Dusk Audio

Plugin User Manual

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Free, professional audio plugins for VST3, LV2, and AU.

Contents

Multi-Comp

Overview

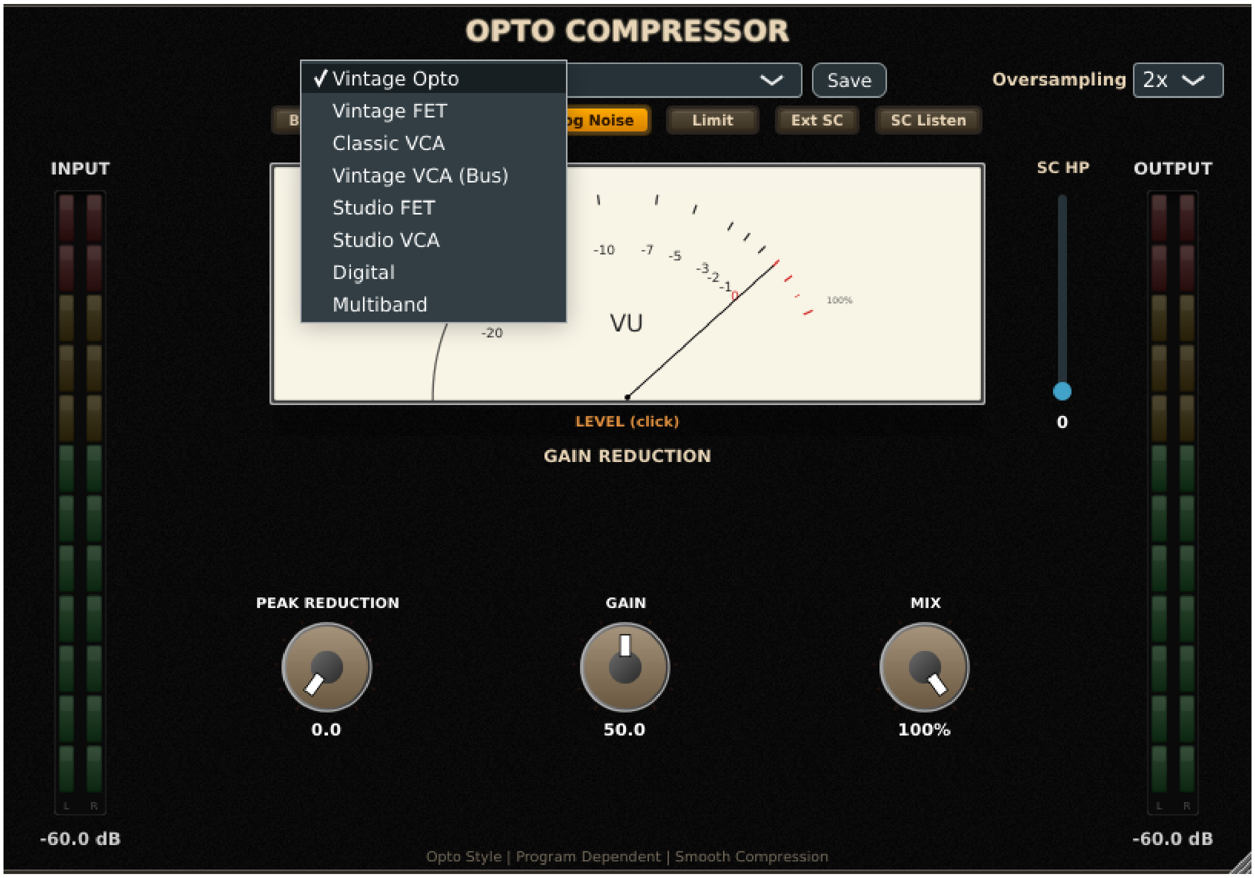
Multi-Comp is a single plugin that switches between eight different compressor flavors plus a four-band multiband. Each mode emulates a distinct hardware design, with its own attack and release behavior, knee shape, and saturation character. The eight modes are Vintage Opto, Vintage FET, Classic VCA, Vintage VCA (Bus), Studio FET, Studio VCA, Digital, and Multiband.

Use it the way you would use any compressor: control dynamic range, glue a bus, tame transients, add audible color. The mode you pick determines how it sounds doing those jobs. Vintage Opto and Vintage FET lean into their character; Studio FET, Studio VCA, and Digital stay out of the way; Vintage VCA (Bus) is purpose-built for mix-bus glue; Multiband splits the signal so you can compress problem frequencies without touching the rest.

Multi-Comp is not a transparent peak limiter (use Digital mode with a low ratio if you need that), and it is not a creative effect plugin (use the Creative preset for the closest thing to one). It is a working compressor that covers most of what a mixing engineer reaches for.

Quick Start

1. Drop Multi-Comp on a track that needs dynamic control. Vocals, bass, and drum bus are good first targets.
2. Set the **Mode** dropdown at the top. If you do not know which mode to pick, start with **Vintage Opto** for vocals and bass, **Classic VCA** for drums, **Vintage VCA (Bus)** for the mix bus.



3. Pull the **Threshold** (or **Peak Reduction** in Opto mode) until you see 3 to 6 dB of gain reduction on the meter. The meter sits to the right of the mode-specific controls. 4. Set **Ratio** to 4:1 for most sources. Ratios above 8:1 are limiting territory. 5. Listen. If the compression sounds choked, lengthen the **Release** (or pick a slower release time on Bus mode). If it sounds soft and lazy, shorten the **Attack**. 6. Use the global **Mix** knob (top of the plugin) to dial in parallel compression. 100% is fully compressed; pulling toward 0% blends the dry signal back in.

You should hear the loud parts come down in level while the quiet parts stay where they are. If you hear pumping on bass-heavy material, raise **SC HP Filter** to 80 to 120 Hz so the compressor stops reacting to the kick fundamental.

Workflows

Lead vocal with Vintage Opto

Source: A pop or rock lead vocal, recorded around -18 dBFS RMS with peaks near -6 dBFS. **Goal:** Even level across the verse and chorus, no audible pumping, the singer still feels alive.

Settings:

- **Mode:** Vintage Opto
- **Peak Reduction:** 50% (this is the sweet spot on the hardware)
- **Gain:** start at 50% (unity), raise to taste once compression is dialed in
- **Limit Mode:** Off
- **SC HP Filter:** 60 Hz
- **Mix:** 100%

Figure 1: Vintage Opto with Peak Reduction at 50%

Why this works. Vintage Opto's optical cell has a program-dependent attack and release, so it does not need you to tune times. It naturally rides level on a vocal. Peak Reduction at 50% gives 4 to 6 dB of gain reduction on a typical lead. The 60 Hz sidechain HP keeps low-frequency rumble from triggering the compressor.

If the vocal still feels uneven, push Peak Reduction toward 70%. If it starts breathing or pumping, drop to 35%.

Drum bus glue with Vintage VCA (Bus)

Figure 2: Bus Compressor with Classic Drum Glue settings

Source: Stereo drum bus with kick, snare, hats, and overheads, summed at around -10 dBFS peak. **Goal:** A cohesive drum kit with the kick and snare punching through, gentle level riding on busy fills.

Settings:

- **Mode:** Vintage VCA (Bus)
- **Threshold:** -15 dB
- **Ratio:** 4:1
- **Attack:** 30 ms (the slowest setting)
- **Release:** Auto
- **Makeup:** 3 dB
- **SC HP Filter:** 90 Hz

- **Bus Mix:** 100%

Why this works. The slowest attack (30 ms) lets the kick and snare transients pass through before the compressor clamps down, which is what makes the drums punch. Auto release adapts to the program material so you do not get pumping on busy passages. The 90 Hz sidechain HP prevents the kick fundamental from over-driving the compressor on every hit.

This is the configuration behind the "Classic Drum Glue" factory preset. If you want more obvious glue, drop Threshold to -18 dB. For a softer, more transparent version, use Ratio 2:1.

Snare punch with Classic VCA

Source: Single snare track, hit hard, peaks at -3 dBFS. **Goal:** Tighter snare with the crack preserved.

Settings:

- **Mode:** Classic VCA
- **Threshold:** -18 dB
- **Ratio:** 4:1
- **Attack:** 15 ms
- **Release:** 80 ms
- **Output:** +3 dB
- **Over Easy:** Off

Why this works. 15 ms is slow enough to pass the initial transient (the "crack") and clamp on the body of the hit. Faster attack (5 ms or less) kills the snap and makes the snare sound dull. 80 ms release recovers in time for the next hit at typical rock tempos.

For a softer, less obvious result, turn **Over Easy** on. This switches the knee from hard to soft (around 6 dB wide), so compression eases in instead of slamming.

De-essing a vocal with Multiband

Source: Lead vocal with harsh sibilance in the 6 to 9 kHz range. **Goal:** Tame the "sss" without dulling the vocal.

Settings:

- **Mode:** Multiband
- Use the **ON** strip toggles to turn off Low, Low-Mid, and Hi-Mid bands. Leave High enabled.
- **Crossover 3:** 6000 Hz (drag the rightmost crossover fader down from 8 kHz)
- **High Threshold:** -28 dB
- **High Ratio:** 6:1
- **High Attack:** 1 ms
- **High Release:** 80 ms
- **MB Mix:** 100%

Why this works. With three bands disabled, the multiband splitter collapses so the High band covers everything from 6 kHz up. A fast attack (1 ms) catches sibilance before it gets loud. Ratio 6:1 is aggressive, but the narrow band limits the audible damage. Watch the High band's gain reduction meter; aim for 4 to 8 dB on the loudest "sss" and near zero on everything else.

Figure 3: Multiband mode default 4-band layout

Figure 4: Multiband collapsed to 1 active band

This same pattern works for kick mud (enable just the Low band, set Crossover 1 around 150 Hz) and for harsh cymbals (just the High band, Crossover 3 around 8 kHz).

Parameter Reference

Global controls (top of the UI)

- **Mode:** Picks the compressor flavor. Eight choices. Mode-specific controls below the drop-down change to match.
- **Bypass:** Reports zero latency to the host while bypassed (no PDC offset on parallel buses).
- **Mix:** 0 to 100%, default 100%. The dry/wet blend at the plugin output. Pull below 100% for parallel compression.
- **Stereo Link:** 0 to 100%, default 100%. How tightly the two channels share gain reduction. 100% means both channels react to the loudest channel; 0% is fully independent (dual mono). Lower values widen perceived stereo on busy material.
- **Link Mode:** Stereo, Mid-Side, or Dual Mono. Mid-Side processes the M and S channels independently; useful for tightening center elements without affecting width.
- **Auto Makeup:** Off or On. When On, automatic level compensation tries to keep output loudness similar to input.
- **SC HP Filter:** 0 to 500 Hz, default 0 (Off). Highpass on the sidechain only (does not affect output). Use 60 to 120 Hz on bass-heavy material to stop low-frequency pumping.
- **Lookahead:** 0 to 10 ms, default 0. Delays the audio by the lookahead time so the detector sees transients before they hit the compressor. Adds latency to the host.
- **Oversampling:** Off, 2x, or 4x, default 2x. Reduces aliasing from the saturation stage. 4x costs about twice the CPU of 2x.
- **External Sidechain:** Off or On. Routes a separate sidechain input from the host (DAW-specific routing required).
- **SC Listen:** Off or On. Sends the sidechain signal to the output for monitoring; turn off before printing.

Mode-specific notes

- **Vintage Opto:** Peak Reduction (0 to 100%) is the only compression control. Limit Mode toggles between gentle (off) and aggressive (on).
- **Vintage FET:** Ratio is fixed in steps (4:1, 8:1, 12:1, 20:1, All). All-Buttons enables the famous extreme compression with non-linear character; Curve Mode picks Modern (the original algorithm) or Measured (matches hardware capture). Transient (0 to 100%) blends raw transient through the compressor.
- **Classic VCA:** Wide threshold range (-38 to +12 dB) and ratio up to 120:1 (limiting). Over Easy switches to soft knee.
- **Vintage VCA (Bus):** Discrete attack (0.1, 0.3, 1, 3, 10, 30 ms) and release (0.1, 0.3, 0.6, 1.2 s, Auto). 30 ms attack plus Auto release is the classic glue setting.
- **Studio FET:** Same control set as Vintage FET but voiced cleaner with less harmonic content.
- **Studio VCA:** Continuous controls, ratio up to 10:1, voiced for transparent dynamics work.
- **Digital:** Continuous everything, with adjustable Knee width (0 to 20 dB) and Adaptive Release (program-dependent). Use this when you want precision.
- **Multiband:** Four bands (Low, Lo-Mid, Hi-Mid, High) with three Linkwitz-Riley 4th order crossovers. Each band has its own threshold/ratio/attack/release/makeup, plus solo and ON. Disabling a band collapses the splitter so the band's frequency range merges into the nearest enabled neighbor (a real two-band or three-band layout, not just a bypass).

Tips and Traps

- **Auto Makeup hides level changes.** When you compare presets or A/B settings, Auto Makeup can mask whether you actually like the compression or just the level boost. Turn it off when auditioning.
- **Oversampling costs CPU.** 2x is the default and is plenty for most material. Drop to Off for tracking; raise to 4x only on a final mix bus where the saturation is doing audible work.
- **Mode switching does not preserve compression amount.** Each mode has its own threshold scale (Opto's Peak Reduction at 50% is not the same as Digital's threshold at -20 dB). Re-set the compression amount when you switch modes.
- **Lookahead adds latency.** The host applies plug-in delay compensation (PDC), so your other tracks stay aligned, but live monitoring through the plugin will sound late. Bypass during tracking.
- **Multiband bands cannot drop below two enabled.** The plugin enforces a minimum of two active bands. Trying to disable a third band locks the toggle.
- **The Bus Compressor's "Auto" release is the magic setting** for glue. If you can't hear the difference between Auto and 0.6 s, listen during a sparse-to-busy transition; that's where Auto earns its keep.

Presets Explained

Figure 5: Preset menu open showing the 6 categories

Multi-Comp ships with 13 factory presets across six categories. Each is a starting point, not a finished sound; expect to nudge threshold and makeup once it is on your source.

