

GrandMaster

Five bands. Two ways to work.

Mixing

Real music · Gentle EQ



Mastering

Real music · Gentle EQ



Contents

Page numbers match the PDF reader. Select a chapter to navigate.

1. Product overview	3
2. The parametric design tradition	4
Engineering choices in high-end analogue designs	4
From design background to this plug-in	4
3. Installation and first use	5
4. Quick start	7
5. Interface map	8
6. Panel parameters	9
7. Q, bandwidth and filter shape	11
8. Modes, routing and output	13
Mono operation	14
Independent style banks and host automation	15
9. Gain, detents and headroom	16
10. Channels, M/S and mono	18
11. Settings, manual and presets	20
Host-managed presets	21
12. Spectrum and EQ response display	22
Band nodes and readout gestures	24
Resolution and valid frequencies	25
13. AUTO static output compensation	26
Calculation definition	27
Scope and limits	27
14. Linear response and harmonics	28
15. Oversampling and latency	30
16. Troubleshooting and support	32
17. Licence and intellectual property	34

Product overview

GrandMaster is a five-band parametric equalizer for mono and stereo tracks, built from measurements of a high-end analog equalizer. Choose Mixing for continuous frequency and Q controls, or Mastering for labelled detents. Both control styles use the same sound model.

A new instance opens in English with the settings below. Existing projects restore their saved state.

Control	Initial setting
Control style	Mixing
Channel fit	Original
Quality	High precision
Band shape	LOW / HIGH: Shelf; middle bands: Q 1.0
Sound extensions	Hardware character, Measured noise and Noise follows EQ on; ADDA off
Routing and output	Stereo: L/R and LINK on; Trim: 0.0 dB. Mono uses the left model
Analysis and compensation	Spectrum closed; AUTO off

Original retains measured left/right differences. Average Fit uses the mean response. These choices change the channel relationship in the model; they are not loudness or EQ-strength shortcuts.

The parametric design tradition

Independent frequency, level and width let engineers decide where, how much and how broadly a correction acts. A 1972 professional audio conference paper described these parametric controls alongside active resistance-capacitance circuits and transient/phase design. The contribution is practical, composable control for both broad shaping and focused correction.

Engineering choices in high-end analogue designs

Architecture	How it works
Discrete Class-A	Discrete uses individual devices; Class-A biases the amplifying stage to conduct throughout the signal cycle. These choices give designers control of bandwidth, headroom and noise.
Electronic balance / DC servo	Differential input handles balanced signals. Direct coupling removes interstage series capacitors; the servo stabilizes output DC offset.
Overlapping bands / end shelves	Bands combine focused corrections; end shelves adjust a broad low or high region.
Separate power supply	An external enclosure separates supply and amplifier thermal design.

Cost factors: selected components, manual assembly, individual calibration and small production batches require parts, labour and testing. These engineering investments help explain high-end equipment costs.

From design background to this plug-in

GrandMaster implements sound processing through digital filters, measured calibration and bounded nonlinear corrections. This circuit background explains design choices. Hardware specifications belong to their original documentation; this plug-in’s implemented scope is defined by the technical chapters in this guide.

03

Installation and first use

Item	Current release requirement
System	macOS 11.0 or later
Architecture	Universal: arm64 and x86_64
Format and channels	VST3, AU and AAX; matched mono 1-in/1-out or stereo 2-in/2-out
Plug-in location	/Library/Audio/Plug-Ins/VST3/GrandMaster.vst3 /Library/Audio/Plug-Ins/Components/GrandMaster.component /Library/Application Support/Avid/Audio/Plug-Ins/GrandMaster.aaxplugin
Licence manager	/Applications/JMECHO Audio License Manager.app

This installer includes JMECHO Audio License Manager 1.3.4. A newer installed manager is preserved; updating this plug-in does not automatically update other products' website downloads.

The minimum build target is macOS 11. The application and plug-in build targets include arm64 and x86_64. Your DAW must support the operating system and plug-in format as well. Build targets and testing on a current Mac do not establish operation on the oldest macOS version or a physical Intel Mac; those environments need separate verification.

1. Run the current official installer and complete standard administrator authentication in the macOS system window.
2. Restart the DAW and scan VST3, AU or AAX using its normal procedure. A running host may continue using the previous loaded version.
3. Obtain your GrandMaster activation code from your website account and enter it in the plug-in activation screen, or start the seven-day trial through an available official entry.
4. After success, read the first-use guide and click Start using GrandMaster. The full EQ panel then appears.

