Rave Generation

PRESENTS

FILTERBANK

USER MANUAL

1.0 Introduction

Welcome to Rave Generation: Filterbank. This state-of-the-art audio plugin brings your tracks to life with advanced analog filter modeling, sophisticated modulation systems, and authentic distortion characteristics. Designed for producers, sound designers, and performers who demand exceptional sound quality and creative flexibility, Filterbank delivers the warmth and character of legendary analog hardware within a precision-engineered digital environment.

Filterbank is the result of extensive research into analog circuit behavior, capturing the subtle non-linearities, component interactions, and distinctive characteristics that make hardware filters so coveted. Whether you're crafting experimental soundscapes, adding movement to static sounds, or bringing aggressive character to your mix, Filterbank provides the tools to elevate your productions with authentic analog warmth and character.

1.1 Key features

Rave Generation: Filterbank offers a comprehensive suite of features that empower you to shape sound with unprecedented detail and character:

• **Advanced analog modeling**: Features meticulously designed State Variable Filter architecture with non-linear feedback paths, capacitor simulation, and component interaction modeling for authentic analog sound.
• **Dual filter system**: Two independent filters with lowpass, bandpass, and highpass modes, each enhanced with Moog-inspired resonance feedback and precise blending capabilities. The filters can be arranged in parallel or serial configurations for maximum flexibility.
• **Multi-stage distortion**: Introduces genuine analog warmth or aggressive harmonic enhancement through multiple distortion algorithms including asymmetric processing, DC offset management, and dynamic component saturation. Includes four Neural modes powered by LSTM machine learning, capturing the exact saturation character of real analog filterbank hardware, Rat, DS-1, and TS9.
• **Modulation matrix**: Integrates LFO, ADSR, and AR envelope generators with multiple trigger sources (audio, sidechain, MIDI) for evolving, dynamic control over all filter parameters.
• **Advanced envelope system**: Features Attack-Release (AR) amplitude control and Attack-Decay-Sustain-Release (ADSR) modulation with analog-inspired timing characteristics and multiple trigger modes including envelope follower functionality.
• **Thermal drift simulation**: Employs subtle modulation of harmonic distortion characteristics to recreate the "breathing" quality of analog hardware, adding organic variation to the sound over time.
• **Precision pitch tracking**: Dynamically adjusts filter frequencies based on input pitch detection, creating musical filter responses that complement the source material.
• **High-quality signal processing**: Employs 4x oversampling with sophisticated anti-aliasing for pristine audio quality, even with extreme settings.
• **MIDI integration**: Full MIDI CC compatibility with specialized note triggering system for live performance and sequenced modulation.

2.0 User interface

The interface of Rave Generation: Filterbank provides intuitive access to its powerful features. The layout follows the signal flow from input through filters to output, with modulation sources clearly organized for both creative experimentation and precise control.

2.1 Top row controls

INPUT: Controls the input gain and drive into the filter system. Higher settings introduce analog-style saturation and harmonic enhancement at the input stage, significantly affecting the overall character of the processing.

FM (Frequency Modulation): Adjusts the amount of frequency modulation applied to both filters, allowing for complex timbral changes, sideband generation, and aggressive sonic transformations. The small triangle next to the FM label is a clickable toggle that switches the modulation source between **Internal** (the plugin's input signal, post-overdrive) and **Sidechain** (an external signal routed via the sidechain input). This mirrors the hardware FB2's FM IN jack - by default FM uses the internal signal; flip the triangle to use a sidechain source as the FM modulator instead.

ADSR controls: The central ADSR section includes:

> **A** (Attack): Controls the attack phase of the ADSR envelope, determining how quickly the modulation rises when triggered.
> **D** (Decay): Sets the time for the ADSR envelope to fall from peak to sustain level after the attack phase completes.
> **S** (Sustain): Determines the held level of the ADSR envelope while the trigger remains active.
> **R** (Release): Controls how long the ADSR envelope takes to return to zero after the trigger ends.