Vocals

- **Smooth Opto Vocal.** Vintage Opto, Peak Reduction 50%, SC HP at 60 Hz. The classic optical vocal sound; reach for this on a lead vocal that needs level control without character changes.
- **Vocal Presence.** Vintage FET, Ratio 4:1, fast attack (~0.5 ms), 60 ms release, SC HP at 100 Hz. Adds the FET-style aggression and presence; good for rock and indie vocals where you want the compressor heard.
- **Modern Pop Control.** Studio FET, Ratio 8:1, very fast attack (0.3 ms), 120 ms release, Auto Makeup on. Tight, transparent peak control for pop and contemporary genres.

Drums

- **Classic Drum Glue.** Vintage VCA (Bus), Ratio 4:1, 30 ms attack, Auto release, SC HP at 90 Hz. The standard drum-bus setting; let the kick and snare punch through, glue the rest.
- **Room Nuke (FET All).** Vintage FET with All-Buttons mode, threshold -24 dB, slightly slower attack to develop the famous plateau. Use on parallel room mics for the smashed-room effect; not a primary compressor.
- **Snare Snap.** Classic VCA, Ratio 4:1, 15 ms attack, lets the snare crack pass before clamping. Apply per-snare, not on the bus.

Bass

- **Rock Bass Anchor.** Vintage FET with a slower attack to keep sub frequencies clean.
- **Vintage Pinned Bass.** Vintage Opto with maximum Peak Reduction for the classic Motown “pinned” feel.

Guitars

- **Acoustic Strum Tamer.** Classic VCA with a fast attack to control strumming spikes.
- **Funk Rhythm Guitar.** Vintage FET with a fast release; emphasizes the “up” stroke.

Mix Bus

- **Console-Style Glue.** The 4:1, 10 ms attack, Auto release, 4 dB GR setting; the canonical mix-bus compressor sound.
- **Gentle Master.** Studio VCA at Ratio 1.5:1; a low-ratio, modern mastering-bus starting point.

Creative

- **EDM Pump (115-130 BPM).** Heavy compression with a release tuned to quarter notes in the 115 to 130 BPM range; the “side-chain pump” effect without an actual sidechain.

Troubleshooting

I hear nothing. Check the Bypass button at the top, check Mix is not at 0%, and confirm your DAW track output is not muted. If on Multiband mode, the GR meter shows the loudest band; if all bands show zero GR even on a loud signal, your Threshold is set above the input level.

The compressor is pumping on bass-heavy material. Raise SC HP Filter to 80 to 120 Hz. The detector stops reacting to the kick fundamental.

The CPU usage is high. Drop Oversampling from 4x to 2x or Off. Disable True Peak detection if it is on (it adds a 4x or 8x detector path). Multiband at 4x oversampling on a busy bus is the worst-case combination.

The audio sounds late through the plugin. Lookahead and Digital mode's internal lookahead both delay output. Your host applies PDC for playback alignment, but live monitoring is still delayed. Bypass during tracking.

The mode dropdown changes my sound dramatically. That is intentional. Each mode has its own threshold scale, attack and release behavior, and saturation profile. After switching, re-set the compression amount on the new mode's controls.

Multi-Q

Overview

Multi-Q is three EQs in one plugin. The mode switch at the top picks between **Digital** (an 8-band parametric with per-band dynamic EQ), **British** (a four-band console EQ with Brown and Black character options), and **Tube** (a vintage passive tube-style EQ with the classic "boost and cut at the same frequency" interaction). Each mode is voiced and operated differently; switching modes is a creative choice as much as a workflow choice.

Use Digital for surgical work and dynamic frequency control. Use British for tonal shaping with console color (it is the same engine as 4K-EQ, integrated). Use Tube for vintage warmth and the unique passive-EQ curves that interact between bands. The real-time spectrum analyzer above the curve display works across all three modes.

It is not a corrective de-noise tool, and it is not a mastering limiter; pair it with a compressor or limiter for level work. It is a clean, capable EQ that swaps personalities at will.

Quick Start

1. Insert Multi-Q on the source you want to shape. Pick a mode at the top: **Digital**, **British**, or **Tube**.
2. The main display shows your EQ curve overlaid on a real-time spectrum. Click and drag a band's control point to set frequency (horizontal) and gain (vertical).
3. Mouse-wheel over a control point to adjust Q (band width).
4. Right-click a control point for band-specific options (filter shape, dynamic EQ activation in Digital mode, Brown/Black switch in British mode).
5. Use the **Bypass** button to A/B against unprocessed audio.
6. The **A/B** preset slots at the top let you compare two settings: load Multi-Q, set up your EQ on slot A, click **B** to switch to a fresh slot, build a different EQ, then toggle between A and B by clicking the letter.

You should hear the source change tonal balance immediately. If the band you are dragging produces no audible change, check the Q (very high Q with low gain is nearly inaudible) and confirm the band is enabled (in Digital mode, the band-strip toggles below the curve let you disable individual bands).

Workflows

Digital mode: surgical resonance notch

Source: A snare drum with a ringy 380 Hz resonance. **Goal:** Remove the ring without dulling the snare.

Settings:

- **Mode:** Digital
- Choose any unused parametric band. Drag its control point to 380 Hz.
- **Filter shape:** Bell (default for parametrics)
- **Gain:** -8 dB
- **Q:** 6 (narrow)

Figure 6: Multi-Q in Digital mode

Why this works. A narrow bell at the offending frequency cuts only the resonance and leaves the rest of the snare unchanged. To find the exact frequency, set Gain to +8 dB temporarily, sweep the band's frequency, and listen for where the ringing gets worse; that is your target. Then flip Gain to -8 dB.

For an even more surgical version, switch the band's filter shape to **Notch** (right-click). A notch is sharper than a high-Q bell and removes the resonance entirely.

Digital mode: dynamic de-essing

Source: A vocal with sibilance peaks in the 6 to 8 kHz range. **Goal:** Reduce the sibilance only when it gets loud, leave the rest of the vocal alone.

Settings:

- **Mode:** Digital
- Pick the High Shelf band or any parametric band. Drag to 7000 Hz.
- **Filter shape:** Bell
- **Gain:** 0 dB (the static cut is zero; dynamics will do the work)
- **Q:** 1.5

Right-click the band to access **Dynamic EQ** controls:

- **Threshold:** -28 dB
- **Range:** -8 dB (max cut applied when threshold is exceeded)
- **Attack:** 1 ms
- **Release:** 80 ms

Figure 7: Dynamic EQ controls on a band

Why this works. The static gain stays at 0, so non-sibilant content passes through untouched. When sibilance pushes the band's energy above -28 dB, the dynamic EQ cuts up to 8 dB at that frequency. Fast attack catches the transient; release back to zero in 80 ms. The "Multiband De-Ess" factory preset uses this approach.

British mode: vocal sweetening with console character

Source: A lead vocal that needs presence and warmth without surgical cuts. **Goal:** Tonal balance with audible analog character.

Settings:

- **Mode:** British
- **EQ Type:** Brown (E-series) for the warmer character
- **HPF:** On, 80 Hz (cleans up rumble)
- **LF Gain:** -3 dB at 300 Hz, Bell mode
- **HM Gain:** +4 dB at 3500 Hz, Q 0.7
- **HF Gain:** +2 dB at 8000 Hz, Shelf
- **Saturation:** 15%

Why this works. British mode in Multi-Q runs the same engine as 4K-EQ, so the "Vocal Presence" approach from 4K-EQ applies directly. The 300 Hz cut clears boxiness, the 3.5 kHz boost adds intelligibility, the 8 kHz shelf adds air. Saturation at 15% adds the slight harmonic content that makes a vocal sit in a mix without surgical work. For a more aggressive character, switch to Black mode.

Figure 8: Multi-Q in British mode

Figure 9: Multi-Q in Tube mode with boost+cut at 60 Hz

Tube mode: vintage bass with the boost-and-cut trick

Source: Bass guitar that needs both fullness and clarity. **Goal:** Add low-frequency body and tighten the muddy lower-mids.

Settings:

- **Mode:** Tube
- **LF Boost:** 60 Hz, +4 dB
- **LF Attenuation:** 60 Hz, +3 dB (boost AND cut at the same frequency)
- **HF Boost:** 5000 Hz, +2 dB
- **Tube Drive:** 25%

Why this works. The vintage passive EQ trick is to boost and cut at the same low frequency. The interaction of the two passive curves creates a unique shape: more energy at the boost peak (around 60 Hz) AND a slight dip just above it (where the boost rolls off and the cut kicks in). The result is a punchy low end that does not muddy the mids. Tube Drive at 25% adds the harmonic warmth of the simulated 12AX7 stage.

This is not a digital trick; it mimics how real passive tube EQs behave. The "Vintage Bass Trick" factory preset shows this exact approach.

Parameter Reference

Mode-independent (always visible)

- **Mode:** Digital, British, or Tube. Each mode has its own parameter set; the visible controls change when you switch.
- **Bypass:** Reports zero latency to the host while bypassed.
- **Output Gain:** Final makeup gain.
- **Auto Gain Compensation:** Keeps output level matched to input as you boost and cut.
- **Spectrum Analyzer Pre/Post:** Pre shows the input signal; Post shows the EQ output.
- **Oversampling:** 2x or 4x. Reduces aliasing in saturation stages (British and Tube modes).
- **A/B Slots:** Two complete parameter snapshots accessible from the top toolbar. Useful for comparing two different EQ approaches on the same source.

Digital mode

8 fully parametric bands (HPF at the low end, Low Shelf, four mid parametrics named Low / Lo-Mid / Mid / Hi-Mid, High Shelf, LPF at the high end). Each band has:

- **Frequency:** 20 Hz to 20 kHz (range varies by band).
- **Gain:** -24 to +24 dB.
- **Q:** 0.1 to 36 (effectively notch-narrow at the high end).
- **Filter shape:** Bell, Shelf, Pass, Notch, or Band Pass (choices vary by band).
- **Enabled:** Per-band toggle in the strip below the curve.
- **Dynamic EQ:** Threshold (-60 to 0 dB), Attack (0.1 to 100 ms), Release (10 to 1000 ms), Range (-24 to +24 dB). Right-click a band to expand the dynamic controls.

The HPF and LPF have selectable slopes (12, 18, or 24 dB/oct). The mid parametrics support Notch and Band Pass shapes for surgical work.

British mode

Same controls as 4K-EQ:

- **HPF / LPF:** Frequency and enabled toggles.
- **Four bands** (LF, LM, HM, HF): gain, frequency, Q (LM/HM only); LF and HF have Bell/Shelf switches.
- **EQ Type:** Brown or Black.
- **Saturation:** 0 to 100%.
- **M/S Mode:** Off or On.

See the 4K-EQ chapter for detailed parameter notes; the engine is identical.

Tube mode

Figure 10: Tube mode controls

- **LF Boost:** Frequency selectable (typically 20, 30, 60, 100 Hz), gain 0 to +12 dB.
- **LF Attenuation:** Frequency selectable, gain 0 to +12 dB (cut amount; passive design).
- **HF Boost:** Frequency selectable (3, 5, 10, 16 kHz), gain 0 to +12 dB, with **Bandwidth** control (broad or narrow).
- **HF Attenuation:** Frequency selectable, gain 0 to +12 dB.
- **Mid Dip / Peak:** A center-frequency band with selectable curve shape; useful for vocal presence shaping.
- **Tube Drive:** 0 to 100%. Drives the simulated 12AX7 triode stage. 0 is clean; 30 to 50% adds audible second-harmonic warmth.

Boost and Attenuation at the same frequency interact in the passive style: the curve becomes neither pure boost nor pure cut but a unique interaction shape.

Tips and Traps

- **Modes do not share band settings.** A 4 kHz boost in Digital mode is not present when you switch to British, even though the curves of both can pass through 4 kHz. Each mode has its own parameter set; the mode switch is a hard switch, not a re-mapping.
- **Right-click for hidden options.** Filter shape, dynamic EQ, M/S routing per-band, all live in the right-click menu. Beginners often miss them and miss half the plugin's capability as a result.
- **The boost-and-cut trick is mode-specific.** Only Tube mode has that interaction. In Digital or British modes, boosting and cutting at the same frequency cancels out (or nearly so).
- **A/B is your friend on long EQ sessions.** Set A as your starting point. Build B to compare. If neither sounds right, click A again to reset. The A/B slots are persistent within the session; they do not save with presets.
- **British mode shares the 4K-EQ engine.** Settings and behavior are identical. If you have a 4K-EQ preset you like, you can recreate it in Multi-Q's British mode by hand.
- **The spectrum analyzer is a guide, not a sound.** Pre/Post toggle shows you what changed; do not let the pretty curve trick you. Listen first, look at the spectrum to confirm.

Presets Explained

Multi-Q ships with around 50 factory presets across roughly 8 categories. They are organized by source (Vocals, Drums, Guitars, Bass, Mix Bus), by mode (each mode contributes presets), and by use case (Mastering, Creative, Dynamic, M/S). Listing every one would balloon this manual; instead, here are the themes you should know about.

Vocals presets

Multiple presets across all three modes. Digital-mode "Vocal Presence" uses static EQ for clarity and air. British-mode "Console Vocal Chain" applies the 4K-EQ Vocal Presence approach plus saturation. Tube-mode "Warm Vocal (Tube)" uses LF boost-and-cut for body and Tube Drive for harmonic warmth. Pick by what character you want; all three are valid starting points for a lead vocal.

Drums presets

Per-source: "Punchy Kick", "Snare Crack", "Overhead Clarity", and "Rock Drums" (a British-mode bus preset). Digital-mode kick presets often pair a low-shelf boost with a narrow mid cut for click. British-mode drum presets add saturation. Tube-mode drum presets are rare; tube character is usually too colored for kit work.

Mastering presets

"Mastering Surgical" uses Digital with narrow bells; "Mastering Air" lifts the top end with shelves; "Mastering Wide" uses M/S processing for stereo width. "Vintage Air" and "Console Warmth" sit in British and Tube modes respectively for analog flavor on a master.

Dynamic-EQ presets (Digital mode only)

"Multiband De-Ess" applies dynamic cut at sibilance frequencies. "Dynamic Bass Control" tames bass build-up dynamically. "Resonance Tamer" is a starting point for cutting unwanted resonances that only show up at higher levels. These presets demonstrate per-band dynamic EQ; right-click a band on any of them to see how the dynamics are wired.

Creative presets

"Telephone Effect" (narrow band-pass for radio voice), "Lo-Fi Warmth" (heavy filtering with British-mode saturation), "Stereo Width" (M/S width tricks). Use these as effects rather than starting points for serious mixing.

