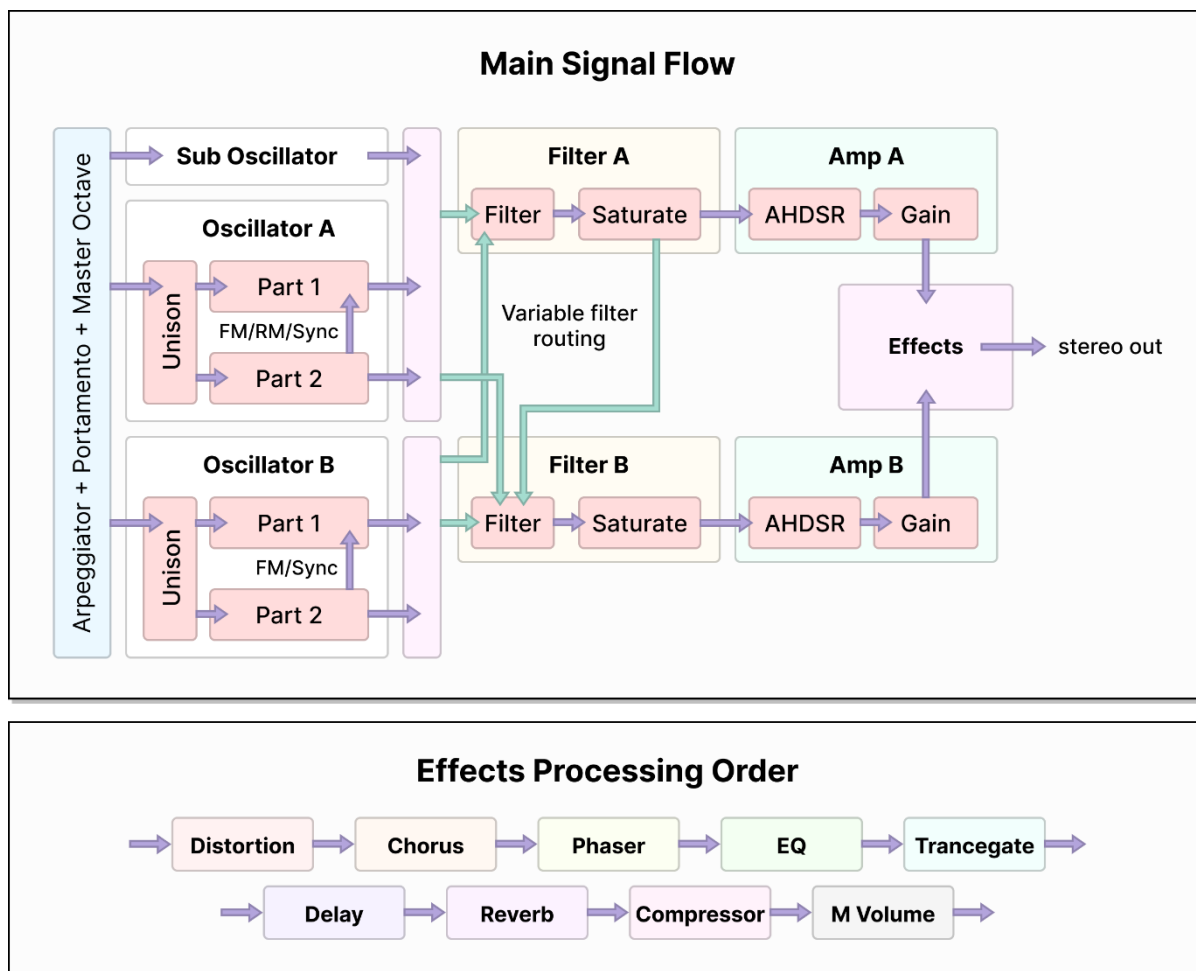
User manual: Opens this document.

View signal path: Shows the signal path popup. Click anywhere to close it.

Visit website: Opens the website for **Integrate** in your default web browser.



Signal flow



You can view the signal path diagram at any time from within the synthesizer by selecting **Menu > View signal path**.

First, if enabled the arpeggiator is applied to incoming MIDI notes. The master octave control is applied to offset the octave of the notes. The portamento glide processing also takes place. The computed pitch is sent to Oscillator A, Oscillator B and the sub oscillator.

Oscillators A and B duplicate the note into several voices as per their unison configuration. Part 2 is processed, and its output is possibly used in the processing of Part 1 if FM, RM or sync are enabled. The output of Oscillator A is combined with the sub oscillator. Note that the signal path is already stereo at this point.