LFO RATE: Controls the frequency of the LFO from extremely slow modulation through typical LFO rates to audio-rate modulation, enabling both subtle movement and FM-like effects.

LFO DEPTH: A bipolar control for LFO intensity. Center position provides no modulation, while turning right or left increases the modulation depth in different phase relationships, allowing complementary modulation of both filters.

AM (Amplitude modulation): Controls the depth of amplitude modulation, creating tremolo, ring-modulation, and gating effects depending on settings. The small triangle next to the AM label is a clickable toggle that switches the modulator source between **Internal** (Filter B's output, meaning RESO B strongly shapes the modulator character) and **Sidechain** (an external signal routed via the sidechain input). This mirrors the hardware FB2's AM IN jack behavior.

PAR<>SER: Controls the routing of the dual filter system. Fully left (PAR) processes audio through both filters in parallel, while fully right (SER) routes the output of Filter A into Filter B for serial processing. Intermediate settings blend between these configurations.

DRY<>WET: Controls the balance between the bypassed (dry) and processed (wet) signals. At center position, both signals are mixed equally, while fully left bypasses all processing, and fully right applies 100% effect.

2.2 Switch controls

HI BOOST/HI CUT/NORMAL: A three-position switch that controls pre-filter equalization. HI BOOST enhances high frequencies before filtering for added brightness, NORMAL maintains a neutral frequency response, and HI CUT reduces high frequencies for a warmer character.

ENV FLW/NORMAL/SOFT: A three-position switch controlling the trigger sensitivity for both AR and ADSR envelopes:

ENV FLW: Increases trigger sensitivity for better response to quiet sources
NORMAL: Standard trigger sensitivity for most audio material
SOFT: Reduces sensitivity to prevent false triggering with complex material

OCT SHIFT/NORMAL/PERF 5TH: A three-position switch for harmonic enhancement:

OCT SHIFT: Raises filter frequencies by one octave
NORMAL: No transposition
PERF 5TH: Raises both filters up a perfect fifth (x1.5)

PITCH/NORMAL/PITCH -OCT: Controls the pitch tracking function:

PITCH: Dynamically adjusts filter frequencies based on the detected pitch of the input
NORMAL: Disables pitch tracking
PITCH -OCT: Similar to PITCH but targets lower frequencies, ideal for bass material

2.3 Filter controls

FILTER A: Sets the cutoff frequency for Filter A. This exponential control provides fine adjustment in the critical lower frequencies while allowing quick access to the full frequency spectrum.

RESO A: Adjusts the resonance feedback for Filter A. Higher settings emphasize frequencies near the cutoff point, creating peaks that can range from subtle enhancement to extreme self-oscillation.

LBH 1 (Low-Band-High): A bipolar control that smoothly blends between lowpass (fully left), bandpass (center), and highpass (fully right) responses for Filter A.

BLH 1 (Band-Low-High Correction): A sophisticated correction control that modifies the filter response by adding or subtracting filter components. At center position (0), no correction is applied. Moving left (-B) reduces bandpass presence, while moving right (-B+LH) adds lowpass and highpass components to the bandpass.

RATIO: A 12-position rotary switch that defines the harmonic relationship between Filter A and Filter B. "FREE" allows independent control, while the numbered positions lock Filter B to specific musical intervals relative to Filter A.

MODE: A two-position switch above the RATIO knob. STEPPED uses the original 12 fixed positions (Free, 1, 1.5 ... 16). CONTINUOUS turns the knob into a smooth sweep of the harmonic ratio from 1x to 16x, for gliding, detuned movement between the two filters. In Continuous mode the FREE label reads CONT.

DIR: A two-position switch above the RATIO knob that sets the direction of the harmonic relationship. BELOW places Filter B below Filter A (the classic behavior). ABOVE places Filter B above Filter A, using the same intervals mirrored upward.

FILTER B: Sets the cutoff frequency for Filter B. When RATIO is active, this control is subordinate to FILTER A according to the selected harmonic relationship.