Vintage presets (Tube mode)

"Vintage Bass Trick" demonstrates the boost-and-cut at the same frequency. "Vintage Air" shelves the top end gently. "Warm Vocal (Tube)" combines LF boost-and-cut with Tube Drive. These are the showcases for what Tube mode can do that the other modes cannot.

If you want to study the presets, open one and right-click each band to inspect its settings. The preset names suggest use cases; the band-by-band layout shows the technique.

Troubleshooting

The plugin sounds the same in all three modes. Confirm you are actually switching modes (the curve display changes when you do; the band labels change too). With all bands flat, the difference between modes is only the saturation and the filter topology of any active filters. To hear the difference, boost or cut a band in each mode and compare.

The spectrum analyzer shows different curves than the EQ shape. That is intentional. The analyzer shows actual frequency content; the EQ curve shows the filter response. They are different views. Pre/Post toggle on the analyzer compares input to output.

Right-click does nothing. On Linux, some hosts intercept right-click for their own menus. Try Ctrl+click or check your host's preferences for plugin context-menu handling.

My British mode preset sounds different from a 4K-EQ preset with the same settings. The engines should be identical. The most likely cause is sample-rate handling or auto-gain compensation; confirm both plugins are at the same sample rate and either both have Auto Gain on or both off.

4K-EQ

Overview

Figure 11: 4K-EQ main UI

4K-EQ is a 4-band parametric equalizer modeled on the British large-format console channel strip. The four bands (Low, Low-Mid, High-Mid, High) cover the full audio spectrum, with separate high-pass and low-pass filters at the input. Two character modes ("Brown" and "Black") swap the EQ curves between the E-series and G-series console designs.

Use it where you would use a console EQ: tonal shaping during the mix, gentle pre-master sweetening, broad corrective work on individual sources. The bands are musical rather than surgical; if you need to notch out a single resonant peak, reach for Multi-Q's Digital mode instead.

The console-style saturation stage adds a small amount of harmonic color when driven, and stays clean when the input gain is conservative. Auto-gain compensation keeps the output level matched to input as you boost and cut, so an A/B comparison reflects tone changes only.

Quick Start

1. Insert 4K-EQ on the source you want to shape. The interface shows four band sections (LF, LM, HM, HF) with HPF on the far left and LPF on the far right.
2. Pick a character with the **EQ Type** switch at the top: **Brown** for the warmer, slightly soft E-series sound, **Black** for the more aggressive G-series with extended highs and lows.

3. Sweep one band's **Frequency** while you boost its **Gain** by about 6 dB. Listen for the area you want to enhance, then back the gain off to 2 to 4 dB.
4. Use the **Q** knob (on LM and HM) to widen or narrow the affected range. The default Q (0.7) is broad and musical; raise toward 2 for surgical work.
5. The **HPF** and **LPF** are off by default. Toggle them on to clean up rumble (HPF at 60 to 100 Hz on most sources) or tame harshness (LPF only when needed).
6. Leave **Auto Gain Compensation** on while you work. It keeps levels matched so your ears focus on tone, not loudness.

You should hear the source change tonal balance without changing apparent level. If you do hear a level change, double-check Auto Gain is on and that you have not pushed any band past 12 dB.

Workflows

Vocal presence and clarity

Figure 12: Vocal Presence preset loaded

Source: A lead vocal that sits in the mix but needs more presence and air. **Goal:** Lift the singer forward without harshness.

Settings:

- **EQ Type:** Brown

- **HPF:** On, 80 Hz
- **LF Gain:** -3 dB at 300 Hz, Bell Mode On
- **HM Gain:** +4 dB at 3500 Hz, Q 0.7
- **HF Gain:** +2 dB at 8000 Hz, Shelf
- Other bands flat.

Why this works. The HPF at 80 Hz removes proximity rumble and headphone bleed without thinning the body. The LF cut at 300 Hz tames the boxy buildup that lives there in close-mic'd vocals. The HM boost at 3.5 kHz adds intelligibility (the consonant range). The 8 kHz shelf adds air. The Brown character keeps the boost gentle. This matches the "Vocal Presence" factory preset.

If the vocal sounds harsh after this, drop HM Q to 0.5 (wider, softer boost) or move the HM frequency down to 2.5 kHz. If it still sounds dull, try Black mode for a touch more clarity.

Kick drum punch

Source: Close-mic'd kick drum. **Goal:** Solid low-end thump and audible click for cut-through.

Settings:

- **EQ Type:** Brown
- **HPF:** On, 30 Hz (removes sub-rumble below the fundamental)
- **LF Gain:** +4 dB at 50 Hz, Shelf
- **LM Gain:** -2.5 dB at 200 Hz, Q 0.8
- **HM Gain:** +3 dB at 2000 Hz, Q 1.5
- **HF Gain:** flat
- **Saturation:** 10%

Why this works. The 50 Hz shelf adds weight; bell mode at the same frequency would only add to a narrow range, and you want the whole low end lifted. The 200 Hz cut removes "boxy" buildup that competes with the bass. The 2 kHz bell with a tight Q (1.5) brings out the beater click. A small amount of console saturation (10%) thickens the body without obvious distortion. This matches the "Kick Punch" factory preset.

For a thumpier kick, raise LF center to 70 Hz. For more click, raise HM gain to +5 dB.

Mix bus glue

Source: Stereo mix bus, near final balance. **Goal:** Add console color and gentle high-end sheen without changing the mix.

Settings:

- **EQ Type:** Black (G-series for slightly tighter low end)
- **LF Gain:** +2 dB at 60 Hz, Shelf
- **HM Gain:** -2 dB at 2500 Hz, Q 0.8
- **HF Gain:** +2.5 dB at 10000 Hz, Shelf
- **Saturation:** 20%
- **Auto Gain Compensation:** On

Why this works. A wide 2 dB shelf at 60 Hz adds weight without muddying. A gentle 2 dB scoop at 2.5 kHz pulls the harshness band back, which can fight a mix when many instruments live in that range. The 10 kHz shelf adds gloss. 20% saturation gives the bus the harmonic glue of a real console without obvious distortion. This is the “Bright Mix” factory preset’s approach. The “Glue Bus” preset uses similar settings with a slightly higher saturation and a touch more low end.

Parameter Reference

Filters (HPF and LPF)

- **HPF Frequency:** 20 to 500 Hz, default 20. The corner frequency of the high-pass filter (18 dB/oct). Use 60 to 120 Hz on vocals and most instruments to remove rumble.
- **HPF Enabled:** Off by default. Truly bypassed when off (no phase or magnitude effect on the signal).
- **LPF Frequency:** 3 to 20 kHz, default 20 kHz. The corner frequency of the low-pass filter (12 dB/oct).
- **LPF Enabled:** Off by default. Use sparingly; over-aggressive low-pass filtering dulls a track.

Low frequency band (LF)

- **LF Gain:** -20 to +20 dB, default 0. Hardware spec is ±15 dB on Brown and ±18 dB on Black, but the plugin lets you go further when you need to.
- **LF Frequency:** 30 to 480 Hz, default 100. Center frequency for bell mode, corner frequency for shelf mode.
- **LF Bell Mode:** Off (shelf) by default. Shelf mode boosts or cuts everything below the frequency; bell mode affects only a band around the frequency.

Mid bands (LM and HM)

- **LM Gain / HM Gain:** -20 to +20 dB. Always bell-shaped (no shelf mode on the mid bands).
- **LM Frequency:** 200 to 2500 Hz, default 600.
- **HM Frequency:** 600 to 7000 Hz, default 2000.
- **LM Q / HM Q:** 0.4 to 4.0, default 0.7. Lower Q is wider and more musical; higher Q is narrower and more surgical.

High frequency band (HF)

- **HF Gain:** -20 to +20 dB.
- **HF Frequency:** 1500 to 16000 Hz, default 8000. Black mode extends further to 16 kHz; Brown is more focused around 7 kHz.
- **HF Bell Mode:** Off (shelf) by default.

Global

- **EQ Type:** Brown (E-series) or Black (G-series). Brown is warmer and slightly softer; Black is brighter with extended low and high response.
- **Bypass:** Reports zero latency to the host while bypassed.
- **Input Gain / Output Gain:** -12 to +12 dB. Use Input Gain to drive the saturation stage harder; Output Gain to compensate.
- **Saturation:** 0 to 100%, default 0. Console-style harmonic distortion. Stays clean at 0; adds warmth at 10 to 25%; gets obviously colored above 50%.
- **Oversampling:** 2x or 4x, default 2x. 4x reduces aliasing in the saturation stage at the cost of more CPU.
- **Spectrum Pre/Post:** Default Post. The spectrum analyzer view shows audio after the EQ; flip to Pre to see the input signal.

- **Auto Gain Compensation:** On by default. Keeps output loudness matched to input so EQ comparisons are level-matched.

Tips and Traps

- **Brown vs Black is not a "better" choice.** They are voiced differently. Brown is the smoother, more musical option for vocals and acoustic sources. Black is more aggressive and works well on drums, bass, and full mixes. Try both on the same source and pick what your ears prefer.

- **Bell vs shelf changes the character a lot.** A shelf at 100 Hz at +4 dB lifts everything below 100 Hz; a bell at 100 Hz at +4 dB only lifts a narrow range around 100 Hz. The wrong choice can sound boxy or thin. - **Saturation is gain-driven.** With Input Gain at 0 dB and Saturation at 50%, you may hear very little change. Push Input Gain to +6 to +9 dB and use Output Gain to compensate; the saturation will be much more apparent.

- **Auto Gain Compensation lies a little.** It level-matches RMS, but EQ changes can shift perceived loudness even at matched RMS. Trust your ears, and disable Auto Gain temporarily if you want to confirm an absolute level change. - **The HPF and LPF are sharp.** 18 dB/oct on the HPF will be obvious; do not set the HPF too high on bass instruments.

Presets Explained

4K-EQ ships with 14 factory presets across 7 categories. Each one is a starting point; expect to nudge frequencies and gains based on your source.

Vocals

- **Vocal Presence.** Brown mode, +4 dB HM at 3.5 kHz, +2 dB HF shelf at 8 kHz, gentle 300 Hz cut, HPF at 80 Hz. The classic vocal-forward setting.

Drums

- **Kick Punch.** +4 dB LF shelf at 50 Hz, -2.5 dB at 200 Hz, +3 dB at 2 kHz with tight Q. Body, mud cut, click.
- **Snare Crack.** Bell-shaped boost at 250 Hz for body, 5 kHz for crack, plus 8 kHz shelf for attack.
- **Drum Bus Punch.** Wide LF lift at 70 Hz, 350 Hz cut, 3.5 kHz boost, gentle 10 kHz shelf, 25% saturation, Black mode.

Bass

- **Bass Warmth.** +4 dB shelf at 80 Hz, slight 400 Hz cut, broad 1.5 kHz bell. For round, full bass.
- **Bass Guitar Polish.** Tighter low end (60 Hz shelf), 250 Hz cut, presence boost at 1.2 kHz, slight 4.5 kHz top.

Guitar

- **Acoustic Guitar.** Slight 100 Hz cut for boxiness, 2.5 kHz boost for fingernoise clarity, 12 kHz shelf for sparkle.

Keys

- **Piano Brilliance.** 80 Hz shelf for foundation, 500 Hz cut to clear the mids, 2 kHz and 8 kHz boosts for definition and shine.

Mix Bus

- **Bright Mix.** Wide 60 Hz shelf, gentle 2.5 kHz scoop, 10 kHz shelf with 20% saturation. Adds clarity without changing the mix.
- **Glue Bus.** Similar to Bright Mix with slightly more low end and saturation. The "console color" preset.

Creative

- **Telephone EQ.** Aggressive HPF and LPF (300 Hz and 3 kHz) with a 1 kHz bell boost. Lo-fi telephone effect.
- **Air & Silk.** Top-end-only setting: 7 kHz boost, 15 kHz shelf. Adds air without touching the body.

Mastering

- **Master Sheen.** Subtle 5 kHz and 16 kHz boosts, 10% saturation. Adds polish without changing the mix.
- **Master Bus Sweetening.** Gentle 50 Hz lift, 600 Hz cut, 4 kHz and 15 kHz boosts, 15% saturation, slight input cut. The most subtle preset; use as a starting point for mastering.

Troubleshooting

The EQ has no audible effect. Check that **Bypass** is off and the band you are adjusting has its **Gain** set to something other than 0 dB. The HPF and LPF must be **Enabled** to do anything; they are off by default.

Boosting any band makes the mix louder. That is expected when **Auto Gain Compensation** is off. Turn it on to compare EQ shapes at matched levels. If Auto Gain is on and you still hear level changes, the EQ is shaping the spectrum in a way that affects perceived loudness even at matched RMS.

The saturation knob seems to do nothing. The saturation stage is gain-driven; it only colors when the signal is hot enough to drive it. Turn up **Input Gain** by 6 to 9 dB and compensate with **Output Gain**, or feed the plugin a hotter input signal.

TapeMachine

Overview

Figure 13: TapeMachine main UI

TapeMachine emulates a professional reel-to-reel tape machine. Two machine models (Swiss 800 and Classic 102) with four tape formulations (Type 456, GP9, 911, 250), three speeds (7.5, 15, 30 IPS), and two EQ standards (NAB, CCIR) cover most of what you would find in a 1970s to 1990s pro studio. On top of that you get separate wow and flutter controls, switchable noise, oversampled saturation, and a four-position signal path so you can hear just the electronics, just the tape, the full chain, or true bypass.

Use it where you would use a real tape machine: subtle bus glue, vocal warmth, drum bus character, gritty character on guitars, or full lo-fi treatment. The machine model and speed affect the high-frequency response (head bump and roll-off); the tape formulation affects the saturation curve and noise floor; bias and calibration determine how hot you can drive before the sound breaks down.

It is not a chorus, flanger, or pitch-shift plugin. The wow and flutter are subtle modulations of the playback speed and behave like real tape; if you want extreme pitch effects, use a dedicated modulation plugin.