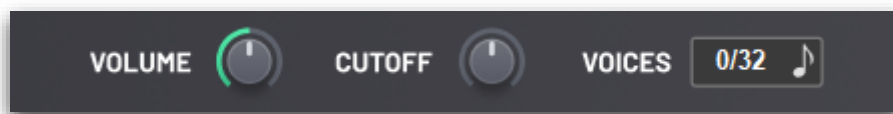
Filters A and B then select their input. Filter A can either have its source connected to the sum of Oscillator A and the sub, or it can be connected to the sum of all oscillators. Filter B can be connected to Oscillator B, all oscillators, or the output of Filter A's saturator. Each filter applies the filter itself followed by a saturator, and then the output is sent to the connected amplifier. Amplifier A receives input from Filter A, and Amplifier B from Filter B.

The amplifiers apply an AHDSR envelope to the volume of the voice, and then apply the final gain adjustment. The output of the amplifiers is combined and sent to the master effects. At this point, the signal path is no longer per-voice.

The master effects are processed in the following order: multiband distortion, chorus, phaser, EQ, trancegate, delay, reverb and compressor. Finally, the master volume is applied, and the signal is sent to the host.

Master controls and pitch

Master controls



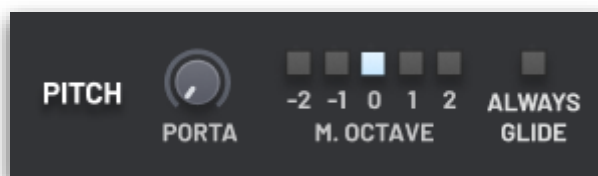
At the top of the synthesizer's interface are the master controls.

Volume: Controls the amplitude of the final signal that will be sent to the host. Notably, this is applied after any effects.

Cutoff: Controls the cutoff points of both **Filter A** and **Filter B** at the same time. This is useful for automation or performances, where the filters control different parts of the patch and you want to control them together.

Voices: This area functions both as a display of the number of currently playing notes (including those that have been released but are still tailing off), and a toggle for monophonic mode. **Left click** to switch between monophonic and polyphonic modes. In monophonic mode, only one note can be played at a time. When a new note is played, its pitch will be used instead of the old note. In polyphonic mode, up to 32 notes can be played simultaneously. This upper limit of 32 is enforced to prevent accidental over-consumption of CPU.

Pitch



Porta: Controls how long the synthesizer takes to glide from the pitch of the previous note to the new note. At zero seconds, this switch takes place immediately.

Always glide: When enabled, the portamento glide always occurs. When disabled, gliding only happens when the new note overlaps the previous note. That is, the second note is played before the first note is released.

M. octave: The master octave. This changes the effective octave of all notes received by the synthesizer. However, it does not affect key tracking, both directly in the filter sections, or via the modulation matrix.

Oscillators

Main parameters



Volume: Controls the amplitude of the waveform generated by the oscillator. At 0%, the oscillator will produce no sound.

Pan: Controls the stereo balance of the oscillator's output. At -100%, sound will only be produced on the left channel, and at 100%, sound will only be produced on the right channel.

Detune: Controls the oscillator's fine tuning. On oscillators A1 and B1, this ranges from -60 to 0 cents (1/100th of a semitone), and on oscillators A2 and B2, this ranges from 0 to 60 cents. Note that if this parameter is modulated then the modulation will apply a consistent offset to the fine tuning of the oscillators regardless of the dial's polarity.

Semitone: Controls the coarse tuning of the oscillator, aligned to semitones. Hold **shift** while dragging this dial to snap to octave intervals.

Wave: Select the waveform of the oscillator. At 7 o'clock is the triangle waveform; at 9 o'clock is the square waveform; at 11 o'clock is the sawtooth waveform; at 1 o'clock is the sine waveform; at 3 o'clock is the noise generator; and at 5 o'clock is the extended waveform selection. The extended waveform is selected using the **Param** control.

Important – In the "Digital" rendering mode, the color of the noise depends on the pitch of the note that it is played at. This is to emulate the effect of early sound chips from video game consoles. You can prevent this by disabling the "Key follow" option on the oscillator.

Param: The behavior of this control depends on the waveform selected into the oscillator. For the triangle waveform, this is a triangle-to-sawtooth morph. For the square waveform, this determines the duty cycle – try modulating this with an LFO! For the sawtooth waveform, this determines the tension (only available in the "Digital" rendering mode; see the **Oscillator Character** section). For the sine waveform, this determines the brightness by adding higher harmonics. For the noise generator, this softens the noise by removing the higher frequencies.

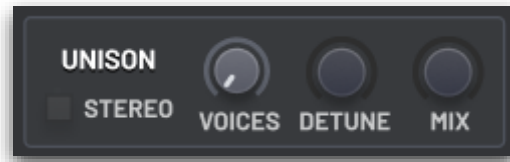
Key follow: Determines whether the oscillator follows the pitch of the played note. When disabled, you should use the **Detune** and **Semitone** controls to select the pitch you want.

Invert: When enabled, the polarity of the oscillator's output is inverted. This can be used to create phasing effects when used in conjunction with other oscillators.