RANGE: A two-position switch above the FILTER B knob, active only when RATIO is engaged. OFF is the classic behavior (FILTER B has no function while synced). TRANSPOSE repurposes the FILTER B knob to slide the whole synced filter pair upward (up to two octaves) while keeping the harmonic interval between the filters intact.

RESO B: Controls the resonance for Filter B with a slightly different character than RESO A, allowing complementary tonal shaping when both filters are used together.

LBH 2: Provides the same filter type control as LBH 1 but for Filter B, enabling complex dual-filter configurations.

BLH 2: Provides the same correction functionality as BLH 1 but for Filter B.

2.4 Envelope and trigger controls

TRIG AR: Selects the trigger source for the AR envelope:

OFF: No external triggering (manual control only)
SIDECHAIN: Uses audio from the sidechain input for triggering
MIDI: Allows MIDI notes to trigger the AR envelope

A (Attack): Controls the attack time for the AR amplitude envelope, shaping how quickly the processed signal reaches full volume.

R (Release): Sets the release time for the AR envelope, determining how the processed signal fades out when the trigger ends.

2.5 Bottom panel controls

DIST TYPE: Choose between different distortion modes:

CLEAN: Minimal distortion with subtle enhancement
WARM: Analog-inspired saturation with even harmonic emphasis
CRUNCH: More aggressive distortion with complex harmonic content
EDGE: Maximum distortion with pronounced frequency-dependent character
DYNAMIC: Input-responsive distortion with signal-dependent bias and saturation memory, creating tube-like sag and harmonic interaction that evolves with playing intensity
NEURAL: Authentic Filterbank drive character captured through neural network modeling (LSTM). Reproduces the real hardware's saturation response with high fidelity
RAT: Classic Rat distortion captured via neural network. Delivers the iconic fuzzy, aggressive character of the original pedal at high gain
DS-1: DS-1 distortion captured via neural network. Reproduces the classic Japanese crunch character at high gain settings
TS9: TS9 overdrive captured via neural network. Delivers the smooth, mid-pushed drive character at full drive settings

Note: All neural modes (Neural, Rat, DS-1, TS9) bypass algorithmic distortion entirely and run real hardware captures through LSTM inference. Oversampling is automatically disabled in these modes as the neural models already capture the full harmonic spectrum of the original hardware. Neural modes run hotter than algorithmic modes, reflecting the authentic output levels of the captured hardware - use the OUTPUT knob to compensate

TRIG ADSR: Selects the trigger source for the ADSR envelope.

OFF: No external triggering (manual control only)
SIDECHAIN: Uses audio from the sidechain input for triggering
MIDI: Allows MIDI notes to trigger the ADSR envelope

TRIG AR: Selects the trigger source for the AR envelope.

OFF: No external triggering (manual control only)
SIDECHAIN: Uses audio from the sidechain input for triggering
MIDI: Allows MIDI notes to trigger the AR envelope

ADSR SENS: Controls the audio threshold level (in dB) for ADSR triggering. Adjust this to fine-tune how the ADSR responds to your audio material.

AR SENS: Controls the audio threshold level (in dB) at which the AR envelope will trigger when using audio input as a trigger source. Lower values make the AR more sensitive to quiet sounds.

Note on the Sidechain input: The sidechain input is shared across the FM source toggle, AM source toggle, and the AR/ADSR sidechain triggers. A single sidechain signal can simultaneously fire your envelopes and drive FM/AM modulation - useful for things like a kick triggering ADSR while also shaping the carrier with the kick's transient.

THD DRIFT: Controls the amount of thermal drift simulation, affecting how the distortion characteristics subtly change over time. Higher settings create more pronounced "breathing" effects typical of analog hardware.

NOISE: Introduces calibrated pink noise into the signal path, useful for adding texture, creating ambient atmospheres, or enhancing percussive elements.

LFO SYNC: Synchronizes the LFO frequency to your DAW's tempo with various rhythmic divisions (1/1, 1/2, 1/4, etc.), ensuring perfectly timed filter modulation.

OUTPUT: Adjusts the final output gain after all processing. This compensates for level changes introduced by filtering and distortion.

