Kombinat User’s Guide
Audio Damage, Inc.
Release 4.0

28 April 2026

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Introduction

Distortion is fundamentally a simple idea: take an audio signal, run it through something nonlinear, and collect the harmonics that come out the other side. The complications begin when you try to do this to a full-bandwidth mix or a complex source. A bass guitar, a drum loop, a synthesizer pad all contain energy spread across the frequency spectrum, and a distortion algorithm applied to the whole signal treats all of that energy equally. In practice, "equally" usually means "dominated by the low end," because the low-frequency content tends to be loudest, and the distortion responds to it most strongly. The result is often muddy, undefined, and sonically uninteresting.

The answer to this problem is the same one that loudspeaker designers, mastering engineers, and DJ mixer manufacturers arrived at independently: divide the signal into frequency bands, process each band separately, and mix the results back together. Kombinat is our implementation of this idea applied to distortion. The original version appeared in 2007. We believe it was the first multi-band distortion plug-in, though we'll admit we didn't conduct an exhaustive survey of the field. Version 4 represents a significant expansion of the concept, and the gap between the two is long enough that it felt like starting fresh in several respects.

Kombinat 4 is the fourth iteration. Like its predecessors, it uses a bank of crossover filters to divide the incoming signal into separate frequency bands, each of which has its own distortion engine and level controls. What's new in version 4 is the degree of flexibility: the number of bands is variable from one to eight, and the distortion algorithm for each band can be selected from a library of 38, ranging from models of specific analog circuits to entirely novel types that have no physical counterpart. A second signal routing mode lets you chain the distortion engines in sequence rather than running them in parallel, which opens up a different set of sonic possibilities entirely. Rounding out the processing chain are a global output filter with twelve modes, a compressor derived from our Rough Rider plug-in, and a feedback path that routes some of the output back to the input.

Kombinat 4 can be subtle. It can also be completely unhinged. The distance between those two things is shorter than you might expect.

System Requirements

The following table summarizes the operating system requirements and formats provided for Kombinat. Kombinat is a 64-bit plugin.

Operating System	**Minimum Version**	**Formats**
macOS	10.13	AAX, AudioUnit, CLAP, VST3; Intel and Apple Silicon
Windows	10.0	AAX, CLAP, VST3
Ubuntu	18.0	CLAP, LV2, VST3
iOS (separate purchase)	iOS 12	AUv3

Demonstration Version

We encourage you to download and try the demonstration version of Kombinat before purchasing it. The demo version of Kombinat is the same as the regular version, but has the following limitations:

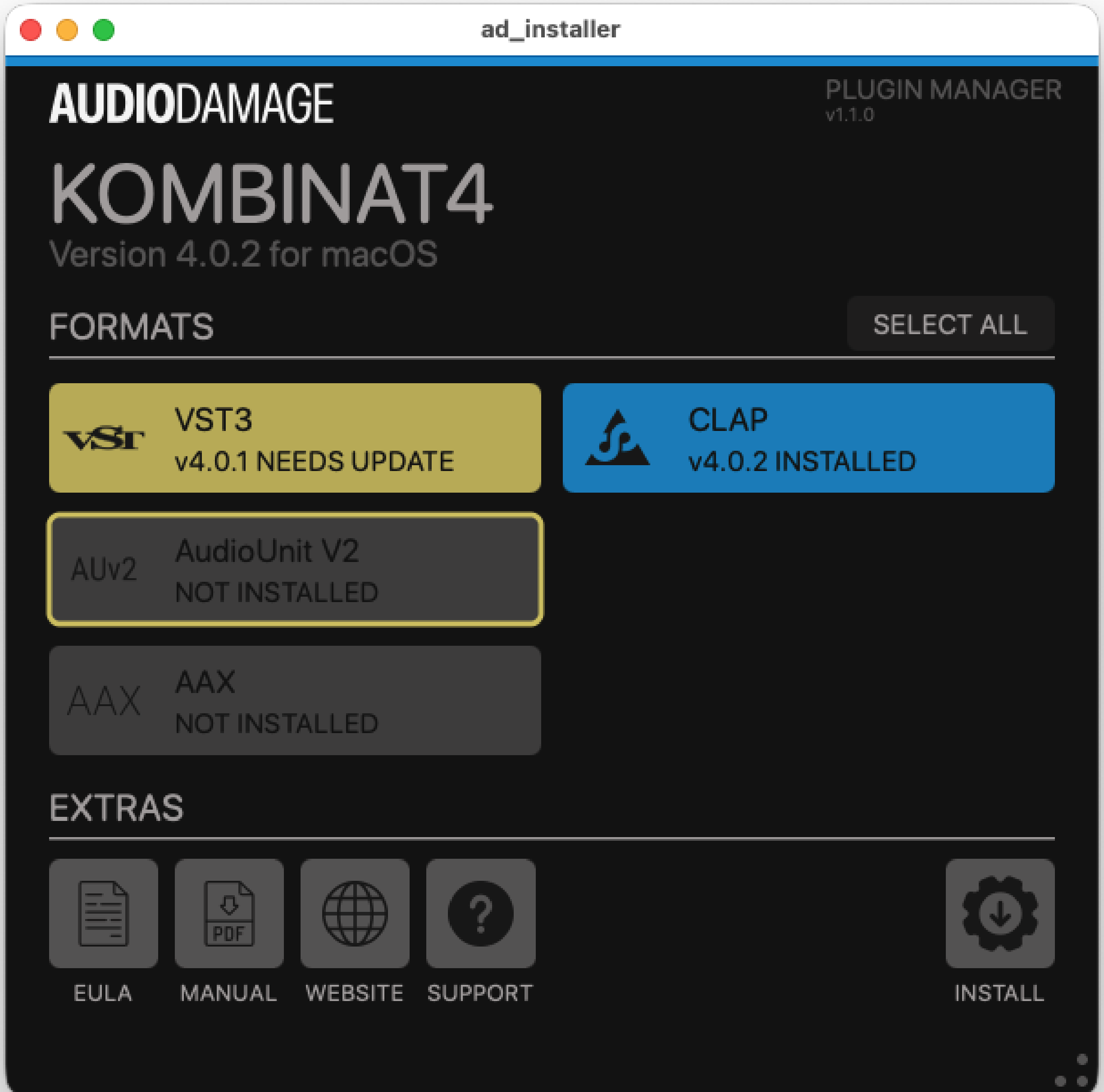
- Presets cannot be saved, nor can parameter values or other settings. This includes the information usually stored by your host DAW. If you save a DAW session with an instance of the demo version of Kombinat, Kombinat will revert to its default state when you reload the session.

