

Vital User Manual

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0.0 Disclaimer

I am not employed by or associated with Vital Audio.

I was not there when this synthesizer was being developed.

I have developed this user manual by using my background of producing music for over 10 years, my ears, oscilloscopes, and frequency spectrum analyzers.

There are certain things in this user manual where I theorized what the synthesizer is doing.

For example, some of the oscillator morph algorithm descriptions may not be 100% accurate.

With that being said this manual should be sufficiently accurate to learn the ins and outs of Vital.

If someone from Vital Audio is reading this.

I have a bachelors degree in mathematics.

I have work experience coding in R, Python, and C.

Feel free to reach out to me.

1.0 General Info

Vital is a versatile wavetable synthesizer that can be used for both additive and subtractive synthesis. Combining the voices, filters, effects and modulation in Vital can create almost any sound imaginable.

1.1 Sizing

Click on the Vital symbol in the upper left hand side of the synth to adjust the viewing size of Vital.



In the sizing menu is an option which allows Vital to automatically search for updates. This can be toggled on or off.

1.2 General Navigation

At the top of the synth are options to navigate to different parts of the synth.



Below is a brief description of each section.

Voice: Oscillators, Sampler, and Filters

Effects: Effects Rack containing 9 potential effects

Matrix: Modulation matrix. Contains a list of all parameter modulation, and fine tuning for modulation curves.

Advanced: Advanced oscillator and global options

1.3 Presets

At the top of the synth are preset options

The components are labeled below



1. Preset Selector

2. Save Preset

3. Preset Options

Preset Selector

Clicking on the preset selector will bring up the preset browser.

The left hand side of the browser contains a description of the currently selected preset as well as preset folders and style tags.

The right hand side of the browser shows the specific presets in a list. The presets can be sorted by name, author, style tag, or date created.

Click the star icon next to a specific preset to add the preset to the favorites folder. Once the preset has been added to the favorites folder the star next to the preset will be shown in purple.

Save Preset

Saves the current settings in Vital as a preset in the user presets folder.

When saving a preset, provide the preset name, author, style tags, and a brief description of the preset in the comments section.

Preset Options

Browse Presets: Opens preset browser

Save Preset: Saves the current settings in Vital as a preset in the user presets folder

Open External Preset: Opens a Vital preset from anywhere your computer has access to

Export Preset: Exports preset so it can be shared with others

Import Bank: Imports many presets as preset folder(bank)

Export Bank: Exports many presets as a preset folder(bank)

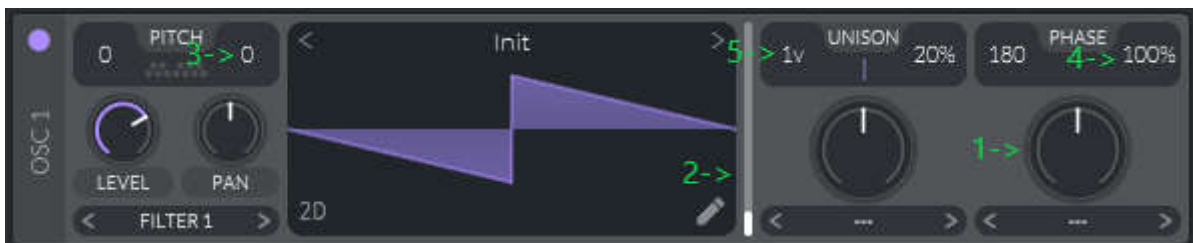
Initialize Preset: Loads default settings for Vital

Load Tuning File: A tuning file can be loaded here. Read advanced settings for more info.

Log Out: Log out of Vital Audio account

1.4 Parameter Adjustments

Vital's parameters can be adjusted with the use of knobs, sliders, or numeric, text, and percentage fields. Examples of these types of parameter adjustments are shown below.



1. Knob
2. Slider
3. Numeric field
4. Percentage field
5. Text Field

Parameters can be adjusted by clicking and dragging with the mouse.

Holding CTRL while clicking and dragging allows for fine tuning.

Holding ALT while clicking a knob will pull up the value of that parameter.

Type over this value to manually enter a new specific value for the parameter.

To cancel the value change after the text box opens, delete all of the characters in the text box.

Alternatively, right click the parameter and select "Enter Value"

Double clicking a parameter will return the knob to it's default value.

Alternatively, rick click the parameter and select "Set to Default value"

2.0 Voice

The voice section of Vital comes equipped with 3 wavetable oscillators, 1 sampler, and 2 filters.

The oscillators and sampler are run in parallel.

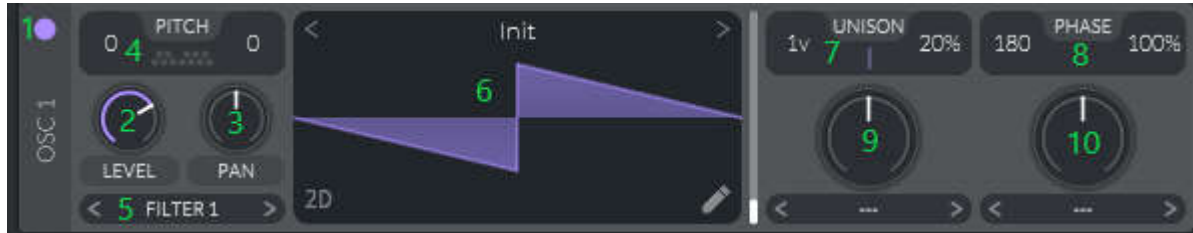
However, each oscillator can receive frequency or ring modulation from other oscillators or the sampler.

The filters can be applied to any of the oscillators, the sampler, or the other filter.



2.1 Oscillators

Below is the Vital Oscillator with its components labeled.



1. Activator
2. Level Knob
3. Pan Knob
4. Pitch Settings
5. Routing Options
6. Wavetable View
7. Unison Settings
8. Phase Settings
9. Left Modulation Knob
10. Right Modulation Knob

Each oscillator can be turned on or off using the activator the upper left corner of the oscillator. When turned on the oscillator will light up, and when off the oscillator will be dim and gray.

The oscillator's volume can be adjusted with the level knob.

The oscillator's output can be panned from left to right with the pan knob.

2.1.a Pitch Settings



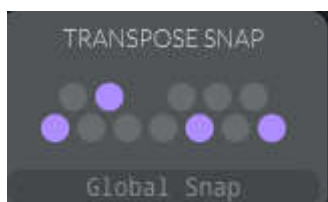
The pitch adjustment on the left will adjust the incoming midi note by that number of semitones.

This is set to 0 by default so that the oscillator will play the incoming midi note as is.

The pitch adjustment on the right will adjust the incoming midi note by that number of cents. With 100 cents being 1 semitone.

This is set to 0 by default so that the oscillator will play the incoming midi note as is.

The keyboard under the word "pitch" will open the transpose snap settings.



Once open, select specific semitones that you would like to limit the left pitch adjustment knob to.

When a semitone is selected it will lite up in purple.

If no semitones are selected, then all semitones are allowed.

To understand this better, here is an illustrative example.

If I want to only be able to adjust my oscillator by 0, or 7 semitones (a perfect 5th) I would light up the C and G keys.

Now when I drag the pitch adjustment on the left the numbers 4 through 9 will adjust my midi note by 7 semitones, and all other numbers will leave the pitch of the oscillator unchanged (In other words a 0 semitone change).

Now if we modulate the pitch adjustment on the left with an LFO we can create an arpeggiated sound.

Global snap is set up similar to transpose snap, but will limit the possible midi output notes. This can be used to force midi notes played to be in a specific scale or key.

2.1.b Routing Options

Below the pitch level and pan controls is a routing control.



Each oscillator can be routed to either filter 1, filter 2, filter 1 & 2, effects, or direct out.

If any of the filter choices are selected, the signal will flow into the effects rack after the filter.

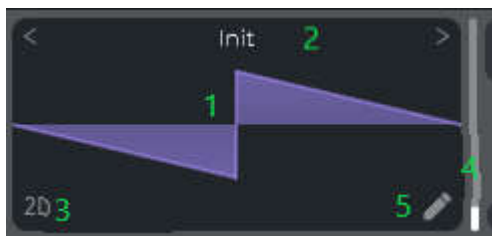
If a filter choice is selected and the filter is not turned on, then the signal will simply flow into the effects rack with no filter applied.

If direct out is chosen then the oscillator's output will come out of the synth with no filters or effects applied.

2.1.c Wavetable View

In the middle of each oscillator is the wavetable view.

Below is the wavetable view with each of its components labeled



- 1: Wavetable Visual
- 2: Wavetable Selector
- 3: Wavetable Visual selector
- 4: Wavetable Slider
- 5: Wavetable Editor

The wavetable can be chosen by clicking on the wavetable selector, selecting a folder on the left and then selecting a specific wavetable on the right.

Click the star icon next to a specific wavetable, and the star will turn purple. The wavetable will be added to the favorites folder.

Change the Wavetable Visual by clicking on the wavetable visual selector.

2D shows a 2 dimensional view of the current frame of the wavetable.

3D shows a birds eye 3 dimensional view of all of the frames of the wavetable with the current frame highlighted.

SP. Spectral. Shows the Harmonics of the wavetable.

The wavetable slider slides between wavetable frames.

2.1.c.i Wavetable Editor

Clicking on the wavetable editor opens a wavetable editing window.

This allows for editing the wave shape or the individual harmonics of any frame in a wavetable.

This section is inherently based on visuals, so it is best described through video.

