

S1988

Vintage Sampler Emulation

Time-Stretch Processor & Looper



RAVE *generation*

USER MANUAL

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1. Introduction

S1988 is a vintage sampler emulation plugin inspired by the legendary AKAI S1000 hardware sampler from the late 1980s. This plugin captures the distinctive character of early digital sampling while adding modern creative capabilities including granular time-stretching, resonant filtering, and lo-fi degradation effects.

The AKAI S1000 was renowned for its warm, slightly gritty character that came from its 16-bit converters, analog filters, and the inherent limitations of early digital audio. S1988 faithfully recreates these characteristics while providing intuitive controls for everything from subtle vintage coloring to extreme sound mangling.

Whether you're adding authentic retro character to drums and samples, creating experimental soundscapes with time-stretching, or designing lo-fi textures, S1988 delivers that coveted late-80s sampler sound with modern flexibility.

2. Key Features

2.1 Vintage Sampler Character

- 18 dB/octave resonant low-pass filter modeled after classic sampler topology
- 18 dB/octave high-pass filter for frequency shaping
- Warm preamp saturation with musical gain compensation
- Filter drive for additional harmonic content

2.2 Lo-Fi Processing

- Variable sample rate reduction (4000 Hz - 44100 Hz)
- Bit depth reduction (4-bit to 16-bit) for authentic digital degradation
- DC blocking to maintain clean output even with extreme settings

2.3 Time Stretch Engine

- Granular time-stretching with variable grain size (2-200 ms)
- Time factor from 25% to 400% for dramatic tempo manipulation
- Optional pitch compensation to maintain original pitch during stretching
- Transpose control (± 24 semitones) for pitch shifting
- Reverse playback mode for creative effects

2.4 Flexible Output

- Equal-power dry/wet mix for parallel processing
- Auto-gain compensation to maintain consistent output levels
- Mono/Stereo processing modes
- Zero latency operation

3. User Interface

The S1988 interface is organized into logical sections that follow the signal flow. Controls are grouped by function for intuitive operation:

- Input Section: Input level and preamp controls
- Filter Section: High-pass, low-pass filters with resonance and drive
- Lo-Fi Section: Sample rate and bit depth reduction
- Time Stretch Section: Granular time manipulation controls
- Output Section: Mix, output level, and processing mode

4. Input Section

The input section provides gain staging and saturation controls that shape the signal before filtering.

Parameter	Range	Default	Description
Input	-20 to +20 dB	0 dB	Input gain control. Sets the level entering the processor.
Preamp	0 - 100%	0%	Warm saturation stage. Adds harmonic content and soft clipping with automatic gain compensation.
Auto Gain	Off / On	Off	Automatic output level compensation based on preamp, filter drive, and resonance settings.

5. Filter Section

The filter section provides S1000-style 18 dB/octave filters with resonance and drive capabilities. These filters respond immediately to parameter changes for dynamic, responsive filtering.

5.1 Filter Parameters

Parameter	Range	Default	Description
HP Filter	20 - 2000 Hz	20 Hz	High-pass filter cutoff frequency. 18 dB/octave slope removes low frequencies.
LP Filter	200 - 20000 Hz	20000 Hz	Low-pass filter cutoff frequency. 18 dB/octave slope with resonance.
Resonance	0 - 100%	0%	Filter resonance (Q). Higher values create emphasis at the cutoff frequency.
Filter Drive	0 - 100%	0%	Post-filter saturation. Adds grit and edge to the filtered signal.

5.2 Filter Topology

The S1988 uses cascaded filter stages to achieve its 18 dB/octave slopes:

- Low-Pass Filter: 12 dB biquad with resonance + 6 dB one-pole cascade
- High-Pass Filter: 6 dB one-pole + 12 dB biquad cascade

This topology matches the behavior of classic hardware samplers while maintaining stability at high resonance settings.

6. Lo-Fi Section

The Lo-Fi section recreates the digital degradation characteristics of vintage samplers with variable sample rate and bit depth reduction.

Parameter	Range	Default	Description
Sample Rate	4000 - 44100 Hz	44100 Hz	Reduces the effective sample rate. Lower values create aliasing and digital artifacts.
Bit Depth	4 - 16 bits	16 bits	Reduces bit resolution. Lower values create quantization noise and stepped waveforms.

6.1 Classic Settings Reference

Hardware Era	Sample Rate	Bit Depth	Character
AKAI S1000 (1988)	44100 Hz	16-bit	Clean, warm
Early Samplers (1985)	22050 Hz	12-bit	Gritty, classic
8-bit Era (1980s)	11025 Hz	8-bit	Crunchy, lo-fi
Extreme Lo-Fi	4000 Hz	4-bit	Destroyed

7. Time Stretch Section

The time stretch engine uses granular synthesis to manipulate time and pitch independently. This section enables creative effects from subtle time adjustments to extreme sonic manipulation.

