SNOP

SONIC NOSTALGIA OUTPUT PROCESSOR

USER MANUAL
Version 1.0

Thank you for supporting my work by purchasing another one of my passion projects!

SNOP started out as a simple experiment with a few effects, and over time, it grew into something much bigger.

While it wasn't meant to aim for sample accuracy–and I still don't claim it's there–the more machines and junk I collected, the more obsessed I became with fine-tuning the details that tug at my heartstrings. I kept adding more types of vintage sounds that evoke that sweet nostalgia.

Your support means more fuel and funds to create even cooler stuff. So, know that you've made a meaningful contribution.

Thank you for allowing me to geek out on what I love and back the next creator.

And no, there are no Easter eggs in this one! I'll see what I can do for a future update. What's your high score on Vocodine, by the way? ;)

I really hope you enjoy the plugin and find it useful!

Blezz Beats

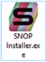
INSTALLATION

1.1 Installing SNOP

SNOP comes with dedicated installers for PC and Mac wrapped in a ZIP-file. The download is the same for both systems, so just make sure you unzip the file first to reveal them.

PC

Unzip SNOP.zip and run the installer called SNOP Installer.exe



Mac

Unzip SNOP.zip and run the installer called SNOP Installer.dmg

A temporary download link is delivered to the email you used on purchase, but remember that you can sign up on blezzbeats.com using the same email address to access it under "My purchases" at any time.

1.2 Requirements

PC

Windows 7 SP1 or later

Mac

macOS 10.13 or later, M1, M2, M3 compatible

64 bit VST3, AAX or AU compatible host software.

1.3 Loading SNOP in your DAW

Using SNOP is very straightforward regardless of which DAW you're using. Simply find Blezz Beats and SNOP in your plugins list and add it as an insert effect on any track.

There is only one requirement - it has to be a **stereo track**.

Some DAWs (such as Ableton Live for example) handle the conversion from mono to stereo automatically, while others will not show SNOP in your list of plugins if you're on a mono track. In some cases you might want to use it anyways, for example if you're using Logic Pro and you're recording vocals on a mono audio track. In that case I suggest you create a stereo bus for this purpose.

1.4 SNOP not showing up in your DAW?

Make sure that you're trying to place it on a stereo track, if it's still not visible try rescanning your plugins.

SNOP is installed in the following locations:

PC
VST3: Program Files/Common Files/VST3
AAX: Program Files/Common Files/Avid/Audio/Plug-Ins

Mac
AU: Macintosh HD/Library/Audio/Components
VST3: Macintosh HD/Library/Audio/VST3
AAX: Macintosh HD/Library/Application Support/Avid/Audio/Plug-Ins

Mac users, note that the Library folder is in the root directory of your hard drive. There is a nearly identical folder in your user directory, for example YOUR USER/Library/Audio and some DAWs may default to searching that directory instead. In that case, add the actual folder before rescanning.

If you're not finding SNOP make sure you have the correct folders added before rescanning your plugins.

2 Overview

Let's get familiar with navigating the interface so you can truly customize your vintage sound.

The interface of SNOP can be divided into 4 sections:

1. Top panel
2. Machines and input section
3. Module section
4. Output section

Each section will be explained in detail in the following pages.

TOP PANEL

3.1 Parameters and Items

The top panel is a simple interface that contains 3 essential parts:

1. Stereo input selection
If you are an avid sampler, you know the importance of being able to choose between stereo, left, right, or left and right merged. This also serves as a means to more accurately emulate certain devices that can only handle a mono input, but I've left the power of choice to you.

2. Master mix
The master mix serves as a macro control for all parameters apart from the Machine section and the bandwidth of each module.

0% = No effect
100% = Full effect

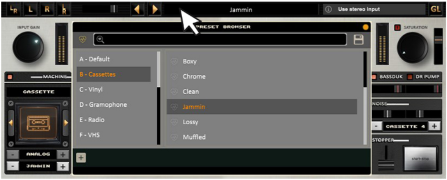
3. Preset browser
You probably know what a preset browser is. Use the left and right buttons to browse quickly between presets or click the preset name to bring up the preset browser. The presets are a useful way to quickly achieve a certain sound, not only because that's what presets do, but also to save you a step since you can combine different noises with different machine emulations. If you want a profoundly cassetty cassette sound for example, you won't need to first select a cassette and then select a cassette noise.

