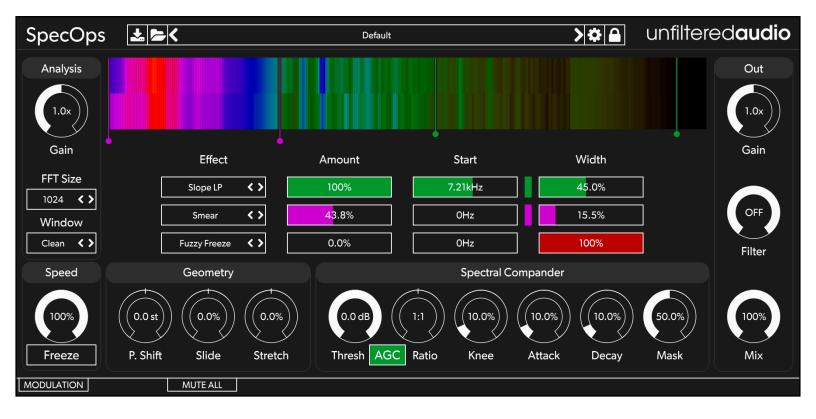
SpecOps Manual by unfiltered**audio**



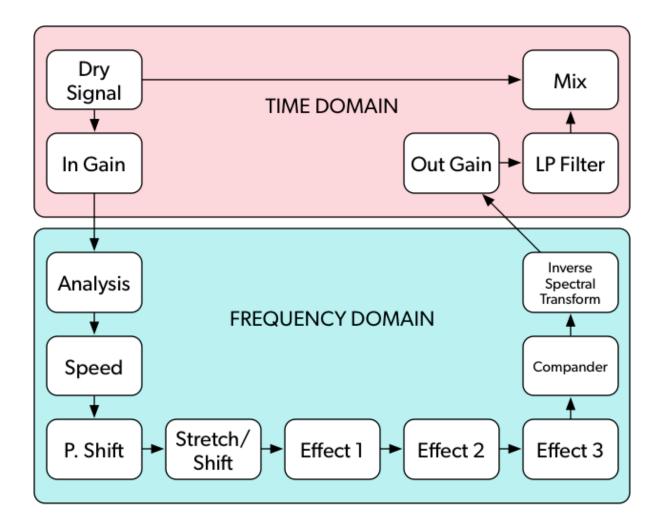


Introduction

SpecOps is the ultimate spectral processor. It consists of 36 performance-focused spectral effects, each optimized for Unfiltered Audio's infinitely deep modulation system.

Features

- 36 unique spectral effects to choose from.
- A multi-mode Spectrograph shows you exactly how every effect changes your signal.
- A Spectral Compander applies independent dynamic processing to every frequency bin: Up to 32,768 simultaneous compressors!
- Patchable modulation system: Our highly acclaimed modulation system is here with full control over every parameter.



A Note on CPU Usage, Latency, and Spectral Artifacts

SpecOps is a spectral plugin, meaning that it uses a Fourier Transform to analyze an incoming signal. Audio is processed in large batches, so CPU use tends to "spike" unevenly based on the window size.

As an example, when the FFT Window Size parameter is set to "32768", SpecOps will wait for 32,768 samples of input before processing them as a spectral frame. This means that it will be idle for 32 times longer than a Window Size of "1024", but when it is active it will use 32 times the processing power in a short period of time.

At a Sampling Rate of 44.1 kHz, SpecOps will run about 44 times a second with a Window Size of "1024", but only about once per second with a Window Size of "32768". With a Window Size of "1024", SpecOps will use a more constant amount of resources, while "32768" will lead to visible spikes on your DAW's CPU meter. Put simply, SpecOps will use roughly the same **average** amount of CPU, no matter the Window Size. However, the **range** of CPU usage will change dramatically based on window size.