Retrigger: When enabled, the oscillator starts at the same phase in its cycle when the note is pressed. You can use the **Initial Phase** control to determine this starting phase. When retrigger is disabled, the oscillator starts at a random phase in its cycle, giving the impression of a free-running analog oscillator. When the unison feature is enabled, you will likely want to

leave retrigger disabled, otherwise a “pluck” sound will occur when the note is first played, as the unison voices suddenly go out of phase.

Unison



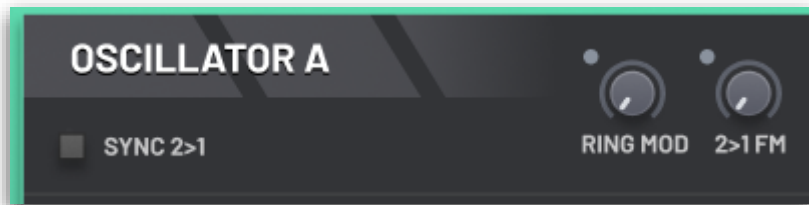
Unison simulates the effect of multiple notes being played at the same frequency. You can then subtly detune these notes to create a very rich and evolving sound.

Voices: The number of notes to simulate. At one voice, the unison effect is disabled.

Mix: One of the unison voices always plays the main frequency of the note, and the other voices are spread in pitch above and below it. The mix control determines how loud the other voices are. The main voice always plays at full volume. At 0% mix, the unison effect is disabled.

Detune: How far apart the unison voices are detuned. At 0%, the voices are all play at the same frequency.

Inter-oscillator modulation

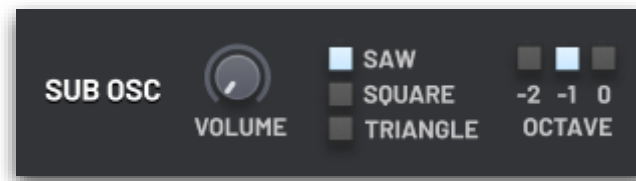


2>1 FM: Uses the output of the second oscillator (A2, B2) to modulate the frequency of the first (A1, B1). Note that this is internally implemented as phase modulation to reduce CPU load, similar to many classic synthesizers.

Ring Mod (only available in **Oscillator A**): Uses the output of the second oscillator (A2) to modulate the amplitude of the first (A1). Note that if Oscillator A1 is disabled then you will not be able to hear this effect. However, Oscillator A2 may have its volume at 0%, and the effect will still work.

Sync 2>1: When enabled, every time the second oscillator (A2, B2) reaches the end of its cycle it will force the first oscillator (A1, B1) to reset to 0% phase. By increasing the pitch of the first oscillator, this can create new timbres by essentially chopping off part of its waveform.

Sub oscillator



Volume: Controls the volume of sub oscillator.

Waveform: Select between a sawtooth, square and triangle waveform. In the “Digital” rendering mode, all these waveforms are low-passed so that the sub oscillator does not noticeably interfere with the output of the main oscillators. In the “Clean” rendering mode, the waveforms are identical to as in the main oscillators.

Octave: Determines the pitch of sub oscillator, relative to the played note, in octaves.

Filters and amplifiers

Filter routing

There are two filters, **Filter A** and **Filter B**.

The output of **Filter A** is connected to **Amp A**. The output of **Filter B** is connected to **Amp B**.

The outputs of **Amp A** and **Amp B** are combined and then sent to the effects.

The input of **Filter A** can be selected from “**Oscillator A** and the **Sub Oscillator**” (default) or “all oscillators” (A, B and the sub).

The input of **Filter B** can be selected from “**Oscillator B**” (default), “all oscillators” (A, B and the sub) or the output of **Filter A**.

When the input of **Filter B** is selected to be **Filter A**, the filters are processed in series. Otherwise, they are processed in parallel.

Within both filter units, there is a filter and saturator. The mode of the filter can be selected from a variety of choices (low pass 12dB, high pass 24dB, etc.), and the mode of the saturator can be selected from a variety of choices (soft, hard, etc.).

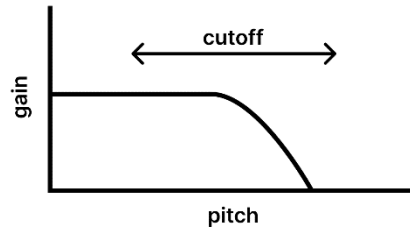
Filters



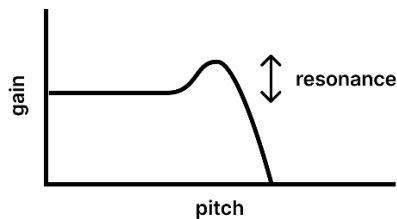
Filter mode: Select the mode of the filter. **LP 12dB** and **LP 24dB** are low pass filters which remove frequencies above the cutoff point. **HP 12dB** and **HP 24dB** are high pass filters which remove frequencies below the cutoff point. **Band 1** and **Band 2** are band pass filters which remove low and high frequencies, keeping only a range of frequencies around the cutoff point. **Notch 1** and **Notch 2** are notch filters (sometimes called band-reject) which remove a range of frequencies around the cutoff point and keep all other frequencies. **Allpass** preserves all frequencies but changes the phase relationship between different frequencies; when both filters are taken from the same source, this can create interesting phasing effects.