Quick Start

1. Insert TapeMachine on a track or bus. Vocal busses, drum busses, and the master are all common targets.
2. Leave the defaults for your first listen: **Swiss 800**, **15 IPS**, **Type 456**, **Repro** signal path, **NAB** EQ, **Auto Calibration** on, **Saturation** 4%.
3. Compare bypassed against active. You should hear a subtle high-frequency softening and a touch of harmonic warmth.
4. Push **Input Gain** by 4 to 6 dB to drive the saturation harder. Compensate with **Output Gain** to keep levels matched.
5. Try the other machine: switch **Tape Machine** to **Classic 102** for a different head response and more pronounced character.
6. If you want the full vintage effect, raise **Wow** to 10 to 15%, **Flutter** to 5 to 8%, and toggle **Noise Enabled** on with **Noise Amount** around 10%.

You should hear the difference immediately on transients (slightly softer attack), on highs (gentle roll-off above 12 to 15 kHz), and on overall tone (subtle low-mid warmth from the head bump). With Saturation above 10% and Input Gain pushed, the harmonic content becomes audible.

Workflows

Subtle warmth on a vocal

Source: A clean studio vocal that needs analog character without coloration. **Goal:** Slight warmth and softening, no obvious tape sound.

Settings:

- **Tape Machine:** Swiss 800
- **Tape Speed:** 30 IPS (cleanest, widest frequency response)
- **Tape Type:** GP9 (low noise, high-output)
- **Signal Path:** Repro
- **EQ Standard:** AES (30 IPS forces this)
- **Input Gain:** +2 dB
- **Saturation:** 8%
- **Wow / Flutter:** 0 / 0 (we want clean speed)
- **Noise Enabled:** Off
- **Output Gain:** -1 dB

Why this works. 30 IPS gives the flattest response and the lowest noise; the high-frequency roll-off is barely audible. GP9 tape is voiced for transparency. A small amount of saturation and input gain delivers the subtle nonlinearity that makes audio "sit" better in a mix without obvious distortion. With wow and flutter at zero, only the EQ shape and saturation contribute. This is the "Mastering Touch" preset's approach.

Drum bus character with 70s grit

Source: Stereo drum bus, mixed and balanced. **Goal:** Pre-internet-era warmth and a touch of grit.

Settings:

- **Tape Machine:** Classic 102

Figure 14: Classic 102 machine selected

- **Tape Speed:** 15 IPS
- **Tape Type:** Type 456
- **Signal Path:** Repro
- **EQ Standard:** NAB
- **Input Gain:** +6 dB
- **Saturation:** 35%
- **Bias:** 42% (slightly under-biased for added harmonic content)
- **Wow:** 12%
- **Flutter:** 6%
- **Noise Enabled:** On, **Noise Amount:** 10%
- **Output Gain:** -3 dB

Why this works. Classic 102 is the more colored of the two machines. 15 IPS at NAB gives the classic 1970s head response with audible high-frequency roll-off. Type 456 saturates earlier than the modern formulations. Driving the input by 6 dB pushes the saturation into clearly audible territory. Under-biasing (Bias below 50%) thins the saturation slightly and adds harmonic distortion. Audible wow and flutter complete the era. This is the "70s Rock" preset.

For a less aggressive version, drop Saturation to 20% and Input Gain to +3 dB.

Cassette tape lo-fi effect

Source: Anything you want to sound like it is playing back from a worn cassette. **Goal:** Audibly degraded, narrow-bandwidth, modulated.

Settings:

- **Tape Machine:** Classic 102
- **Tape Speed:** 7.5 IPS (worst high-frequency response)
- **Tape Type:** Type 250 (highest noise, most coloration)
- **Signal Path:** Repro
- **EQ Standard:** NAB
- **Input Gain:** +4 dB
- **Saturation:** 50%
- **Bias:** 35% (heavily under-biased, distorted)
- **Highpass Frequency:** 80 Hz
- **Lowpass Frequency:** 8000 Hz (thin top end)
- **Wow:** 30%
- **Flutter:** 20%
- **Noise Enabled:** On, **Noise Amount:** 30%
- **Output Gain:** 0 dB

Why this works. 7.5 IPS at the most aggressive tape (Type 250) with heavy under-bias and 50% saturation produces the thick, distorted sound of a cassette running too hot. Audible wow and flutter pitch-shift the playback, completing the cassette illusion. The 8 kHz low-pass narrows the bandwidth dramatically. This roughly matches the "Cassette Deck" preset.

Mastering bus polish

Source: A finished stereo mix. **Goal:** Inaudible analog flavor, no obvious processing.

Settings:

- **Tape Machine:** Swiss 800
- **Tape Speed:** 30 IPS
- **Tape Type:** GP9
- **Signal Path:** Repro
- **EQ Standard:** AES
- **Input Gain:** +2 dB
- **Saturation:** 6%
- **Bias:** 50% (perfectly biased)
- **Auto Calibration:** On
- **Wow / Flutter:** 0 / 0
- **Noise Enabled:** Off
- **Auto Compensation:** On
- **Oversampling:** 4x
- **Output Gain:** -1 dB

Why this works. Conservative everything: cleanest machine, fastest speed, lowest-noise tape, perfectly biased, no modulation, no noise. Just enough saturation and input gain to add second-harmonic content. 4x oversampling keeps the saturation aliasing-free. A 1 dB cut on the output prevents any chance of clipping into the next plugin. The result should be inaudible on its own; the difference is only obvious on bypass A/B. This matches the "Master Bus Glue" preset.

Parameter Reference

Machine and tape

- **Tape Machine:** Swiss 800 (default) or Classic 102. Swiss 800 is the cleaner, more linear machine; Classic 102 has more pronounced head response and saturation character.
- **Tape Speed:** 7.5, 15 (default), or 30 IPS. Faster speed gives flatter frequency response, less noise, and less saturation per dB of input. 7.5 is the most colored and noisy; 30 is the cleanest.
- **Tape Type:** Type 456 (default), GP9, 911, or 250. Different tape formulations have different saturation curves, noise floors, and tonal characters. 456 is the classic Ampex; GP9 is a modern low-noise; 911 is the popular European; 250 has the most character (and the most noise).
- **Signal Path:** Repro (default), Sync, Input, or Thru. Repro is the full signal chain (electronics into tape into reproduce head); Sync uses the record head as the playback head (slightly different EQ); Input is the electronics only with no tape (no saturation or modulation); Thru is true bypass.
- **EQ Standard:** NAB (default), CCIR, or AES. American (NAB) and European (CCIR) tape EQs differ above 1 kHz; AES is mandatory at 30 IPS. The plugin will switch to AES automatically when you pick 30 IPS.

Drive and tone

- **Input Gain:** -12 to +12 dB. Drives the tape stage. Higher values produce more saturation and harmonic distortion.
- **Saturation:** 0 to 100%, default 4%. The depth of the tape saturation curve. Default is barely audible; 20% is "pleasant"; 50% and above is distortion-heavy.
- **Bias:** 0 to 100%, default 50%. Tape bias level. 50% is perfectly biased (cleanest). Below 50% is under-biased and adds distortion. Above 50% is over-biased and dulls the highs.

- **Calibration:** 0 dB (default), +3, +6, or +9 dB. The reference operating level. Higher calibration means the tape is set to handle louder signals; useful when feeding hot mixes.
- **Auto Calibration:** On (default) or Off. When on, bias and calibration auto-adjust based on tape type and speed for optimal recording. Turn off to set bias manually for creative under- or over-biasing.

Filtering and modulation

- **Highpass Frequency:** 20 to 500 Hz, default 20. The corner of the input high-pass filter.
- **Lowpass Frequency:** 3 to 20 kHz, default 20 kHz. The corner of the input low-pass filter.
- **Wow:** 0 to 100%, default 7. Slow pitch drift (around 0.3 to 0.8 Hz). Adds vinyl-like wobble at higher values.
- **Flutter:** 0 to 100%, default 3. Faster pitch modulation (3 to 7 Hz). Adds tape machine character at higher values.
- **Noise Amount:** 0 to 100%, default 0. Tape noise floor level.
- **Noise Enabled:** Off (default) or On. Turn on to introduce noise; Noise Amount is the level.

Output

- **Output Gain:** -12 to +12 dB. Final makeup gain.
- **Auto Compensation:** On (default) or Off. Compensates output level when Input Gain changes, so A/B comparisons are level-matched.
- **Oversampling:** 1x, 2x, or 4x (default). Reduces aliasing from the saturation stage.
- **Bypass:** Reports zero latency to the host while bypassed.

Tips and Traps

- **Auto Calibration is your friend until you want it not to be.** With Auto Calibration on, switching tape type or speed re-biases the tape for you. Turn it off when you want to creatively under-bias (40 to 45% for grit) or over-bias (55 to 60% for a darker, dulled tone).

- **30 IPS forces AES.** This is hardware-accurate behavior; real machines also lock the EQ standard at 30 IPS. Pick 15 or 7.5 IPS if you want NAB or CCIR.

- **Signal Path is not Bypass.** "Thru" is true bypass (signal passes unchanged). "Input" runs the electronics but no tape (no saturation or modulation). "Sync" uses the record head for playback (slightly different EQ). "Repro" is the full chain. Use Sync to compare what the engineer hears during tracking versus the printed tape.

- **The default Saturation of 4% is conservative.** If you cannot hear any change versus bypass, raise it to 15 to 25% and pull Input Gain up by a few dB.

- **Wow and Flutter are subtle by default.** 7% Wow and 3% Flutter approximate a well-maintained machine. Push to 15 to 30% for the obviously-vintage sound; below 5% you may not hear them on most material.

- **Noise is a creative tool, not a problem to fix.** When you want the analog floor sound, enable Noise and set Noise Amount to 5 to 15%. For clean work, leave it disabled (the default).

Figure 15: Bias control with Auto Calibration off

Figure 16: Signal Path dropdown

Figure 17: Preset menu open

Presets Explained

TapeMachine ships with 15 presets across 5 categories. Each is a starting point; expect to nudge Input Gain and Saturation to taste.

Subtle

- **Gentle Warmth.** Swiss 800, 30 IPS, GP9, low saturation. Almost-inaudible analog flavor for transparent work.
- **Transparent Glue.** Similar to Gentle Warmth but slightly more saturation; for bus duty.
- **Mastering Touch.** The cleanest setting in the plugin: 30 IPS, GP9, AES, perfectly biased, no modulation. For mastering bus or any "I want a hint of analog" job.

Warm

- **Classic Analog.** 15 IPS, Type 456, NAB, moderate drive. The textbook warm tape sound.
- **Vintage Warmth.** Classic 102, 15 IPS, more pronounced head bump, slight wow. For sources that need to feel older.
- **Tube Console.** Heavier saturation with audible second-harmonic content; pairs well with a bus EQ.

Character

- **70s Rock.** Classic 102, 15 IPS, Type 456, under-biased (42%), audible wow and flutter, noise on. The drum-bus and rock-mix preset.
- **Tape Saturation.** Heavy drive, full saturation. Think parallel-tape bus sound.
- **Cassette Deck.** 7.5 IPS, narrow bandwidth, audible modulation. Lo-fi territory.

Lo-Fi

- **Lo-Fi Warble.** Heavy wow and flutter, narrow bandwidth, audible noise. Creative-effect preset.
- **Worn Tape.** Even more degraded; for atmospheric or sound-design work.
- **Dusty Reel.** The most extreme; obvious tape grit, modulation, noise. Use sparingly.

Mastering

- **Master Bus Glue.** Cleanest mastering setting; minimal saturation, no modulation, no noise. Subtle bus glue.
- **Analog Sheen.** Adds a touch of harmonic content to brighten a master without obvious processing.
- **Vintage Master.** A more colored mastering option, with a touch of head bump and very subtle modulation.

Troubleshooting

The plugin sounds the same with bypass on and off. With **Saturation** at 4% and **Input Gain** at 0 dB, the difference is genuinely subtle. Push Saturation to 25% and Input Gain to +6 dB to confirm the plugin is doing something, then dial back to taste.

Switching to 30 IPS changed my EQ Standard automatically. That is intentional. Real tape machines lock to AES EQ at 30 IPS; the plugin matches. Pick 15 or 7.5 IPS to use NAB or CCIR.

The Bias knob does nothing. Auto Calibration is on by default and overrides manual bias. Turn off **Auto Calibration** in the calibration section, then Bias is yours to drive.

The CPU usage is high. 4x oversampling is the default and is the heaviest setting. Drop to 2x for tracking; the audible difference is small but the CPU savings are significant. 1x is appropriate only if you cannot afford the 2x cost; aliasing in the saturation stage becomes audible on bright sources.

I hear pitch drift even with Wow and Flutter at zero. Confirm Signal Path is Repro or Sync; if you set everything to zero but still hear drift, restart the plugin (a stuck modulator is rare but possible).

DuskVerb

Pre-release. DuskVerb is currently in pre-release status. Parameter ranges, preset names, and engine behavior may change before the 1.0 release. This manual reflects the current 0.5.x line; check the website for updates if you are reading an older copy.

Overview

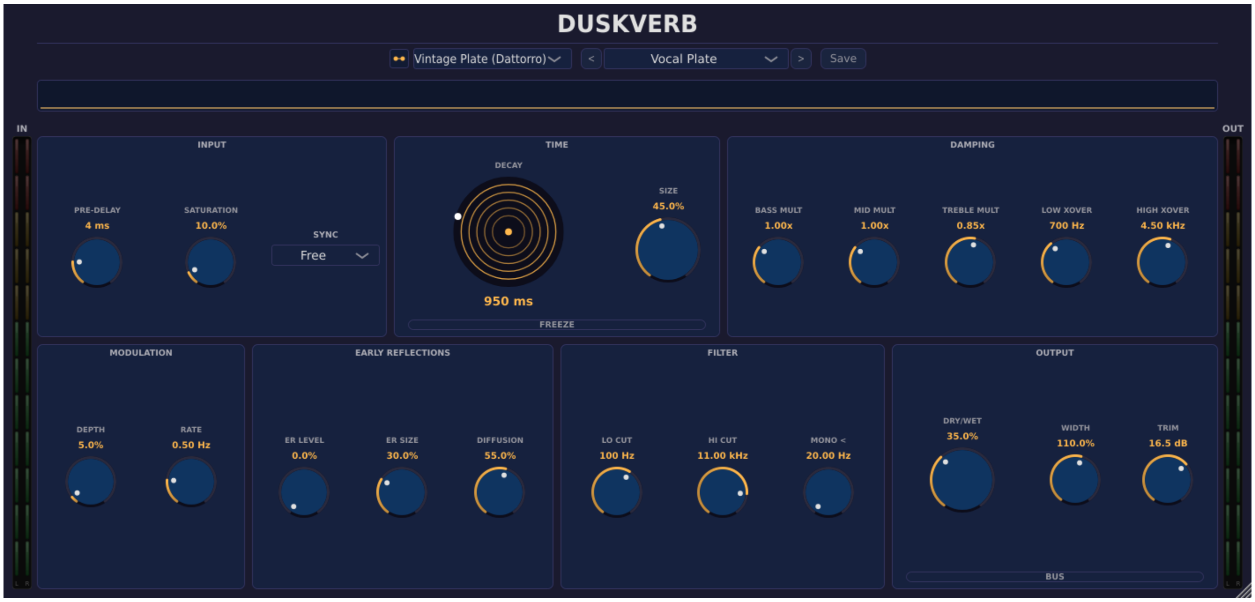


Figure 18: DuskVerb main UI

DuskVerb is an algorithmic reverb with six distinct engines under one user interface. Each engine targets a different reverb territory: **Vintage Plate (Dattorro)** captures the classic plate sound; **High Density (6-AP)** is dense and modern; **Quad Room (QuadTank)** is for tight rooms and short ambience; **Realistic Space (FDN)** is the largest, longest, and most realistic; **Spring Tank (6G15)** is the surf-guitar spring sound; **Non-Linear (RMX16)** is the gated 1980s drum sound and other non-natural curves.