Check out the following tutorials to learn more on Vital's wavetable editor.

https://www.youtube.com/watch?v=kco25cSiXBQ&t=908s&ab_channel=MarulaMusic

https://www.youtube.com/watch?v=ycNhgG0IBos&ab_channel=QB%21K

https://www.youtube.com/watch?v=1Bwk1B-LmXU&ab_channel=DataBroth

2.1.d Unison and Phase Settings



On the left side of the unison settings, the number of unison voices can be set to any integer between 1 and 16.

Note that using many unison voices will increase CPU usage.

On the right side of the unison settings, is the % of detune applied to the unison voices.

A value of 0% will apply no detune. (However unison may still have an impact on the sound due to the phase of different voices.)

A value of 100% will apply the full amount of detune as set in the advanced oscillator settings.

The default detune range is 2 semitones. Meaning 2 semitones up and 2 semitones down.

On the left side of the phase settings, the starting position of the waveform can be selected.

A phase of 180 degrees will start in the middle of the waveform.

A phase of 0 degrees will start at the leftmost side of the waveform.

A phase of 360 degrees will start at the rightmost side of the waveform. Since the waveform repeats, we have come full circle and this is identical to 0 degrees.

On the right side of the phase settings is a phase randomization setting.

The oscillator will start the wave form at a random phase within a range of x% around the phase starting degree.

For example if the phase starting degree is set to 180 degrees, and phase randomization is set to 50%.

Then the waveform will randomly start anywhere from degree 90 to 270.

If the phase randomization is set to 100% like it is by default, the phase starting position selector becomes irrelevant.

If the phase randomization is set to 0%, the phase will always start at the exact degree of the waveform that is selected.

2.1.e Oscillator Modulation Knobs



The left modulation knob morphs the shape of the waveform using different algorithms.

The right modulation knob has some additional algorithms to morph the shape of the waveform, but also includes the option to perform frequency modulation or ring modulation on the oscillator by using another oscillator or the sampler as input.

Left Knob Morphing Algorithms

Vocode: Vocoder formant shifting (Less extreme than formant scale)

Formant Scale: Formant shifting (more extreme than vocoder)

Harmonic Stretch: Keeps fundamental harmonic in place while shifting all other harmonics up and down the harmonic series. (Does not allow harmonics to go below fundamental harmonic)

Inharmonic: Keeps fundamental harmonic in place while shifting all other harmonics up or down not in line with the harmonic series. (Does not allow harmonics to go below fundamental harmonic)

Smear: Removes low ordered harmonics by morphing wave shape.

Random Amplitudes: Randomizes the amplitude of each part of the waveform while holding onto some continuity. Not pure randomness.

Low Pass: Applies a low-pass filter to the waveform. This effectively removes the higher order harmonics until the knob is turned all the way to left when only the fundamental harmonic remains, which results in a sine wave.

High Pass: Applies a high-pass filter to the waveform. This effectively removes all the lower order harmonics until all of the harmonics in the human hearing range have been removed which results in silence when the knob is turned all the way to the right.

Phase Disperse: Disperses phase of the waveform. Think about it as randomly spreading out the waveform horizontally.

Shepard Tone: A shepherd's tone is a sound that appears to be continuously increasing in pitch forever. This auditory illusion occurs when a sound is increasing in pitch and slowly fades out as another pitch an octave below it slowly fades in. The knob when dragged slowly from the left to the right will perform this auditory illusion.

Spectral Time Skew: Morphs harmonics between wavetable positions.

Right Knob Morphing Algorithms

Sync: Repeats the waveform and squishes the waveform. This results in the pitch increasing in octaves relatively smoothly with out playing other notes.

Formant: Repeats the waveform and squishes the waveform but also makes the amplitudes of the waveform approach 0 the closer they get to 0 degrees or 360 degrees.
This is similar to sync, but it avoids introducing dc offset

Quantize: Quantizes the waveform to a grid. The grid gets less refined the more the knob is turned to the right. This is essentially bit/sample reduction.

Bend: Pinches waveform at 180 degrees and morphs the rest of the wave form to the right or the left respectively depending on which way the knob is turned.

Squeeze: Pinches waveform at 0, 180, and 360 degrees and morphs the rest of the wave form to the right or the left respectively depending on which way the knob is turned.

Pulse: Pulse width modulation. Pinches waveform 0 degrees. Squishes waveform into this point. Added area of waveform is flat.

Frequency Modulation: Plays the current oscillator at the frequency of another oscillator or sampler. **This results in complex sounds with distinct harmonic characters and is widely used in modern sound design.**

To learn more watch the following videos

https://www.youtube.com/watch?v=r3EQQ-XF3jA&ab_channel=FloydSteinberg

https://www.youtube.com/watch?v=LnoqZ45ooB8&t=385s&ab_channel=ZenWorld

Ring Modulation: Plays the current oscillator with the amplitude modulated from another oscillator or sampler.

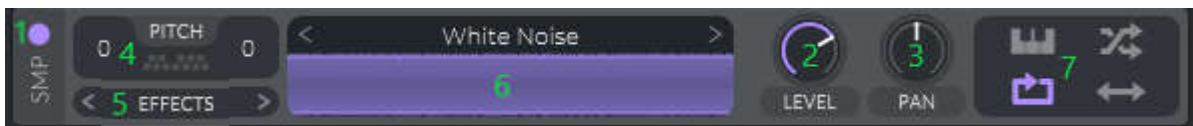
To learn more watch the following videos

https://www.youtube.com/watch?v=rYdBHWF1_2k&t=230s&ab_channel=Runningonair

https://www.youtube.com/watch?v=DCPxe6P1KWo&ab_channel=Sweetwater

2.2 Sampler

Below is the Vital Sampler with its components labeled.



1. Activator
2. Level Knob
3. Pan Knob
4. Pitch Settings
5. Routing Options
6. Sample View
7. Sample Playback Options

The sampler can be turned on or off using the activator the upper left corner of the sampler.

When turned on the sampler will light up, and when off the sampler will be dim and gray.

The sampler's volume can be adjusted with the level knob.

The sampler's output can be panned from left to right with the pan knob.

2.2.a Pitch Settings



The pitch adjustment on the left will adjust the incoming sample by that number of semitones.

This is set to 0 by default so that the sampler will play the incoming sample as is.

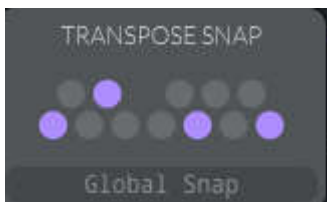
The pitch adjustment on the right will adjust the incoming sample by that number of cents. With 100 cents being 1 semitone.

This is set to 0 by default so that the sampler will play the incoming sample as is.

Note that samples will be sped up or slowed down to increase or decrease pitch.

There are no granulization options available.

The keyboard under the word "pitch" will open the transpose snap settings.



Once open, select specific semitones that you would like to limit the left pitch adjustment knob to.

When a semitone is selected it will lite up in purple.

If no semitones are selected, then all semitones are allowed.

To understand this better, here is an illustrative example.

If I want to only be able to adjust my sample by 0, or 7 semitones (a perfect 5th) I would light up the C and G keys.

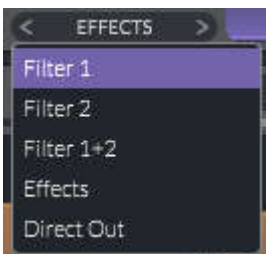
Now when I drag the pitch adjustment on the left the numbers 4 through 9 will adjust my sample note by 7 semitones, and all other numbers will leave the pitch of the oscillator unchanged (In other words a 0 semitone change).

Now if we modulate the pitch adjustment on the left with an LFO we can create an arpeggiated sound.

Global snap is set up similar to transpose snap, but will limit the possible midi output notes. This can be used to force midi notes played to be in a specific scale or key.

2.2.b Routing Options

Below the pitch controls is a routing control.



The sampler can be routed to either filter 1, filter 2, filter 1 & 2, effects, or direct out.

If any of the filter choices are selected, the signal will flow into the effects rack after the filter.

If a filter choice is selected and the filter is not turned on, then the signal will simply flow into the effects rack with no filter applied.

If direct out is chosen then the oscillator's output will come out of the synth with no filters or effects applied.

2.2.c Sample View

The sample view contains a sample chooser as well as an image of the chosen sample's waveform.



When the sample is played the amount of the waveform that has been played will be highlighted.

The sample can be chosen by clicking on the title of the sample, selecting a folder on the left and then selecting a specific sample on the right.

Click the star icon next to a specific sample, and the star will turn purple. The sample will be added to your favorites folder.

2.2.d Sample Playback Options

On the right hand side of the sampler are sample playback options.



Keytracking is the keyboard in the upper left corner.

Activating this will make the sample be played back at a pitch according the midi note played.

Note that samples will be sped up or slowed down to increase or decrease pitch.

There are no granulization options available.

If keytracking is deactivated like it is by default the sample will play at it's original pitch and speed.

Shuffle is the crossing arrows in the upper right corner.

Activating shuffle will start the sample from a random spot within the sample each time a midi note is played.

Loop is the circular arrow in the lower left corner.

Activating the loop will loop the sample endlessly

Bounce Back is the double sided arrow in the bottom left corner.

Activating Bounce Back will play the sample in reverse after the sample finishes playing forward.

2.3 Filters

Below is a Vital filter with its components labeled.



1. Activator
2. Filter Type Chooser
3. Filtered Frequency Response Curve
4. Cutoff Frequency
5. Resonance
6. Filter Type Blend
7. Filter Input Routing
8. Drive Knob, Mix Knob, and Keytracking Knob

Each filter can be turned on or off using the activator in the upper left corner of the filter. When turned on the filter will light up, and when off the filter will be dim and gray.