Activation and licensing

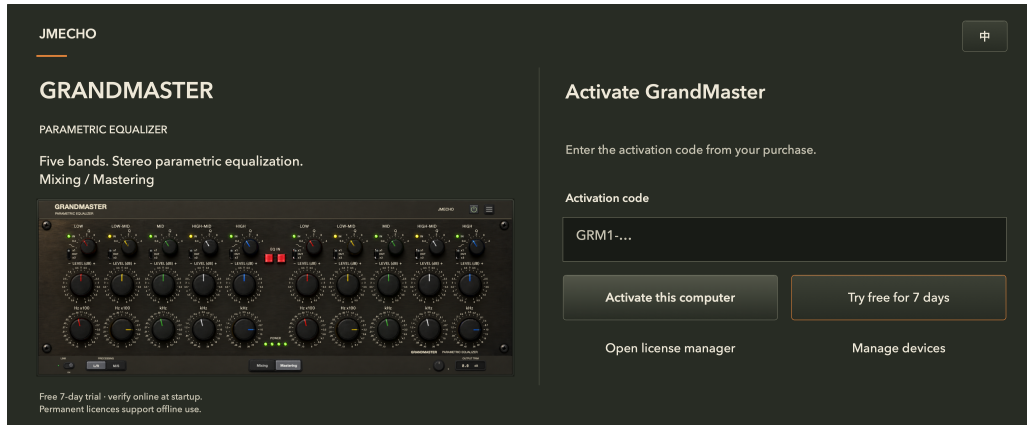


Figure 1. GrandMaster activation screen. No customer activation code is shown.

Initial perpetual activation needs Internet access. Once valid device-bound credentials are stored locally, a perpetual licence works offline without a continuous network heartbeat. The trial runs for seven consecutive days from first successful issuance. Each new host process needs online validation; a short lease lasts at most 24 hours and never extends beyond trial expiry. The trial does not automatically purchase or charge.

Documentation and notarization do not confirm that sales, trials or downloads are available. Use current official channels and your valid entitlement.

Without a valid licence and completed first-use confirmation, the plug-in, including internal Power bypass, is silent. Host bypass routes around the plug-in and does not prove activation.

04

Quick start

1. Load GrandMaster on a mono or stereo track or bus and confirm licensing is ready.
2. Keep Original selected and choose Mixing or Mastering for your preferred control style.
3. Adjust one band's frequency, LEVEL and Q, starting with a small gain change.
4. LOW and HIGH start as shelves. Drag their Q upwards for a bell; drag below 0.4 to return to Shelf.
5. Optionally enable AUTO for an estimate, then use OUTPUT TRIM to match processed and unprocessed loudness on the actual source.
6. Click the upper-right power button or the bottom POWER lamps to compare processing and bypass.

Drag vertically to adjust a knob; hold Shift for finer movement. The cursor hides during the drag and returns on release. Frequency values use the unit-label slot above the knob; other controls show their own readouts. Double-click a knob to type. Frequency entries display Hz and also accept kHz. Press Enter or click outside to apply a valid value; press Esc to cancel.

Click the OUTPUT TRIM number to type a value, then click outside to apply it. Mixing GUI gain and Trim use a 0.1 dB grid. Mastering gain steps are 0.5 dB at x1 and 1 dB at x2; the visible value and printed scale show effective gain. One decimal on screen does not rewrite continuous values restored from a project or sent by host automation.

At your fingertips

Action	Result
Drag up or down	Adjust the knob; hide the cursor during the drag
Shift + drag	Make finer adjustments
Option + click	Restore this knob's default
Double-click a knob	Enter a value directly
Enter / click outside	Apply a valid entry
Esc	Cancel the entry
Click a printed value	Jump to that position

05

Interface map



Figure 2. Numbered control map. The two band banks share the same structure; each number identifies one example.

No.	Control or area	Reference chapter
1	Q and shelf icon	Panel parameters
2	Frequency (middle row)	Panel parameters
3	LEVEL (bottom row)	Panel parameters
4	Band IN lamp	Panel parameters
5	x1 / OUT / x2	Panel parameters
6	Central EQ IN	Panel parameters
7	LINK	Modes and output
8	L/R / M/S	Modes and output
9	Upper Mixing / Mastering	Modes and output
10	OUTPUT TRIM	Modes and output
11	POWER lamps	Modes and output
12	Three-line settings menu	Settings, manual and presets

Both Power controls change the same state. Lamps show Power, not licensing; internal bypass remains gated when unlicensed.