- Kombinat will cease to emit audio altogether 20 minutes after you add it to your DAW session. You can remove it and add it again, but it will revert to its default state.

If you purchase Kombinat after using the demonstration version, simply run the installer provided to you after your purchase to replace the demo version with the full version.

Installation

Kombinat uses our custom plugin manager application for installation. Launch it as usual on your operating system of choice and you'll be presented with a window like this:



Near the top of the window, beneath the large word KOMBINAT, you'll see the version number of the software carried by the installer. This is distinct from the version of the plugin manager itself, which is shown in the upper right and usually not of much interest.

Under the heading FORMATS are large buttons corresponding to the plugin formats which can be installed: AAX, AU, CLAP, LV2 and/or VST3, depending on the operating system. If the plugin is already present on your system in one or more formats (i.e. if you're upgrading from a previous version), the corresponding button is drawn in blue. When possible, the version number of the existing plugin is also shown.

Click a button to select the format for installation. A yellow outline appears around the button to indicate that its format will be installed. In the above screenshot, VST3 and CLAP are installed, AAX and AudioUnit are not installed, VST3 is one revision behind that contained in the installer, and the AudioUnit is selected for installation. Clicking a button a second time removes the yellow outline, and the corresponding format will not be installed. Clicking the SELECT ALL button selects all available formats for installation.

No changes to your system's storage device take place until you click the INSTALL button near the lower-right corner of the window. Click that button and you'll receive visual confirmation that the formats you've selected have been installed. (Yes, it happens quickly. Our products are not encumbered by any DRM or anti-piracy baggage and hence install more quickly than many others.) On Windows and Linux, if you hold down the Shift key on your keyboard, the INSTALL button's label switches to UNINSTALL, and clicking it will remove the selected formats from your system[1]. Once you're installed and/or removed the formats you need, simply close the application in the usual manner for your operating system. You're done. There is no license code or other authorization necessary; we'd rather assume we can trust you than burden you with an onerous DRM system.

You'll find some handy buttons under the EXTRAS heading, all of which are pretty self-explanatory:

EULA – presents the End-User License Agreement for our products. By clicking the INSTALL button you're implicitly agreeing to these terms, but we expect that you'll find them reasonable should you take the time to read them.

MANUAL – opens the current version of this user manual, in PDF form, in your web browser.

WEBSITE – opens the product's web page in your browser.

SUPPORT – displays information for contacting us, either via our Discord presence or through email.

[1] Blame Apple, not us, for the lack of this feature on macOS. On macOS just manually delete the plugin(s) from your plugin folder(s).

Operation

Kombinat can be used in either a stereo or mono context in your host DAW software. If you use a stereo input, the left and right channel separation of the input signal is preserved in the output signal. Kombinat is meant to be used as an insert effect but there's no reason you can't use it as a send/return effect if doing so suits you.

Kombinat 4's window is organized into three rows. Across the top is the band display, where you can see and adjust the frequency ranges processed by each distortion engine. The middle row contains the engine and filter controls, along with the input level and oversampling settings. The bottom row contains the output-side processing: the compressor, feedback, and output controls.

Here is a screenshot of Kombinat:

Bands

Kombinat 4 uses a series of crossover filters, similar to those found in a DJ mixer, to divide the incoming audio into separate frequency ranges. Each range has its own distortion engine, and we call these ranges bands. Kombinat 4 supports between one and eight bands.

The band display at the top of the window shows the current bands arranged left to right, corresponding to their position in the frequency spectrum: bands on the left handle lower frequencies, bands in the middle handle the midrange, and bands on the right handle the high frequencies. A spectrum analyzer runs behind the band display, giving you a visual indication of the frequency content of the incoming signal.

Each band occupies a numbered rectangular region in the display. At the top of each region are M and S buttons, which mute and solo the band respectively. The vertical slider in the center of each region sets the output level of that band, with a range of -80 to +12 dB. The small x button at the upper left of a band's region removes that band. Clicking the + button at the lower right of the display adds a new band.

You can also use the + and - buttons at the top left of the window to add and remove bands. Kombinat 4 will not let you reduce the number of bands below one, nor increase it above eight.

To change the crossover frequency between two adjacent bands, click and drag the divider between them. The crossover frequency can be set anywhere between 60 Hz and 20 kHz.

Clicking on a band in the display selects it, and the engine and filter controls in the middle row will update to show that band's settings.

Engine Mode

The ENGINE MODE control at the top of the window switches between Kombinat 4's two signal routing modes: MULTIBAND and SERIAL.

In MULTIBAND mode, the crossover filters divide the incoming audio into separate frequency ranges, each band's engine processes its corresponding range, and the results are mixed back together before passing through the filter and output processing. This is the standard multi-band configuration and is the most useful for treating different parts of the frequency spectrum differently; for example, applying gentle saturation to the low end while using a more aggressive algorithm on the highs.

In SERIAL mode, all bands receive the full-bandwidth signal and are connected in series: the output of band 1 feeds the input of band 2, the output of band 2 feeds band 3, and so on. The crossover dividers in the band display still appear, but in SERIAL mode they do not filter the signal; only the engines and their settings matter. SERIAL mode lets you chain up to eight distortion engines in a single signal path, stacking their effects.

Input

The INPUT knob sets the level of the signal entering Kombinat 4. It has a range of -40 to +6 dB. Since many of Kombinat 4's distortion engines respond to the level of the signal, the INPUT knob is one of the most powerful tools for shaping the character of the sound — a small adjustment here can dramatically change what you hear coming out.