Latency

On the same topic, increasing the Window Size increases the latency introduced by SpecOps. A Window Size of "32768" adds 32,768 samples of latency, or nearly a full second at a Sampling Rate of 44.1 kHz. Changing the Window Size during playback can result in pops, brief silence, or other audible artifacts depending on how the DAW deals with latency changes. As an example, in Live and Cubase, you will hear silence. In Logic, latency reports aren't acknowledged until playback restarts. This means changing the Window Size during playback in Logic can knock SpecOps out of time alignment with other tracks.

We've optimized SpecOps as much as possible. Many DSP sections will be skipped depending on parameter settings. Here are some tips for making SpecOps more efficient:

- When Pitch Shift is set to "0.0 st," CPU usage is reduced since the pitch shifting algorithm will be skipped for that spectral frame.
- If "Slide" and "Stretch" are both at 0%, CPU usage is reduced since the Stretch & Slide algorithm will be skipped for that spectral frame. This algorithm is separate from the Pitch Shift algorithm.
- When any of the individual Effect Amount parameters are set to "0%," CPU usage is reduced since that effect slot will be skipped for that spectral frame.
- When the Compander's Threshold parameter is at "0 dB," CPU usage is reduced since the companding algorithm will be skipped for that spectral frame.
- The "Smear" and "Decapitate" effects use more CPU as the Amount parameter increases.
- In general, decreasing the active Width of an effect reduces its CPU usage.

Spectral Artifacts

The process of converting a time domain signal to-and-from the frequency domain can lead to a number of spectral artifacts.

FUZZINESS WITH NEUTRAL SETTINGS

This artifact is only obvious when using extremely simple inputs to SpecOps (i.e. sine waves). This is a byproduct of our "overlap-add" technique.

To avoid: For a completely neutral default patch, change the "Clean" window to "Glitch". *Why on earth would we call it "Clean" if it's not... clean? And why is "Glitch" neutral?!:* We chose the "Clean" window based on its performance when actively using spectral effects. The "Glitch" window is a rectangle window which performs well at neutral settings but quickly falls apart with any active transformation.

BRIEF PHASING WHEN HITTING "PLAY"

You may hear a strange phasing artifact when restarting playback in some DAWs with specific SpecOps settings. If you experience this, you are most likely using a Speed parameter below 50% or a Freeze effect with an Amount above 50%.

To Avoid: When starting playback, use automation to start SpecOps at Speed = 100% (or Freeze = 0%), then automate to the desired parameter settings.

Why is this happening?: Most of our Speed/Freeze effects are based on spectral frame feedback, meaning that the magnitude and phases of the previous frame are blended with the current frame. When restarting playback in some hosts, hitting stop twice, or starting an offline render, SpecOps will clear its spectral frames for predictable playback performance. With non-default Speed/Freeze settings, the first spectral frame will be blended with a flat-phase spectrum before spectral data is acquired.

PHASE SMEARING WHEN RESETTING PITCH SHIFT

This is a similar artifact to the last. Sometimes, when changing Pitch Shift to "0.0 st," you may hear some spectral phasing. The duration of this can be dependent on the Window Size and the type of input.

To Avoid: Use a smaller Window Size. Use an input with more transients. Use a higher Input Gain setting. Alternatively, use the Stretch parameter.

Why is this happening?: The Pitch Shifting algorithm in SpecOps uses a flux detection algorithm for identifying transients. When a transient occurs, we do some phase manipulation to preserve the transient. If the incoming signal is drone-like (i.e. lacking transients), there may be some spectral smearing when changing the Pitch Shift amounts.

Getting Started

Installing SpecOps

Simply run the provided installer. On Windows, you may need to select your VST directory if the installer doesn't pick your preferred directory by default.

Browsing Presets

SpecOps comes with a wide array of presets to get you started. The preset manager at the top of SpecOps's interface is used to load, save, or browse presets. If you want to get a good feel for the power of SpecOps, spend some time with these presets before browsing the manual.

Standard Unfiltered Audio Features

CONTROLS

-All knobs and vertical sliders are controlled in a smooth up-and-down motion.