Source: Controls where the filter’s input is taken from.

Cutoff: Changes the cutoff frequency of the filter. The diagram below shows the effect of changing the cutoff on a low pass filter. The frequency response curve is moved left as the cutoff is decreased, and moved right as it is increased.



Resonance: Boosts the frequencies near the cutoff frequency. The diagram below shows the effect of changing the resonance on a low pass filter. The peak in the frequency response curve is moved down as the resonance is decreased, and moved up as it is increased. At 0%, there is no peak, and the curve looks like the diagram above.



Key track: Uses the pitch of the played note to offset the cutoff frequency automatically. For example, with a low pass filter this has the effect that higher notes are brighter in timbre and lower notes are duller. At 0%, this effect is disabled. At 100%, there is a 1-1 mapping of pitch frequency to cutoff offset. For example, increasing the pitch of the played note by one octave will increase the cutoff frequency by one octave.

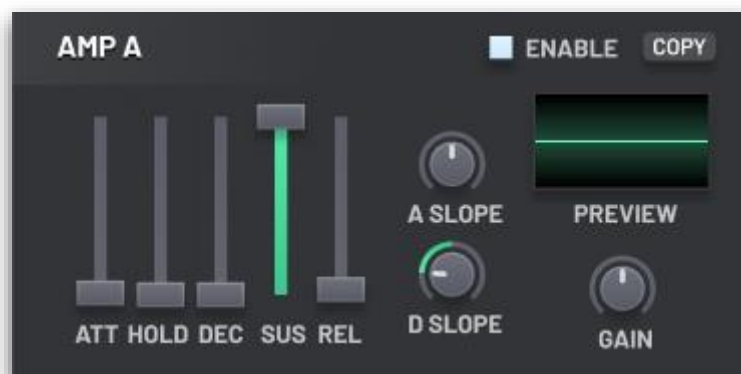
Saturation

The saturator is a form of distortion that is applied on a per-note basis, rather than to the final mixed signal (as is done by the multiband distortion effect).

Saturation mode: Changes the waveshaping formula used by the saturator. It is recommended that you choose this by ear!

Drive: Amplifies the signal before it is passed into the saturator. This has the effect of increasing the strength of the distortion.

Amplifier



The amplifiers feature an AHDSR envelope to control the volume of the sound over the lifetime over the note press, as well as a final gain parameter. When modifying any of the parameter that affect the envelope, a visual diagram of its curve over time is shown.

Attack: Controls the time it takes for the note to fade in to maximum volume, after it has been pressed.

Hold: Controls the time at which the note remains at maximum volume.

Decay: Controls the time it takes for the note to decay from maximum volume to the sustain level.

Sustain: The volume at which the note will play at after the attack-hold-decay period, until the note is released.

Release: Controls the time it takes for the note to decay to silence from whatever volume it is was at when it was released.

Attack slope: Controls the slope of the attack period. At 0%, the note begins increasing in volume quickly at first, and then slows down; this emulates the effect of the capacitors in an analog synthesizer. At 100%, the note begins increasing in volume slowly at first, and then speeds up.

Decay slope: Controls the slope of the decay and release periods. At 0%, the note begins decreasing in volume quickly at first, and then slows down; this emulates the effect of the capacitors in an analog synthesizer. At 100%, the note begins decreasing in volume slowly at first, and then speeds up.

Gain: The gain applied to the output of the amplifier before it is sent to the effects section.

Modulation matrix

LFOs



There are three LFOs. Each has four dedicated modulation destination slots, however you can use the **Matrix Assign** section for additional assignments. The light in the top-left corner of each LFO shows its current value. The “C” button in the top-right corner is a copy button, allowing you to copy the settings from another LFO.

The LFOs are bipolar. That is, when they modulate a parameter, they will both increase and decrease from its original value, depending where they are in their cycle.

Clock switch: Toggles whether the LFO’s rate is synchronized with the host’s tempo.

Rate: The rate at which the LFO oscillates. When synchronization is disabled, this is a frequency; at its minimum setting, the LFO is stopped. When synchronization is enabled, this is a number of beats.

Wave: The waveform the LFO is generating. The first few of these are self-explanatory, and a small preview of the waveform shape is shown in the control. “Sample and hold” produces random values at the given rate. “Random” similarly produces random values at the given rate, but it slides between them rather than suddenly stepping from one to the next. “Random (smooth)” is similar again but the sliding between values is additionally smoothed, so that changes in direction of movement are less noticeable. “Chaos” ignores the rate parameter, and generates a new random value every millisecond.