Oversampling

The OVERSAMPLING control sets the internal sample rate at which Kombinat 4 processes audio. The options are Off (processing at your host's sample rate), 2x (twice your host's sample rate), and 4x (four times your host's sample rate). Oversampling reduces aliasing artifacts that some of the more aggressive distortion algorithms can produce at higher frequencies, at the cost of additional CPU load. Whether it makes an audible difference depends on the algorithm and the material you are processing; for many low-frequency applications it will have little effect.

Be aware that many of Kombinat 4's distortion algorithms are inherently sample-rate dependent: the same settings can produce a different sound at 44.1kHz than at 96kHz, because the nonlinear processing interacts differently with the signal at different resolutions. This is also true of oversampling: oversampling may reduce the undesireable aliasing in soft-clipping modes, but aliasing is part of the charm of more aggressive "digital-sounding" distortion processes. In short, oversampling may or may not be useful depending on how you're using Kombinat. Keep in mind if you record or mix at one sample rate and render at another the rendered result may not sound identical to what you heard during the session.

Engine

The central portion of the middle row shows the controls for the currently selected band's distortion engine. Click on a band in the top display to select it; the selected band is highlighted.

The band number is shown at the left of the engine section. Next to it is the MUTE button, which silences the current band, followed by the SOLO button, which silences the other bands.

The ALGORITHMS menu selects the distortion algorithm for the current band. Kombinat 4 has 38 algorithms, from subtle harmonic coloration to complete sonic destruction. The algorithms are described in detail in the Algorithms section below.

Each algorithm has up to three parameter knobs. The labels beneath the knobs change to show the parameter names for the selected algorithm. Not all algorithms use all three knobs; unused knob positions are indicated by a dash. The meaning of each knob is described in the Algorithms section.

The GAIN knob at the right of the engine section sets the output level of the current band. Adjusting it here is the same as moving the vertical slider for that band in the display at the top of the window.

Filter

The filter processes the combined output of all of the bands. Its controls are at the right side of the middle row. The filter follows the distortion engines in the signal path, so it shapes the tone of the distorted signal.

The filter type selector chooses from twelve filter modes:

4P LP — A four-pole low-pass filter, the type most commonly found on analog synthesizers. It attenuates high frequencies above the cutoff, and passes low frequencies below it. The attenuation slope is 24 dB per octave.

2P LP — A two-pole low-pass filter with a gentler slope of 12 dB per octave, suitable for less dramatic high-frequency shaping.

4P HP — A four-pole high-pass filter. It attenuates low frequencies below the cutoff and passes high frequencies above it, at a slope of 24 dB per octave.

2P HP — A two-pole high-pass filter with a 12 dB per octave slope.

4P BP — A four-pole band-pass filter, which passes a range of frequencies centered on the cutoff and attenuates both higher and lower frequencies.

Notch — A notch filter, which attenuates a narrow band of frequencies at the cutoff while leaving frequencies above and below it largely unaffected.

OTA LP — A low-pass filter based on an operational transconductance amplifier (OTA) circuit topology, similar to the filters found in certain classic synthesizers. It has a character distinct from the standard 4P LP filter.

OTA HP — A high-pass filter using the same OTA topology.

OTA BP — A band-pass filter using the OTA topology.

914 BP — A band-pass filter modeled on the fixed filter bank circuit of the Moog 914 module. It has a vintage, distinctive character that differs from a standard band-pass filter.

Filterpod — An Audio Damage original filter design that combines a four-pole low-pass filter and a biquad peak filter running in parallel, both tuned to the same frequency and resonance. The two outputs are summed together. The result is a filter with the gentle low-pass character of the 4P LP overlaid with a strong resonant peak at the cutoff frequency — a combination with a distinctive, vintage-leaning character unlike any of the other filter types.

Bypass — The filter is removed from the signal path.

The FREQ knob sets the cutoff frequency of the filter, from 20 Hz to 20 kHz. Turning it clockwise raises the cutoff frequency.

The RESO knob controls the resonance. As resonance increases, the filter emphasizes frequencies near the cutoff, giving the sound a more pronounced tonal character. At high resonance settings some filter types will self-oscillate, producing a sustained tone even in the absence of an input signal.

Compressor

Kombinat 4 includes a compressor derived from our Rough Rider plug-in, placed after the filter in the signal path. The ON button at the left of the compressor section enables and disables it.

The THRESH knob sets the signal level at which the compressor begins to act. Signals above this level are reduced in gain. Lower threshold values cause the compressor to engage more readily and affect more of the signal.

The RATIO knob determines how much gain reduction is applied to signals that exceed the threshold. A ratio of 4:1 means that for every 4 dB the signal exceeds the threshold, the output increases by only 1 dB. Higher ratios produce more aggressive compression, with ratios above roughly 10:1 approaching limiting behavior.

The ATTACK knob controls how quickly the compressor responds when a signal exceeds the threshold. A fast attack clamps down on transients immediately; a slower attack lets them pass through before the compression takes hold, which can help preserve the punch of percussive sounds.

The RELEASE knob sets how quickly the compressor stops reducing gain after the signal falls below the threshold. If the release is too short, you may hear the level pumping in an obvious way; if it is too long, the compressor may not recover quickly enough between loud events. Let your ear be your guide.

The MAKEUP knob applies gain after the compression to compensate for the overall level reduction. It has a range of 0 to 30 dB.

The SC HP knob sets the cutoff frequency of a high-pass filter applied to the compressor's sidechain signal — that is, the signal that the compressor uses to decide when and how much to compress. With the SC HP frequency set higher, the compressor becomes less sensitive to low-frequency content. This can prevent a heavy bass signal from causing unwanted gain reduction across the full frequency range.

The SC button, located below the ON button on the left of the compressor section, switches the sidechain signal between two sources. When unlit, the compressor uses Kombinat 4's own processed output as its sidechain — that is, the signal that is being compressed is also the signal driving the compression decisions. When lit, the compressor uses an external sidechain input instead, allowing a separate audio source in your host DAW to control the compression. To use the external sidechain, route a signal to Kombinat 4's sidechain input using your host software's standard method for auxiliary inputs.