-Use the Command key (on OS X) or Control key (on Windows or OS X) to fine-tune controls.

-Double-click or Alt-click on a control to return it to a default setting.

PRESETS

-All presets are saved with a .uap file extension. These presets are compatible across all platforms and plug-in formats. They use a simple, non-encrypted XML format for easy editing. -Use the "Lock" menu to prevent specific parameters from changing while browsing presets.

INTERFACE

-Click the Gear icon on the preset manager to bring up interface options.

-Dark and Light skins provide a comfortable view for all studio situations.

-Resizable interface for easy viewing on all monitors.

Controls

Analysis

This section affects how the incoming signal is analyzed. Simple adjustments to this section can completely change the timbre of the current preset.

Gain: Sets the level of the input signal. At 1.0x, no gain is applied.

FFT Size: Changes the number of samples used for frequency analysis. Large values increase latency, increase frequency accuracy, and decrease rhythmic accuracy. If you are working with percussive material and want to preserve transients, use a smaller value. At extremely low values, the effects become very lo-fi. At extremely high values, your audio is turned into a smeared frequency cloud.

Window: Changes the type of "window" used for frequency analysis. The various windows can have dramatic impacts on the tonality of every effect in this plugin.



Speed

This section occurs after the Analysis stage and before the Geometry stage.

Speed: Master "speed" control. This creates feedback between frequency analysis frames, decreasing the rate at which new spectral frames are acquired.

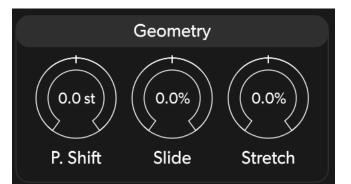
Freeze: Locks the current frame and prevents new samples from being analyzed. It is identical to turning the Speed control to 0%. This will essentially take the last group of samples and turn it into an oscillator. Try freezing some audio and turning the Pitch Shift control to create a spectral synthesizer.



Geometry

This section occurs after the Speed stage and before the master Effects section. In this section, spectral effects are created by "moving" each frequency bin.

Pitch Shift: Applies a harmonic pitch shift



to the signal. At 12 o'clock, the signal is unchanged. This control is quantized to semitones.

Slide: Applies an inharmonic frequency shift to the signal. At 12 o'clock, the signal is unchanged.

Stretch: Applies compression or expansion to the spectral bins. This algorithm is similar to the pitch shift control, but is unquantized and not phase aware. At 12 o'clock, the signal is unchanged.

Effect	Amount	Start	Width
Slope LP < 📏	0.0%	OHz	100%
Smear < >	0.0%	OHz	100%
Fuzzy Freeze 🖌 🔪	0.0%	OHz	100%

Effects

This section occurs after the Geometry section and before the Spectral Compander. For every effect, **Start** controls what frequency the effect is first applied to, and **Width** controls how many frequency bins are affected. The affected frequencies will be referred to as the "Effect Region".

FILTERS

Slope LP: This is a gentle low-pass filter. The stop-band frequency is set by Start. The Slope of the filter is set by Width. The intensity of the gain reduction is set by Amount. Every frequency higher than the last frequency in the region receives the maximum amount of gain reduction. For instance, if the low-pass Amount is at 100% and the Effect Region is 22Hz to 5kHz, every frequency above 5kHz will be fully attenuated.

Slope HP: This is a gentle high-pass filter. The Slope of the filter is set by Width. The intensity of the gain reduction is set by Amount. The frequency bin set by Start and all frequencies before that receive the maximum amount of gain reduction. For instance, if the high-pass Amount is at 100% and the Effect Region is 80 Hz to 2kHz, every frequency below 80Hz will be fully attenuated.

Region LP/HP: These filters behave identically to their Slope counterparts except they only affect frequencies within their Effect Region.

Brickwall LP/HP: These are the only two effects in SpecOps without Start and Width controls. These filters have an infinite slope, meaning that the stopband frequencies are completely attenuated.

Noise Filter: This is a simple filter that removes all frequency bins that are below a certain amplitude. This amplitude is determined by the Amount control.