2.6 Additional controls

LFO MODE: A three-position switch selecting between RESTART (LFO restarts with each new trigger), NORMAL (standard sine wave modulation), and SAWTOOTH (changing the LFO waveform from sine to sawtooth for different modulation characteristics).

ADSR MODE: A three-position switch selecting between:

ADSR: Standard envelope generator mode with full ADSR control
FREEZE: Sustains the current envelope state regardless of trigger input
ENVELOPE FOLLOWER: Dynamically tracks the amplitude of the input signal

OVERSAMPLE: Toggles between standard processing and 4x oversampling. When enabled, this significantly reduces aliasing artifacts, particularly with high resonance settings and upper-frequency content, at the cost of increased CPU usage. Note: Oversampling is automatically disabled when any neural distortion mode is active (Neural, Rat, DS-1, TS9), as the neural models already capture the full harmonic spectrum of the original hardware.

2.7 MIDI controls

When the AR Trigger or ADSR Trigger is set to MIDI mode, you can control the triggering behavior via MIDI note messages. These controls allow precise timing and triggering from your MIDI controller, integrating Filterbank into live performance or MIDI-based production setups.

MIDI Note	Function	Behavior
60 (C4)	Unblock ADSR	Allows ADSR to trigger from audio or MIDI.
61 (C#4)	Block ADSR	Prevents ADSR from triggering (audio or MIDI blocked).
62 (D4)	Unblock AR	Allows AR to trigger from audio or MIDI.
63 (D#4)	Block AR	Prevents AR from triggering (audio or MIDI blocked).
66 (F#4)	Normal Trigger ADSR	Triggers the ADSR envelope normally.
70 (A#4)	Normal Trigger AR	Triggers the AR envelope normally.
68 (G#4)	Normal Trigger Both	Simultaneously triggers both AR and ADSR envelopes.
65 (F4)	Trigger ADSR	Triggers ADSR with instant attack.
67 (G4)	Gate-Off ADSR	Forces gate-off of ADSR with instant release.
69 (A4)	Trigger AR	Triggers AR with instant attack.
71 (B4)	Gate-Off AR	Forces gate-off of AR with instant release.

Note on Octave Labeling:
- MIDI note 60 corresponds to Middle C (C4).
- Some DAWs (e.g., Ableton, Logic, Studio One) may label this note as C3. This is simply a labeling difference, and the functionality remains the same.

3.0 Technical overview

Filterbank employs several advanced analog modeling techniques to achieve its distinctive sound quality. This section provides insight into the technical innovations that make this plugin unique.

3.1 SVF implementation

At the heart of Filterbank is a State Variable Filter (SVF) architecture that accurately reproduces the behavior of analog hardware filters. Unlike typical digital filters, our implementation:

- Uses separate state variables that correspond to physical elements in analog circuits
- Provides simultaneous lowpass, bandpass, and highpass outputs with precise blending
- Employs bilinear transform prewarping for accurate frequency response at all settings
- Calculates coefficients dynamically based on cutoff frequency and resonance parameters

3.2 Analog circuit modeling

Several techniques are used to faithfully emulate analog circuit behavior:

Diode pair simulation: The feedback path incorporates non-linear diode modeling that creates the characteristic asymmetric distortion found in analog filters, enhancing harmonic content and providing warmth.

Moog-style resonance feedback: Our recursive resonance system closely emulates the behavior of Moog ladder filters, creating the distinctive "analog" resonance character when pushed to self-oscillation.

Non-linearity control: A specialized non-linearity factor parameter allows smooth transition between clean digital response and analog-like behavior, particularly important at high resonance settings.

Resonance delay modeling: Filter parameters incorporate subtle delays that simulate component-level interactions, creating a more organic response to rapid parameter changes.