Feedback

The FEEDBACK section routes some of Kombinat 4's output back to its input, creating a feedback loop through the entire processing chain. At moderate settings this adds resonance and tonal color; at higher settings the sound becomes increasingly unstable and self-sustaining.

The AMOUNT knob controls how much of the output is fed back. At low settings the effect is subtle; as you increase it, the feedback becomes more apparent. Exercise caution at high AMOUNT settings — it is straightforward to create very loud sustained tones.

The ATTACK knob controls how quickly the feedback signal rises. The DECAY knob controls how quickly it falls. Together these shape the envelope of the feedback contribution and can be used to give the feedback a more rhythmic or transient character.

Output

The OUTPUT section at the bottom right of the window contains the final output controls.

The AGC button enables automatic gain control, which adjusts the output level to compensate for the overall loudness change introduced by the distortion engines. This is useful when comparing the effect of different algorithms or settings at a consistent listening level.

The DC BLOCK button enables a DC-blocking filter on the output. Some distortion algorithms, particularly those involving rectification or asymmetric clipping, can introduce a DC offset into the signal — a shift in its average value that is inaudible on its own but can cause problems with downstream processing or cause output clipping. Enabling DC BLOCK removes this offset.

The BYPASS button bypasses all of Kombinat 4's processing, passing the dry input signal to the output unaltered. This is useful for A/B comparisons between the processed and unprocessed sound.

The OUTPUT knob sets the overall output level, with a range of -60 to +6 dB.

Algorithms

Kombinat 4 has 38 distortion algorithms. The first, bypass, passes the signal through without alteration and has no controls. The remaining 37 are described below.

The knob labels shown in the descriptions below are those displayed in Kombinat 4's engine section when that algorithm is selected. Knob positions shown as — are not used by the algorithm.

BYPASS

Bypass passes the signal through without any processing. It has no parameter knobs. Use it in MULTIBAND mode when you want a particular frequency band to pass through unaffected, or in SERIAL mode when you want a slot in the engine chain to have no effect.

FUZZ

The FUZZ algorithm is derived from a model of an analog fuzz pedal circuit. It produces the warm, slightly woolly distortion character that fuzz pedals are known for — distinct from the harder edge of a clipper or the smoothness of a saturator.

Knob 1 — AMOUNT: increases the fuzz intensity and overall signal level.

SATURATE

The SATURATE algorithm simulates the distortion produced by overloading a transistor-based gain stage, such as those found in preamplifiers. At low settings it adds a subtle harmonic warmth; at higher settings it rounds off the peaks of the signal in a smooth, musically pleasing way.

Knob 1 — AMOUNT: increases the pre-gain and hence the amount of saturation.

DISTORT

The DISTORT algorithm uses a cubic sigmoid transfer function combined with zero-crossing distortion. The zero-crossing component adds a slight gate-like effect at quiet passages, of the kind associated with contemporary boutique guitar pedals. It is more aggressive than SATURATE.

Knob 1 — AMOUNT: increases the distortion intensity.

Knob 2 — GAIN: increases the signal level before distortion, which also increases the distortion amount.

CLIP

The CLIP algorithm is a hard clipper whose positive and negative clipping thresholds can be set independently. Since audio signals are rarely perfectly symmetric, treating the two halves differently can produce a range of tonal results. Some diode-based guitar pedals clip asymmetrically by design; this algorithm lets you do the same intentionally.

The clipping thresholds are set relative to zero. Turning either knob up moves the corresponding threshold away from zero. Leaving both at zero produces silence, since the signal is clipped at zero on both sides. Turning both all the way up raises the thresholds well beyond the peaks of the signal, at which point you will need to use the GAIN knob to push the signal hard enough to clip.

Knob 1 — GAIN: sets the pre-gain applied before clipping, from 1x to 10x.

Knob 2 — LOW: sets the clipping threshold for the negative half of the signal.

Knob 3 — HIGH: sets the clipping threshold for the positive half of the signal.

WARP

The WARP algorithm distorts the signal by using a sinusoid as a non-linear transfer function. In less technical terms, it wraps the signal around a sine wave. The result can range from harsh distortion that sounds like hard clipping to metallic, ring-modulation-like overtones, depending on the FREQ setting and the level of the input.

Knob 1 — FREQ: controls the frequency factor of the sine wave and hence the timbre of the distorted signal.

DEGRADE

The DEGRADE algorithm applies several forms of digital signal degradation. It is the successor to the BITZ engine found in earlier versions of Kombinat.

The RATE knob controls a sample-rate reducer. Turning it up causes the signal to be resampled at progressively lower rates, introducing the aliasing and lo-fi character associated with early digital hardware. The effect is most audible in the high-frequency bands, since those frequencies are the first to alias as the sample rate drops.

The BITS knob controls bit-depth reduction. At its minimum, the signal passes at full resolution. Turning it up reduces the number of bits used to represent the signal, first to 16 bits, and then all the way down to one bit as the knob reaches its maximum.

The ERROR knob introduces errors into the bits used to represent the signal. The knob controls how long the errors persist and hence how much they damage the audio. The BITS and ERROR knobs are somewhat complementary: the effect of ERROR is less noticeable at high BITS settings.

Knob 1 — RATE: controls the sample-rate reduction.

Knob 2 — BITS: controls the bit-depth reduction.

Knob 3 — ERROR: introduces and sustains bit errors.

RING MOD

The RING MOD algorithm is a ring modulator. Ring modulators produce inharmonic, clangorous tones by multiplying the input signal by an oscillator. The oscillator in this engine has a variable waveform shape, from sinusoidal to square-like, giving you control over the character of the modulation.

Knob 1 — AMOUNT: controls the depth of the ring modulation. At zero, the dry signal passes through; at maximum, only the ring-modulated signal is heard.