06

Panel parameters

Band	Mixing range	Mastering labels
LOW	15 to 800 Hz	13 to 800 Hz
LOW-MID	15 to 800 Hz	13 to 800 Hz
MID	120 Hz to 8 kHz	130 Hz to 8 kHz
HIGH-MID	400 Hz to 26 kHz	420 Hz to 26 kHz
HIGH	400 Hz to 26 kHz	420 Hz to 26 kHz

1. Q / Shelf

Sets bell-filter width.

Range: 0.4 to 4.0; Mastering uses 0.4, 1, 2, 3 and 4. / Default: LOW and HIGH: Shelf; middle bands: Q 1.0 / Bell. / Unit: —

Effect: A higher Q narrows the affected range; a lower Q broadens it. LOW and HIGH can become low and high shelves.

Conditions: Shelf is available on LOW/HIGH only. Mastering GUI Q edits use 0.4, 1, 2, 3 and 4; Mixing adjusts continuously. Moving LOW/HIGH below minimum Q can enter Shelf; moving upwards returns to Bell. Finer Q values in legacy or host-saved states keep their saved interpretation rather than being unconditionally rewritten.

Tip: A wide Q is a useful starting point for broad tonal changes.

2. Frequency

Sets the band's frequency position.

Range: See the range table. Mastering has 24 labelled positions per band. / Default: New Mixing instance: 100, 300 Hz, 1, 3 and 10 kHz, ordered LOW to HIGH. The first Mastering visit maps these to 92, 320, 920 Hz, 3 and 10 kHz. / Unit: Hz / kHz

Effect: Mixing adjusts continuously; Mastering snaps to the nearest labelled value.

Conditions: A nominal label is not every unit's exact electrical centre. A label can exceed current Nyquist; it does not then represent a displayable audio frequency, although the actual engine still computes its skirt inside the valid band.

Tip: 6000, 6k and 6 kHz all enter 6 kHz.

3. LEVEL (dB)

Sets effective band gain, shown in the bottom control row.

Range: Mixing: -15 to +15 dB. Mastering effective gain: x1 -6 to +6 dB; x2 -12 to +12 dB. / Default: 0.0 dB / Unit: dB

Effect: Positive values boost; negative values cut. Mastering x2 doubles the gain range, displayed numbers and printed scale. Numeric entry is in effective dB.

Conditions: The band does not contribute EQ gain while its IN or the channel EQ IN is off.

Tip: Click a printed value to jump to it, or double-click the knob to type.

4. Band IN

Enables or disables a single band's EQ.

Range: On / Off / Default: On / Unit: —

Effect: Click the IN lamp. When off, that band's EQ gain contribution is zero.

Conditions: This does not disable Hardware character, ADDA or Output Trim for the whole plug-in.

5. x1 / OUT / x2

Selects the Mastering gain range.

Range: x1 / OUT / x2 / Default: x1 / Unit: —

Effect: x1 uses the dial value; OUT sets effective band gain to 0 dB; x2 doubles the dial value, for an effective -12 to +12 dB range.

Conditions: Mixing always processes as x1. The range switch does not change Mixing gain.

6. Central EQ IN

Enables or disables five-band EQ for its channel.

Range: Stereo: independent left/right switches. Mono: left switch only. / Default: Both on / Unit: —

Effect: When a channel is off, its bands do not contribute EQ gain.

