

# Sandman Pro 1.1 Manual

by unfilteredaudio



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## ***Introduction***

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Sandman Pro is the delay of your dreams: a multi-mode delay workstation with unmatched loop-freezing capabilities. Sandman Pro builds on the legacy of Sandman by introducing six new delay modes while refining the original's tape mode. These delays include an instant delay that jumps to new delay times without clicks or pitch artifacts, a flexible stereo pitch shifter with feedback, a 16-tap multi delay, a unique reverse mode, and more.

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## **Features**

- Seven incredible delay modes.
- Visualize your loops: a new waveform display shows your loop's Start/End points, along with current playback position.
- Full stereo: Left and Right channels can have linked or independent delay times. A Spread mode allows you to create subtle or massive offsets between the two. A Stereo Width control adds some contrast to the final output, along with an optional Mid/Side mode.
- Patchable modulation system: Our highly acclaimed modulation system is here with full control over every parameter, including Sleep, Reverse, and Kill.

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# ***Getting Started***

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## **Installing Sandman Pro**

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

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## **Browsing Presets**

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

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## **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.

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# Controls

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## Input

**In/Out Gain:** Controls the amount of gain applied to the incoming and outgoing audio signals. Both gains only affect the Wet signal. In Gain occurs before any processing, while Out Gain occurs after. It can be useful to turn down the In Gain to assist with feedback-heavy patches, while Out Gain can help with final mixing and saturation.

**Mute:** Mute's the input in a click-free manner. This can be very useful if you want to monitor only the current feedback tail. It can also be modulated for rhythmic gating.

**Pan:** Pans the input between the left and right delay channels. To create a traditional ping-pong delay, you can pan the signal hard in one direction and turn up X-Feedback.

**Mono:** Averages a stereo input into dual mono. This only affects the plug-in's input. To make the output mono, look at the "Width" control.

**S. Rate:** A signature feature of Unfiltered Audio's plug-ins. This control determines the sampling rate of the \*entire\* plug-in, not just the input. Changing the sampling rate will affect the shifter frequency, the length of the delay units (when not locked), the behavior of the filters, and more. Needless to say, entire worlds of unusual sounds and artifacts are contained within this knob!

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## Delay Time

**Delay Time:** In regular (Linked) mode, this will appear as one large control. In Independent or Spread modes, this will appear as separate L and R controls. These controls determine the length of the delay buffer.

**Time Mode:** Changes how the Delay Time knobs work.

1. **Linked:** One large delay knob controls the left and right delay times simultaneously.
2. **Independent:** Left and right delay times are set separately.
3. **Spread:** Left sets a "master" delay time. The right delay time is then given as an offset of this. Try Setting a long master delay time with a tiny offset to create Steve Reich-style phasing loops in Sleep mode.

**Tempo Sync:** When active, delay times are locked to rhythmic divisions of your DAW's tempo.

**Lock Time:** When active, changing the sampling rate will not affect the delay time. Turn this on if you want to manipulate the sampling rate for lo-fi sounds, but also want to maintain a steady rhythm.

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## Delay Mode

Sandman Pro contains seven awesome delay modes. We've spent a lot of time ensuring that each delay mode is special in its own way!

### **CLASSIC TAPE**

This mode emulates tape delays, and is the closest in behavior to the original Sandman. The most noticeable aspect of tape delay is slew: there is an audible pitch change when delay times are changed (this is because the read and write heads are moving towards or away from each other, a process that cannot happen instantly).

**Saturate:** This control introduces an amount of saturation to louder signals that are being written to the tape. This controls will affect the frozen buffer in Sleep Mode, so you can record a clean loop and saturate it later.

**Wow:** This control emulates unpredictable fluctuations in tape-based systems. When turned up, this will produce pitch warbling effects as the playback speed loses steadiness. This control will affect the frozen buffer in Sleep Mode, so you can use this to break up static loops in subtle ways.

**Slew:** This determines the length of time it takes for the tape to catch up when delay times are changed. This also affects the amount of slew present when changing the length of the Sleep buffer.