In the middle of each filter is a visual of the filter's frequency response curve.

Surrounding the filter frequency response curve are 3 sliders which control the filter's cut off frequency, resonance, and filter type blend.

Note that Formant, Comb, and Phase filters may have different sliders.

The cut off frequency sets the frequency where filtering will begin to be applied.

The resonance adjusts the level of a peak at the cutoff frequency.

The resonance can be set high enough that the filter begins self oscillating and the pitch of the cutoff frequency becomes audible.

Cutoff frequency and resonance can be adjusted simultaneously by clicking and dragging inside the filter frequency response curve box.

Filter type blend will smoothly transition from one type of filter to another.

Typically each specific filter will have 2 or 3 types of filters for different positions on the slider.

For example, the default filter "Analog : 12dB", smoothly transitions from low-pass, to band-pass, to high-pass filter when dragging the slider.

In the bottom left of the filter are filter input routing options.

Any oscillator, the sampler, or the other filter can be routed into the filter using these options.

If something is routed into the filter the text will light up in orange, and the routing options in the oscillator or sampler will be updated.

If an oscillator or the sampler is unrouted from a filter, it will automatically be routed to the effects rack.

The drive knob will add a small amount of saturation after the filter is applied.

The mix knob adjusted the dry/wet signal of the filter and the drive knob.

Note that there may be some strange shapes on the frequency response curve when adjusting the mix knob. This is normal, and has to do with the way filters work regarding phase.

The keytracking knob will adjust the filter cutoff frequency according to the incoming midi note. If set to 100%, the cutoff frequency will perfectly correlate to the incoming midi note.

To demonstrate this, here is an illustrative example.

Set the cutoff frequency to 1046.4hz (note this is 4 times the default value of 261.6hz which is middle C), and set keytracking to 100%.

Now play a note and slowly turn the resonance to 100%.

You will hear the self oscillation of the filter playing 2 octaves up perfectly in tune with any note you play.

For self oscillation one octave set the cutoff frequency to 523.2hz.

For self oscillation in unison leave the cutoff frequency at it's default value.

For self oscillation a perfect 5th up set the cutoff frequency to 392.4hz.

*Note this wont work for very high midi notes due to rounding error.

Negative values for keytracking will set the cutoff frequency lower for higher pitches and higher for lower pitches.

Filter types

The different filter types achieve the same goal but do so with different algorithms.

The differences in the filter types become significantly more apparent when using self oscillation, turning up the filter drive, or applying post processing distortion.

Analog | Dirty | Ladder | Digital

Ladder is supposed to mimic the transistor ladder filters from Moog.

12db: Frequencies are 12 decibels quieter for each octave distance from the cutoff frequency.

Note that an octave is either half or double a frequency value.

Blend, blends between a low-pass, band-pass, and high-pass filter.

24dB: Frequencies are 24 decibels quieter for each octave distance from the cutoff frequency.

A 24 db filter has twice the slope of a 12 db filter.

Blend, blends between a low-pass, band-pass, and high-pass filter.

Notch Blend: Blend, blends between a low-pass, notch, and high-pass 24 dB filter.

Notch Spread: Blend, blends between a band-pass, and notch 24 dB filter.

Note that in the middle of this blend is an interesting double notch filter.

B/P/N: Band Peak Notch. Blend, blends between a band-pass, peak and notch filter.

Diode

Diode is supposed to mimic the diode ladder filter from Roland.

Think of an Acid House Bass sound.

Lowshelf: Blend, blends between a low-pass, and a band-pass filter.

Lowcut: Blend, blends between a low-pass, and a band-pass filter.

If Cutoff frequency is set high enough, sub frequencies will be cut with a slight bump at the cutoff frequency.

Formant

A formant filter is supposed to mimic how the human voice produces vowel sounds. This is typically achieved with high peaks and low valleys which approach each other.

Many of the filter controls are different for the formant filter compared to the previous filter types.

The blend slider is replaced with a formant transpose slider.

This will shift all of the peaks and valley's simultaneously in an exponential fashion.

The resonance and frequency cutoff sliders, are replaced with formant X and Y sliders.

These will alter the relationship of the peaks and valleys which can be used to produce different vowels.

The drive knob is replaced with a peak knob.

This is essentially a resonance knob for the peaks and valleys.

The key track knob is replaced with a spread knob.

When turned to the left, this will spread out all of the peaks and valleys.

When turned to the right, the peaks and valleys will converge to a band-pass filter.

Comb

Comb filters are a series of notch filters.

Typically placed exponentially apart or as part of the harmonic series.

Comb filtering can achieve the same sound of a flanger due to the way a flanger works with phase cancellation.

However, you should not think of a comb filter as the same as a flanger, because they do two different things.

They are two separate tools and approaches that can be used to achieve the same result.

Note that the drive knob is replaced with a cut knob.

The cut knob will act differently depending on the specific comb filter algorithm chosen.

This knob can be used to control a low pass, band-pass or high pass filter after the comb-filter.

Or this knob can be used to control a dry/wet parameter for the comb filter starting at a specific cutoff frequency.

Negative comb filters mimic a flanger where the delayed sound is inverted from the original sound.

Positive comb filters mimic a flanger where the delayed sound is not inverted from the original sound.

Positive results in more even order harmonics, while negative results in more odd order harmonics.

Phaser

A comb filter with a less notches.

Comb filtering can achieve the same sound of a phaser due to the way a phaser works with phase cancellation.

However, you should not think of a comb filter as the same as a phaser, because they do two different things.

They are two separate tools and approaches that can be used to achieve the same result.

Blend blends between the amount of notches in the filter (goes hand in hand with poles in a phaser).

Negative mimics a phaser where the delayed sound is inverted from the original sound.

Positive mimics a phaser where the delayed sound is not inverted from the original sound.

Positive results in more even order harmonics, while negative results in more odd order harmonics.

3.0 Modulation and Macros

Parameters in Vital can be modulated in a number of different ways including but not limited to envelopes, LFOs, randomness, and macros.

A modulator can be added to almost any parameter in Vital.

Modulation squares are the empty boxes in vital.

A few examples of modulation squares are shown below.



To add modulation to a parameter hover over a modulation square, then click and drag the modulation on top of any parameter.

Note that the modulation will be auditioned on any parameter while dragging the modulation over it, but will not actually be applied until the mouse click is released.

Once modulation is applied to a parameter a few things will happen.

A modulation circle will appear in the modulation square.

The same modulation circle will appear when the mouse hovers over the modulated parameter.

The second corresponding modulation circle will also appear when the modulation circle in the modulation square is clicked.



While a midi note is played, a cyan line or curve is shown on the parameter to create a visual of how much modulation is being applied to the parameter.



If the mouse hovers over a modulated parameter with multiple modulation sources, a modulation circle for each modulation source will appear with a descriptive label.

Clicking and dragging a modulation circle will increase or decrease the modulation amount.

The pie in the modulation circle is a visual of the modulation amount.

Additionally a cyan line or curve on the parameter will be shown to visualize the modulation amount.

Holding CTRL while clicking and dragging a modulation circle will allow for fine tuning.

Holding ALT while clicking and dragging a modulation circle will allow for modulation amount to be typed in as a specific value.

Right clicking on a modulated parameter

will allow the option of disconnecting individual modulation sources, or disconnecting all modulation sources at once.

Right clicking on a modulation square will allow the option of disconnecting individual modulation destinations, or disconnecting all modulation destinations at once.

Right clicking a modulation circle will allow the following options

Remove: Removes modulation permanently

Bypass: Modulation is bypassed temporarily until unbypass is applied

Make Bipolar: The parameter moves symmetrically above and below the original parameter value

Make Unipolar: The parameter moves in one direction away from the original parameter value and back.

Make Stereo: Modulate the left and right parameters in opposite directions for the left and right channels

3.1 Envelopes



Envelope 1 is mapped to oscillator and sampler volume by default.

This can only be unmapped by applying a different modulation source to an oscillator or sampler volume.

In the upper right hand corner of the envelope view is a magnifying glass.

Click and drag on this magnifying glass to zoom in and out of the envelope.

Dragging up will zoom in, and dragging down will zoom out.

Note that time is labeled in seconds or milliseconds on the x axis and has vertical grid lines.

Each envelope has 6 parameters at the bottom of the envelope view.

Delay, attack, hold, decay, sustain, release.

These parameters can be adjusted by clicking and dragging their knobs

Alternatively, these parameters can be adjusted by clicking and dragging the dots in the display.

Closed dots adjust the endpoints of the line segments.

Open dots adjust the curves.

Delay is the amount of time in seconds after the midi note plays until the envelope begins.

Attack is the amount of time in seconds that it takes for the envelope to get from it's starting value to its maximum value.

Hold is the amount of time in seconds that the envelope will stay at its maximum value if the midi note is still being played.

Decay is the amount of time in seconds that it takes for the envelope to get from its maximum value to its sustain level after the amount of hold time if the midi note is still being played.

Sustain is the % envelope value between the minimum and maximum.

The sustain level will be reached and held after the delay, attack, hold and decay time if the midi note is still being played.

The level will quickly move to the sustain level if the midi note is released early.

Release is the amount of time in seconds after the midi note is released that it takes for the envelope to move from the sustain level to the minimum level.

3.2 LFOs

LFO stands for low frequency oscillator.

At its core, an LFO is a modulation parameter that repeats at a specific frequency.