Use it where you would use any reverb. Vocals, drums, guitars, and full-mix space all have an engine in DuskVerb that is voiced for them. The 34 factory presets are anchored to specific hardware references (Lexicon 480L, PCM 90, EMT 140, Bricasti M7, AMS RMX16, others) and serve both as starting points and as a tutorial in what each engine can do.

It is not a convolution reverb (use Convolution Reverb for IR-based work), and it is not a delay/multitap effect. It is a six-in-one algorithmic reverb that lets you pick the right engine for the job.

Quick Start

1. Insert DuskVerb on a return bus or directly on a track. For most workflows, an aux/return at 100% wet is the cleanest setup.
2. Pick an **Algorithm** from the top dropdown. If you are not sure, start with **Vintage Plate (Dattorro)**; it works on almost everything.

3. Open the preset menu. Each engine has its own preset list. Pick one that matches your source (Vocal Plate for vocals, Tight Drum Room for drums, etc.). 4. Adjust **Decay Time** to taste. The preset gives you a starting point; longer decays sustain more, shorter decays sit more discreetly. 5. Use **Pre-Delay** (0 to 250 ms) to push the reverb tail later, which keeps the dry signal forward and the reverb perceived as space rather than smear. 6. The **Dry/Wet** knob controls the mix. On a return bus, leave at 100%. On an insert, dial back to 20-40% wet.

You should hear a tail when you stop the source. If the tail is too dark, raise **Treble Multiply**; too bright, lower it. If the tail rings or sounds metallic, drop **Diffusion** below the preset value or pick a different engine.

Workflows

Vocal plate on a lead vocal

Source: A lead vocal that needs space without sounding wet. **Goal:** Smooth, classic plate tail that thickens the vocal without obscuring it.

Settings (or load the **Vocal Plate** preset and tweak from there):

- **Algorithm:** Vintage Plate (Dattorro)
- **Decay Time:** 1.8 s
- **Pre-Delay:** 30 ms
- **Size:** 0.6
- **Mod Depth:** 0.15
- **Mod Rate:** 1.0 Hz
- **Treble Multiply:** 0.8 (slight high-frequency rolloff in the tail)
- **Lo Cut:** 200 Hz (keeps the tail out of the bass range)
- **Hi Cut:** 8000 Hz
- **Dry/Wet:** 100% (use as a send return)

Why this works. Plate reverbs are dense from the first reflection; they do not have audible early reflections. Pre-delay of 30 ms separates the tail from the dry signal so the vocal stays forward. Lo

Cut at 200 Hz prevents the tail from muddying the low end. Subtle modulation (depth 0.15, rate 1 Hz) keeps the plate from sounding static. The 1.8-second decay is a typical pop-vocal length.

Tight drum room

Source: Drum bus or a snare track that needs room without long tail. **Goal:** Short, punchy ambience that adds size without smearing transients.

Settings:

- **Algorithm:** Quad Room (QuadTank)
- **Decay Time:** 0.6 s
- **Pre-Delay:** 0 ms
- **Size:** 0.4
- **Diffusion:** 0.5 (less than fully diffuse keeps transients distinct)
- **Early Ref Level:** 0.7 (early reflections are most of what you hear)
- **Treble Multiply:** 0.95 (keep highs in the room)
- **Lo Cut:** 80 Hz
- **Dry/Wet:** 25% (insert) or 100% on a return

Why this works. Short decay plus low diffusion plus prominent early reflections gives a “room” sound rather than a “reverb” sound. The drums still hit hard; the room just adds a bit of three-dimensional space. The “Tight Drum Room” preset uses these proportions.

Realistic concert hall on piano

Source: Solo piano or piano in a sparse mix. **Goal:** A long, lush hall tail that sounds like a real space.

Settings (or use the **Smooth Concert Hall** preset):

- **Algorithm:** Realistic Space (FDN)
- **Decay Time:** 4.5 s
- **Pre-Delay:** 60 ms
- **Size:** 0.85
- **Diffusion:** 0.85
- **Early Ref Level:** 0.4
- **Early Ref Size:** 0.7
- **Bass Multiply:** 1.2 (emphasize the low frequencies in the tail, characteristic of large rooms)
- **Mid Multiply:** 1.0
- **Treble Multiply:** 0.7 (concert halls absorb highs)
- **Lo Cut:** 60 Hz
- **Hi Cut:** 12000 Hz
- **Dry/Wet:** 100% on a return at -10 to -6 dB send level

Why this works. The FDN engine produces realistic late reverberation with audible per-frequency decay differences. Bass multiply above 1 emphasizes the long bass tail typical of real halls. Treble multiply below 1 captures the high-frequency absorption you hear in rooms with absorptive surfaces. Long pre-delay (60 ms) maintains piano clarity.

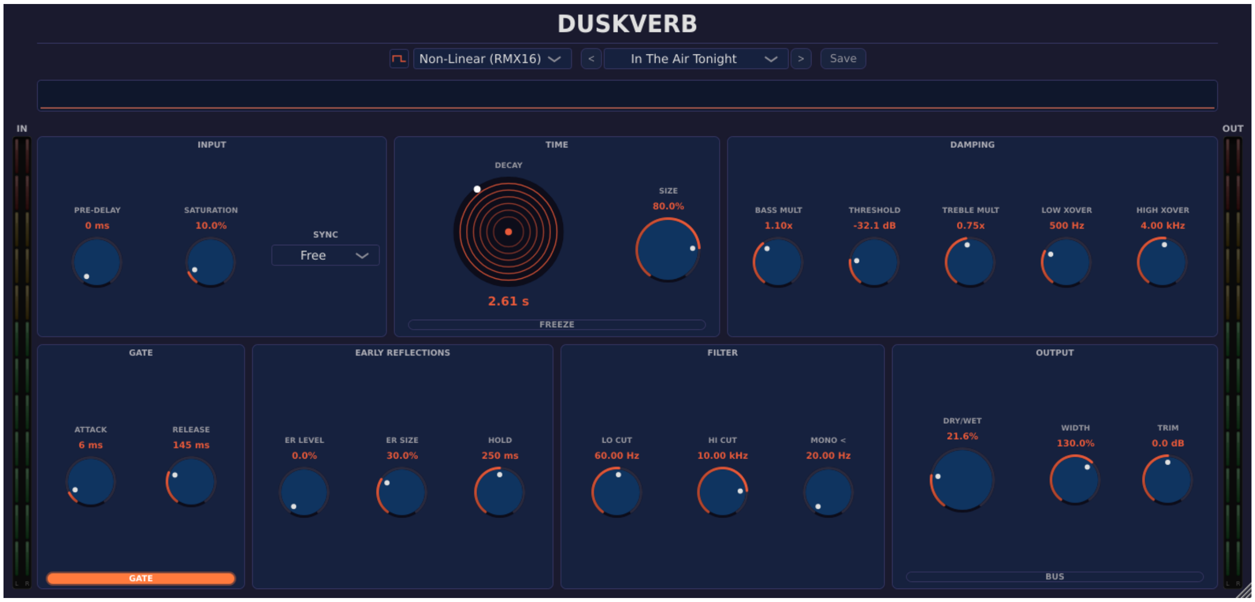


Figure 19: Non-Linear engine with Gate enabled

Gated 80s snare

Source: Snare drum that needs the classic 1980s gated sound. **Goal:** A non-natural reverb shape that cuts off after a fixed time.

Settings:

- **Algorithm:** Non-Linear (RMX16)
- **Decay Time:** 1.2 s (the gate length, not natural decay)
- **Pre-Delay:** 0 ms
- **Size:** 0.6
- **Treble Multiply:** 1.0
- **Gate:** Enabled (default; this is the parameter that makes the engine non-linear)
- **Dry/Wet:** 35% (insert)

Why this works. The Non-Linear engine in this mode mimics the AMS RMX16's "Non-Lin 2" algorithm: the reverb has constant level for a fixed time (set by Decay Time) and then cuts to silence rather than fading naturally. This produces the 1980s snare sound that defined a decade of records. The "Snare Plate XL" preset is plate-based; for true gated character, switch to Non-Linear.

Parameter Reference

Algorithm

- **Algorithm:** Selects which DSP engine processes the audio. Six choices: Vintage Plate (Dattorro), High Density (6-AP), Quad Room (QuadTank), Realistic Space (FDN), Spring Tank (6G15), Non-Linear (RMX16). Switching engines crossfades over a few hundred milliseconds.

Mix and routing

- **Dry/Wet:** 0 to 100%. Wet/dry mix at the plugin output. 100% on returns; lower on inserts.

- **Bus Mode:** On or off. When on, the plugin assumes a 100% wet send/return setup and bypasses the dry path entirely (slight CPU savings).
- **Bypass:** Reports zero latency to the host while bypassed.

Time and size

- **Pre-Delay:** 0 to 250 ms. Delay before the reverb tail begins. Longer pre-delay separates the dry signal from the reverb.
- **Pre-Delay Sync:** Free, 1/32, 1/16, 1/8, 1/4, 1/2, 1/1. When set to a note value, pre-delay locks to the host tempo.
- **Decay Time:** 0.2 to 30 s. Total tail length. Different engines interpret this differently; the FDN engine reaches the highest decays; the Non-Linear engine treats this as a gate length.
- **Size:** 0 to 1. Perceived room size. Engine-dependent: in Quad Room, low values are cabinet-sized and high values are arena-sized.

Modulation

- **Mod Depth:** 0 to 1. Amount of pitch modulation in the tail. Subtle values (0.1 to 0.2) keep the tail moving; high values (0.5 plus) produce chorus-like effects.
- **Mod Rate:** 0.1 to 10 Hz. Speed of the modulation. Slower rates (around 0.5 Hz) sound natural; faster rates produce flanger-like motion.

Frequency shaping

- **Bass Multiply:** 0.3 to 2.5. Decay-time multiplier for low frequencies. Above 1 makes the bass decay longer than the rest (typical of large halls). Below 1 shortens the bass tail (typical of small studios with absorbers).
- **Mid Multiply:** 0.3 to 2.5. Decay multiplier for the mid range.
- **Treble Multiply:** 0.1 to 1.5. Decay multiplier for the high frequencies. Below 1 produces high-frequency damping (the "warm" tail).
- **Low Crossover:** 200 to 4000 Hz. Frequency where Bass Multiply gives way to Mid Multiply.
- **High Crossover:** 1000 to 12000 Hz. Frequency where Mid Multiply gives way to Treble Multiply.
- **Lo Cut / Hi Cut:** Filters the input into the reverb. Lo Cut keeps low frequencies dry; Hi Cut keeps highs dry. Useful for keeping the reverb out of the kick and the air bands respectively.

Color and density

- **Saturation:** 0 to 1. Drives a soft saturation stage in the reverb feedback path. Adds warmth and harmonic content.
- **Diffusion:** 0 to 1. Density of the reverb tail. High diffusion is dense and smooth; low diffusion sounds grainier and more echo-like.
- **Early Ref Level:** 0 to 1. How loud the early reflections are relative to the main tail.
- **Early Ref Size:** 0 to 1. The simulated room size for the early reflections.

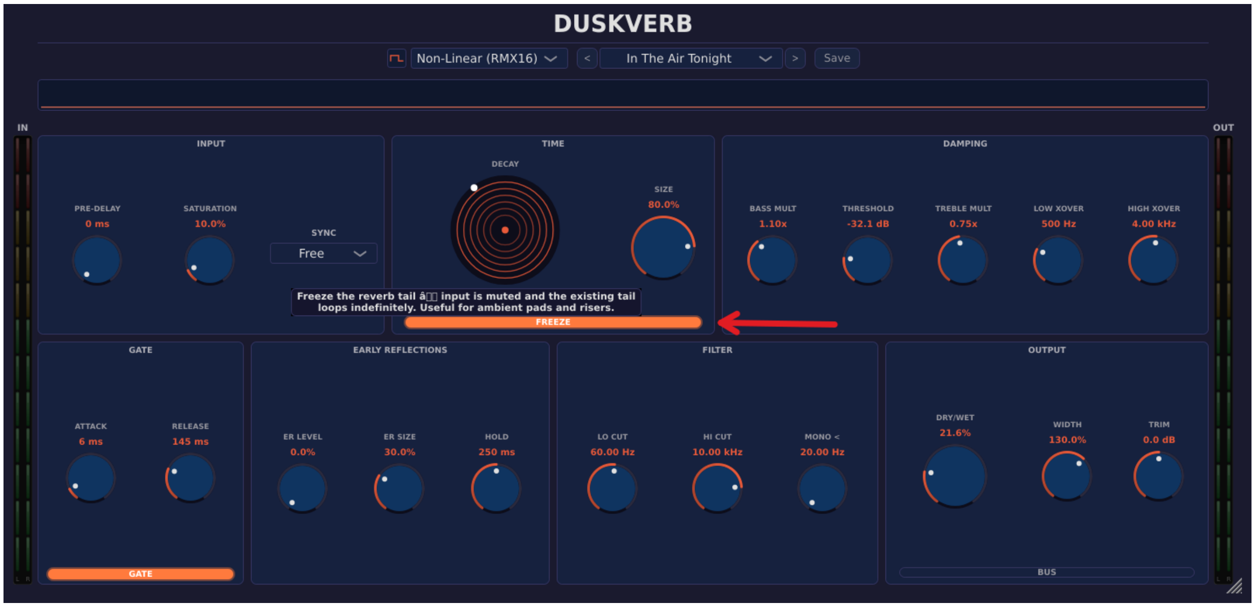
Output and stereo

- **Width:** 0 to 2. Stereo width of the reverb output. 1 is unity; 2 is double-wide (M/S based).
- **Freeze:** Off or On. When on, the reverb tail cycles infinitely.
- **Gate:** Off or On. Enables the Non-Linear engine's gating behavior.