We will explore the preset browser further on the next page.

The top-right displays a tooltip for any component you hover on, just in case you want to know what it does.

3.2 Preset browser

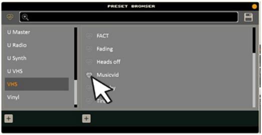
To get familiar with SNOP, it's a good idea to start by browsing some of the presets.



You can quick-browse the presets by clicking the left and right arrows in the top bar, or open up the preset browser by clicking on the title of the preset.

Banks are located in the left column, and presets in the right column. Click the names to load them.

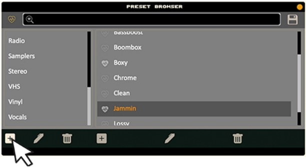
Click the little heart-icon to add it to your favorites, and search all presets using the search bar.



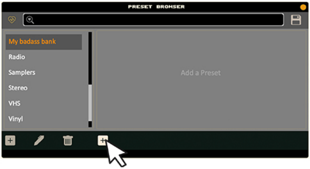
To list all favorited presets, click the golden heart icon to the left of the search bar.

3.3 Saving presets

The factory banks are write protected and can not be saved to. For this reason, you first need to create a bank to save your preset to:

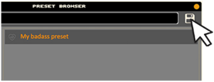


Click the plus icon to your bottom left to create a new bank, then click the plus icon to the right to save your preset.



You can rename it by clicking the pencil or delete it by clicking the trash can.

The floppy disk icon saves changes that have been made to your preset:



MACHINES

4.1 What's going on in there?

A machine emulation that affects the signal pre, mid, and post. This means it passes the audio signal through several stages depending on the selected machine. Subsequently, the input gain affects the sound of the final output, any activated module will color the sound differently, and the sound will also be manipulated at the final stage before reaching the output.

If a digital device such as a sampler is chosen, this may include things such as sampling rate, bit depth, DC quantization method, interpolation, pre and post-filtering (anti-alias filtering), noise floor and fluctuations, dithering, and frequency response by convolution.

These emulations are not 1:1 recreations of the devices they are inspired by, so if you bring out your spectrum analyzers and measuring equipment you will find discrepancies in the frequency responses, etc. But, it will be difficult for most people to tell the difference unless you listen to them side by side.

Analog devices have different processing steps to emulate the beautiful flaws of mediums such as cassette tapes, VHS tapes, or records in a non-linear fashion. Examples of the methods used are wow, flutter, bias, azimuth shifts and other phase-related errors, noises of different kinds, filters, dropouts, tape saturation, filters, stereo imaging, and also a set of convoluted impulse responses from various devices to capture different tonalities and timbres.

Rather than filling the screen by exposing 300 parameters, I've aimed to capture the soul of both the artifacts produced by the medium itself and by the device used for playback in these machine presets so each one behaves a bit differently.

Each machine has different settings that can be chosen depending on the context of the machine preset.

This section will **NOT** be affected by the main mix if activated, the reason for this is mainly to avoid unwanted phase issues with IR convolution.

4.2 Selecting a machine emulation

Selecting a machine emulation can be done in two ways. Either cycle through them using the arrows on the side of the little CRT display, or click the middle of it to bring up the machine selection list:

4.3 Analog machines

Cassette

Emulates the artifacts and frequency responses of different cassette tapes in different players.

Control 1: N/A (Analog)
Control 2: Select cassette and player

Vinyl

Emulates the artifacts and frequency responses of different vinyl records in varying shape.

Control 1: N/A (Analog)
Control 2: Select vinyl record and player

Gramophone

Emulates different gramophone records, including some non-existing lined hybrids.

Control 1: N/A (Analog)
Control 2: Select gramophone record/player

Radio

Emulates different radios, from decks to handheld pocket radios.

Control 1: N/A (Analog)
Control 2: Select radio

VHS

Emulates different VHS tapes and players in different shapes.

Control 1: N/A (Analog)
Control 2: Select VHS tape/machine

4.4 Digital machines

Each machine in SNOP has pre- and post-filtering, bit rate reduction, noise floors, DC quantization etc. For example the SB-12 has a fixed sample rate of around 26kHz, 12-bit bit depth, an impulse response based anti alias filter, several up and down-sampling steps and sample and holds and more.