Vital's LFO can modulate parameters in an essentially infinite amount of different shapes by using built in LFO shapes or creating new ones.

Below is the Vital LFO with its components labeled.



1. LFO Modulation Squares
2. LFO Shape Chooser
3. LFO Shape Visual
4. Brush Tool
5. Grid Options
6. LFO Phase Slider
7. LFO Mode Chooser
8. Frequency Settings

9. Smooth Knob, Delay Knob, and Stereo Knob

LFO Shape Chooser

A specific LFO shape can be chosen by clicking in the upper right hand corner on the LFO Shape Chooser. Select a folder on the left and then select a specific LFO shape on the right.

Click the star icon next to a specific LFO shape, the star will turn purple. The LFO shape will be added to your favorites folder.

LFO Shape Visual

The LFO Shape Visual shows how a parameter will be modulated by the LFO.

The minimum value is at the bottom of the display and the maximum value is at the top.

As a note is played, the LFO Shape Visual will highlight the part where the LFO is at in it's cycle.

Click and drag on open dots to control points in the LFO shape.

Note that open dots will snap to the grid when dragged close to the grid.

Click and drag on closed dots to control the curve between two open dots.

Hold shift while clicking and dragging on any closed dot to control all curves at once.

Double click an empty space to add a new closed dot.

Double click an existing closed dot to delete it.

Click the squiggly line to the left of the LFO Shape Chooser to smooth out all angles in the LFO shape.

Grid Options

The grid number on the left will determine the amount of horizontal grid lines.

The grid number on the right will determine the amount of vertical grid lines.

Brush Tool

Click on the brush icon to activate the brush tool.

Note that the brush icon will become highlighted when activated.

Click on the shape to the right of the brush icon to choose a brush shape to draw with.

While the brush tool is activated, clicking in the LFO Shape Visual will draw the brush shape in between each horizontal grid with the level at the height of the mouse.

Note that clicking and dragging across the LFO visual will draw across the entire visual with ease.

Holding CTRL while the mouse is hovering over the LFO display will toggle the brush tool on and off.

LFO Phase Slider

Directly below the LFO visual is the LFO phase slider.

By default the LFO phase slider will determine where the LFO shape begins.

However this depends on the LFO mode.

LFO Mode Chooser

In the bottom left of the LFO display is the LFO Mode chooser.

The following LFO modes are available in Vital.

Trigger: Starts LFO at LFO phase slider position each time midi note is played. When the LFO reaches the end of the shape, the LFO starts from the beginning of the shape.

Sync: Starts LFO at LFO phase slider position according to the BPM in the project. When the LFO reaches the end of the shape, the LFO starts from the beginning of the shape.

Envelope: Starts LFO at LFO phase slider position each time midi note is played. When the LFO reaches the end of the shape, the modulation stops.

Sustain Envelope: Starts LFO at beginning of LFO each time midi note is played. When the LFO reaches the LFO phase slider, the modulation stops.

Loop Point: Starts LFO shape at beginning of LFO each time midi note is played. When the LFO reaches the end of the shape, the LFO starts from the LFO phase slider position.

Loop Hold: Starts LFO shape at beginning of LFO each time midi note is played. When the LFO reaches the LFO phase slider position, the LFO starts from the beginning of the shape.

LFO Frequency Settings

To the right of the LFO mode chooser are the LFO frequency settings.

Drag the left side of these settings to adjust the frequency rate.

Click the right side of these setting to adjust the frequency rate style.

Below are descriptions of the frequency rate styles.

Seconds: Self explanatory. Repeats are in seconds.

Tempo: Repeats are in fractional intervals of project BPM. For example eighth notes (1/8)

Tempo Dotted: Repeats are in fractional intervals of project BPM. For example dotted eighth notes (1/8)

Tempo Triplets: Repeats are in fractional intervals of project BPM. For example eighth note triplets (1/8)

Key track: Frequency is set to the frequency of the incoming midi note. The frequency can be offset from the incoming midi note by semitones or cents on the left side of the LFO frequency settings for this style.

LFO Knobs

On the bottom of the LFO display are three knobs.

The smooth knob will cross-fade the beginning and end of the LFO together over the course of a specified time in seconds.

Right clicking the word "SMOOTH" allows for replacing the smooth knob with a fade in knob.

The fade in knob determines the amount of time in seconds for the LFO to be faded from a dry unaltered signal to a fully wet modulated signal.

The delay knob determines the amount of time in seconds after the midi note is played before the LFO will be applied.

The stereo knob allows for an offset in the LFO phase slide between the left and right channels.

3.3 Random

Random allows for random modulation to be applied to parameters.

True randomness is non continuous.

Modulating a parameter in a non continuous way will most likely introduce digital artifacts and clicks.

For things to sound musical we would prefer to use specific randomness algorithms that allow for continuity.

In general these algorithms will move a small amount from their current position to their next position to allow for a more continuous type of randomness.

To be clear consider the following example.

The goal is to generate 10 values between -10 and 10

True randomness would generate something like the following vector.

[5, -8, -2, 9, 10, 4, 8, -6, 8, 0]

Note the numbers jump around a lot.

The biggest jump is the jump from -6 to 8 which is a jump of 14.

No consider another algorithm to generate random numbers where each number is the previous number plus a random number between -2 and 2.

This algorithm would generate something like the following vector.

[5, 6, 5, 3, 2, 3, 3, 2, 4, 6]

Note that now the numbers move in a smoother more continuous fashion.

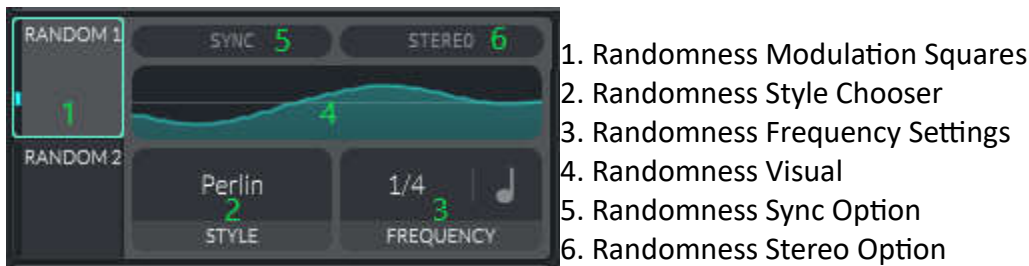
Think about a speaker trying to play different amplitudes.

The speaker has a built in digital signal processor that allows for small jumps to be played smoothly.

However, large jumps can start to cause issues that will result in digital artifacts or clicks.

This is why the second type of algorithm would be preferred if we would like continuous musical results free of artifacts and clicks.

Below is the Vital Rantom Modulation Generator with its components labeled



Randomness Style Chooser

The Randomness Style Chooser is located in the bottom left.

Clicking on this will allow for the following randomness algorithms.

Perlin: Perlin Noise. Developed by Ken Perlin to create natural forms of randomness in textures for graphics in the movie Tron. Perlin received an Academy Award for Technical Achievement for creating this algorithm.

Sample and Hold: Holds at an amount then randomly jumps to another held amount, then randomly jumps to another held amount, and so on and so forth. Between randomly held amounts the modulation has a quick (but not completely vertical) slope to allow for smooth transitions free of digital artifacts and clicks.

Sine Interpolate: Randomness based around a sine wave oscillation

Lorenz At tractor: Non repeating randomness shaped like a butterfly's open wings. At first there may appear to be repeating a pattern, but there is actually not. Created by Edward Norton Lorenz. Originally used to try and describe the atmosphere and weather. Resulted in the butterfly effect theory.

https://www.youtube.com/watch?v=aAJkLh76QnM&ab_channel=It%27ssoblatant

Randomness Visual

The randomness visual shows the amount of modulation applied as midi notes are played.

Randomness Frequency Settings

In the bottom right of the randomness view are the randomness frequency settings.

Note that frequency settings will make the most sense for the sample and hold randomness style.

Other randomness style frequencies will be relative to each other.

Drag the left side of these settings to adjust the frequency rate.

Click the right side of these setting to adjust the frequency rate style.

Below are descriptions of the frequency rate styles.

Seconds: Frequency is in seconds.

Tempo: Frequency is in fractional intervals of project BPM. For example eighth notes (1/8)

Tempo Dotted: Frequency is in fractional intervals of project BPM. For example dotted eighth notes (1/8)

Tempo Triplets: Frequency is in fractional intervals of project BPM. For example eighth note triplets (1/8)

Key track: Frequency is set to the frequency of the incoming midi note. The frequency can be offset from the incoming midi note by semitones or cents on the left side of the LFO frequency settings for this style.

Randomness Sync Option

In the top of the randomness view is a Sync option.

Clicking on the word Sync will highlight the word and enable the option.

If the sync option is on, then randomness starts with the value it previously left off on at the end of the previous midi note.

If the sync option is off, then randomness starts from the beginning of the randomness algorithm each time a midi note is played.

Randomness Stereo Option

In the top of the randomness view is a Stereo option.

Clicking on the word Stereo will highlight the word and enable the option.

If the stereo option is on, then a randomness algorithm will be generated for the left and right channels separately.

3.4 Other Modulation Sources

On the bottom right of the LFO menu is a 4x2 grid of additional modulation sources.



Note: Modulates parameter by an amount depending on the midi note played at the beginning of the midi note played.

Velocity: Modulates parameter by an amount depending on the velocity of the midi note played at the beginning of the midi note played.

Lift: Modulates parameter by an amount depending on the lift of the midi note played at the end of the midi note played.