- **Mono Below:** 20 to 300 Hz. Frequencies below this cutoff are summed to mono in the reverb output. Default 20 Hz (effectively bypass); typical settings 80 to 150 Hz to keep low-frequency reverb mono-compatible.
- **Gain Trim:** -48 to +48 dB. Final output level adjustment.

Tips and Traps

- **Pre-Delay is your most important parameter for clarity.** Without pre-delay, the reverb tail starts on the same sample as the dry signal and the result smears. 20 to 60 ms of pre-delay keeps vocals and drums distinct from their reverb.
- **Hardware anchors are real.** The presets are tuned to specific hardware references. "Vintage Vocal Plate" is anchored to the EMT 140; "Blade Runner Concert" to the Lexicon 224; "Cathedral" to a long Lex 480L hall. If you are familiar with the source hardware, the preset name tells you what to expect.
- **Engine switching is not parameter-preserving.** Each engine has its own internal state. Switching engines while a tail is decaying produces a crossfade; do not expect identical-sounding results across engines at the same parameter values.
- **Freeze is loud.** Freeze captures the current reverb tail and loops it indefinitely. Levels can build dramatically; pull Gain Trim back before enabling Freeze on a busy mix.



- **Mono Below preserves bass mono compatibility.** Stereo reverb on bass frequencies often phases on mono fold-down. Set Mono Below to 80 to 120 Hz on mastering or mix-bus reverb. - **The Non-Linear engine treats Decay Time as a gate length.** It does not behave like other engines for that parameter. Refer to the gated-snare workflow above.

Presets Explained

DuskVerb ships with 34 factory presets across 7 categories. Each is hardware-anchored and serves as a starting point.

Plates

Six plate presets ranging from short and bright (**Vocal Plate**, anchored to PCM 90 P2 1.0) to long and lush (**Gold Plate**, PCM 90 P2 0.2). **Vintage Vocal Plate** is the EMT 140 anchor for

Figure 20: Preset menu by category

darker, steely plate character. **Snare Plate XL** is the long plate sound that helped define 1980s snare reverb on records.

Halls

Eleven halls, from utility (**Utility Hall**, PCM 90 P0 2.9) through bright studio (**Bright Hall**, PCM 90 P0 2.8) to lush concert (**Smooth Concert Hall**, Lex 480L Smooth Hall) to massive cathedral (**Cathedral**, Lex 224 Concert Hall A 6.5 s). **Blade Runner 224** and **Blade Runner Concert** capture the long-decay extended-tail Lex 224 sound. **Lush Dark Hall** is the Lex 480L Hall A warm-dark variant. Pick by length and brightness.

Rooms

Five rooms covering tight (**Tight Drum Room**) through medium (**PCM Drum Room**, **Studio Room**) to atmospheric (**Reverse Taps**, **In The Air Tonight** for the Phil Collins gated drum effect). Use **Tight Drum Room** as your default for drum bus ambience.

Chambers

Realistic Chamber captures a typical studio chamber sound; longer than a room but shorter than a hall.

Springs

Surf ’63 Spring is the Dick Dale “Misirlou” reverb; aggressive, bouncy. **Tank Drip** is a shorter spring tank for subtler surf and reggae work.

Ambient

Four ambient presets: **Black Hole** (Eventide-anchored), **Infinite Blackhole** (huge sustaining ambience), **Mobius Pad** (Strymon-anchored), **Ambient Swell**. Use these for sound design and pad-like effects rather than realistic space.

Shimmer

Cascading Heaven, **Deep Blue Day** (the Brian Eno track that the engine is named after). Currently the Shimmer engine is hidden in the dropdown but the presets remain accessible. Re-enabled for 1.0.

Troubleshooting

The tail sounds metallic or rings. Drop **Diffusion** to around 0.5 to 0.7 if the tail is buzzing or sounds like a ring modulator. Try a different engine; the FDN engine is the smoothest at long decays, the Vintage Plate engine is naturally more dense.

The reverb is too loud at low frequencies. Raise **Lo Cut** to 100 to 200 Hz so the reverb input is high-passed before reaching the engine. Bass frequencies in reverb tails build up quickly and muddy a mix.

Switching engines drops the tail. That is intentional. Each engine has its own internal state; the crossfade between engines is brief (a few hundred milliseconds) and the new engine starts with a clean tail. Plan engine switches between sections, not within a phrase.

The sound is the same regardless of which preset I load. Confirm you are loading from the **Preset** menu and not just changing the **Algorithm**. Loading a preset sets all parameters including the algorithm; changing only the algorithm leaves the other parameters at their previous values.

My host shows extra latency on insert. DuskVerb has a small fixed latency from its early-reflection delay line. The host applies plug-in delay compensation automatically; check that PDC is enabled in your DAW preferences if other tracks sound out of sync.

Spectrum Analyzer

Overview

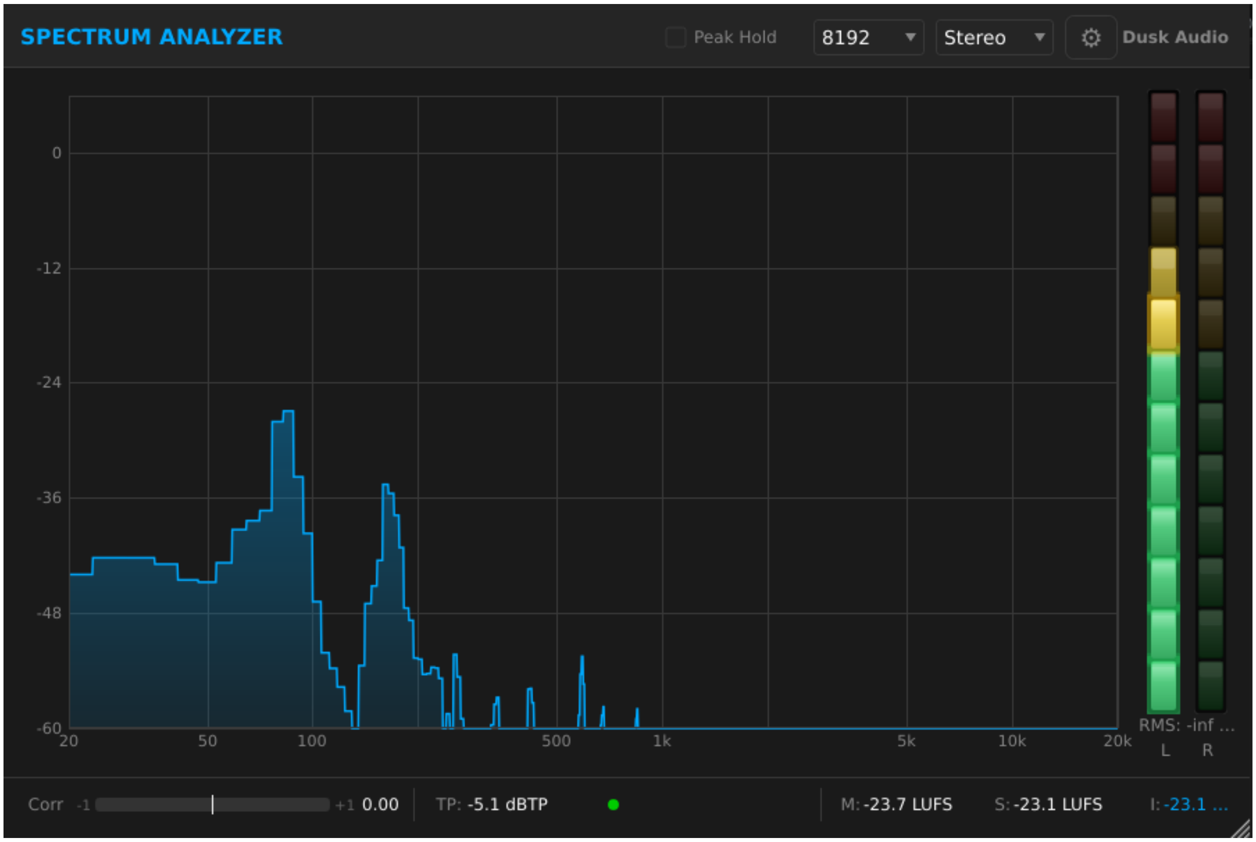


Figure 21: Spectrum Analyzer main UI

Spectrum Analyzer is a measurement plugin, not a sound-shaping plugin. Drop it on any track or bus to see what is actually there: a real-time FFT spectrum, four kinds of loudness reading (Momentary, Short-term, Integrated, LRA), an ITU-R BS.1770-4 compliant true-peak meter, a stereo correlation meter, and K-System headroom display.

Use it to verify what your ears are telling you. The spectrum shows tonal balance and resonance; LUFS shows perceived loudness for streaming-platform targets; true-peak catches inter-sample peaks that a regular sample-peak meter misses; correlation flags phase issues before they cost you a mono fold-down.

It is not a corrective EQ, and it is not a sonic-fingerprint matching tool. It is a precise set of meters that gives you the numbers you need to decide what to do next.

Quick Start

1. Insert Spectrum Analyzer on the track or bus you want to measure. Mastering bus is a common place to put it; it works equally well on a single instrument.

2. Play the project. The spectrum draws across the main display. Loudness readings appear next to it.

3. Click the gear icon (top right) to open the settings overlay. The defaults (4096 FFT, 0.5 smoothing, 0 dB/oct slope, K-14 metering) work for most program material.

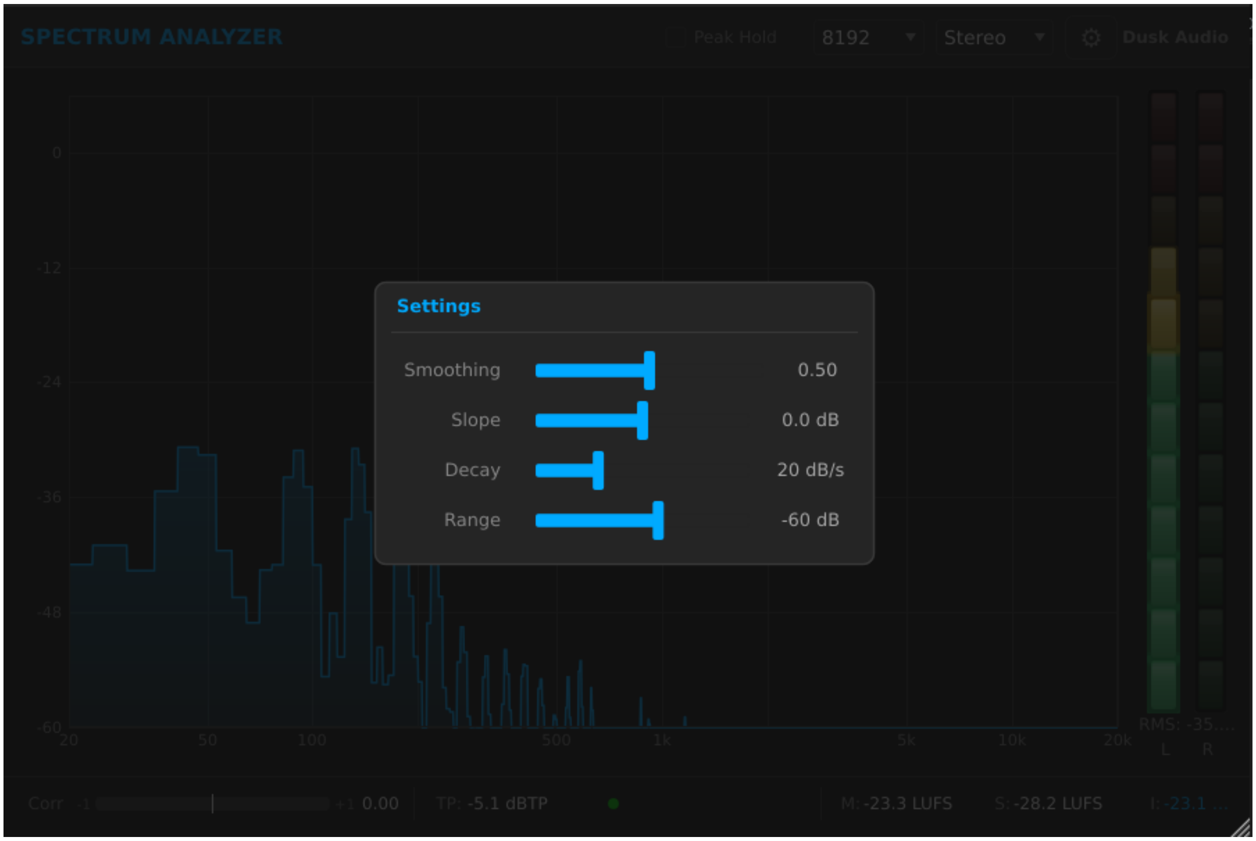


Figure 22: Settings overlay open

4. Look at the Integrated LUFS readout once the section you are checking has played for at least 10 seconds. That is your loudness number.

5. Watch the True Peak indicator. If it lights red, your master is clipping inter-sample; pull the output back at least 1 dB.

6. Use the Channel Mode dropdown in the header to switch between Stereo, Mono, Mid, and Side views.

You should see frequency content where you expect it (kicks under 100 Hz, vocals around 1 to 3 kHz, cymbals above 8 kHz). The correlation bar should sit between 0 and +1 on most material; negative values mean the stereo image is fighting itself.