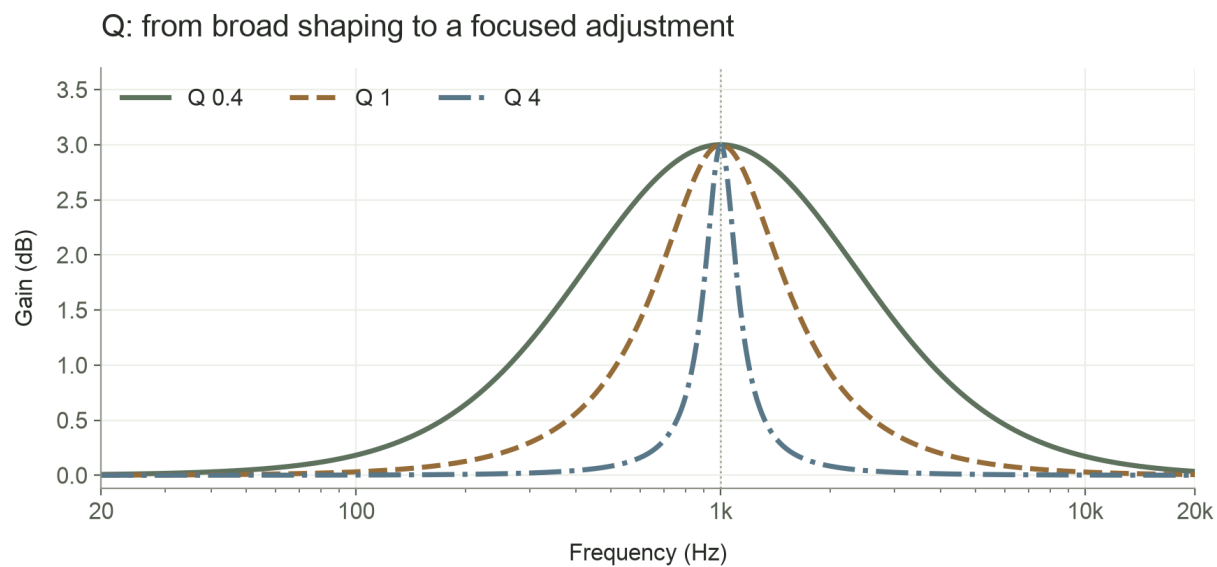
Conditions: Hardware character, ADDA and Output Trim have separate controls. EQ IN is not whole-plug-in power bypass.

Click a printed Q, frequency or LEVEL value to jump to that position. Clicking a Q value selects Bell. Option-click a knob, or press Home while it is focused, to restore its default: gain/Trim 0 dB, frequency its default position, LOW/HIGH Shelf and middle Q 1.0. Arrow keys adjust the focused knob. In stereo, LINK mirrors these GUI edits.

07

Q, bandwidth and filter shape

Frequency positions the adjustment. Gain sets its boost or cut. Q controls how concentrated a Bell adjustment is. Identify the part of the sound to change, then refine its width.



Concept illustration · not a measured response · teaching biquad, 1 kHz / +3 dB

Figure 3. A teaching Bell at 1 kHz and +3 dB, using Q 0.4, 1 and 4. Concept illustration, not a measured GrandMaster or hardware response.

Q	Effect and use
0.4	Broad; a starting point for overall tonal balance.
1.0	Between broad and focused; the initial setting for the three middle bands.
4.0	Narrow; locate the frequency and compare the local problem against nearby tone.

Q is not a fixed bandwidth in Hz. A resonant-filter definition relates Q to centre frequency divided by bandwidth; a peaking EQ also needs a gain and filter definition. The figure illustrates increasing concentration without replacing this product’ s measured fit with a teaching formula.

Bell and Shelf

Bell boosts or cuts a region around the selected frequency and tends toward the original level farther away. A low shelf changes the low-frequency end; a high shelf changes the high-frequency end, tending toward a new level on that side. Shelf is neither an infinitely wide Bell nor a high-pass or low-pass cutoff.