### **MODERN INSTANT**

This mode uses granular windowing to create a delay that can change times instantaneously without pitch or click artifacts. This mode is perfect for heavy modulation, and can be used to create time-stretching effects (seriously, modulate the delay time with a slow sawtooth!).

**Dirt:** This adds a subtle amount of noise to the input. At extreme settings, it introduces the same type of bitcrushing present in Dent's CRISPR mode. This control will affect the frozen buffer in Sleep Mode, so you can record a clean loop and destroy it later.

**Jitter:** This randomizes the delay time with every "loop". This is different from randomizing the delay time with an S/H noise modulator, as the amount of randomization is dependent on the delay time, and the point of randomization is determined after each randomized "loop" is completed.

### **PITCH SHIFTER**

This is a delay-based pitch-shifting algorithm far different from the FFT-based shifter on Fault. This uses a decidedly more vintage approach with modulated micro-delays. With positive pitch shift amounts and feedback, this can produce the classic "shimmering ice cavern" effect. With X-Feedback and unequal pitch shift amounts and delay times... well...

**Pitch L/R:** Determines the amount of pitch shifting present on each channel. This does not affect the Sleep buffer.

**Window Size:** Determines the size of each grain/micro-delay. At small values, this will introduce a metallic effect.

**Link:** When active, the right pitch shift control will always equal the left.

### **GLITCH SHIFTER**

This is a pitch shifting algorithm... implemented entirely incorrectly. This can produce vibrant color explosions, alien drums, swirling echo clusters, and more. Note that changing the delay times in this mode can have a dramatic effect on the sound. Smaller delay times lead to more coherent sounds. Or something.

**Glitch L/R:** \\_(ツ)\_/

**Link:** When active, the right glitch shift control will always equal the left.

### **MULTI-TAP**

This is a 16-tap delay line. Like Modern Instant mode, all controls can be modulated without artifacts. Even the number of active taps can be manipulated rapidly.

**Spacing:** This controls the spacing of the taps. At 12 o'clock, the taps are evenly spaced. To the left, the taps create a bouncing ball effect by grouping later taps more closely. To the right, earlier taps cluster together.

**Amp:** This controls the amplitude curve applied to the individual taps. At 12 o'clock, the taps will all have equal amplitude. To the left, the taps will die-off. To the right, the taps will produce a bloom effect as they increase in volume.

**Taps:** Controls the number of active taps.

**Cluster Fdbk:** When active, \*all\* active taps are sent to the feedback. There's an amplitude compensation algorithm on this mode that prevents it from running away rapidly. When inactive, only the last tap will be sent to feedback. The last tap's position distance from the write head is always equal to the delay time of that channel.

### **REVERSE**

This mode provides a reverse delay effect using windowed grains. A lot of care was taken in avoiding clicks or pops. This may be the cleanest reverse delay algorithm that we've ever heard!

**Dirt:** This is identical to the Dirt control in Modern Instant mode.

**Slew:** This is identical to the Slew control in Classic Tape mode.

## **NO ECHO**

The final delay mode is... not a delay! In this mode, the input is written directly to the Sleep buffer. You can use this mode and modulate the Sleep button to produce crazy stutter and chop effects.

**Dirt:** This is identical to the Dirt control in Modern Instant mode.

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## Sleep Buffer

The Sleep Buffer is Sandman Pro's signature feature. This is a frozen delay buffer with tons of control (instead of a boring old "Freeze" button). When Sleep is activated, Sandman Pro's large waveform display will stop displaying real-time input data and will instead show the contents of the frozen buffer. The waveform display will also show where the loop starts and ends, along with the current position of the read head. Note that most special mode-specific delay controls (such as Pitch Shifter's pitch) will not affect the frozen buffer. For specific details on each delay mode, take a look at the Delay Mode section.

**Sleep:** When engaged, the delay buffer is frozen and looped. No new information will be written to it. The length of the frozen buffer is determined by the Delay Time, Start, and End controls.

**Kill:** Immediately empties the delay buffer. This works regardless of whether or not Sleep is active. This control will not produce a click, so this can be used for interesting rhythmic effects.