Parameter	Range	Default	Description
Stretch	Off / On	Off	Enables the time stretch engine. When off, audio passes through normally.
Time Factor	25 - 400%	100%	Time stretch ratio. 50% = half speed, 200% = double speed.
Grain Size	2 - 200 ms	50 ms	Size of granular windows. Smaller = smoother transients, larger = more artifacts.
Pitch Comp	Off / On	On	Pitch compensation. When on, maintains original pitch during time stretching.
Transpose	-24 to +24 st	0 st	Pitch shift in semitones. Independent of time factor.
Reverse	Off / On	Off	Reverses playback direction through the grain buffer.

7.1 Grain Size Guidelines

- 2-10 ms: Very smooth, minimal artifacts, best for subtle stretching
- 20-50 ms: Balanced, good for most material
- 50-100 ms: Audible granular artifacts, creative textures
- 100-200 ms: Extreme granular effects, rhythmic stuttering

7.2 Important: Time Stretch Buffer & Rendering Workflow

Understanding the Real-Time Buffer

When the Stretch feature is enabled, S1988 uses a real-time granular buffer of approximately 7 seconds. This means:

- Audio passes through a circular buffer before being processed by the time-stretch engine
- There is inherent latency introduced by this buffer
- The plugin processes audio in real-time, which differs from offline/rendered processing

Initial Buffer Warm-Up

Important: When you first enable the Stretch function, the granular buffer needs a moment to fill with audio data. During this initial period, the output may sound incomplete or glitchy.

To get optimal sound quickly:

1. Enable the **Stretch** button
2. Let audio play for a few seconds to fill the buffer
3. **Move the Time Factor and Grain Size knobs slightly** - this helps the granular engine settle into a stable state
4. The sound will soon become smooth and consistent

This is normal behavior for real-time granular processing and only occurs when first activating the stretch engine or after significant parameter changes.

Why Rendering Is Required

Because S1988's time stretch operates in real-time on a live buffer, the processed audio must be **rendered to a new audio track** (also called "bouncing in place", "committing", or "freezing") to:

1. **Lock in the timing** - Ensure the stretched audio aligns correctly with your project timeline
2. **Capture the exact processing** - Commit the granular artifacts, pitch compensation, and reverse effects permanently
3. **Free up CPU resources** - Remove the real-time processing load once you're happy with the result
4. **Enable further editing** - Work with the stretched audio as a standard audio file

Recommended Workflow

1. Set up your time stretch parameters (Time Factor, Grain Size, Pitch Comp, Transpose, Reverse)
2. Adjust until you achieve the desired sound
3. **Render/Bounce the track to a new audio track** using your DAW's built-in function
4. Disable or remove S1988 from the original track once rendered
5. Continue mixing with the rendered audio

DAW-Specific Render Commands

Below is a quick reference for rendering/bouncing audio in place across popular DAWs:

Studio One

Function	Access Method	Shortcut
Transform to Rendered Audio	Right-click track → Transform to Rendered Audio	(Custom)
Bounce to New Track	Event menu → Bounce to New Track	Ctrl+Alt+B / Cmd+Option+B

Ableton Live

Function	Access Method	Shortcut
Freeze Track	Right-click track → Freeze Track	Ctrl+Alt+Shift+F
Bounce Track in Place	Right-click track → Bounce Track in Place	---
Bounce to New Track	Right-click clip → Bounce to New Track	---

Logic Pro

Function	Access Method	Shortcut
Bounce Regions in Place	File → Bounce → Regions in Place	Control+B
Bounce Track in Place	File → Bounce → Track in Place	Control+Cmd+B
Freeze Track	Right-click track → Freeze Track	---

Pro Tools

Function	Access Method	Shortcut
Freeze Track	Click snowflake button / Right-click → Freeze	---
Commit Track	Right-click track → Commit	Alt+Shift+C
Track Bounce	Track menu → Bounce	Shift+Alt+Ctrl+B

Cubase / Nuendo

Function	Access Method	Shortcut
Render in Place	Edit → Render in Place	(Custom)
Bounce Selection	Audio → Bounce Selection	(Custom)
Freeze Track	Right-click → Freeze Instrument Track	---

FL Studio

Function	Access Method	Shortcut
Consolidate	Playlist → Tools → Consolidate Tracks	Ctrl+Alt+C
Render to Audio	Right-click mixer track → Render to Audio	---
Export as WAV	File → Export → WAV	Ctrl+R

Note: FL Studio does not have a native "Freeze" function. Use Consolidate to create a rendered audio version of your track.

Reaper

Function	Access Method	Shortcut
Render/Freeze Track	Track → Render/Freeze tracks	---
Glue Items	Item → Glue items	---
Render to File	File → Render	Ctrl+Alt+R

Bitwig Studio

Function	Access Method	Shortcut
Bounce in Place	Right-click clip → Bounce in Place	Shift+B
Bounce	Right-click clip → Bounce	Ctrl+B / Cmd+B
Freeze Track	Right-click track → Freeze	---

8. Loop Section

The loop section enables real-time audio looping with tempo synchronization capabilities. When enabled, incoming audio is captured and played back in a continuous loop, allowing you to freeze and manipulate audio material.