Mode: Every voice has its own set of LFOs. The mode determines how these are related. “Poly” allows the LFOs of each voice to run independently. When a note is pressed, its LFOs start at zero phase. “Mono” and “Free” instead keep the LFOs synchronized between the different voices (except when the rate parameter is modulated by a polyphonic source). “Mono” resets all the LFOs to zero phase when the first note is pressed after all notes having been released. “Free” keeps all LFOs running freely, and their phase will not be reset by any note presses or releases.

Mod envelopes

There are two mod envelopes. Each has four dedicated modulation destination slots, however you can use the **Matrix Assign** section for additional assignments. The light in the top-left corner of each LFO shows its current value. The “C” button in the top-right corner is a copy button, allowing you to copy the settings from another envelope.

The mod envelopes are unipolar. That is, if the modulation amount is positive, then the envelope will only increase the target parameter, and if the modulation amount is negative, it will only decrease the target.

These are attack-decay envelopes. Releasing a note does not have an effect on them. The envelopes are polyphonic; that is, each voice has its own set of envelopes. They are triggered when a note is first pressed. You can configure whether arpeggiated notes retrigger envelopes in the **Arpeggiator** section. The envelopes will not retrigger if the synthesizer is in monophonic mode, “Always glide” is enabled, and two notes overlap.

Attack: The time for the offset to increase from zero to its maximum value, after the note is pressed.

Decay: The time for the offset to decrease back to zero from the maximum value, after the attack stage has completed.

Tension: Controls the slope of the decay portion of the envelope. At 100%, the decay first decreases quickly and then slows down as it nears zero. At -100%, the decay slowly decreases at first and then speeds up after a while.



Matrix assignments



This section allows you to make additional modulation assignment using the LFO and mod envelopes as source, and it also allows you to modulate parameters to using a few additional sources. There are five slots in the matrix, each with an amount parameter.

Here is a description of the modulation sources only available in the matrix section.

Mod Wheel: The modulation wheel available on most MIDI keyboards. Monophonic.

Important – In the FL Studio native version of the plugin, this source is replaced with “Mod X”, a polyphonic source that can be controlled in the Piano Roll as with other native FL Studio plugins. As with any parameter, you can permanently link this to a control on your MIDI keyboard using the following procedure: open **Integrate** in FL Studio; in the Browser, select Generators > Integrate, and right click on “Mod X”; select “Override generic link”; and finally, adjust the control on your MIDI keyboard you want to link it to. (Note that FL Studio will, somewhat confusingly, not let you link it to the mod wheel itself!)

Key Track: Keyboard tracking. This is a polyphonic source that uses the pitch of the played note to determine the modulation amount. Note that the filters already have a key tracking option built-in to their cutoff point.

Velocity: Note velocity. This is a polyphonic source that uses the velocity of the played note to determine the modulation amount. Note that the velocity is hard-wired to determine the volume of the voice, but this can be used for additional effects, such as opening up a low pass filter for notes played with greater velocity.

Constant: A constant value of “1”. This can be used with a few parameters to extend their range. For example, the oscillator coarse tuning parameters can be set to higher values than their dials will allow by additionally modulating them with the constant source at some positive modulation amount.

Effects

Delay



The delay effect can run in one of two modes: polyrhythmic or ping-pong.

Polyrhythmic mode provides two separate delay lines of different lengths. The signal enters both lines in parallel, and the output is taken by summing the samples at the other end of the lines. By varying the length of the delay lines, different “polyrhythms” can be achieved, creating complex echoing patterns.

Ping-pong mode is a more traditional delay effect. It contains a signal delay line, but each stereo channel is taken from a different point in the line, creating a simple rhythmic pattern between the left and right channel.

Feedback: Controls the volume of the first echo, and the difference in amplitude between the following echoes. At 100%, the echoes will not decay away. At 0%, the first echo will not be audible.

Low pass: Controls the cutoff point of a low pass filter applied to the feedback. The higher this value is set, the lower the cutoff point is moved, thus increasing the apparent damping of the signal.

High pass: Controls the cutoff point of a high pass filter applied to the wet signal. The higher this value is set, the higher the cutoff point is moved. This filter is only applied to the final wet signal being mixed into the result; it does not affect the feedback signal.

Mix: Controls how much of each delay line to mix back into the dry signal. Unlike the mix controls on other effect units, changing this only affects the amplitude of the wet signal, and the dry signal remains at full volume.

Time: Controls the length of each delay line, relative to the host’s tempo.

Modulate (ping-pong only): Slightly modulates the length of the delay line using a LFO to reduce phasing issues that can occur when echoes stack up on top of each other when a note is held long enough.

L-R ratio (ping-pong only): Controls the ratio of the positions in the delay line where the left and right channel samples are read from.

Pan (polyrhythmic only): Controls the initial pan of the signal sent into both delay lines. When the signal is fed back into the start of the delay lines, its stereo channels are swapped.