3.3 Capacitor simulation

The distinctive response of analog filters is partly due to capacitor charge/discharge characteristics:

- Different time constants for charging and discharging phases create non-linear response curves
- Switched capacitor behavior emulates how analog circuits naturally shift between states
- Sign-dependent reset behavior accurately captures real capacitor response to signal polarity inversions
- Dynamic parameter adjustment based on resonance and frequency settings creates the characteristic "squeeze" heard in analog filters

3.4 Distortion and saturation

Filterbank incorporates multiple stages of analog-inspired distortion:

- Signal path saturation based on the hyperbolic tangent curve creates natural compression effects
- Asymmetric distortion processing generates the even harmonics characteristic of analog hardware
- DC offset management enhances harmonic character while preventing unwanted artifacts
- Multiple distortion modes (Clean, Warm, Crunch, Edge, Dynamic, Neural, Rat, DS-1, and TS9) provide a wide range of tonal options, from algorithmic saturation to neural network-captured hardware character from four iconic devices

3.5 Thermal drift and noise

To capture the subtle imperfections that give analog equipment its character:

- THD (Total Harmonic Distortion) drift simulation uses low-frequency modulation to create the "breathing" quality of analog circuits
- Multi-stage filtered noise generation produces authentic pink noise for added texture
- Adaptive noise that responds dynamically to signal levels emulates how real components behave
- Weight-based distortion coloration creates distinctive sonic signatures at different frequency ranges

3.6 Advanced modulation system

The modulation system builds upon the analog modeling techniques:

- LFO modulation with variable waveforms (including Sawtooth and Sine) for evolving filter sweeps
- ADSR envelope with analog-inspired attack, decay, sustain, and release characteristics
- Envelope follower mode that dynamically tracks input signal for organic filter modulation
- Sophisticated cross-modulation between filters for complex, evolving sounds
- Pitch tracking that adapts filter frequency based on input signal's harmonic content

3.7 Analog-inspired gain staging

Proper gain staging is crucial to analog sound quality:

- Input gain structure dynamically affects distortion characteristics and filter behavior
- Resonance compensation prevents excessive gain at high resonance settings
- Adaptive gain scaling based on filter settings maintains consistent output levels
- Analog-inspired dynamic range compression at signal path transitions
- Subtle high-frequency enhancement through carefully calibrated high-shelf filtering

3.8 Component interaction modeling

Real analog circuits exhibit complex interactions between components:

- Cross-modulation between state variables creates subtle harmonic interactions
- Deliberate asymmetry in feedback paths mirrors real circuit behavior

- Accurate modeling of component saturation under varying signal conditions
- Dynamic interaction between resonance and cutoff frequency settings
- Self-oscillation characteristics that behave like genuine analog counterparts

Note on self-oscillation: Use caution when using high resonance.

Note on UI resizing: The plugin's UI resizing is supported in most formats and DAWs. However, in Logic Pro, resizing may not work correctly due to limitations in how the DAW handles third-party plugins. This is a known Logic Pro issue, not a problem with the plugin itself.

For more resources, updates, and preset packs, visit ravegeneration.io. Dive deeper into the world of audio manipulation and discover new ways to bring your tracks to life.

4.0 Installation & troubleshooting

4.1 System requirements

Before installing Rave Generation: Filterbank, please ensure that your system meets the following requirements:

- Operating system:
    - macOS 10.13 or later
    - Windows 10 or later
- Software: Digital Audio Workstation (DAW) that supports VST3, or AU plugins (e.g., Ableton Live, Logic Pro, Studio One, FL Studio, etc.).
- Processor: Intel Core i5 (or equivalent) or higher for optimal performance.
- RAM: 4 GB minimum (8 GB or more recommended for larger projects).
- Disk Space: 200 MB of free disk space for installation.

4.2 Installation process

1. Download the installation file from the official website or the platform where you purchased the plugin.
2. Run the installer and follow the on-screen instructions.
3. Launch your DAW and locate Rave Generation: Filterbank in your plugin list.
4. If prompted, activate the plugin using the license key provided upon purchase.

4.3 Troubleshooting

If you encounter any issues during installation or operation, try the following solutions:

- Plugin not showing in DAW: Ensure that the plugin folder path is correctly set within your DAW's plugin manager.
- Activation issues: Double-check your internet connection and ensure you are entering the correct license key.