Most of the digital machine emulations on SNOP have two settings in common as opposed to the analog machines where you simply select a medium. Those are the sample rate and pitch shift:

Control 1: Sample rate
On the machine emulations which have variable sample rates, this can be used to define which input sample rate should be simulated. Some machines have fixed sample rates, in which case this setting is disabled.

Control 2: Pitch shifting
In SNOP, pitch shifting **does not actually mean shifting the pitch of the sound**, but rather reproducing the damage done to the audio if it were sampled and tuned to the defined setting. For example, if your source audio is playing at its original pitch it would sound something like if you sampled it at a higher speed/pitch and tuned it back down to its original pitch. To pitch it AND get the artifacts, simply repitch your source audio.

Drop sample pitch shifting

Drop sample pitch shifting is probably the simplest and most violent way of shifting pitch digitally. In practice on an actual sampler with a recorded audio sample, it works by reading the buffer of samples at specific intervals depending on which pitch is set. For example, to pitch a sample down it will repeat some of the samples, and to pitch it up it will skip some of the samples. This causes artifacts such as aliasing and imaging because the sampled waveform is essentially not reproduced smoothly.

To get around this, slightly more advanced samplers would use interpolation to calculate the in-between steps and smooth out the soundwave in the best way possible. The complexity of interpolation algorhithms can vary greatly.

Variable clock speed pitch shifting

Another "primitive" way of manipulating the pitch and playback speed of a sample is by changing the interrupt speed of the clock which updates the sample values, simply put.

In SNOP, the methods vary between emulated machines. Since we're not dealing with pre-recorded buffers of audio but "live" input I had to use a plethora of effects including up/down/resampling to match the sample rates and sample dropouts as closely as possible, simulating the pre and post filtering and recording impulse responses from the original machines to get the frequency responses closer to the real things. Again, the output is not identical but should be close enough to spare you the expense of buying these vintage gems for now. It hurt.

There are a couple of exceptions such as the MPSEE 60 which has a fixed sample rate and instead lets you control whether the audio is played back at the original input sample rate, or compressed by sampling the audio 6 or 12 semitones higher and pitching it back down (or using the compression function in newer OS versions). The ESS950 lets you control the input bandwidth/sample rate, and the output lowpass filter.

4.4 Digital machines

A600

Emulating the sound of a classic personal computer with a standard 8 bit parallel port sound sampler, without filter.

Control 1: Select input sample rate
Control 2: Select clock speed variation pitch shift rate

A600 (filtered)

Same as the above, but with an output low pass filter captured from the machine.

Control 1: Select sample rate
Control 2: Select clock speed variation pitch shift rate

PSX ADPCM

The amount of time I spent on this was rather pointless. But now you can create bad voice overs that sound like Resident Evil. Some games would compress the sound even further, so add some Decimator to it to make the effect more dramatic.
Control 1: Select sample rate
Control 2: N/A

Game Man

You can probably guess what this is trying to emulate. Sampled audio is not typical for Game Boy games, but thanks to brilliant software like LSDJ by Johan Kotlinski we're able to move ultra lofi samples to the Game Boy. It's quite the process though, so you could just cheat and use this.
Control 1: N/A
Control 2: Select full or half speed

SB-12

Emulates the AD/DA, anti-aliasing (didn't really get that right), and drop sample pitch shift of a legendary drum sampler. Keep in mind again that it doesn't change the pitch, so to accurately approximate the sound you should pitch the source audio by the same amount of semitones. These machines were also mono, so switch the L, R, or combined LR switches on for more authenticity..

Control 1: N/A
Control 2: Control 2: Select drop sample pitch shift rate

4.4 Digital machines

MPSEE 60

Emulates the classic sequencer/sampler with 3 different modes. Original mode emulates the sound when it's sampled straight into the machine and played back, compressed mode emulates pitching the sample up by 6 or 12 semitones and pitching it back down to original speed, or using the compress function on newer OS versions.

Control 1: Original/Compressed
Control 2: -60 or -120 cents in compressed mode

ESS950

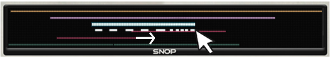
Emulates everyone's favorite 12-bit rack sampler. The first parameter controls the bandwidth (effectively the sample rate), which doesn't have as steep of an anti-aliasing filter as the original machine as you will hear in the lower rates. The second parameter controls the value of the low pass filter and its profile.