Knob 2 — FREQ: controls the frequency of the modulator oscillator, from roughly 40 Hz to 840 Hz.

Knob 3 — SHAPE: controls the waveform of the oscillator. At its minimum the waveform is sinusoidal; turning it up progressively squares off the waveform, which makes the modulation more aggressive.

X NOISE

The X NOISE algorithm multiplies the input signal by band-pass filtered noise. The result is a signal that retains the dynamics and general character of the input but with a noisy, gritty texture added, the character of which is shaped by the filter settings.

Knob 1 — AMOUNT: controls the depth of the noise modulation.

Knob 2 — FREQ: controls the center frequency of the band-pass filter applied to the noise, from 50 Hz to roughly 3 kHz.

Knob 3 — Q: controls the resonance of the filter, which narrows or widens the band of noise frequencies.

OCTAVER

The OCTAVER algorithm generates harmonic content at octave relationships to the input. It does this using half-wave and full-wave rectification, which are the classical methods for producing sub-octave and upper-octave harmonics from an audio signal. The three knobs set the relative levels of the original signal, the half-wave rectified signal (which produces content an octave below), and the full-wave rectified signal (which produces content an octave above).

Knob 1 — DRY: sets the level of the original, unprocessed signal.

Knob 2 — 1/2: sets the level of the sub-octave signal.

Knob 3 — 2/1: sets the level of the upper-octave signal.

WAVY

The WAVY algorithm adds a sine-shaped waveshaping component to the signal: the output is the input plus a scaled sine function of the input multiplied by a frequency factor. At low settings this adds subtle harmonic interest; at higher settings the distortion becomes more complex and unpredictable.

Knob 1 — AMOUNT: controls the depth of the waveshaping component.

Knob 2 — FREQ: controls the frequency factor of the sine function, which changes the character of the harmonics added.

RECTIFY

The RECTIFY algorithm applies half-wave and full-wave rectification with adjustable slope and rounding. Rectification reflects part or all of the signal above or below zero, producing a distinctive asymmetric distortion character.

Knob 1 — SLOPE: blends between negative and positive reflection of the signal.

Knob 2 — ROUND: controls the degree of rounding at the zero-crossing point, varying the hardness of the rectification.

TUBE CLIP

The TUBE CLIP algorithm models the asymmetric soft-clipping behavior of an overdriven tube gain stage. Unlike simple clippers, the tube-style circuit creates a softer knee at the clipping threshold, with a character that varies with the BITE setting.

Knob 1 — DRIVE: sets the pre-gain applied to the signal, from 1x to 10x.

Knob 2 — BITE: controls the sharpness of the clipping knee. Lower values produce a rounder, softer knee; higher values make the transition to clipping more abrupt.

Knob 3 — MIX: blends between the dry signal and the distorted signal.

NERD RAGE

The NERD RAGE algorithm is unlike any conventional distortion type. When the signal exceeds the threshold set by the THRESH knob, it is recorded into an internal buffer. When the signal falls below the threshold, the buffer is played back in reverse, with its level decaying at the rate set by the DECAY knob. The result is a reactive, unpredictable effect that transforms quiet passages into ghostly echoes of loud ones. The sonic character is difficult to describe and best understood by experimenting with it.

Knob 1 — THRESH: sets the signal level above which audio is recorded into the buffer.

Knob 2 — DECAY: controls how quickly the replayed buffer signal fades.

Knob 3 — GAIN: sets the output level of the algorithm.

TANH

The TANH algorithm applies a hyperbolic tangent transfer function to the signal. The tanh function is a classic soft-clipping curve: it is linear for small signals and smoothly limits large ones, producing symmetric, primarily odd-order harmonic distortion. It is a mathematically clean, smooth form of saturation.

Knob 1 — DRIVE: controls the amount of drive applied before the tanh function, from subtle saturation to heavy limiting.

ASYM TANH

The ASYM TANH algorithm is a variation of the TANH algorithm with an offset applied before the transfer function, producing asymmetric clipping. Asymmetric clipping emphasizes even-order harmonics, which gives the distortion a character somewhat similar to a tube amplifier.

Knob 1 — DRIVE: controls the drive amount.

Knob 2 — OFFSET: shifts the operating point of the transfer function, controlling the degree of asymmetry.

OVERDRIVE

The OVERDRIVE algorithm uses an arctangent transfer function. The arctan function is a natural choice for overdrive modeling — it is smooth and continuous, it limits the signal symmetrically, and it produces a character similar to many classic overdrive pedal designs.

Knob 1 — DRIVE: controls the amount of drive.

POLYNOMIAL

The POLYNOMIAL algorithm applies a polynomial transfer function of the form $m / (m^2 \times \text{feedback} + 1)$, where m is the driven input signal. This produces a smooth, compression-like saturation whose character is shaped by the FEEDBACK knob.

Knob 1 — DRIVE: sets the pre-gain, from 1x to 45x.

Knob 2 — FEEDBACK: controls the coefficient that shapes the polynomial curve, changing the hardness and character of the limiting.

LINEAR DISTORT

The LINEAR DISTORT algorithm applies the transfer function $x \times a / (|x \times a| + b)$, a form of soft clipping that is gentler and more linear than the tanh or arctan varieties. The DRIVE and BIAS knobs give independent control over the numerator and denominator of the function.

Knob 1 — DRIVE: sets the input gain coefficient, from 1x to 85x.

Knob 2 — BIAS: controls the denominator coefficient, which shapes the softness of the limiting.

SINE DRIVE

The SINE DRIVE algorithm uses a sine function as a non-linear transfer function. At low drive settings, the sine function acts as a gentle soft clipper. As the drive increases, the signal is large enough to wrap around the sine wave, producing complex harmonic content. At very high drive settings the output becomes increasingly chaotic and noise-like.

Knob 1 — DRIVE: controls the drive amount.

VAR HARD CLIP

The VAR HARD CLIP algorithm uses an arctan power-law function to create a variable-hardness clipping characteristic. At low HARD settings the clipping knee is relatively gradual; at high settings it approaches a true hard brick-wall limiter.