Noise Focus: This is the inverse of the Noise Filter effect. All frequency bins above a certain amplitude are removed. This amplitude is determined by the Amount control.

MIXING

Region Cut: All frequency bins within the Effect Region are reduced in amplitude. The amount of attenuation is determined by the Amount control.

Region Solo: All frequency bins *outside* of the Effect Region are reduced in amplitude. The amount of attenuation is determined by the Amount control.

Region Amplify: All frequency bins within the Effect Region are boosted in amplitude. The amount of amplification is determined by the Amount control.

Thinner: This is a strange filter that cuts the amplitude of frequency bins that are less prominent than their neighboring frequencies. It is a very subtle effect, but it can help thin out a signal and emphasize its prominent frequencies. The amount of attenuation is determined by the Amount control.

Contrast: This is a filter that reduces the amplitude of quiet frequencies more rapidly than louder frequencies. The Amount control determines the intensity of the effect.

GEOMETRY

Expand/Compress: These effects are almost identical to unipolar versions of the Stretch control in the Geometry section. However, they can be applied to a specific region instead of the entire signal.

Slide Up/Down: These effects are almost identical to unipolar versions of the Slide control in the Geometry section. However, they can be applied to a specific region instead of the entire signal.

FREEZERS

Freeze: This effect behaves in a similar manner to the Speed control (below the Analysis section). However, the control curve is tweaked to produce a different feel and the effect can be applied regionally. The Amount control determines the intensity of the freeze effect.

Glitchy Freeze: This is a variation of the Freeze effect. The Amount control determines the probability that each individual frequency bin will be frozen from frame to frame. This probability is different for each channel, so this can produce very wide stereo freeze effects.

Random Freeze: The Amount control determines the probability that the entire Effect Region will be frozen this spectral frame. This probability is different for each channel, so this can produce an interesting stereo stutter effect.

Threshold Freeze: The Amount control sets an amplitude threshold. If the Effect Region contains a frequency bin that is louder than the threshold, than the entire Effect Region will be frozen for the current spectral frame.

Resonant Freeze: The Amount control sets the intensity of the freeze effect *for the loudest frequency this frame*. The intensity of the freeze effect for every other frequency bin is determined by their relative intensity compared to the loudest frequency. As a result, this will produce a strange ringing effect that is very different from the other freeze effects.

Fuzzy Freeze: This is a compound effect that combines the Freeze effect with the Noisify effect (described in the next section). This adds noise to the spectral frame to help animate frozen drones.

EFFECTS

Topmost Solo: This effect reduces the amplitude of every frequency bin *except for the loudest frequency*. The amount of attenuation is determined by the Amount control.

Smear: In this effect, louder frequency bins "leak" into their neighboring bins. Previously clear frequencies become washes of noise. The Amount control determines how much spectral leakage occurs. This is the most computationally expensive effect in SpecOps, so a high Amount setting will use noticeably more CPU.

MP3ify: This effect reduces the amount of possible information that each frequency bin can contain. Turning up the Amount control will lead to sounds that may bring back fond memories of the Napster era.

Decapitate: This effect removes the loudest frequency bin and amplifies the second-loudest bin. Turning up the Amount will repeat this effect multiple times.

Duststorm: This is a mutation of the MP3ify algorithm that corrupts the bin information through a mismatched encoder and decoder. The Amount control determines the intensity of the effect.

Noisify: This control multiplies the amplitude of every frequency bin by a random amount, the range of which is determined by the Amount control.

Clipper: This effect reduces the maximum amplitude of each bin while simultaneously amplifying each bin. This produces a heavily artifacted spectral distortion. The Amount control sets the maximum amplitude and amplification amount.

Reverse: This effect swaps the amplitude of the first bin in the Effect Region with the last. Increasing the Amount control will swap more bins at each end. At 100%, the Effect Region will be completely reversed.