Control 1: Select sample rate
Control 2: Select filter value

MODULES

5.1 Modules Introduction

SNOP has 6 modules that can be enabled or disabled by you. The frequency range of each module can be set either with the knobs or by clicking and dragging the display when the currently selected module is active:



5.2 Selecting and activating modules

Clicking a module will highlight it and set the current edit mode on the display to it. If you click the Bandpass module for example, clicking and dragging the display will set the frequency range and edge level:

To activate or deactivate the effect, click the small button on top of the module:

5.3 Decimate

Decimate is a resampling effect with two upsampled zero-order holds to precisely dial in the type of aliasing you want. You will notice that it differs from the other modules in that it has two bands on the display:

This means you can crush your audio at different rates on different bands, fun.

Parameters

Low
0-100%

The amount of "sample rate reduction" from 20 Hz and up until the currently set low band cutoff, in other words up until the frequency set for High.

High
0-100%

The amount of "sample rate reduction" from the currently set low band cutoff and up until the currently set high band cutoff.

Finetune
-100%-100%

Fine-tunable resampling rate of the High part of the selected frequency band.

5.4 Wobble 1 & 2

Used to create wow, warble, flutter, and other types of pitch modulations on the selected band.

PRO TIP:
Different frequency bands in combination with different wobble speeds will require different depths before you really start to hear the effect.

Low selected frequency range and low wobble speed - Less depth required before effect is audible.
High selected frequency range and low wobble speed - Less depth required before effect is audible.
Low selected frequency range and high wobble speed - More depth required before effect is audible.
Etc.

It's a good idea to crank up the depth as you're tuning the wobble speed, and dial it back to a sweetspot when you're satisfied.

Parameters

Depth
0-100%

How deep the oscillator modulating the delay line causing the pitch shift should go, in other words how pronounced the effect is.

Speed
0.15-15 Hz

The speed of the oscillator - how fast the pitch oscillations are.

Jitter
0-100%

How much jitter/variations are in the speed of the modulation. Most noticable with higher depth values.

5.5 Delay

A feedback delay which can be used to create everything from regular delays on a specific bandwidth to dub echoes, phase distortions, or even old school real time time-stretch effects.

PRO TIP #1:
Activate the delay with full mix and set the bandpass within the same frequency range to use it on a send channel as a dedicated delay effect, with the advantage of all the other colorations that SNOP brings.

The presets in the "Delay" category are set up in this way to demonstrate this principle.

PRO TIP #2:
Activate the delay with full mix and automate the speed. Going from quick to slow will create an old school time stretch effect, very cool when its limited to a frequency range.

Parameters

Speed
10-1000ms (1 second)

How deep the oscillator modulating the delay line causing the pitch shift should go, in other words how pronounced the effect is.

Feedback
0-100%

How many times the delayed signal is repeated, and its decay envelope.

Mix
0-100%

A dry/wet mix between the original signal and the delayed signal within the selected band.

5.6 Bandpass

Bandpass can be used both as a regular bandpass filter and an EQ of sorts. It has steep low- and high frequency cutoffs so pulling the edge level all the way down will completely isolate the selected band, but you can also keep a bit of the low and high spectrum if you'd like.

There is also a self-modulating notch filter that acts on the selected frequency range, controlled by the threshold.

Parameters

Edge level
-100dB-0dB

The volume of the bands below and above the selected frequency range.

Notch filter
The volume envelope of the selected frequency range is always measured and used as the trigger for a notch filter which bounces up and down starting from the selected lower cutoff.
The threshold is the minimum volume required for the modulation to trigger, and Notch is a combined morph of the Q and gain of the notch filter. This effect can be used to create cool self-modulating effects like auto-wahs, the animated lowpass-filtered samples of old school hip hop akin to the works of Pete Rock, or wherever your imagination takes you.

Keep in mind that the notch filter is at work even when Edge level is full, so you **don't necessarily need to bandpass your audio/pull the edge level down to use this effect**.

Threshold
-30dB-0dB

Sets the minimum volume that needs to be reached in the selected frequency range to trigger the notch filter modulation. The light indicator to the right of the knob helps you discover the sweet spot for the currently playing audio.

Notch peak
0-15dB

Sets the gain of the notch, and compensates equally by attenuating the whole signal.