Knob 1 — DRIVE: sets the pre-gain applied to the signal.

Knob 2 — HARD: controls the hardness of the clipping knee. Turning it up makes the transition from linear to clipped progressively sharper.

GLOUBI BOULGA

The GLOUBI BOULGA algorithm implements a classic nonlinear distortion function developed by Laurent de Soras. It produces a complex, asymmetric harmonic distortion with a distinctive character that is difficult to categorize as any standard type.

Knob 1 — DRIVE: controls the pre-gain, from 1x to 48x.

MXR

The MXR algorithm models the MXR Distortion+ pedal using a lookup table derived from the pedal's circuit. The Distortion+ is a classic hard-clipping distortion circuit whose sound has been an important part of rock guitar since the 1970s. This implementation uses a high-resolution table interpolation, using values measured directly from a hardware unit, to reproduce the circuit's characteristic clipping behavior.

Knob 1 — DIST: controls the distortion amount.

SOFT CLIP

The SOFT CLIP algorithm is a straightforward soft clipper with a variable threshold. As the AMOUNT knob is increased, a progressively larger portion of the signal is pushed into a boosted soft-clip region, increasing both the apparent loudness and the harmonic content.

Knob 1 — AMOUNT: controls the clip level.

CUBIC NLD

The CUBIC NLD algorithm applies a cubic non-linear distortion function of the form $x - x^3/3$. This function naturally produces odd-order harmonics — primarily the third harmonic — with a smooth, continuous curve. An OFFSET control shifts the input before the function is applied, introducing asymmetry and even-order harmonics.

Knob 1 — DRIVE: sets the pre-gain, from 1x to approximately 100x on a logarithmic curve.

Knob 2 — OFFSET: shifts the operating point of the function, producing asymmetric distortion.

WAVEFOLD

The WAVEFOLD algorithm is a wavefolder. When the signal exceeds the bounds of the transfer function, instead of being clipped it is reflected back in the opposite direction, producing a folded waveform. Wavefolding is associated with the Buchla synthesizer tradition and produces a dense, harmonically rich sound that increases in complexity as the drive is increased.

The SYMM knob introduces an offset before the folding, making the fold asymmetric. At its center position the folding is symmetric; moving it in either direction shifts the fold point.

Knob 1 — DRIVE: controls the fold depth, from subtle shaping to extreme folding.

Knob 2 — SYMM: controls the symmetry of the fold.

CHEBYSHEV

The CHEBYSHEV algorithm applies Chebyshev polynomial functions to the signal. Chebyshev polynomials have the mathematical property of adding specific harmonics to a signal with a high degree of precision. The ORDER knob blends between the second through seventh Chebyshev polynomials (T2 through T7), each of which adds a progressively higher harmonic. Unlike most distortion algorithms, whose harmonic content varies with input level, the Chebyshev polynomials add harmonics that are specific to the order selected.

Knob 1 — ORDER: blends between Chebyshev orders T2 through T7. At its minimum, the second polynomial is applied; at its maximum, the seventh.

Knob 2 — DRIVE: sets the pre-gain applied before the polynomial function.

BITWISE

The BITWISE algorithm applies binary logic operations to the signal after quantizing it to a given bit depth. The three operations available are XOR (exclusive or), AND, and OR, each of which interacts with a bit mask to alter specific bits of the quantized signal value. The results are characteristically harsh and digital, producing artifacts that have little in common with any analog distortion type.

Knob 1 — OP: selects the operation. The lower third of the knob's range selects XOR, the middle third selects AND, and the upper third selects OR.

Knob 2 — MASK: sets the bit mask applied by the operation.

Knob 3 — BITS: sets the bit depth to which the signal is quantized before the operation is applied, from 8 to 16 bits.

SLEW LIMIT

The SLEW LIMIT algorithm limits the rate at which the signal is allowed to change. Limiting the slew rate attenuates high-frequency content and rounds off transients; at more extreme settings, the signal is reduced to a slow ramp between values. The RISE and FALL knobs set the slew rate limits for upward and downward changes independently, so the positive and negative transitions of the signal can be shaped differently.

Knob 1 — RISE: controls the maximum rate at which the signal can increase. Turning it anti-clockwise reduces the rise rate, producing a slower, more smeared attack on transients.

Knob 2 — FALL: controls the maximum rate at which the signal can decrease.

COMPARATOR

The COMPARATOR algorithm compares the input signal to an internal ramp oscillator. When the ramp's value is above a threshold — which is modulated by the amplitude of the input signal — the signal passes; when below, it is inverted. The result is a harsh, frequency-modulated distortion whose character depends on the relationship between the oscillator frequency and the frequency of the input.

Knob 1 — FREQ: controls the frequency of the internal ramp oscillator.

Knob 2 — THRESH: sets the base threshold for the comparison.

DIODE CLIP

The DIODE CLIP algorithm models diode-style clipping with an asymmetric soft knee. Real diode clippers used in guitar pedals often use different diode types for the positive and negative halves of the signal — for example, a germanium diode on one side and a silicon diode on the other — which produces asymmetric clipping with different knee characteristics for each half. The ASYM knob controls this asymmetry.

Knob 1 — DRIVE: sets the pre-gain, from 1x to 20x.

Knob 2 — ASYM: controls the asymmetry of the clipping between the positive and negative halves of the signal. At the center position the clipping is symmetric; moving the knob in either direction makes one half clip harder than the other.

FEEDBACK FM

The FEEDBACK FM algorithm applies frequency modulation distortion using a sine function with feedback. The output of each sample is fed back into the modulation of the next, creating a self-influencing distortion that produces FM-like harmonic complexity. At low FBACK settings the algorithm produces smooth harmonic distortion; as the feedback increases the behavior becomes progressively more chaotic and self-resonant.

Knob 1 — DRIVE: controls the modulation index, which sets the depth of the FM effect.

Knob 2 — FBACK: controls how much of the previous output is fed back into the current sample's modulation.