**Reverse:** Immediately reverses the playback direction of the frozen buffer.

**Start/End:** Determines the start and end points of the looping, frozen buffer.

**Lock Length:** When active, the playback length of the frozen buffer will always equal the delay time, no matter what the Start and End settings currently are.

**Smooth:** When active, a tiny envelope is used to crossfade the start and end points of the frozen buffer. This will suppress clicks. This effect may not be wanted when loop lengths are extremely small or sampling rates are very low.

**Sleep Filter:** When active, the frozen buffer's output will run through the LP and HP filters.

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## Echoes

**Feedback:** Controls the amount of each delay channel's output that is fed back into its own input.

**X-Feedback:** Controls the amount of each delay channel's output that is fed back into the other channel's input.

**Diffuse:** Smears the delay outputs by running them through an extra-chunky all-pass filter network. At low sampling rates, you can use this as a cluster echo effect.

#### **FILTERS**

Sandman Pro contains two 12 dB/Oct filters: High-Pass and Low-Pass. These filters can dramatically shape the sound of the delay's feedback path.

**LP:** Sets the cutoff frequency of the low-pass filter.

**HP:** Sets the cutoff frequency of the high-pass filter.

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## Output

**Soft Clip:** Turns on soft clipping at the output. The signal is saturated using a parabolic equation. This will actually affect the sound of the entire signal in a subtle way. This clipping is more pleasant and analog-sounding. When soft clipping is off, the signal will hard clip if it exceeds the plug-in's headroom.

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

**M/S:** When active, Sandman Pro will output in Mid/Side mode. This can be used for advanced mixing and processing techniques. If you are unfamiliar with Mid/Side processing, the simplest explanation is that "Mid" can be thought of as the mono or "shared" component of the two channels, while "Side" can be thought of as the stereo or unique components of the two channels. In M/S mode, the first channel output will be Mid, and the second channel output will be Side. You can output M/S signals onto a L/R configuration without negative side effects. It will definitely sound unusual!

**Width:** Controls the perceived width of the output signal. In L/R mode, 0% is mono, 50% (12 o'clock) is stereo, and 100% is super-wide (Note that components of the left and right signals will bleed over into each other). In M/S mode, 0% is mid-only, and 100% is side-only.



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# **Modulation System**

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## **Introduction**

Sandman Pro 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 show 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.

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## 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.

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## 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 primitive wave types which are useful for classic “envelope” style parameter modulation.

The Amplitude knob is common to all of them, corresponding to the depth of modulation. 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 always 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 the speed of modulation. 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 DAW's global tempo setting- for instance setting the frequency to  $\frac{1}{4}$  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 which 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. In this way you can control many knobs 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 (<https://roli.com/products/blocks>) 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).

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## **Options**

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Sandman Pro 1.1 introduces an expanded options menu. The following unique options are available in Sandman Pro:

**Enable Display:** This enables or disables the waveform display at the bottom of Sandman Pro'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.

**Balanced Dry/Wet:** In Sandman Pro 1.0, the Mix control behaved the same way as the Mix control on Fault, Dent, and Indent. At 50% Mix, the Wet and Dry portions of the mix appear at 50% of their regular amplitude. With this option enabled, at 50% Mix, the Wet and Dry portions of the mix appear at 100% of their regular amplitude. If you don't like the amplitude drop that occurs when Sandman Pro is first loaded, you should enable this option.

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## ***Recipes and Ideas***

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### Steve Reichian Phased Loops

- Set Delay Time Mode to Spread.
- Set a long primary (L) delay time. Set R to a very small offset (5-40 ms).
- Run some audio into Sandman Pro and freeze it by activating Sleep.
- Listen to your loops go slowly out of phase with each repetition.

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### Time Stretcher

- Set Delay Mode to Modern Instant.
- Turn off Tempo Sync.
- Set Delay Time to 5-50 ms.
- Connect a rising sawtooth LFO modulator to the Delay Time control.
- Tweak the modulation depth and sawtooth speed until you find a sound you like.

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### Ping-Pong Delay

- Turn on “Mono” mode.
  - Pan the input hard left or hard right (100% in either direction).
  - Turn up X-Feedback.
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## ***Credits and Thanks***

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<http://www.unfilteredaudio.com>