5.7 Stereo

The stereo module allows you to modify the width and modulation of the stereo image on the selected band. Used moderately it can be utilized for tastefully adjusting the stereo placement of the entire image, abused it can be used to misplace it in psychadelic pseudo-3D realms and breaking traditional phase rules.

Because of how the nature of stereo perception works, the results will differ quite dramatically depending on if you're using a mono or stereo input, which frequency range you've marked, and the frequency range of your audio. Applying more than one of the 3 stereo effects at once will start bending reality, or just make it sound bad in a shitty way.

PRO TIP:
If your input signal is mono, wideness won't have any effect until it passes 150%.

Parameters

Wideness
0-200%

0%: Mono
200%: Ultra-wide and misplacing
The Wideness parameter uses a combination of techniques moving from M/S level adjustments into Haas-effect phase misplacement in the extreme values.

Phaser
0-100%

How much phaser do you want? Your choice.

Chorus
0-100%

The Chorus knob changes the depths, rates, and mix of the chorus.

OUTPUT SECTION

Output section

The output section is great for adding some final touches to your sound. **Control the output gain with the orange slider.**
Every processor here is not necessarily the last in the chain internally, but it makes sense to think of it as the last stop when tweaking.

Saturation

Two modes can be used for saturating the signal, and you can toggle between them using the small button next to the knob.

Off

When the button is toggled off, the saturation behaves more like the tape compression that happens when you drive the volume into a cassette deck. It's a soft clipping form of saturation that moves from slight to heavy compression-like processing and eventually a "kind" type of distortion.

On (careful!)

Toggling the button on, there is a more drastic saturation of the signal that goes into overdrive and clipping on higher levels.

Knob

0-60dB

0 dB keeps the signal at the input volume coming from the previous processing. Keep in mind that a lot of things including compression, convolution, and gain compensation can happen along the way, so the signal will not necessarily be balanced with the original input volume.

Push this value up to start hearing more tape compression, saturation or distortion.

OUTPUT SECTION

Bassduk

I invented this effect, but most likely so did plenty of other people, so there might be a proper name for it.

It's a 2-band frequency split with an envelope follower on the low band which controls the volume of the high band.

When the low end of your audio signal exceeds the set threshold, the high end will bow down to it.

This can be used to create an effect reminiscent of sidechaining on a single audio signal, or to emphasize the effects of a busted analog medium where the bass is overbearing.

Threshold
-100-0dB

Defines the limit of the low band at which the high band should start ducking.

Dr. Pump

Dr. Pump is a single-parameter combined transient designer and expander. It works great on drum breaks where you want to remove some reverberation and add extra punch, and also on AI-generated drum loops with ugly artifacts that need removing (fun fact).

The analysis and processing happens early in the chain to make the most use of it, but there are a few things to keep in mind:

As with any basic transient shaper, mid/side analysis is a key component. Therefore, it's beneficial to input the audio in stereo if possible.

Very busy audio where the frequency spectrum is packed and saturated makes peak-picking more difficult, and the effect less pronounced.

If the signal is driven, saturated, or overly compressed after the effect has been applied, you might begin returning to square one since the attenuated parts will be amplified again.

Amount
0-100%

More amount means sharper transients and attacks, with the spaces in between being attenuated and mild saturation of the signal.

OUTPUT SECTION

Noise

SNOP has a wide selection of lovingly recorded noises from different mediums and devices that can be used to sprinkle on top of your audio, or rather merge with it. Each noise has its own tuning which affects how it interacts with the original audio signal in a certain way, from dynamically attenuating the volume to cutting the high end or just simmering on top, however the noise would affect the signal in real life.

Select a noise by using the arrow buttons next to the noise display, or click the display itself to bring up the overviewing menu of noises:

Control the pitch and playback speed with the left fader, useful for making slight adjustments to suit your soundscape better or purposefully making weird sounds.

Control the volume using the right fader. Increasing the volume may also increase the interaction with the original sound.

Stopper

What would a plugin like SNOP be if it didn't have the popular "tape stop" effect? Not sure why it's called that in general, no tape I ever stopped has sounded like that. But it slows down and sounds cool just the same.

Speed

0.6-2 seconds

Defines how long it takes from the time you press the button until it slows down and comes to a full stop.

Start/Stop

Toggles the stopping effect.