Workflows

Tonal balance check on a mix

Source: A near-final mix bus, played at performance level. **Goal:** Confirm the mix has the spectral shape you intended.

Settings:

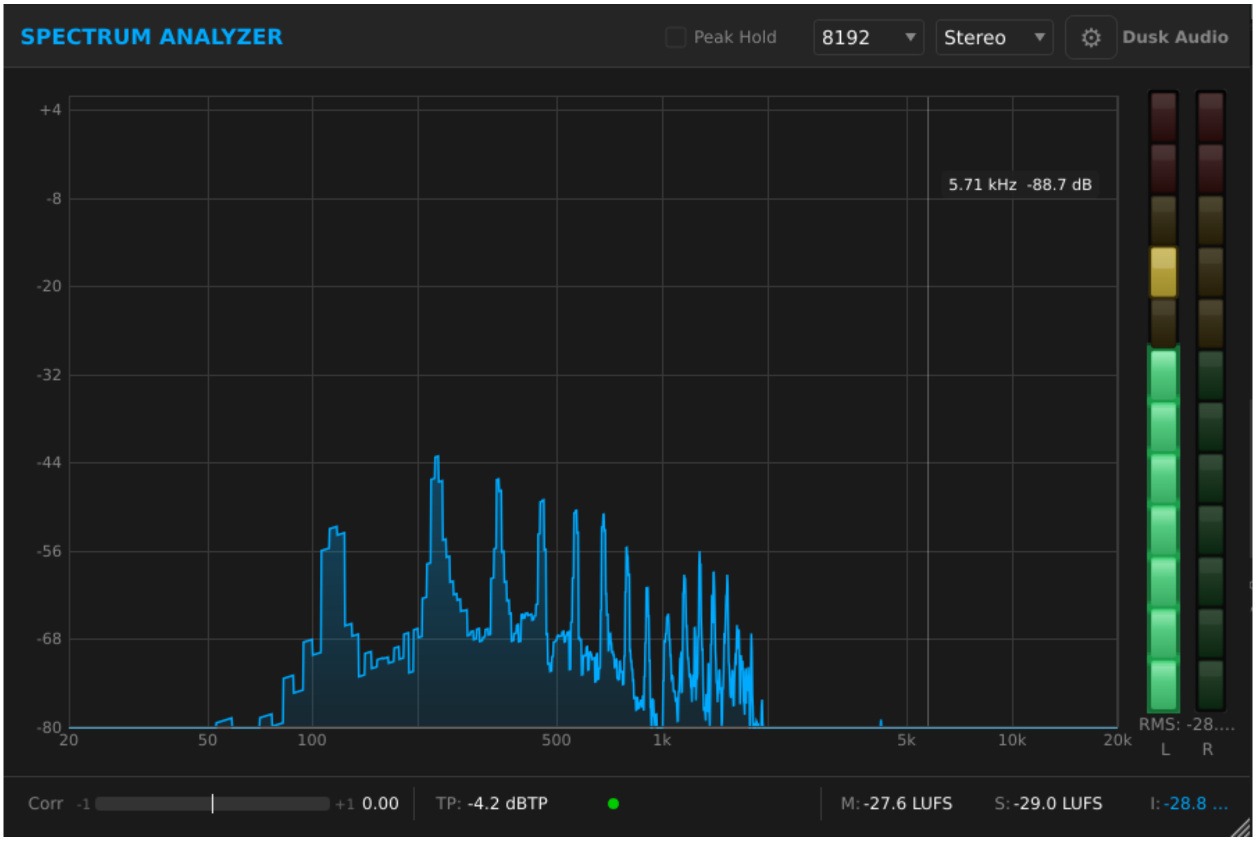


Figure 23: Tilted display with 4.5 dB/oct slope

- **Channel Mode:** Stereo
- **FFT Resolution:** 4096
- **Slope:** 4.5 dB/oct
- **Smoothing:** 0.7
- **Display Min:** -60 dB
- **Display Max:** +6 dB
- **Peak Hold:** On
- **Peak Hold Time:** 5 sec

Why this works. A 4.5 dB/oct slope tilts the display so a balanced "pink-noise" mix appears roughly flat across the spectrum. With Slope at 0, low end always dominates the visualization and you cannot eyeball mid and high content at a glance. Higher smoothing (0.7) damps out per-frame noise so you see steady tonal shape, not transients. Peak Hold with a 5-second decay shows you the loudest frequencies during the chorus.

Look for: a roughly even hill across 60 Hz to 10 kHz, with a small bump in the kick range (50 to 100 Hz) and a gentle roll-off above 12 kHz. Big resonant spikes (a single peak that sticks 6 dB above its neighbors) usually mean a problem frequency you can fix with a narrow EQ cut.

Streaming loudness target

Figure 24: LUFS readouts

Source: A finished master, ready to upload. **Goal:** Confirm Integrated LUFS hits the streaming target (Spotify, YouTube, Amazon Music, and Tidal: -14 LUFS integrated; Apple Music: -16 LUFS integrated; podcasts: -16 LUFS).

Settings:

- **Channel Mode:** Stereo
- **K-System Type:** K-14
- **Display Min:** -60 dB
- **Display Max:** 0 dB
- Other settings: defaults

Why this works. Set Spectrum Analyzer post-fader, post-everything, on the master bus. Play the entire song from start to end without stopping (LUFS Integrated only updates while audio is above the gate threshold; pausing pollutes the reading). The Integrated reading at the end is your loudness number.

If Integrated LUFS reads -10, your master is louder than the streaming target and the platform will turn it down. If it reads -18, you have headroom to push the level up. The True Peak meter must stay at or below -1 dBTP for safe transcoding to lossy formats. Watch the LRA (Loudness Range) reading; for most modern pop and rock, LRA between 5 and 10 LU is typical. Above 15 indicates very dynamic material.

Mid/Side imaging check

Source: A stereo mix that feels “narrow” or “off-center”. **Goal:** Identify what is in the middle versus the sides.

Switch the Channel Mode dropdown between Mid and Side while the mix plays. The spectrum updates to show only that component.

- **Mid mode:** kick, snare, bass, lead vocal, anything panned center.
- **Side mode:** room mics, reverb returns, doubled guitars, anything panned wide.

What to look for. If the Side spectrum has heavy low-frequency content (below 150 Hz), the mix has phasey low end; consider high-passing the side channel or summing low end to mono. If the Mid spectrum shows nothing above 5 kHz, the highs are all in the sides; that can sound airy on speakers but vanishes on mono playback.

Phase correlation check

Source: Anything stereo, especially overhead drum mics, room mics, or stereo synths. **Goal:** Catch phase issues before mono fold-down loses the signal.

Watch the correlation bar at the top of the meter strip. Positive (green) values mean the channels mostly agree; this is good. Values near zero (yellow) mean stereo content with low correlation; this is fine for ambient material. Values below zero (red) mean the channels are partially out of phase; if you sum to mono, those frequencies cancel.

If you see persistent negative correlation, swap the polarity of one channel of the offending stereo source and check again. Briefly negative values during transients are normal for very wide stereo material.

Parameter Reference

Spectrum Analyzer has 10 user-facing parameters, all exposed through the settings overlay (gear icon) except Channel Mode which sits in the header. The LUFS, True Peak, and correlation meters are always running and have no parameters of their own.

Display

- **FFT Resolution:** 2048, 4096 (default), 8192. Higher resolution gives finer low-frequency detail at the cost of slower update rate. 2048 is the most responsive; 8192 reveals individual harmonics down to about 30 Hz. 4096 is the right starting point.
- **Smoothing:** 0.0 to 1.0, default 0.5. Damps the per-frame fluctuations of the spectrum line. 0 is maximally responsive (jittery); 1 is heavily smoothed (sluggish). Use 0.7 to 0.9 for tonal-balance work; 0.2 to 0.4 for catching transient resonances.
- **Slope:** -4.5 to +4.5 dB/oct, default 0.0. Tilts the display. +4.5 makes pink-noise read flat; great for mixing reference. 0 shows the raw spectrum.
- **Decay Rate:** 3 to 60 dB/sec, default 20. How fast the spectrum line falls when audio energy drops. Slower decay (lower number) holds peaks visible longer.
- **Display Min:** -100 to -30 dB, default -60. Bottom of the visible vertical range.
- **Display Max:** 0 to +12 dB, default +6. Top of the visible vertical range.

Peak Hold

- **Peak Hold:** On / Off, default Off. Overlays a “peak hold” trace that captures the highest level seen at each frequency.

- **Peak Hold Time:** 0.5 to 10 sec, default 2.0. How long the held peaks stay visible before they decay.

Channel routing

- **Channel Mode:** Stereo, Mono, Mid, or Side. Stereo overlays the L and R spectra; Mono sums both; Mid shows the (L+R)/2 sum; Side shows the (L-R)/2 difference.

Metering

- **K-System Type:** K-12, K-14 (default), K-20. Sets the reference level for the K-System scale. K-14 is the typical mastering reference; K-20 is broadcast/film; K-12 is loudness-targeted music.

Tips and Traps

- **Pre-fader vs post-fader matters.** Spectrum Analyzer measures whatever signal reaches it. Insert it pre-fader to measure the source independent of automation; post-fader to measure what the listener hears.
- **LUFS Integrated needs a continuous play.** The Integrated reading uses gating to ignore silence and very-low-level passages. If you stop and start playback, the gate window resets and the reading drifts. Play through the section in one go.
- **True Peak with 4x oversampling is on by default.** This catches inter-sample peaks that a regular sample-peak meter misses. The cost is small CPU; leave it on for mastering.
- **The correlation meter can mislead on quiet sources.** When the signal is below about -50 dBFS, correlation readings get noisy and can swing wildly. Trust correlation only when there is real audio playing.
- **Slope shifts the curve, it does not change the audio.** This is a display setting only. Do not confuse it with an EQ.
- **Mid/Side mode is for monitoring, not for routing.** Switching to Mid does not change the audio leaving the plugin; it changes what the spectrum draws.
- **The plugin does not write to the audio path.** Spectrum Analyzer is a pure analyzer; output equals input regardless of settings.

Recommended Setups

Spectrum Analyzer does not ship with factory presets, but a few starting-point configurations are worth saving as DAW-level “saved states” or default presets you can recall.

Mixing reference

Slope 4.5, Smoothing 0.8, Decay 20, FFT 4096, Channel Mode Stereo, Peak Hold on with a 3-second decay. Drop this on the mix bus while building a song; the tilted display lets you eyeball whether the mix is generally balanced or skewing one way.

Mastering target

Slope 0, Smoothing 0.5, Display Min -100, Display Max 0, K-System K-14, FFT 8192. The longer FFT gives the low-end resolution you need to spot 50/60 Hz hum and sub-bass issues. K-14 is the standard mastering reference scale.

Imaging analysis

Channel Mode Side, Slope 4.5, Smoothing 0.7. Quickly switches the display to the side component so you can see what is contributing to stereo width. Pair with a second instance set to Mid mode if your DAW supports parallel routing.

Live tracking

Smoothing 0.2, Decay 40, FFT 2048, Peak Hold off. Maximally responsive setup for catching transients and quick spectral changes during recording. The shorter FFT keeps latency-related drawing artifacts to a minimum.

Troubleshooting

The spectrum is completely flat. No audio is reaching the plugin. Check that you have inserted it in the signal path (not on a return) and that the track is playing. If the signal is very quiet (below -60 dB), it can fall below the bottom of the default display range; lower Display Min to -100 dB.

LUFS reads -inf or never settles. Integrated LUFS uses gating; if the audio is mostly below -70 LUFS the gate filters everything out. Make sure the track is at a reasonable level and play through a longer section. The reading needs at least a few seconds of audio above -50 LUFS to give a useful number.

The correlation bar swings between green and red constantly. That is normal on busy stereo material. Watch the average position over a few seconds, not the instantaneous value. Persistent red on quiet material can indicate phase issues with stereo doubling or short-delay effects.

Chord Analyzer

Overview

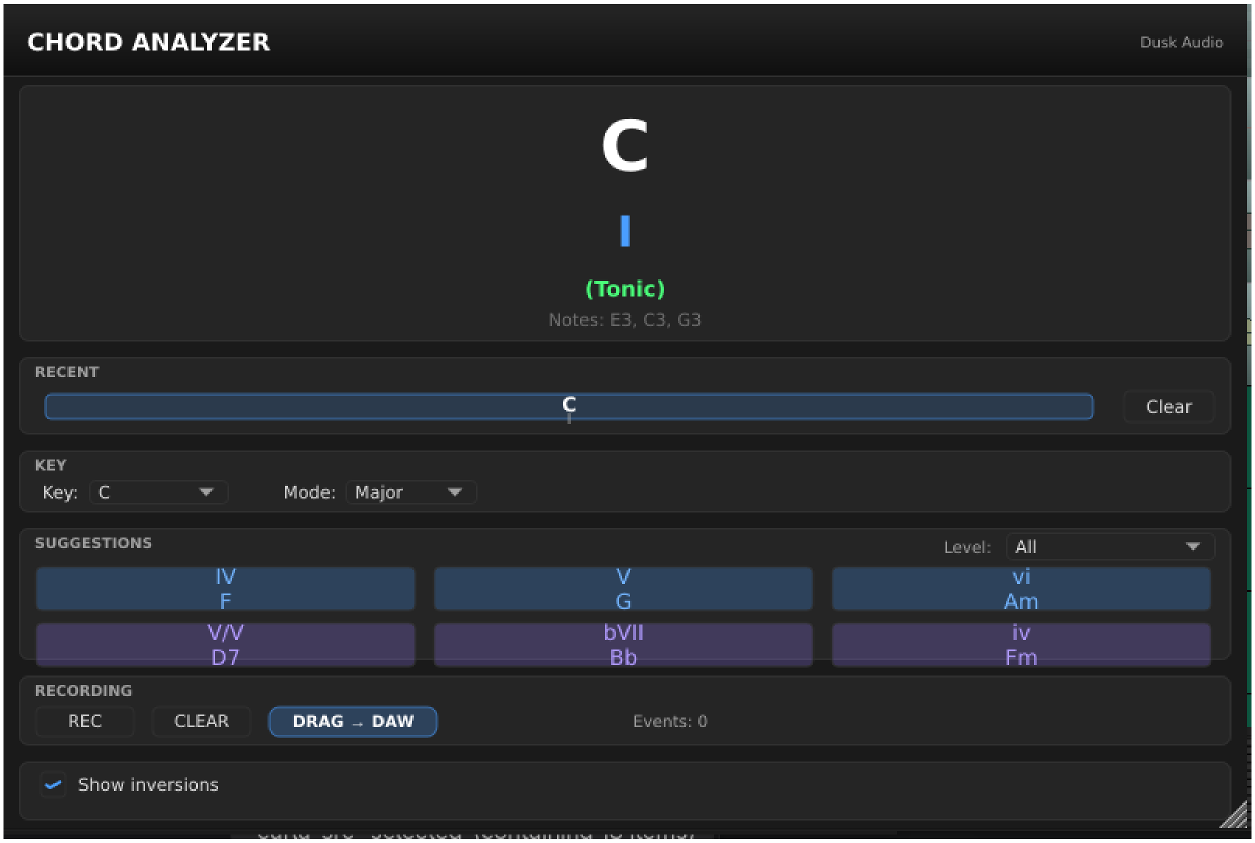


Figure 25: Chord Analyzer main UI showing a detected chord

Chord Analyzer is a MIDI plugin that watches the notes coming into your DAW and tells you what chord you are playing. It detects the chord root, quality (major, minor, 7th, sus, etc.), bass note for slash chords, and inversion. It also shows theory-aware suggestions: which chords are typical in the key you have selected, and which substitutions are available.