GLITCHERS

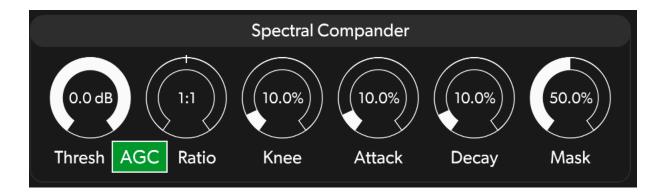
Glitchy Reverse: This is a broken version of the Reverse effect. Instead of completely swapping each frequency bin, the process only swaps phase information between each bin.

Glitchy HP: This is a very crispy HP filter with a lot of artifacts. It doesn't cut the low frequencies but boosts the high frequencies in an unpredictable manner.

Glitchy Shift: This is a totally broken pitch shifting algorithm. It will behave unpredictably when Start and Width change.

Glitch Bubbler: This effect will randomly select filter bins and duplicate them multiple times across their neighbors. The Amount control will determine how large these clumps are.

Glitch Fire: This effect will randomly swap frequency bins in the spectrum. The Amount control determines how many swaps occur each frame.



Spectral Compander

After all of the Effects are applied, the signal runs through the Spectral Compander. Unlike a typical compander (compressor + expander), *every single frequency bin* has its own compander. This means that if you have an FFT size of 1024, there are 1,024 separate companders running inside of SpecOps simultaneously!

Threshold: Sets the decibel level at which each frequency bin's compander becomes active.

Ratio: Sets how much each bin is compressed or expanded when the bin's amplitude exceeds the Threshold. At 12 o'clock, the bins are unaffected.

AGC: When active, make-up gain based on the Threshold and Ratio settings will be applied to every frequency bin.

Knee: Changes how severely the compander's gain curve transitions from linear to non-linear response.

Attack: Determines how quickly the compander reaches its target gain level when the amplitude goes above the threshold. Higher percentages lead to longer attack times.

Decay: Determines how quickly the compander returns to a neutral gain level when the amplitude drops below the threshold. Higher percentages lead to longer decay times.

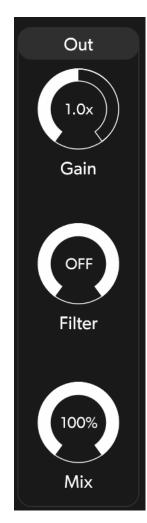
Mask: Determines the amount of spectral bleed. At 0%, the spectral compander effect can have a lot of spectral artifacts. At 100%, a lot of spectral bleed is applied to the bins, leading to a less severe, digital sound.

Output

Gain: Sets the level of the output signal. At 1.0x, no gain is applied. As gain is increased, loud signals will be softly saturated instead of hard clipped.

Filter: Applies an analog-style (non-FFT) two pole LP filter to the wet output. This can help to reduce harsh spectral content in the upper frequencies. This filter will not appear on the spectral display as it is not part of that process.

Mix: Controls the balance between the incoming dry signal and the outgoing wet (processed) signal.



Modulation System

Introduction

SpecOps utilizes Unfiltered Audio's flagship modulation system, with which you can quickly patch anything from simple parameter changes to complex and even self-evolving systems. Starting off is very easy due to the direct cable-routing user interface.

The first step is to reveal the modulation manager by clicking the "Modulation" button in the bottom corner of the interface. You'll see the plugin interface get taller, revealing modulation sources, outputs, and inputs on each modulatable plugin parameter.

Patching

Input and Output Ports

Outputs and inputs are both denoted by a simple circle. To avoid confusion, just remember that outputs are always located on the top left of each modulator. Everything else is an input.

Output ports have a small circular scaling knob below them which can be used to alter its calculated output value. This knob is set to 1.0 by default (fully clockwise), but can be turned down to 0.0 (center) to zero out its values entirely, or -1.0 (fully counter clockwise) to invert them.

Creating Connections

To create your first modulation "patch" simply click and drag on one of the outputs. You'll see a cable appear next to your mouse, the end of which you can drag to any input on the interface. When you get close enough you'll see the cable "lock" into place, showing you that the path is valid. Once you release the mouse, the connection completes and you'll see the modulation start to animate in real-time.