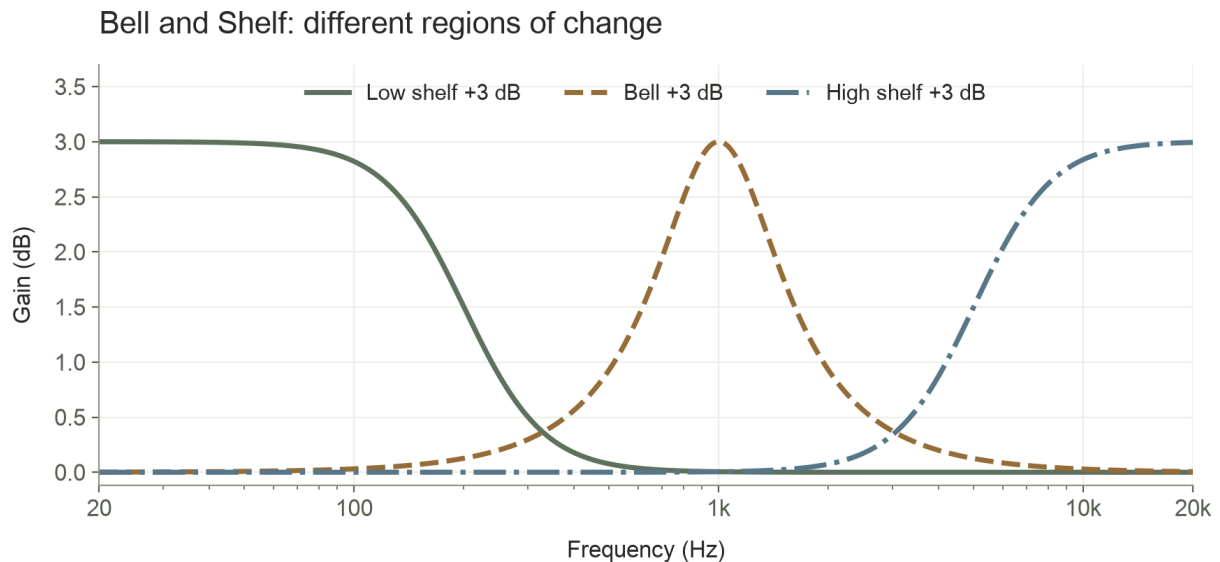


Figure 4. Teaching +3 dB low-shelf, Bell and high-shelf shapes. Concept illustration; corners and slopes do not specify the measured model.

LOW and HIGH offer Shelf; the middle three bands offer Bell only. A new instance starts with both end bands in Shelf. Shelf does not use the Bell bandwidth meaning of Q. Return to Bell to use Q from 0.4 to 4.0; Mastering detents are 0.4, 1, 2, 3 and 4.

A printed frequency can reach 26 kHz, while sample rate still limits the frequencies digital audio can represent. Nyquist is 22.05 kHz at 44.1 kHz and 24 kHz at 48 kHz. Shapes near that limit depend on the digital implementation; a 26 kHz label does not mean a low-rate stream can reproduce a 26 kHz sine.

08

Modes, routing and output

7. LINK

Links GUI editing of the current left/right control banks.

Range: On / Off / Default: On / Unit: —

Effect: Enabling LINK in the GUI copies the selected side's five-band and EQ IN values to its partner. Subsequent corresponding GUI edits affect both.

Conditions: Raw host automation of LINK or channel parameters does not retrospectively synchronize them in the audio callback. LINK does not remove Original's measured calibration differences. In M/S it links Mid/Side controls; it is disabled in mono.

8. L/R / M/S

Selects stereo or mid/side processing.

Range: L/R / M/S / Default: L/R / Unit: —

Effect: In L/R, the left and right banks process left and right. In M/S, the left bank processes Mid and the right bank processes Side; output is decoded back to stereo.

Conditions: M/S is a plug-in routing feature, not a claim that the measured hardware includes this route.

9. Mixing / Mastering

Selects a control style and recalls its independent GUI parameter bank.

Range: Mixing / Mastering / Default: Mixing / Unit: —

Effect: Mixing offers continuous frequency and Q with 0.1 dB GUI gain. Mastering uses frequency and Q detents; effective gain steps are 0.5 dB at x1 and 1 dB at x2.

Conditions: GUI Mixing/Mastering switches retain separate frequency, gain, Q, shape, range, Band IN, EQ IN and Q-detent states. The first visit to an unused style projects current effective values to legal positions; later returns restore its saved bank. Both banks are saved with the project. Raw host mode automation changes the parameter style without an implicit GUI bank recall or mirroring.

10. OUTPUT TRIM

Adjusts mono or stereo output level at the end of the chain.

Range: -12 to +12 dB; 0.1 dB GUI edits. / Default: 0.0 dB / Unit: dB

Effect: Higher values raise output level; lower values reduce it.

Conditions: It follows M/S decoding. AUTO adds an independent common gain without rewriting manual Trim. Neither is applied while Power is off; output gain cannot produce unlicensed audio.

Tip: Click the number to type. Enter or an outside click applies; Esc cancels.

11. Power / POWER

Enables or bypasses the whole plug-in chain.

Range: On / Off / Default: On / Unit: —

Effect: The upper power button and bottom POWER lamps are synchronized. Off bypasses EQ, Hardware character, ADDA, noise, manual Trim and AUTO.

Conditions: A valid licence and first-use confirmation are prerequisites. Host external bypass is separate.

Mono operation

A mono instance accepts one input and one output and uses the unaveraged left measured model throughout EQ, Hardware character, noise and ADDA. The right controls, LINK, L/R-M/S and channel-fit selection are disabled. They retain their stored values without affecting mono audio; a restored stereo layout can use those settings again. Mode, Power, the left bank and Output Trim remain available. Mono-to-stereo layouts are not supported.