Reverb



The reverberation unit creates a simulation of the blurring of a sound's tail when played in a large acoustic space, where individual echoes are in such high number they become indistinct.

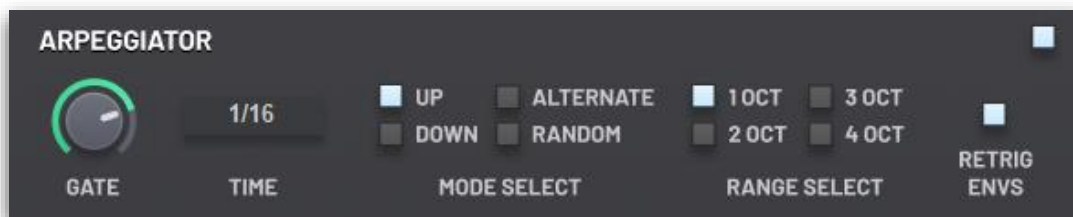
Pre-delay: Controls how long before the reverberations begin.

Time: Determines the approximate length of the reverb tail, in seconds.

Mix: The amount of the output from the reverb to be mixed with its input. 0% effectively disables the reverb, while 100% takes only the signal from the reverb.

Algorithm: Selects the acoustic space to simulate. This affects the length, brightness and complexity of the reverb tail. It is recommended that you mostly ignore the names and details of the algorithms, and instead choose whichever sounds best for your patch by ear!

Arpeggiator

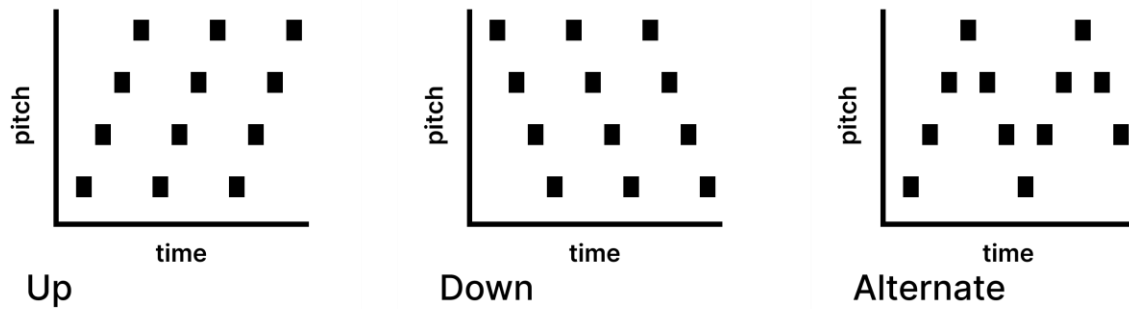


The arpeggiator converts the held notes into a repeating, arpeggiated pattern, synchronized with the host's tempo.

Gate: Determines the gap between two notes in the arpeggio. The lower the value, the shorter the notes, and the higher the value, the longer the notes. This does not change the times at which notes will start being playing (and thus does not change the rate). Instead, it changes when they end.

Time: Determines the time from the start of one note to the next, in fractions of a bar.

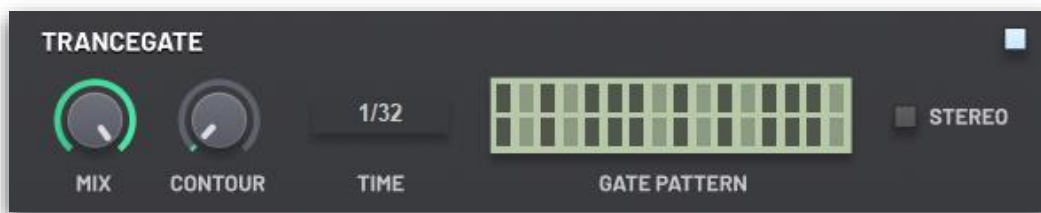
Mode select: Determines the order in which the held notes will be played in the repeating pattern. "Up" will play the note of lowest pitch first, then work its way to the note of highest pitch, and finally reset back to the note of lowest pitch and repeat. "Down" does the opposite. "Alternate" goes from the lowest note to the highest note, then back down to the lowest note, and then it repeats. "Random" picks a random note each time out of the held notes.



Range select: Duplicates the pattern between multiple octaves. The octaves are placed above the octave of the first note.

Retrigger envelopes: When enabled, each arpeggiated note will start the amp and mod envelopes from the beginning. When disabled, these envelopes will only be retriggered when a new note is actually sent from the host. Disabling this can be useful for chiptune-like effects, where the volume of note decreases over time, while the pitch is quickly switched between different octaves.

Trancegate



The trancegate applies an on-off patterns of steps to the volume, producing a sharp, rhythmic effect. You can pick between a mono or stereo pattern; the latter lets you control each stereo channel separately.

Mix: The amount of the output from the trancegate to be mixed with its input. 0% effectively disables the trancegate, while 100% takes only the signal from the trancegate.