Oct Note: Similar to note, but modulation amounts for each note repeat every octave.

Pressure: Responds to MPE pressure

Slide: Responds to MPE slide

Stereo: Modulates sets different parameter values for left and right channels.

Random: Modulates parameter by a random amount at the beginning of each midi note played.

3.5 Macros

Macros can be used to control multiple parameters at once.

Vital comes equipped with 4 macros on the left hand side of the synth.

Each macro has a macro knob, and associated modulation square beneath it.

The only difference between macros and other modulation sources is that the macro knob needs to be turned to modulate the associated parameters.

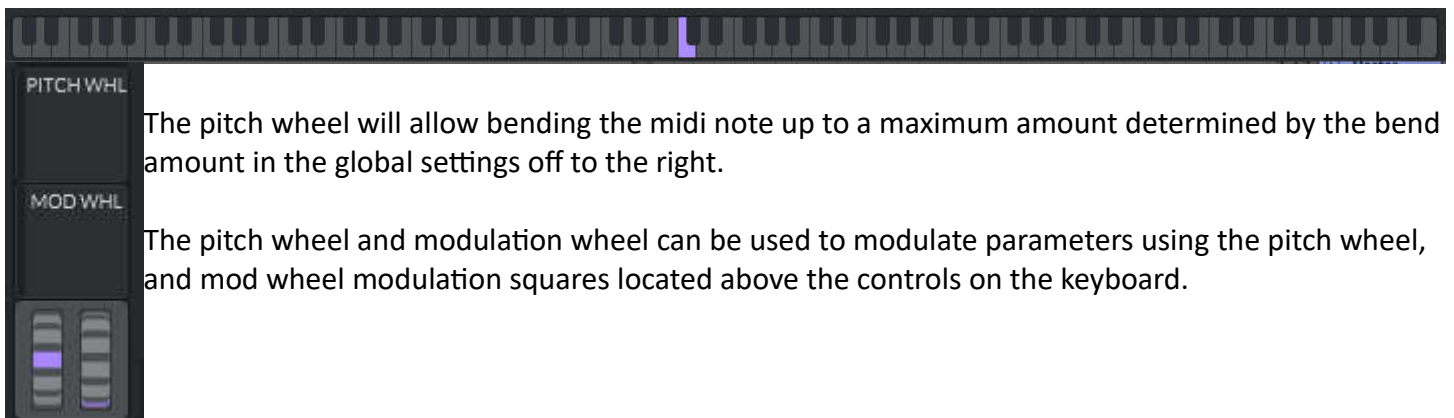
The macro knob can be turned manually by clicking and dragging, using another Vital modulation source such as LFO, using DAW automation, or midi controller mapping.

3.6 Keyboard

On the bottom of Vital is a keyboard with a pitch wheel and a modulation wheel.

When a midi note is played it will be highlighted in purple on the keyboard.

The keys can also be clicked with the mouse and will play.



The pitch wheel will allow bending the midi note up to a maximum amount determined by the bend amount in the global settings off to the right.

The pitch wheel and modulation wheel can be used to modulate parameters using the pitch wheel, and mod wheel modulation squares located above the controls on the keyboard.

4.0 Matrix

The matrix menu shows a list of all active parameter modulations, and a modulation remap curve for the selected modulation



4.1 Modulation List

The list view has the following columns

#: Row number. Clicking on the row number will change the number to an X and will bypass this modulation mapping until the X is clicked again.

Source: Modulation source (LFO, Envelope, Macro, Velocity, Pitch Bend, etc...)

Bipolar: Contains a switch that decide if the modulation is bipolar or unipolar.

Bipolar will move the modulation equally in both directions away from the original value.
Unipolar will move the modulation in one direction away from the original value.

Stereo: Contains a switch that decides if the modulation is stereo or mono. Stereo will move the modulation in opposite directions from the original value for the left and right channel.

Morph: Contains a curve control for how the modulation will be applied to the parameter. Note that this curve and the curve the modulation remap curve are different curves and will both be applied for the final result. This curve can be used for quick simple curves, while the modulation remap curve can be used for finer tuning, or they can be used in combination.

Amount: Modulation amount. Negative values will move the modulation in the opposite direction of positive values.

Destination: Modulation destination. Can be any parameter in vital. While the modulation is being applied a cyan line visual appears in this cell.

4.2 Modulation Remap Curve

Click on any modulation in the list to view the modulation remap curve in the bottom of the modulation view. This view can be used to create unique modulation curves with interesting results including arpeggiated, and percussive sounds.

Below is the modulation Remap Curve with its components labeled.



1. Modulation Remap Curve Chooser
2. Modulation Remap Curve Visual
3. Brush Tool
4. Grid Options

Modulation Remap Curve Chooser

A specific modulation remap curve can be chosen by clicking in the upper right hand corner of the modulation curve display on the title of the modulation curve.

Select a folder on the left and then select a specific modulation curve shape on the right.

Click the star icon next to a specific modulation curve, the star will turn purple and the modulation curve will be added to the favorites folder.

Modulation Remap Curve Visual

The minimum value is at the bottom of the display and the maximum value is at the top.

As a note is played, the modulation curve will highlight the part where the modulation is at.

Click and drag on open dots to control points in the modulation curve.

Note that open dots will snap to the grid when dragged close to the grid.

Click and drag on closed dots to control the curve between two open dots.

Hold shift while clicking and dragging on any closed dot to control all curves at once.

Double click an empty space to add a new closed dot.

Double click an existing closed dot to delete it.

Click the squigly line to the right of the grid options to smooth out all angles in the modulation curve.

Grid Options

The grid number on the left will determine the amount of horizontal grid lines.

The grid number on the right will determine the amount of vertical grid lines.

Brush Tool

Click on the brush icon to activate the brush tool.

Note that the brush tool will become highlighted when activated.

Click on the shape to the right of the brush icon to choose a brush shape to draw with.

While the brush tool is activated, clicking in the modulation curve visual will draw the brush shape in between each horizontal grid with the level at the height of the mouse.

Note that clicking and dragging across the modulation curve visual will draw across the entire visual with ease.

Holding CTRL while the mouse is hovering over the modulation curve display will toggle the brush tool on and off.

5.0 Effects

Vital has a total of 9 effects which can be added to the effects rack.

These are some of the most common effects used in sound design and can be used to create very complex, rich sounds.

The effects rack signal flows from top to bottom with the output of each effect flowing into the input of the next.

The effects rack is setup in alphabetical order by default.

All effects are disabled by default.

To use effects simply click the activator button in the top left.

To reorder the effects rack simply click and drag the effect up or down in the chain.

5.1 Chorus

Chorus is used to make a single instrument sound like a group of the same instrument playing at once. This is why it is called a chorus. Because it turns a singer into a group of singers (AKA a chorus).

Chorus duplicates the original sound at different modulating delay times with slightly modulating pitches. This is to mimic a group of instruments playing the same sound at slightly different times and pitches.

Below is Vital's Chorus with its components labeled.



1. Activator
2. Delay Visual
3. Delay Voices Parameter
4. Delay Modulation Frequency
5. Delay Timing Knobs
6. Feedback Knob
7. Filter Frequency Response curve
8. Filter Knobs
9. Dry/Wet Mix Knob

Delay Visual

Shows a visual of the spacing of the delayed voices for the chorus.

Voices

Chooses the amount of pitch modulated delays for the chorus.

Can be set to 4, 8, 12, or 16.

Think about this as how many instruments will be playing in the group.

Note that this setting is visualized in the delay visual.

Delay Modulation Frequency.

Determines the amount of time before delay time modulation repeats itself.

Can be set as a time in seconds, or a fraction of the songs BPM.

Can be set to freeze to remove delay time modulation.

Note that this setting is visualized in the delay visual.

Delay Timing Knobs

Delay 1: Relative control for delay time for half of the chorus voices.

Delay 2: Relative control for delay time for the other half of the chorus voices.

Depth: Multiplies each delayed voice by a different fractional amount as the knob is turned to the left.

Adds a different increasing amount of delay time to each delayed voice as the knob is turned to the right.

Note that these settings are visualized in the delay visual.

Feedback Knob

Feeds a percentage of the chorus output back into the chorus input.

Negative percentages will feed a phase inverted version of the chorus output back into the chorus input.

Negative feedback can be useful for solving phase cancellation issues especially for short delays with a lot of feedback.

Filter Knobs

A filter can be applied to the chorus output.

Spread: mixes between a band-pass filter and no filter.

Cutoff: Sets a cutoff frequency for the band-pass filter.

Note that these settings are reflected in the filter frequency response curve

Mix Knob

Mixes the dry unaltered sound with the chorus output.

5.2 Compressor

The compressor in vital allows for multi-band or single band downward or upward compression.

This allows for gentle compression, all out OTT squashing, or anything in between.

Downward compression turns a signal down by a specific ratio once the signal is above a certain threshold.

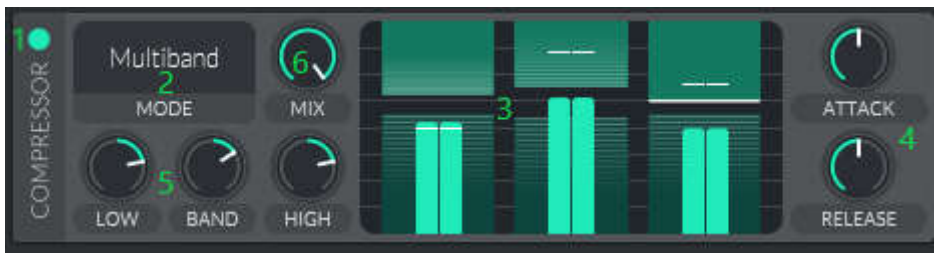
For example if a signal is 4db above its threshold and has a 4:1 ratio, then the signal will be turned down 3db so the output is only 1 db above the threshold.