You can create a second connection from the same output source by clicking on it again. Likewise, try clicking and dragging from an input to an output, which works in the same fashion.

Moving or Removing Connections

In order to move a connection, click on one of the thicker ends of the cable, near the input or output port. You'll see the cable lock onto your mouse, allowing you to move it to a different input/output source. If you release the mouse while the cable is not "locked" onto any port, it will be removed entirely, which is how connections are deleted.

Inter-Modulation

As you might have noticed, all of the knobs on each modulator have input ports of their own, capable of receiving modulation just like parameters on the main section of the interface. Any output port can be connected to these inputs, meaning that a modulator can even modulate its own parameters! Experimenting with complex inter-modulation chaining can lead to dynamically evolving behavior, and even emergent systems.

Automation

Our modulation system provides 16 automation "slots" that can be assigned to one or more modulation parameters. Every knob and button on a modulator can be assigned to one of these automation slots by right clicking on it and following the "Add X to Automation Slot…" menu. Once assigned to a slot, this element will mirror the automated value sent in through your DAW. You'll also see a small number appear next to the element, denoting the automation slot number to which it is assigned.

Adding Modulators

You can add a new modulator by clicking the "+" button in the modulation header. At this time a maximum of 6 modulators can be used at once, although that is a somewhat arbitrary maximum that we plan to increase in the future.

Removing and Duplicating Modulators

Modulators can be removed or duplicated by right clicking on them and then selecting the appropriate option.

Mute Modulation

The "Mute All" button in the modulation header allows you to quickly mute and un-mute all modulation sources.

Saving and Loading

All modulation routing, values, automation, and general state is saved with the preset and/or with the DAW's session. Presets are a great way to quickly share interesting modulation patches with other users.

Modulator Types

You can change a modulator's type by clicking the dropdown menu next to its symbol. There are currently six types to choose from, each with its own characteristics and use case. Common parameters such as frequency will remain at their selected value, making the process of auditioning different modulator types both quick and easy.

LFOs: Sine/Saw/Tri/Square

The LFO family of modulators all use common waveforms which are useful for classic "envelope" style parameter modulation.

The Amplitude knob is common to all of them, corresponding to modulation depth. This can be either Bipolar or Unipolar which is determined by the state of the +/- button in the modulator's bottom left corner. Bipolar modulation fluctuates between negative and positive, whereas unipolar modulation is always positive when the amplitude is above 0% and always negative when it is below 0%.

The Frequency knob corresponds to modulation speed. It is set in Hz (cycles per second) when in non-tempo-synced mode. Tempo Sync can be activated by clicking the "Hz" button on the left side of the modulator, which will turn into a musical note symbol indicating that it has been initiated. In tempo sync mode, modulation speed is synchronized to divisions of the DAWs global tempo setting- for instance setting the frequency to ¼ will correspond to quarter-note modulation cycles.

The Phase knob determines the value of the LFO at its starting point. The starting point occurs when the DAW's transport (playback) resets. The LFO can be manually reset with the trigger button on its left side, or via the reset modulation input below that. An input signal transitioning above zero here will trigger the reset.

The Saw/Tri modulator has an additional Shape knob which determines the slope of the triangular wave. At the default value of 0.5, you get a normal triangle wave. At 0.0, you get a downward sloping triangle wave and at 1.0 you get an upward sloping triangle.

The Square modulator has a Width knob that changes the pulse width of the square wave. At the default value of 0.5, you get a normal half-up, half-down square wave. As you approach 0.0, the wave will be almost entirely down, and conversely as you approach 1.0, the wave will be almost entirely up.

Input Follower

The input follower modulator allows you to easily "sidechain" parameters to the incoming audio stream. The Smooth knob determines the speed of the rms meter used to track the incoming audio. Lower values correspond to a more instantaneous reading, while higher values smooth everything out.