Independent style banks and host automation

On the first GUI visit from Mixing to Mastering, an effective gain magnitude up to 6 dB uses x1 and the nearest 0.5 dB position. Above 6 dB it uses x2, the nearest 1 dB position and a 12 dB limit. For example, Mixing +11 dB maps to Mastering +11 dB / x2; +14 dB maps to +12 dB / x2, while returning to Mixing restores +14 dB. Existing OUT is retained, and later Mastering visits restore the saved range. Initial projection can alter sound; check effective readouts.

Original / Average Fit, quality, L/R / M/S, LINK, ADDA, sound extensions, AUTO and manual Output Trim are shared rather than separately stored in the two style banks. Host automation may change raw mode alongside other parameters; the plug-in does not infer a complete transaction or recall a hidden bank in its audio callback. For repeatable automation, record the relevant parameters explicitly in the host.

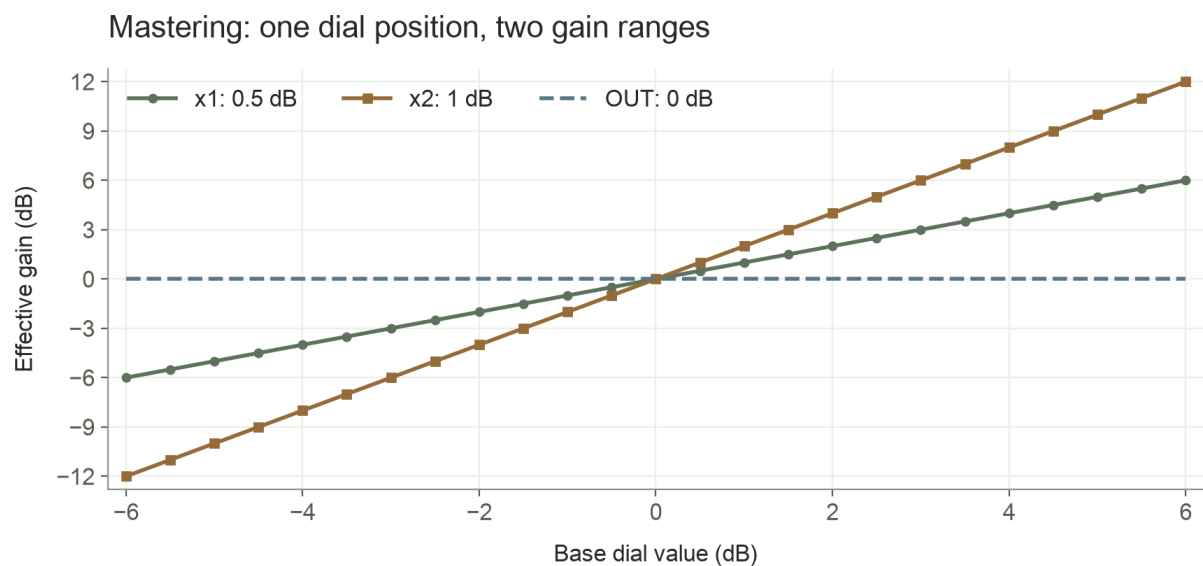
09

Gain, detents and headroom

Mixing and Mastering change the control style. They share the current sound model; they do not select two separately measured devices. Original / Average Fit and High precision / Low latency are separate settings.

Control	Mixing	Mastering
Frequency	Continuous	24 labelled positions per band
Q	Continuous, 0.4 to 4.0	0.4 / 1 / 2 / 3 / 4
GUI gain	-15 to +15 dB; 0.1 dB	x1: -6 to +6 dB; 0.5 dB
Gain range	Processes as x1; range switch has no effect	x2: -12 to +12 dB; 1 dB

0.1 dB is the Mixing knob's GUI edit step. Host automation can retain finer floating-point values; the single displayed decimal is not the processing resolution. Existing projects preserve their saved values and compatibility behaviour.



Concept illustration · not a measured response · current control mapping, not a response measurement

Figure 5. Mastering maps the base dial to effective gain: x1 retains it, x2 doubles it and OUT gives zero. Drawn from current control rules, not a measured response.