Upward compression turns a signal up by a specific ratio once the signal is below a certain threshold.

For example if a signal is 4db below its threshold and has a 4:1 ratio, then the signal will be turned up 3db so the output is only 1 db below the threshold.

Both downward and upward compression reduce the dynamic range of the original signal, which can be used to make an instrument sound hefty and large.

Below is the Vital compressor with its components labeled.



1. Activator
2. Compressor Banding Options
3. Compressor Visual
4. Attack and Release Knobs
5. Post Compressor EQ
6. Compressor Mix Knob

Compressor Banding Options

Input signal can be separated into different frequency bands such that each frequency band can have its own threshold and ratio settings for downward and upward compression.

The low frequency band cuts off at approximately 150 Hz.

The high frequency band cuts off at approximately 3000 Hz.

The frequency band cutoffs can not be changed.

Compressor Visual

The opaque bright green meters in the forefront of the visual represent the output levels of the compressor. The small horizontal white lines hovering around the peak of the meters represents the input level of the compressor.

The distance between the meter peak and the horizontal white lines is the amount of compression attenuation being applied.

The transparent green rectangles in the background control the compressor threshold and ratio.

Note that upwards compressors are slightly more transparent than downwards compressors.

Click and drag on the upper/bottom edge of these rectangles to adjust the threshold for a given compressor.

Hold shift while clicking and dragging to adjust the threshold for all compressors at once.

Click and drag within a rectangle to adjust the ratio a given compressor.

Note that ratios are not shown numerically. The ratio is determined by the amount of grid lines in the rectangle compared to the amount of grid lines on the background.

When the grid lines for a downward compressor match the grid lines in the background, the ratio is 1:1 and the compressor is effectively bypassed.

When the grid lines for one of the compressors on the bottom half of the display become larger than the grid lines in the background the compressor becomes more opaque. This is because having a ratio of less than 1:1 on an upward compressor results in downward compression.

Attack and Release Knobs

Attack adjusts the amount of time that it takes after the signal hits its threshold to fully attenuate the sounds. Turning the knob all the way to the left will immediately attenuate the sound.

Turning the knob to the right allows some time for transients to pass through unaltered while fading in the attenuation.

Release adjusts the amount of time that it takes after the signal returns below the threshold to stop attenuating the sound.

Turning the knob all the way to the left will immediately stop attenuating the sound.

Turning the knob to the right allows some time to pass while attenuation is faded out.

Note that attack and release very important controls on a compressor. Adjusting them can allow for a sound to be flattened, bring out the transients, or even creating pumping sounds.

Post Compressor EQ

Allows for post compression signal to be EQed using a Low Shelf, Mid Shelf, and High Shelf.

The low frequency shelf cuts off at approximately 150 Hz.

The high frequency shelf cuts off at approximately 3000 Hz.

The frequency band cutoffs can not be changed.

Mix Knob

Mixes the dry unaltered sound with the post compressor post eq output.

5.3 Delay

Delay repeats the original signal at a specific frequency.

Optionally a filter can be applied to each repeat.

This can be a great tool for adding depth to a sound.

Below is Vital's Delay with its components labeled



1. Activator
2. Delay Visual
3. Delay Time Frequency
4. Delay Mode Options
5. Feedback Knob
6. Filter Frequency Response curve
7. Filter Knobs
8. Dry/Wet Mix Knob

Delay Visual

Shows visual of the delayed repeats of the original signal for the left and right channels.

Left is on top. Right is on bottom.

Delay Time Frequency

Determines the amount of time before the signal repeats itself.

Can be set as a time in seconds, or a fraction of the song's BPM.

Note that this setting is visualized in the delay visual.

Delay Mode Options

Determines method playing repeats of the original signal.

Note that this setting is visualized in the delay visual.

Also for any mode other than mono there are two delay time frequency settings.

Mono: Repeats the left and right signals at the same time.

Stereo: Repeats the left and right channels at potentially different times which can be set in the two delay time frequency settings.

Ping Pong: Repeats a mono summed version of the original signal alternating on the left and right side.

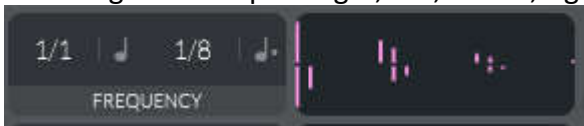
The delay time frequency setting on the right determines the time until the first repeat (on the right channel), and the time after the repeat on the left channel to repeat on the right channel.

The delay time frequency setting on the left determines the amount of time after the repeat on the right channel to repeat on the left channel.



Mid Ping Pong: Similar to ping pong, but channels are not summed to mono. Additionally a repeat that plays both the left and right channel repeats simultaneously is added.

So the signal will repeat right, left, stereo, right, left, stereo, etc.



Feedback Knob

Feeds a percentage of the delay output back into the input.

Negative percentages will feed a phase inverted version of the output back into the input.

Negative feedback can be useful for solving phase cancellation issues especially for short delays with a lot of feedback.

Filter Knobs

A filter can be applied to the delay output.

Spread: mixes between a band-pass filter and no filter.

Cutoff: Sets the cutoff frequency for the band-pass filter.

Note that these settings are reflected in the filter frequency response curve

Mix Knob

Mixes the dry unaltered sound with the delay output.

Note that this setting is visualized in the delay visual.

5.4 Distortion

Vital's distortion is a waveshaping distortion.

Waveshaping distortion shapes the sound waves of a signal by mapping amplitude input levels to amplitude output levels. Additionally a gain knob is used to boost or lower the incoming input signal.

This can be a great tool for adding textures and harmonics.

To learn more about how wave shape relates to harmonics, watch the following video on the Fourier transform. https://www.youtube.com/watch?v=spUNpyF58BY&ab_channel=3Blue1Brown

Below is Vital's waveshaping distortion with its components labeled.

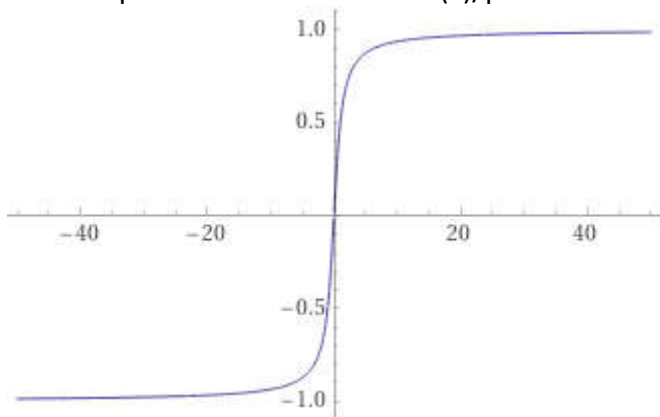


1. Activator
2. Waveshaping Input Vs. Output Chart
3. Drive Knob
4. Waveshape Type Chooser
5. Filter Placement Option
6. Filter Frequency Response Curve
7. Filter Control Knobs
8. Dry/Wet Mix Knob

Waveshapping Input Vs. Output Chart

This chart shows the relationship between input amplitude and output amplitude. Input amplitude is shown on the x axis while. Output amplitude is shown on the y axis.

Typically a function that maps numbers from an infinite range to a small fixed range will be used. For example the chart of $2 \cdot \arctan(x) / \pi$ is shown below.



Note that this function can take an infinite range as input, but the output lives between -1 and 1. These types of functions are useful because in digital audio the amplitude has to live between the values of -1 and 1.

Drive Knob

The drive knob increases or decreases the amplitude of the input signal before the waveshaping is applied.

Note that the Waveshaping Input Vs Output chart will react to changes in the drive knob. This is because the x axis will reflect the input after it has been turned up or down by the drive knob.

Waveshape Type Chooser

The will choose the specific function that will map input amplitude to output amplitude. A brief description of each type of waveshaping distortion is below.

Soft Clip: Loud signals proportionally become infinitely close to full amplitude, but never actually reach full amplitude. This results in a round curve before the signal is flattened when it reaches it's amplitude limits. This is useful for adding some harmonics without changing the original sound too much.

Hard Clip: Signals above full amplitude are set to full amplitude. This results in the signal abruptly flattening with a hard angle when it reaches it's amplitude limits. This is useful for adding odd order harmonics and changing the original sound to be more aggressive.

Linear Fold: Signals above full amplitude are reflected back the other way. This is useful for adding a unique set of harmonics, and significantly changing the original sound.

Sine Fold: Similar to linear fold, but the edges of the reflections are rounded off. This is useful for adding a unique set of harmonics, and significantly changing the original sound, but less aggressive than linear fold.

Bit Crush: Output Amplitude Levels are restricted to certain values. The drive knob controls how many amplitude levels are allowed. This is essentially the same as bit reduction. Bit depth is the amount of possible amplitude levels an audio file can read. For reference, 16 bit files contain $2^{16}=65,536$ possible amplitude levels.

Down Sample: Same as bit crush, but the drive knob has a different sensitivity curve.

Filter Placement Option

A filter can optionally be applied before or after the distortion. This is useful because distortion introduces a lot of harmonics, so it may be nice to reduce the amount of harmonics before or after distortion is applied. This can be used to reduce the harshness that sometimes comes with distorted sounds.

Note that when the filter is turned on, the Filter Frequency Response Curve is highlighted in red.