Contour: Controls how sharp the gate is. At 0%, the gate immediately opens and closes with each step. At 100%, the gate smoothly moves from one step to the next.

Time: The length of each step, in fractions of a bar. This is always synchronized to the host's BPM.

Gate pattern: The pattern of the gate. **Left click** a step to toggle it on or off.

Stereo: When enabled, the gate pattern can be set differently for the left and right stereo channels. The top row of steps on the gate pattern represents the left channel, and the bottom row represents the right channel. When disabled, the gate pattern can only be edited so that both channels are either on or off at the same time for a given step.

Oscillator character



The oscillator character section lets you control how the oscillators behave. This is a very important section, since it lets you shape the sound of your patch at the very beginning of its synthesis, before other sections like unison and filters take place.

There are two modes the oscillators can run in: "Clean" or "Digital". The "Clean" mode provides rendering in a similar manner to other modern synthesizers, with a clean sound with little aliasing. On the other hand, the "Digital" mode replicates the more aliased signals of early virtual analogs, and lets you tweak different parameters of the sound such as its brightness and sharpness, which can help you model the character of existing synthesizers. Although the digital aliased waveforms can sound harsh when played alone, they come into a world of their own when layered with effects like unison and chorus. Ultimately, it is up to you to choose what is best for each individual patch.

Sharp and **Bright**: Controls a combination of filters and EQ to adjust the initial coloration and timbre of the high frequencies in the sound.

Bass: Boosts the low frequencies.

Bits: Reduces the bit depth of the sound, simulating the hardware of old computers and video game consoles.

Parametric EQ



The EQ allows you to boost or cut the gain of three different frequency bands. You can configure the frequencies selected by each band, and you can also change the band width of the mid frequency band.

Bass gain: The amount in decibels to boost or cut the bass frequencies by. Specifically, this is a low-shelf EQ.

Bass frequency: The frequency cutoff point selected for the bass gain. All frequencies below this point will be affected.

Mid gain: The amount in decibels to boost or cut the mid frequencies by. Specifically, this is a peaking EQ.

Mid frequency: The frequencies selected for the mid gain. All frequencies around this point will be affected.

Mid band width: The size of the mid band. This controls how far away from the **Mid Frequency** value frequencies will still be affected.

High gain: The amount in decibels to boost or cut the high frequencies by. Specifically, this is a high-shelf EQ.

High frequency: The frequency cutoff point selected for the high gain. All frequencies above this point will be affected.

Phaser



The phaser sends the signal through a variable number of all pass filters. The cutoff is modulated by a built-in LFO. This creates interesting phasing effects, causing different frequencies to be attenuated over time.

Mix: The amount of the output from the phaser to be mixed with its input. 0% effectively disables the phaser, while 100% takes only the signal from the phaser.

Stages: The number of all pass filters to send the signal through.

Base frequency: The base cutoff frequency of the filters.

LFO gain: The range around the base frequency that the LFO will sweep in.

LFO rate: How quickly the LFO runs.

Stereo: Adjusts the base frequency between the stereo channels slightly.

Feedback: Sends the phaser's output back into itself. This creates a different sound depending whether the feedback is positive or negative.

Chorus



The chorus effect simulates multiple voices playing the same note with slightly different pitch and stereo spread, achieving a similar result as unison (see the section on oscillators) but with far less CPU usage.

It works by keeping a small buffer of the input signal, and then averaging from multiple positions in the buffer, with the specific positions modulated by LFOs.

Mix: The amount of the output from the chorus to be mixed with its input. 0% effectively disables the chorus, while 100% takes only the signal from the chorus.

Voices: The number of voices to simulate.

Stereo: Offset the reading position in the buffer between the left and right channels.

Speed: Controls the speed of the LFOs. With large values, you will likely want to decrease the **Depth**, otherwise the resulting sound will be highly chaotic and noisy.

Feedback: Mix some of the output from the chorus back into itself.

Depth: Controls the size of the buffer kept, and in turn, the range of the LFOs. Larger values will produce a more intense chorusing result.

Multiband distortion



The multiband distortion unit applies distortion separately to three frequency bands of the signal before mixing them back together. It uses the same distortion algorithms as described in the section on filters.

Mix: The amount of the output from the distorter to be mixed with its input. 0% effectively disables the distorter, while 100% takes only the signal from the distorter.

Low: Pre-gain for the low frequency band, before it is passed through the distortion algorithm.

Mid: Pre-gain for the mid frequency band, before it is passed through the distortion algorithm.

High: Pre-gain for the high frequency band, before it is passed through the distortion algorithm.

Algorithm: Changes the waveshaping formula used by the distorter. It is recommended that you choose this by ear!