Macro Control

The macro modulator can act as a manual control knob capable of outputting values to multiple parameters. This allows control over many parameters with a single gesture. Alternatively, you can patch many modulation sources into the amplitude knob's input port, turning the macro control into a summing modulation bus.

Sample and Hold Noise

The sample and hold noise modulator chooses random values at the chosen frequency rate. The Slew knob controls the amount of interpolation time whenever a new value is set. At a slew of 0.0 values change instantly, whereas at 1.0 they interpolate over the entire cycle period. Tempo Sync and Uni/Bipolar can be set on this modulator just like LFOs (see above).

Step Sequencer

The step sequencer is a modulator that cycles through up to 16 manually chosen values. **T**: "Trigger" input. This includes both a manual button and a modulation input. If the button is clicked or the modulation input transitions to a positive signal, the sequencer's active stage will increase by one.

R: Reset input. This includes both a manual button and a modulation input. If the button is clicked or the modulation input transitions to a positive signal, the sequencer's active stage will reset immediately to the stage set by the START control.

AMP: Sets the maximum amplitude of the modulation outputs.

FREQ: Sets the rate at which the sequencer automatically increments. With tempo sync off, this can be set to 0 Hz, making the sequencer only increment based on the state of the T input. **LENGTH**: Sets the number of active steps in the sequence.

START: Sets the first stage in the sequence. This stage is selected when playback restarts or the Reset input is triggered.

SLEW: Determines how quickly the output values change. At 0.0, the modulation output is stepped. As SLEW increases, new values are smoothly selected, adding a "slide" effect between values.

ROLI Lightpad

This modulator connects to the excellent ROLI Lightpad (<u>https://roli.com/products/blocks</u>) for use as a performance controller. Please note that because this is part of the modulation system, this modulator is not intended for writing automation. If you want to use the Lightpad for automation, you can use the ROLI Blocks Dashboard for loading the interface of your choice and sending MIDI CC.

To get started, connect the Lightpad to your computer using either its USB-C cable or Bluetooth. For instructions on setting up a Bluetooth connection, see the Lightpad's manual.

Once a Lightpad is connected to your computer, it should appear in the drop-down menu on this modulator. Select it and click "Connect". The interface should turn white. You are now in Location + Pressure Mode. To change the active mode, click the large control button on the side of the Lightpad.

LOCATION + PRESSURE MODE

In this mode, the modulator will track the position of one finger on the Lightpad along with its pressure. There are separate outputs for X, Y, Z (Pressure), and On (Touch Gate: on when a finger is touching the Lightpad, otherwise off).

SLIDERS

In this mode, the modulator will output the value of four vertical sliders. Due to the resolution of the Lightpad's screen, it may appear that there are only 15 possible values for the slider. However, the information that is read is higher resolution.

TOGGLES

In this mode, the modulator will output the value of four toggle switches.

QUAD PRESSURE

In this mode, the modulator will output four separate pressure values (one for each quadrant of the Lightpad).

Options

The following unique options are available in SpecOps:

Enable Display: This toggles the spectral display at the top of SpecOps's interface. If you have multiple open interfaces and want to reduce the workload on your graphics hardware, it can be useful to turn this off.

Color-Coded Regions: When active, each effect slot will have a unique color associated with it. When off, every effect control will be the standard green.

Recipes and Ideas

2+ Octave Pitch Shift

- · Set both Pitch Shift and Stretch to their maximum values.
- For even deeper shifting, add more Expand or Compress effects to the effect slots.

Amplitude Sensitive Freeze

- Create an Input Follower modulator.
- Attach the Follower to the Speed control.
- To freeze only the loudest portions of the incoming signal, set Speed to 100% and use the Follower to create negative modulation on the control.
- To freeze only the quiet portions of the incoming signal, set Speed to 0% and use the Follower to create positive modulation.

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Our families, for putting up with the constant stream of weird sounds coming out of our offices (and random stress over esoteric bugs).

http://www.unfilteredaudio.com