Filter Control Knobs

Blend: Blends between low-pass, band-pass, and high-pass filter type.

Cutoff: Sets cutoff frequency for filter

Resonance: Set peak amount at the cutoff frequency.

Mix Knob

Mixes the dry unaltered sound with the distorted output.

5.5 EQ

An equalizer with 3 attenuation points that can be set as a low-pass, low-shelf, high-pass, high-shelf, notch, or peak.

Below is Vital's EQ with its components labeled.



1. Activator
2. EQ Visual
3. Attenuation Point Chooser
4. Attenuation Point Type Chooser
4. Attenuation Point Control knobs

EQ Visual

The EQ frequency response curve is shown as an off-white curve.

The present harmonics in the output of the EQ are shown in the background.

The attenuation points are shown as off-white circles.

The currently selected attenuation point will be a filled in circle.

Clicking and dragging on an attenuation point will adjust the cutoff frequency and gain of the attenuation point.

Attenuation Point Chooser

Clicking on “LOW”, “BAND”, or “HIGH” will select the low, mid, or high attenuation point respectively.

Selecting an attenuation point will fill in the respective circle on the EQ visual and will pull up the respective attenuation point control knobs.

Attenuation Point Type Chooser

The low and high attenuation points are set to shelves by default.

Clicking on the attenuation point chooser will change the point to a low-pass or high-pass.

The mid attenuation point is set to a peak by default.

Clicking on the attenuation point chooser will change the point to a notch.

Attenuation Point Control Knobs

These knobs will control the currently selected attenuation point

Gain: Selects the gain in db for the attenuation point

Note that gain is disabled for low-pass, notch, and high-pass attenuation points.

Cutoff: Selects the cutoff frequency for the attenuation point.

Resonance: Set peak amount at the cutoff frequency.

5.6 Filter

An additional filter identical to the filters in the voice section of the synth.

The only difference is that this filter is routed in the effects chain.

5.7 Flanger

A flanger is a sound produced when the original sound is played with a duplicate of itself at a low but changing delay time.

This was originally done by putting a finger on a tape machine, so that the duplicate track would play just a tiny bit late at a changing amount of delay. This starts to introduce notch filters because the duplicate track starts to cancel some of the frequencies of the original track. These notches naturally start at the top of the frequency range and work their way down as the delay time is increased. This results in a comb filter.

For more information on the history of flangers and how they work watch the following videos.

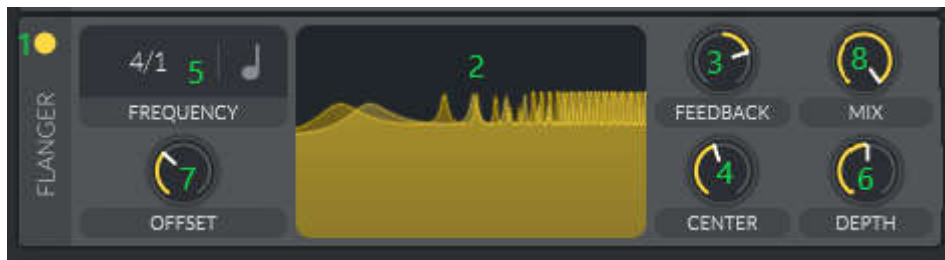
https://www.youtube.com/watch?v=Q1dB7skqrks&ab_channel=OmegaStudiosSchool

https://www.youtube.com/watch?v=Ici_YOVDI_0&t=203s&ab_channel=ProAudioFiles

https://www.youtube.com/watch?v=gYF2h5ry3kY&ab_channel=EmpressEffects

For simplicity, the Vital flanger is the specific type of comb filter that would be the result of an actual flanger, rather than an actual flanger itself.

Below is the Vital flanger with its components labeled.



1. Activator
2. Flanger Visual
3. Feedback knob
4. Center Frequency Knob
5. Center Frequency Modulation Frequency Settings
6. Center Frequency Modulation Depth Knob
7. Stereo Offset
8. Dry/Wet Mix Knob

Flanger Visual

The flanger visual shows the frequency response curve for the comb filter being applied to the sound.

Note that when the offset knob is not set to 0 or 360 degrees, there are two transparent frequency response curves. One is for the left channel and the other is for the right channel.

Feedback Knob

Feeds a percentage of the flanger output back into the input.

Negative percentages will feed a phase inverted version of the output back into the input.

Positive percentages result in more even order harmonics, while negative results in more odd order harmonics.

Center Frequency Knob

Sets the center frequency of the comb filter

Center Frequency Modulation Frequency Settings

Center frequency of comb filter is modulated up and down to create a flanging sound.

This parameter sets the amount of time it takes for the comb filter to move down and up once.

Can be set as a time in seconds, or a fraction of the songs BPM.

Can be set to freeze to remove center frequency modulation.

Center Frequency Modulation Depth Knob

Sets how far the comb filter modulation moves down and up

Stereo Offset

Offsets the comb filter center frequency for the left and right channels by a degree amount.

The degree represents the degree of the center frequency modulation cycle.

Mix Knob

Mixes the dry unaltered sound with the flanger output.

5.8 Phaser

A phaser duplicates the original sound runs it through an all-pass filter and then plays that back with the original sound.

All-pass filters have a flat frequency response curve, but adjust the phase of certain frequencies.

Thus, a phaser causes frequencies to cancel out in a few bands which is the same as a few notch filters.

This process can be repeated multiple times to create more frequency bands that have been canceled out.

For more information on the history of phasers and how they work watch the following video.

https://www.youtube.com/watch?v=gYF2h5ry3kY&ab_channel=EmpressEffects

For simplicity, the Vital phaser is a series of notch filters that are commonly the result of a real phaser, rather than an actual phaser itself.

Below is the Vital phaser with its components labeled



1. Activator
2. Phaser Visual
3. Notch Slider
4. Feedback Knob
5. Center Frequency Knob
6. Center Frequency Modulation Frequency Settings.
7. Center Frequency Modulation Depth Knob
8. Stereo Offset
9. Dry/Wet Mix Knob

Phaser Visual

The phaser visual shows the frequency response curve for the notch filters being applied to the sound.

Note that when the offset knob is not set to 0 or 360 degrees, there are two transparent frequency response curves. One is for the left channel and the other is for the right channel.

Notch Slider

Sets the number of notches in the series of notch filters.

In a classic phaser, adding more all pass filters would result in more notches.

On many phasers the number of all pass filters is called “poles”.

Feedback Knob

Feeds a percentage of the phaser output back into the input.

Center Frequency Knob

Sets the center frequency of the series of notch filters.

Center Frequency Modulation Frequency Settings

Center frequency of the series of notch filters is modulated down and up.

This parameter sets the amount of time it takes for the series of notch filters to move down and up once.

Can be set as a time in seconds, or a fraction of the songs BPM.

Can be set to freeze to remove center frequency modulation.

Center Frequency Modulation Depth Knob

Sets how far the series of notch filters modulation moves down and up

Stereo Offset

Offsets the series of notch filters center frequency for the left and right channels by a degree amount.

The degree represents the degree of the center frequency modulation cycle.

Mix Knob

Mixes the dry unaltered sound with the phaser output.

5.9 Reverb

Used to create the sound of a room.

Digital reverbs are typically a fast delay, with a high amount of feedback and different comb and all pass filters on each delay.

This is trying to replicate sound traveling through the air and bouncing off different surfaces within a room.

The delay replicates the time it takes for the sound to travel from the source to your ear, then to a wall, then back to your ear, then to another wall, and so on and so forth.

The comb filters represent the frequencies that are lost each time the sound bounces off a different surface.

The all pass filters represent the change in phase for different frequency bands as they travel through the air.

Each digital reverb can work in different ways, but these are the key ideas.

Below is Vital's Reverb with its components labeled



1. Activator
2. Pre Reverb EQ
3. Post Reverb EQ Visual
4. Post Reverb EQ Controls
5. Reverb Chorus Controls
6. Reverb Space Controls
7. Pre-Delay Knob
8. Dry/Wet Mix

Pre Reverb EQ

Pre Low Cut: Applies a high-pass filter to the input before reverb is applied.

Pre High Cut: Applies a low-pass filter to the input before reverb is applied.

Post Reverb EQ

A low-shelf or a high-shelf can be applied after the reverb to apply damping.

The EQ frequency response curve is shown in the visual as a purple curve.

The attenuation points are shown as purple circles.

The currently selected attenuation point will be a filled in circle.

Clicking and dragging on an attenuation point will adjust the cutoff frequency and gain of the attenuation point.

Note that gain is restricted to values between -6db and 0db.

Clicking on "LOW", or "HIGH" will select the low, or high attenuation point respectively.

Selecting an attenuation point will fill in the respective circle on the EQ visual and will pull up the respective EQ control knobs.

Reverb Chorus Controls

Optionally a chorus effect can be applied to the reverb.

Chorus Amount: Determines the amount of pitch modulation for the chorus.

Chorus Frequency: Determines the frequency for the pitch modulation on the chorus.

Reverb Space Controls

Size: Makes the room sound like a small closed space or an open large hall

Time: Time of the reverb tail in seconds

Pre-Delay Knob

Reverb can be put on a delay such that the original signal can be played by itself for a small amount of time before the reverb starts. This allows the original signal to be heard more clearly.

The pre-delay knob determines the delay time for the reverb signal in seconds.

Mix Knob

Mixes the dry unaltered sound with the reverb output.

6.0 Global Settings

At the bottom of the right hand panel are some of the global setting for Vital.