Compressor



The compressor automatically reduces the volume of your patch once it exceeds a threshold value. You can configure the time it takes to engage once the threshold is exceeded and how long it takes to disengage after the volume goes below the threshold. Overall, this has the effect of reducing the dynamic range of your sound; loud parts will no longer be as loud, and thus quiet parts will be less quiet in comparison. The compressor is a powerful tool. To produce pleasing results, remember that moderation is key.

Threshold: The amplitude in decibels which must be exceeded for the compressor to engage.

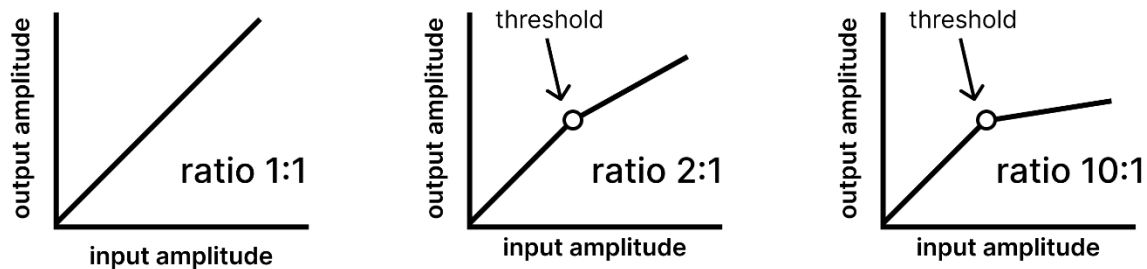
Ratio: The ratio of amplitude over the threshold in the input vs output signal. When the input signal is below the threshold the output and input amplitude are the same. When the input signal is above the threshold, the following formula is used to calculate the output amplitude from the input amplitude:

$$\text{out} = (\text{in} - \text{threshold}) \times \text{ratio} + \text{threshold}$$

For instance, when the ratio is set to 1:1, the compressor will have no effect:

$$\text{out} = (\text{in} - \text{threshold}) \times 1/1 + \text{threshold} = \text{in} - \text{threshold} + \text{threshold} = \text{in}$$

As the ratio dial is turned to larger ratio, the compressor will reduce the volume of signal more, and thus the effect will be greater. But note that regardless of the ratio, as long as the signal stays above the threshold, the compressor will not reduce the volume of the signal below the threshold.



Makeup: Gain applied to the output signal, after it has been processed by the compressor. This is used to “correct” the amplitude of the signal, so that the maximum amplitude is the same as it was before the compressor was applied, meaning that the remaining resultant effects is that the quiet parts have become louder, while the louder parts have remained constant.

Attack: The time it takes for the compressor to engage once the signal passes the threshold. The longer the attack time, the longer the “snap” at the start peaks as they are brought down in volume.

Release: The time it takes for the compressor to disengage once the signal is no longer above the threshold. Making the release time very short can cause a distortion-like effect. Making the release time too long nullifies the effect of the compressor, as it will apply a constant gain to the signal, even in the quiet parts.

Tips and tricks

Interesting modulation destinations

The modulation sources can be used to create interesting, evolving patches. If you're having trouble deciding what you can do, here are a few ideas you should try in your sounds:

- Try modulating the pan of the oscillators with a slow LFO.
- Try modulating the fine tuning of the oscillators with a fast LFO.
- Try modulating the duty cycle of a square waveform with a medium speed LFO.
- Try modulating the FM amount with an envelope.
- Try modulating the coarse tuning of a synced oscillator with an envelope.
- Try modulating the cutoff frequency of a low pass filter with an envelope.
- Try modulating the rate of an LFO with an envelope.
- Try modulating the fine tuning of an oscillator with a slow LFO set to random noise generator, to emulate the slight pitch variances of an analog oscillator.
- Try modulating the volume of an oscillator with a synced LFO set to a ramp waveform, to create a pumping effect.

And of course, you are highly recommended to browse through the factory ROMs and dissect their patches to understand how they work. Remember to use the "Bypass FX" button to quickly disable all the effects on the patch.

Reducing CPU usage

Although **Integrate** has already been optimized to keep CPU usage at a minimum, sometimes you may want to adjust patches to make them a bit lighter on your processor – especially if you're layering lots of synthesizer instances.

- Decrease the release time. This will help reduce the number of notes being synthesized, and in turn reduce the amount of calculations the CPU needs to perform.
- Switch to monophonic mode.
- Decrease the number of unison voices. Unison is quite an expensive effect, as each oscillator must synthesize many waveforms to produce the final output. Typically, you will not notice if the number of voices is decreased from 7 to 4, especially when the sound is not at the forefront of the mix. You can also use a very fast LFO to slightly modulate the fine tuning of an oscillator to create a similarly "large" sound, or you can use the **Chorus** effect from the effects page.
- Don't modulate the filter cutoff or resonance. It is relatively expensive to prepare a new cutoff and resonance, compared to applying the filter with constant parameters. Of course, this is not always possible, as many patches rely on filter modulation.
- Disable any unimportant effects on the effects page, especially those with low mix percentages.

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