FRACTAL

The FRACTAL algorithm applies a Mandelbrot-style iterative calculation to the signal. Each sample is used as the initial value in an iteration sequence, and the output blends the result of the iteration with the original signal based on how many iterations were completed before the sequence diverged. The result is a complex, unpredictable distortion whose character changes dramatically with input level.

Knob 1 — ITER: sets the maximum number of iterations, from 1 to 10. More iterations generally produce a smoother output.

Knob 2 — ESCAPE: sets the escape radius for the iteration. Smaller values cause the iteration to terminate sooner, producing different output characteristics.

DROPOUT

The DROPOUT algorithm probabilistically silences the signal in passages where the signal level is below a threshold, simulating the effect of magnetic-tape dropout — the occasional loss of audio that occurs when oxide particles leave the surface of recording tape. The probability of a dropout occurring increases as the signal level falls further below the threshold.

Knob 1 — THRESH: sets the signal level below which dropout can occur.

Knob 2 — DENSITY: controls the probability of a dropout occurring when the signal is below the threshold. At maximum, the signal is almost always silenced below the threshold; at lower settings, dropouts occur only occasionally.

SHAPESHIFT

The SHAPESHIFT algorithm applies independent power-law curves to the positive and negative halves of the signal. A power-law curve of 1.0 leaves the signal unchanged. Values below 1.0 produce a convex curve, rounding the signal upward (similar to tube saturation). Values above 1.0 produce a concave curve, pushing the signal toward zero before it rises. By setting the positive and negative curves differently, you can create a wide range of asymmetric distortion characteristics.

Knob 1 — POS: sets the power-law exponent for the positive half of the signal, from 0.2 to 5.0.

Knob 2 — NEG: sets the exponent for the negative half.

PHASE DIST

The PHASE DIST algorithm is inspired by the phase distortion synthesis technique used in the Casio CZ series of synthesizers. The algorithm tracks the zero crossings of the signal to establish a pseudo-phase, then warps that phase through a non-linear curve before using it to modulate the signal. The effect is a complex reshaping of the waveform that adds harmonics in a way that differs from conventional transfer-function distortion.

Knob 1 — AMOUNT: controls the depth of the phase warping, from 0 to near-maximum.

Knob 2 — RESO: applies a resonance-like feedback within the distorted phase calculation, increasing the intensity of the effect.

STUTTER

The STUTTER algorithm continuously records the incoming audio into a buffer and periodically replays a segment of it — a grain — in place of the live signal. The result is a rhythmic, repeating stutter effect. The SIZE knob determines how long a segment is replayed, and the RATE knob determines how frequently the stutter is triggered. The DECAY knob reduces the level of the replayed grain each time it repeats, causing it to fade.

Knob 1 — SIZE: sets the length of the grain that is replayed, from approximately 1 millisecond to 100 milliseconds.

Knob 2 — RATE: controls how frequently the stutter is triggered.

Knob 3 — DECAY: controls how much the grain's level decreases with each repetition.

EROSION

The EROSION algorithm adds band-pass filtered noise to the signal, with the amount of noise modulated by the input signal's amplitude. The result is an effect that adds grit and texture to louder signals while leaving quieter passages more nearly unaffected. The algorithm is inspired by Max/MSP's erosion~ object.

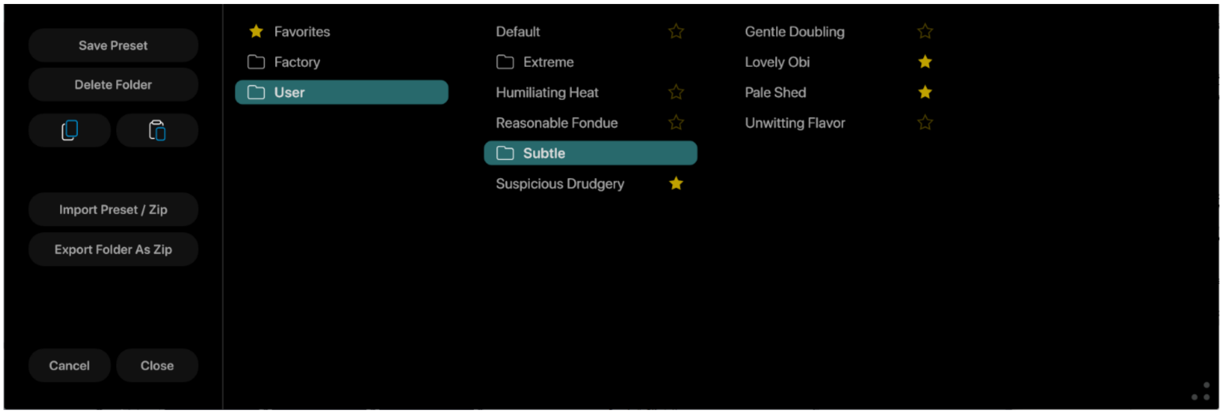
Knob 1 — FREQ: sets the center frequency of the band-pass filter applied to the noise, from 200 Hz to approximately 15 kHz.

Knob 2 — AMOUNT: controls how much filtered noise is added to the signal.

Presets

Kombinat includes a collection of presets to serve as a demonstration of its capabilities and inspirations for your own creations. There are a few controls at the top of the window associated with presets:

The name of the current preset appears in the center. (You probably figured that out yourself.) Clicking the little dots on the left and right loads presets in alphabetical order. Clicking the star outline marks the preset as a favorite to help you find it again in the future. To examine all of the presets, click the name of the current to open the preset browser.



The browser displays presets and folders in scrollable lists, arranged in columns. The leftmost list shows the folders within Kombinat's preset collection, grouped in two categories: Factory and User. Clicking any of these folders reveals its contents in the next list. Clicking on a preset name loads the settings into Kombinat. Click the Close button in the preset browser to dismiss it. If you click the Cancel button instead, the browser closes and Kombinat's settings revert to their previous state. Double-clicking a preset name loads the preset and dismisses the preset browser. Loading a preset irretrievably erases the current settings, so if you have created something that you want to use again, save it as a new preset before loading another preset.