Voices: Determines the amount of midi note that can be played at once. Note that this is different than Oscillator Unison voices

Bend: Determines the amount of possible midi pitch bend in semitones from original pitch. For example a bend amount of two would allow the midi note to be bent two semitones up or down.

Vel Trk: Velocity Tracking.

Spread: Setting this knob all the way to the right will not impact the output of the synth. Turning this knob to the left will sum the stereo signals to mono.

Right click spread to switch spread knob for Rotate Knob.

Rotate: Rotates the left and right pan in a circle.

To understand this, here is an illustrative example.

Turn on oscillator 1 and the sampler.

Pan oscillator 1 all the way to the left.

Pan the sampler all the way to the right.

Now slowly move the rotate knob to hear a rotating stereo field

Glide: Determines the amount of time in seconds it takes for one midi note to glide into the next midi note.

Slope: Determines the curve of the pitch glide between two midi notes.

Always Glide: When activated each midi note will pitch glide starting from the previous midi note played no matter what. When deactivated, each midi note will only pitch glide while two midi notes are overlapping.

Octave Scale: When activated glide time will depend on the distance of the two midi notes. For example gliding 3 octaves will take 3 times as long as it takes to glide 1 octave.

Legato: When voices is set to 1, and legato is activated, modulations such as envelopes or LFOs wont be retriggered when a new overlapping midi note is played.

7.0 Advanced Settings

The advanced settings in Vital are separated into individual oscillator advanced settings and Global advanced settings

7.1 Oscillator Advanced Settings

To light up the oscillator advanced settings, activate the oscillator.

To light up the unison section add more than one voice to the oscillator.

If the oscillator is deactivated or only uses one voice, the advanced setting will be dim and gray.

Below are the advanced oscillator settings with the components labeled.



1. Note Tracking Option
2. Hi Resolution Wavetable Option
3. Oscillator Unison Stack Options
4. Oscillator Unison Detune Range
5. Oscillator Unison Dry/Wet Knobs
6. Oscillator Unison Spread Knobs

Note Tracking Option

Deactivating note tracking will make this oscillator play the same note regardless of the midi note played.

C3 will be played for default oscillator pitch settings.

To play a different note, adjust the oscillator pitch settings.

Hi Resolution Wavetable Option

Wavetables will use a high resolution.

This is probably unnecessary under most circumstances but may have an impact for wavetables with lots of high end frequencies.

Oscillator Unison Stack Options

Determines how unison voices will be stacked in terms of pitch

Below are brief descriptions of the unison stack options.

Unison: Plays voices with pitches ranging from the bottom of the detune range to the top of the detune range around the incoming midi note.

Center Drop 12: The same as unison, but one voice is dropped down an octave and played in mono.

Center Drop 24: The same as unison, but one voice is dropped down two octaves and played in mono.

Octave: Plays voices with pitches ranging from the bottom of the detune range to the top of the detune range around the incoming midi note and the octave above it.

2x Octave: Plays voices with pitches ranging from the bottom of the detune range to the top of the detune range around the incoming midi note, the octave above it, and two octaves above it.

Power Chord: Plays voices with pitches ranging from the bottom of the detune range to the top of the detune range around the incoming midi note and a perfect 5th (7 semitones) above it.

Major Chord: Plays voices with pitches ranging from the bottom of the detune range to the top of the detune range around the incoming midi note, a major 3rd (4 semitones) above it, and a perfect 5th (7 semitones) above it.

Minor Chord: Plays voices with pitches ranging from the bottom of the detune range to the top of the detune range around the incoming midi note, a minor 3rd (3 semitones) above it, and a perfect 5th (7 semitones) above it.

Harmonics: The first voice plays the incoming midi note. Each voice added will play the next pitch in the harmonic series (up to 16 harmonics for 16 voices). Detune will randomly move these harmonics up or down which will result in an inharmonic sound.

Odd Harmonics: The first voice plays the incoming midi note. Each voice added will play the next pitch in the odd ordered harmonic series (up to 16 harmonics for 16 voices). Detune will randomly move these harmonics up or down which will result in an inharmonic sound.
Note that the odd ordered harmonic series excludes octaves.

Oscillator Unison Detune Range

Determines the boundary for how far a voices pitch can be detuned in semitones.

Oscillator Unison Dry/Wet Knobs

Unison Blend Knob: Adjust the volume of detuned voices.

Stereo Unison: Sums stereo signal to mono when turned all the way to the left.

Oscillator Unison Spread Knobs

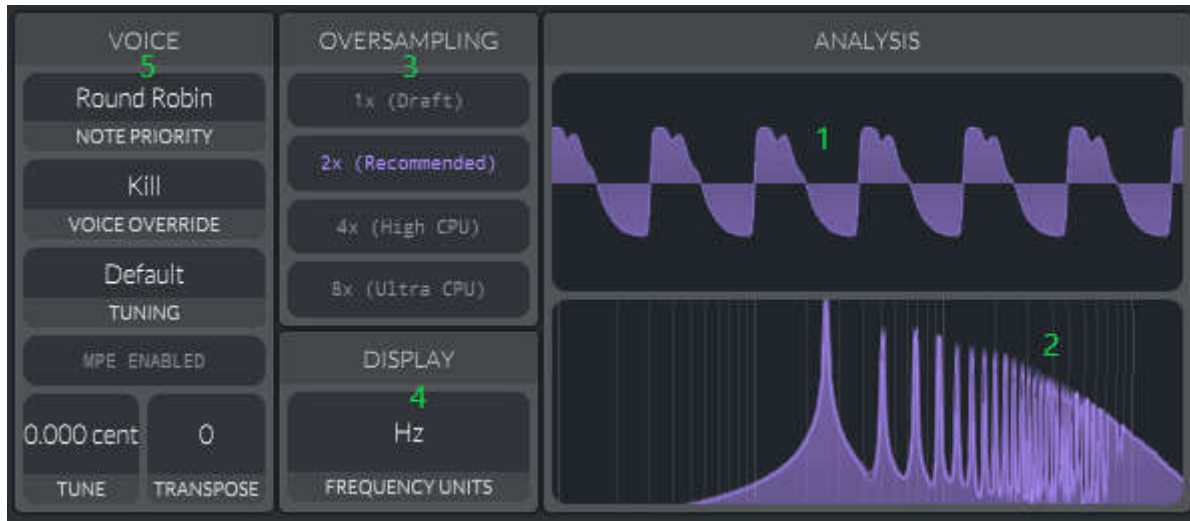
Table Spread: Spreads voices throughout different wavetable positions. The range of wavetable positions is shown visually above.

Spect Spread: Spreads voices throughout different positions on the Oscillator's Left Modulation knob. The range of knob positions is shown visually above.

Dist Spread: Spreads voices throughout different positions on the Oscillator's Right Modulation knob. The range of knob positions is shown visually above.

7.2 Global Advanced Settings

Below are Vital's advanced global settings with the components labeled.



1. Oscilloscope
2. Frequency Spectrum Analyzer
3. Global Oversampling
4. Frequency Units Option
5. Global Voice Settings

Oscilloscope

The Oscilloscope shows a current view of the waveform being played through the synth.

Frequency Spectrum Analyzer

Displays all harmonics present in the current sound.

Global Oversampling

Sets amount of oversampling for the synth.

According to the nyquist theorem 2x oversampling should be sufficient for practically all sounds that a human can hear.

However, on certain sounds with a lot of emphasis on high end frequencies, additionally oversampling may be a little bit helpful at the expense of CPU usage.

1x oversampling will likely result in some aliasing for high end frequencies.

Synths with no high end frequencies will not be impacted when set to 1x oversampling and will save some CPU useage.

Additionally, aliasing could be used as an experimental sound design technique.

Frequency Units Option

Frequencies throughout Vital can be viewed in hertz (Hz) or semitones.

Hertz are cycles per second.

Semitones are measured in distance from middle C.

Note that this setting will stay put even when presets are changed or Vital is closed and opened.

Global Voice Settings

Note Priority: When more midi notes are played than global voices, this option will decide which notes are played and which are not.

Note that a newly added midi note will always play.

Priority is taken from the pool of existing midi notes.

Below are the note priority options

Newest: Newer midi notes are prioritized over older midi notes

Oldest: Older midi notes are prioritized over newer midi notes

Highest: Higher pitch midi notes are prioritized over lower pitch midi notes

Lowest: Lower pitch midi notes are prioritized over higher pitch midi notes

Round Robin: Similar to Highest

Tuning

Loads a tuning file to determine the pitch spacing of the midi notes.

Pythagorean is the ancient original standard tuning based entirely on perfect 5ths.

Try pythagorean tuning for orchestral songs or experimental songs.

This tuning system contains some oddities and some rules left open to interpretation.

Some older instruments even used to have two pitches for the same note.

This relates to the harmonic series and how some of the higher order harmonics are “out of tune”.

https://en.wikipedia.org/wiki/Pythagorean_tuning

Just Intonation is an intonation system that was common in non-western pentatonic music.

Try just intonation for songs built around the pentatonic scale or experimental songs.

https://en.wikipedia.org/wiki/Just_intonation

Default is the modern standard tuning. AKA Equal Temperament.

Default tuning will probably sound the most comfortable for most modern music.

https://www.youtube.com/watch?v=wUBkbrvCmGA&ab_channel=NathanNokes

Tune

Adjust the output of the synth by this number of cents. With 100 cents being 1 semitone.

This is set to 0 by default so that the oscillator will play the incoming midi note as is.

Transpose

Adjust the output of the synth by this number of semitones.

This is set to 0 by default so that the oscillator will play the incoming midi note as is.