Parameter	Range	Default	Description
Loop	Off / On	Off	Enables loop capture and playback. When turned on, audio is captured and looped continuously.
Loop Sync	Free / 1/1 / 1/2 / 1/4 / 1/8 / 1/16 / 1/32 / 1/64 / 1/128	Free	Loop timing mode. Free uses milliseconds; other values sync to host tempo using note divisions.
Loop Length	10 - 4000 ms	500 ms	Loop duration in milliseconds when Loop Sync is set to Free. Ignored when using tempo-synced divisions.

8.1 Loop Behavior

When Loop is enabled, the plugin immediately begins capturing incoming audio. Once the capture buffer reaches the specified loop length, playback begins automatically. The loop continues to play until Loop is turned off.

The loop is processed BEFORE the time stretch engine, which means you can apply time stretching, pitch shifting, and reverse effects to the looped audio in real-time. This creates powerful sound design possibilities.

8.2 Tempo Sync

When Loop Sync is set to a note division (1/1 through 1/128), the loop length is calculated from the host DAW tempo. For example, at 120 BPM with Loop Sync set to 1/4, the loop will be exactly one beat (500 ms). This ensures your loops stay perfectly synchronized with your project.

9. Output Section

The output section provides final gain staging and processing mode options.

Parameter	Range	Default	Description
Mix	0 - 100%	100%	Dry/wet blend using equal-power crossfade. 0% = dry only, 100% = wet only.
Output	-20 to +20 dB	0 dB	Final output level with safety limiting to ± 1.0 .
Processing	Mono / Stereo	Stereo	Processing mode. Mono sums L+R before output.

10. Signal Flow

Understanding the signal flow helps optimize your settings and troubleshoot any issues:



10.1 Key Signal Flow Notes

- Loop is processed after time stretch, so loop changes take effect immediately
- Time stretch is processed first, so filters respond to stretched audio
- Preamp saturation feeds into filters, creating harmonic interactions
- Filter drive is post-filter for raw edge on clean-filtered signal
- Lo-fi processing occurs after filtering to preserve filter character
- DC blocker removes offset from bit crushing artifacts
- Mix blending uses equal-power curves for musical parallel processing

11. Complete Parameter Reference

This section provides a complete reference of all automatable parameters in S1988.

Parameter	Range	Default	Notes
Input	-20 to +20 dB	0 dB	Input gain
Preamp	0 - 100%	0%	Warm saturation
Auto Gain	Off / On	Off	Level compensation
HP Filter	20 - 2000 Hz	20 Hz	18 dB/oct HPF
LP Filter	200 - 20000 Hz	20000 Hz	18 dB/oct LPF
Resonance	0 - 100%	0%	Filter Q
Filter Drive	0 - 100%	0%	Post-filter saturation
Sample Rate	4000 - 44100 Hz	44100 Hz	Rate reduction
Bit Depth	4 - 16 bits	16 bits	Bit reduction
Stretch	Off / On	Off	Enable stretch
Time Factor	25 - 400%	100%	Stretch ratio
Grain Size	2 - 200 ms	50 ms	Granular window
Pitch Comp	Off / On	On	Maintain pitch
Transpose	-24 to +24 st	0 st	Pitch shift
Reverse	Off / On	Off	Reverse playback
Loop	Off / On	Off	Enable loop
Loop Sync	Free / 1/1 - 1/128	Free	Loop sync mode
Loop Length	10 - 4000 ms	500 ms	Loop duration (Free mode)
Mix	0 - 100%	100%	Dry/wet blend
Output	-20 to +20 dB	0 dB	Output level
Processing	Mono / Stereo	Stereo	Channel mode

12. Tips & Tricks

11.1 Authentic Vintage Sampler Sound

- Set Sample Rate to 22050 Hz and Bit Depth to 12 for classic late-80s character
- Add subtle Preamp (10-20%) for warmth without obvious distortion
- Roll off highs with LP Filter around 12-15 kHz for that "sampled" quality

11.2 Filter Techniques

- High Resonance (70-100%): Creates acid-style squelchy filter sweeps
- Filter Drive with Low LP: Aggressive bass tones with controlled highs
- HP + LP together: Create bandpass effects by bringing both filters toward center

11.3 Creative Time Stretch

- Extreme slow (25-50%): Creates ambient pads from any source material
- Large grain (100-200ms) + Reverse: Granular textures with reversed grains
- Pitch Comp Off: Classic varispeed-style pitch/time relationship
- Transpose without Stretch: Quick pitch shifting for harmonies

11.4 Lo-Fi Drums

- Bit Depth 8-10 bits adds crunch while maintaining punch
- Sample Rate around 16-22 kHz removes harsh transients naturally
- Use Mix at 50-70% to blend lo-fi character with clean attack

11.5 Gain Staging

- Enable Auto Gain when using heavy Preamp or Resonance settings
- Watch output levels when stacking Preamp + Filter Drive
- Use negative Output gain to compensate for boost from saturation

Support

For technical support, updates, and additional information:

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