Use it for transcription (figure out what a recording is doing), for learning (see chord names while you play), for songwriting (browse suggested next chords), and for live performance (display the chord on a screen so the rest of the band can follow).

It is not a chord-generation tool (it does not play chords for you), and it is not a MIDI processor (it passes notes through unchanged). It analyzes; it does not modify.

The plugin ships in three variants because different DAWs handle MIDI-only plugins differently. Picking the right variant for your DAW is the most important first step.

Quick Start

1. Identify the right variant for your host:
    - **Chord Analyzer MIDI** for desktop DAWs that recognize MIDI-only plugins (Bitwig, Reaper, Logic, Cubase, Ardour, Carla).

  - **Chord Analyzer** (the instrument version) for DAWs that need a plugin to claim audio output (Ableton Live, FL Studio).
  - **Chord Analyzer Headless** for headless LV2 hosts that have no plugin GUI (Zynthian and similar).
2. Insert the plugin in front of your synth or instrument track. Detailed routing per-DAW is in the next section.
3. Set the **Key Root** and **Key Mode** at the top of the plugin window to match the song you are playing in. C Major is the default.
4. Play notes on your MIDI controller. The detected chord appears in the main display, with inversions and theory suggestions below.

You should see the chord name update as you play (for example, holding C, E, and G shows "C major"; adding B shows "C7"). If the chord display stays empty, check that MIDI is reaching the plugin and that you are holding at least three notes.

Workflows

The right setup depends on your DAW. The plugin always sits in front of your synth or instrument; what differs is how the DAW routes MIDI through it. Pick your DAW from the list below.

Bitwig Studio

1. Add your synth to a MIDI track as normal.
2. In the device chain, click + before your synth.
3. Search for **Chord Analyzer MIDI** in the **Note FX** category and add it.
4. MIDI passes through automatically; your synth still receives all the notes.

Reaper

1. Open the FX chain on your MIDI/instrument track.
2. Click **Add** and search for **Chord Analyzer MIDI** (VST3).
3. Drag it above your synth in the chain.
4. MIDI passes through to your synth automatically.

Logic Pro

1. On your instrument track, click the **MIDI FX** slot (above the Instrument slot).
2. Choose **Chord Analyzer MIDI** under Dusk Audio.
3. The plugin receives MIDI from the track and passes it through to your instrument.

Cubase / Nuendo

1. On your instrument track, add **Chord Analyzer MIDI** as a **MIDI insert**.
2. The plugin receives MIDI from the track automatically.
3. Your instrument remains unchanged.

Ableton Live

Ableton does not list MIDI-only plugins in MIDI tracks the same way other DAWs do, so use the **instrument version** with an Instrument Rack:

1. Load your synth on a MIDI track as normal.
2. Select your synth and press **Cmd+G** (Mac) or **Ctrl+G** (Windows) to group it into an **Instrument Rack**.

3. Click the **Show Chain List** button (three horizontal lines on the left side of the rack).
4. Right-click in the empty space below your synth's chain and choose **Create Chain**.
5. Drag **Chord Analyzer** (the instrument version, not MIDI) from the browser into the new empty chain.
6. Both chains receive the same MIDI; your synth produces sound, the Chord Analyzer displays chords.

FL Studio

Use the **instrument version** with Patcher:

1. Add a **Patcher** instance to your channel rack.
2. Inside Patcher, add both **Chord Analyzer** and your synth as nodes.
3. Route the MIDI input to both plugins in parallel.
4. Route only your synth's audio output to the Patcher output.
5. Both plugins receive MIDI; your synth produces sound, the Chord Analyzer displays chords.

Desktop LV2 hosts (Ardour, Carla)

Use **Chord Analyzer MIDI**; it declares no audio ports and includes the full custom visualizer in the host plugin window.

1. Add **Chord Analyzer MIDI** to the MIDI chain before your synth.
2. MIDI passes through to the next plugin in the chain.
3. Open the plugin window to see the full chord display.

Headless LV2 hosts (Zynthian)

Use **Chord Analyzer Headless**; it exposes the detected chord through native LV2 output control ports so the host can render the values in its own parameter view.

1. Install the bundle from the `chord-analyzer-headless-*.zip` (separate download).
2. Add **Chord Analyzer Headless** to your MIDI chain before your synth.
3. The detected root, quality, bass, and inversion appear as live values in your host's plugin parameter view.

Parameter Reference

Chord Analyzer exposes five user-facing parameters plus four read-only detection outputs.

Key context

- **Key Root:** 12 choices, C through B (with enharmonic spellings shown). Default C. Sets the tonal center for theory-aware suggestions. The detector itself is key-agnostic (it identifies chord names regardless of key), but the suggestion display uses this to highlight chords that fit your key.
- **Key Mode:** Major or Minor, default Major. Combined with Key Root, defines the key for suggestions. Set this to match the song you are playing or transcribing.

Detection settings

- **Suggestion Level:** Basic Only, Basic + Intermediate, or All (+ Advanced). Default All. Controls how many tiers of suggested chord substitutions appear below the detected chord. Basic shows only diatonic triads (the seven chords in the key). Intermediate adds 7th chords

and common borrowed chords. Advanced adds tritone substitutions, secondary dominants, and modal interchange options.
- **Show Inversions:** On by default. When on, the display tells you which inversion of the detected chord you are playing (root position, first inversion, second inversion, third inversion for 7ths). Turn off if you want a less cluttered display.
- **Respect Sustain:** On by default. When on, MIDI CC 64 (sustain pedal) holds the detected chord on screen until you release the pedal, even if you let go of the keys. Useful for transcription workflows where you want to lift your hands to type or take notes. Turn off if your controller sends sustain CCs you do not want the plugin reacting to.

Detection outputs (read-only)

These four parameters expose the detection result for host automation, screen-recording overlays, and the Headless variant's parameter-driven display:

- **Detected Root:** Current chord root (12 values).
- **Detected Quality:** Current chord quality (major, minor, 7, maj7, m7, sus2, sus4, dim, aug, etc.).
- **Detected Bass:** Bass note for slash chords (like C/E or G/B).
- **Detected Inversion:** Which inversion is currently being played.

You cannot set these from the host; they update automatically as the plugin detects chords.

Tips and Traps

- **The variant matters more than the version.** Most "I cannot find the plugin" issues are users looking for the MIDI variant in Ableton (which only sees the instrument variant) or the instrument variant in Logic (which puts the MIDI variant in the MIDI FX slot).
- **The plugin needs at least three notes to detect a chord.** Holding a single note shows nothing; holding two shows an interval but not a full chord name. Three or more notes give a full chord identification.
- **Detection runs on whatever notes are currently held.** If you arpeggiate a chord one note at a time without holding the previous notes, the display flickers between intervals. Use the sustain pedal (with **Respect Sustain** on) to hold notes for chord detection while you arpeggiate.
- **The Key setting does not affect detection.** It only affects which suggestions appear. C Major and A minor produce identical chord names; the suggestions shown below differ.
- **The Headless variant has no GUI.** It exposes detection through host parameters only. If you open it in a desktop DAW, you will see only the parameter list, no visualizer. Use the MIDI or instrument variant if you need the visualizer.
- **MIDI passes through unchanged.** Notes, velocities, channel messages, sustain, modulation, all forwarded to the next plugin in the chain. Chord Analyzer does not insert, suppress, or alter any messages.

Working with Suggestion Levels

Chord Analyzer does not ship with factory presets, but the **Suggestion Level** parameter behaves like a preset for the suggestion display.

Basic Only

Shows only the seven diatonic triads of your selected key. Useful if you are learning theory and want to see only the "in-key" chord options. If you play a chord that is not in your key, it is still detected and displayed; only the suggestions below restrict to in-key options.

Basic + Intermediate

Adds 7th chords (Imaj7, ii7, IV7, V7, vi7, etc.) and common borrowed chords (a iv minor in major, a III major in minor). This is the right setting for most pop, rock, and jazz songwriting work.

All (+ Advanced)

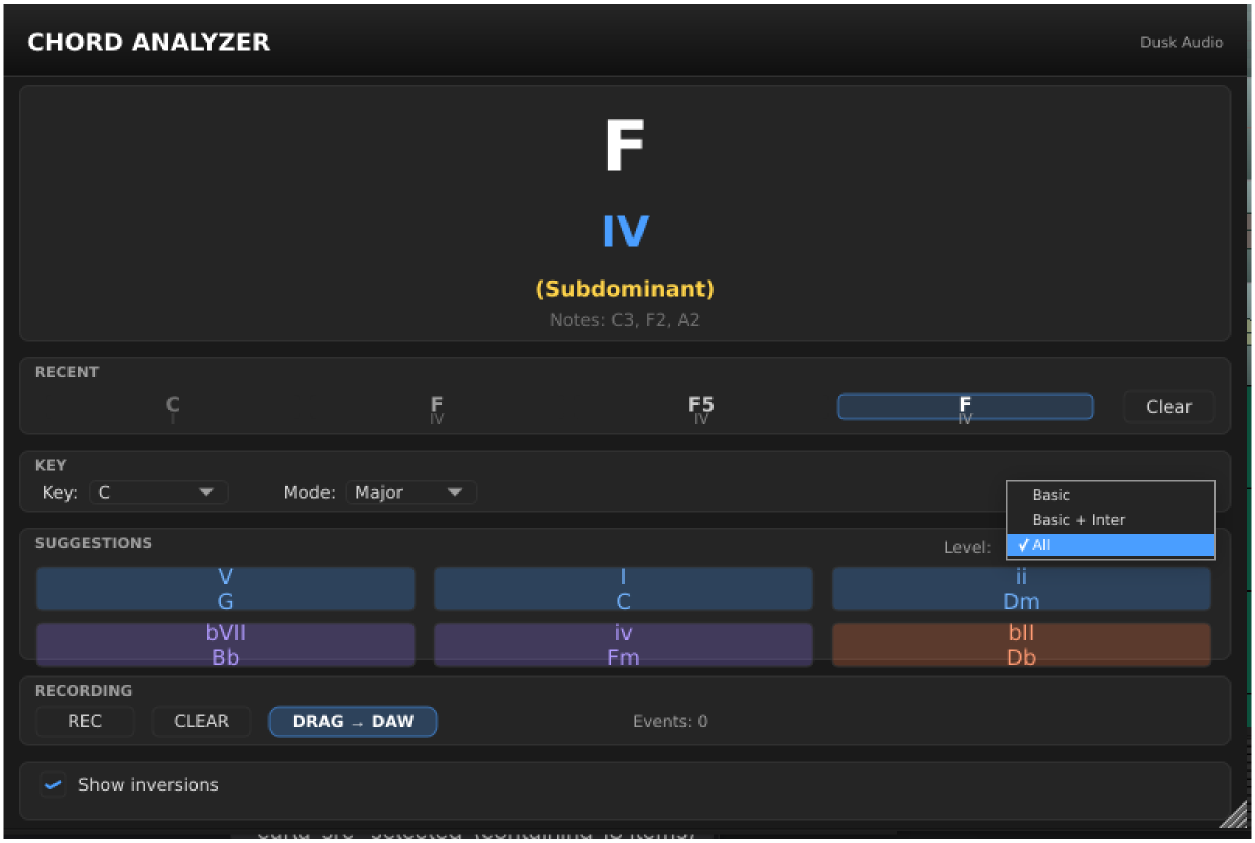


Figure 26: Suggestions panel with All tier visible

Adds tritone substitutions (where bII7 can replace V7), secondary dominants (V7/V, V7/vi, etc.), and modal interchange options drawn from parallel modes. The default; useful for jazz transcription and substitution exploration.

If the suggestion list feels overwhelming, drop to Basic + Intermediate. If you are working through a Real Book chart and the substitutions feel limiting, stay on All.

Troubleshooting

The chord display is empty even though I am playing notes. Confirm the plugin is receiving MIDI. In most DAWs, the MIDI variant must be inserted before the synth on the same track. In Ableton and FL Studio, the instrument variant must be in a parallel chain receiving the same MIDI.

If MIDI is reaching it but no chord shows, you are probably holding fewer than three notes; the plugin needs at least a triad to identify a chord.

My DAW does not show the plugin in its menu. Ableton and FL Studio do not list the MIDI variant in their MIDI plugin menus; use the instrument variant with the routing described above. Logic Pro lists the MIDI variant only in the **MIDI FX** slot, not in the regular instrument menu. If you cannot see either variant anywhere, your DAW has not yet rescanned plugins; force a plugin scan in the host preferences.

MIDI is not reaching my synth after I insert Chord Analyzer. The plugin passes MIDI through unchanged in all desktop variants. If your synth stops receiving notes, the plugin is being inserted as the only destination rather than as a pass-through. Check your DAW's MIDI routing; in Ableton's Instrument Rack approach, both chains receive the same MIDI in parallel, not sequentially.

The detected chord disappears when I let go of the keys. That is expected behavior unless you have **Respect Sustain** on and are pressing the sustain pedal. With the pedal pressed, the detected chord persists until you release the pedal.