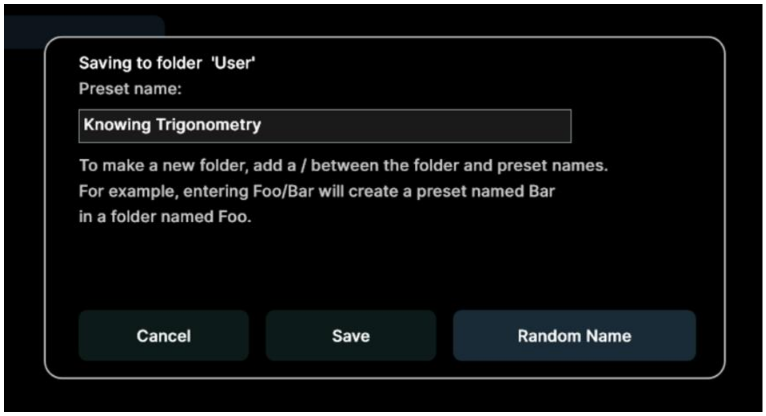
Once you have clicked on any item in the panel, you can navigate within the preset browser with the keys on your keyboard. The left and right arrow keys move the selection between columns, and the up and down arrow keys move it within the list. Tapping the ESC key has the same effect as clicking the Cancel button.

Just like the star at the top of the window which we mentioned previously, the stars to the right of preset names mark presets as favorites. Clicking a filled start removes the mark from a favorite. Once you've marked at least one favorite, a correspondingly named category appears in the leftmost column. Clicking it shows you all the presets you've marked, and clicking their names loads them as usual.

The folders and presets in the browser correspond to folders and files within Kombinat's own folder on your storage device (i.e. your computer's hard drive or SSD). This folder is located at C:\ProgramData\Audio Damage\Kombinat4\ on Windows, and ~/Music/Audio Damage/Kombinat4/ on macOS and Linux. You can store your presets anywhere you like, but for them to show up in the User list they must be placed in the User folder within Kombinat's folder. Also, to avoid possible collisions during future updates, do not store your presets within the Factory folder.

Any folders you create within the User folder will show up as folders in the User list. You can create sub-folders within the User folder, and sub-folders within those folders. You can't nest folders deeper than that because the preset browser has only four lists.

To save your presets, click the Save Preset button at the left edge of the window. This invokes a dialog box with a couple of helpful features. As the text therein describes, you can create a folder within the destination folder (whose name is given at the top of the dialog box) by adding the folder's name to the beginning of the preset's name, separated by a slash mark.



Clicking the Random Name button replaces the preset's name with a pair of words chosen at random from two lists. While the resulting names won't have any connection with what the plugin is doing, you may find this feature useful for coming up with alternatives to routine names like "My Preset 12".

Potential pitfall: once you've saved a preset, clicking its name in the list loads the preset, overwriting whatever changes you've made since you saved the preset. Hence if you want to save the preset again to preserve the changes you've made, do not click on its name before saving it.

You can delete presets and folders from the lists by clicking their name and then clicking the Delete Preset or Delete Folder button. Kombinat will give you a chance to confirm this action or cancel it. If you confirm, the preset/folder will be removed from your storage system and is gone for good.

Importing and Exporting Presets

Preset files are plain-text XML files so that you can exchange them online in forums, copy them between a Windows computer and a Macintosh (and even between an iPad and a regular computer), email them to your friends, etc.

The two buttons with icons representing copying and pasting (copy on the left, paste on the right) copy Kombinat's current settings to the system clipboard and paste settings from the clipboard. You can use the copy and paste commands to transfer settings between two instances of Kombinat or paste the settings into an email message or text editor. When copied to the clipboard, presets are presented in the same XML text as used in preset files.

The Import Preset / Zip button provides a way to add presets to Kombinat without manually moving them into the appropriate folders in your file system. Clicking this button produces a file-browser window wherein you can select either a single preset file or a .zip file containing one or more presets. After you select the file, Kombinat copies the preset(s) into whichever folder you have selected in Kombinat's preset list, unzipping the file first if necessary.

Depending on whether you've selected a preset or folder, the Export Single Preset or Export Folder As Zip button performs the complementary functions of the Import button. First select either a preset or a folder in Kombinat's list, then click the export button. A file-save window appears; choose a location in your file system, give the file a name, and click Save. If you have chosen a folder in Kombinat's preset list, the plugin places it and all of the presets it contains in a .zip file.

Default Preset

If you save a preset with the special name "Default" in the User folder, new instances of Kombinat will load it automatically when you add it to your DAW session. You can use a default preset file to give yourself the same starting point with Kombinat whenever you use it. The plugin installer creates a default preset file for you but feel free to replace it with your own.

Settings

The Settings panel appears when you click the SETTINGS button in the top bar. Clicking anywhere outside the panel closes it without saving; to preserve your changes, click the SAVE button at the bottom of the panel. These settings are global preferences — they are not stored with your project and apply to Kombinat 4 across all sessions.

SHOW TOOLTIPS — enables or disables the tooltip text that appears when you hover over a control. Tooltips are on by default and are a good way to get a quick reminder of what a control does while you are learning the plug-in.

ENABLE ANIMATION — enables the animation of the spectrum analyzer display in the band area. Disabling animation reduces the CPU load of Kombinat 4's editor, which may be useful on slower computers or when running many instances simultaneously.

ENABLE GLOW EFFECTS — enables the glow rendering applied to various UI elements. Like the animation, disabling this reduces the graphical workload of the editor.

AGC ATTACK — sets the attack time of the automatic gain control, in milliseconds. This controls how quickly the AGC responds when the output level rises above its target. The range is 10 to 500 ms.

AGC RELEASE — sets the release time of the automatic gain control, in milliseconds. This controls how quickly the AGC allows the output level to rise again after a loud event. The range is 10 to 500 ms.

And Finally...

Thanks for purchasing Kombinat. We make every effort to ensure your satisfaction with our products and want you to be happy with your purchase. Please write to support@audiodamage.com if you have any questions or comments.