For example, a +3 dB base position gives +3 dB at x1 and +6 dB at x2. Printed gain numbers, readouts and direct entries show effective dB. x1 / x2 multiplies neither frequency nor Q. OUT or Band IN Off removes this band's EQ gain, while Hardware character, ADDA and Output Trim remain separate.

Compare with Trim, without favouring louder audio

dB expresses an amplitude ratio: multiplier = $10^{(dB / 20)}$. +3 dB is about 1.413x, +6 dB about 1.995x and -6 dB about 0.501x. These are amplitude conversions, not a promise that arbitrary music peaks change by the same dB; band overlap, phase and source content determine the peaks.

Position	Effect
Level before the plug-in	Changes linear-path amplitude and the level seen by the nonlinear kernel, potentially changing harmonic ratios.
Band gain	Changes the spectrum and level reaching the later character path.
OUTPUT TRIM	Adjusts the chain output after M/S decoding; it does not change harmonics already generated upstream.

Keep peak headroom at the input and through the processing path, then use Output Trim to match bypassed and processed loudness. There is no automatic total-output safety limiter. A lower final peak alone does not establish an unchanged operating level in the upstream nonlinear stage.

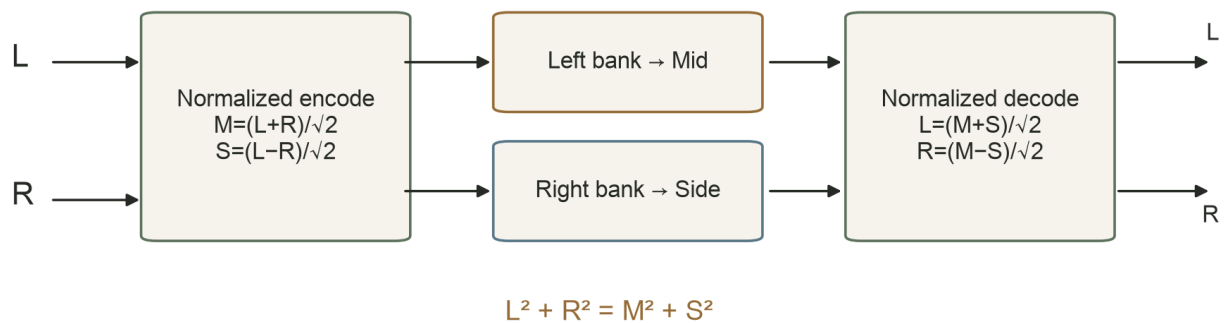
The weak increment's input bound limits the calculation of that small correction, not the main signal. It is not a complete overload, clipping or saturation model. Large input boosts are not a separately measured drive feature.

10

Channels, M/S and mono

L/R processes left and right directly. M/S transforms them into Mid and Side, processes Mid with the left bank and Side with the right bank, then reconstructs stereo. It separates stereo components, not automatically vocals from instruments.

M/S: transform the channels, preserve input energy



Concept illustration · not a measured response · routing illustration; processing can change output energy

Figure 6. M/S uses $1/\sqrt{2}$ -normalized encoding and decoding. The figure shows routing only; processing between the matrices can change output energy. Concept illustration, not a response measurement.

The normalized relations are $M=(L+R)/\sqrt{2}$ and $S=(L-R)/\sqrt{2}$; decoding uses $L=(M+S)/\sqrt{2}$ and $R=(M-S)/\sqrt{2}$. The transform alone satisfies $L^2+R^2=M^2+S^2$. If L and R are identical, Side is zero and Mid is about 3.01 dB above either single channel; total energy has not increased.

Turn LINK off before adjusting Mid and Side independently. Enabling LINK in the GUI copies the selected side's five-band and EQ IN values to the other side; subsequent corresponding GUI edits are mirrored. Raw host automation writes its own parameters without this implicit GUI copy.

Original, Average Fit and mono

Setting	Calibration and scope
Original	Uses each measured calibration and retains full left/right differences. Equal controls do not guarantee identical channel responses.
Average Fit	EQ and hardware-character paths average the two calibrations. It does not sum the left/right audio or collapse it to mono. ADDA remains a separate converter-channel model.
Mono	One input and one output; uses the original left EQ, character, noise and ADDA model throughout, without blending left/right differences.

A mono instance disables the right bank, LINK, L/R-M/S and channel-fit selection. Mixing / Mastering, the left Band IN / EQ IN, Power and Output Trim remain available. Stored disabled-control values do not affect mono audio and are retained if a stereo layout is restored. Mono-input to stereo-output layouts are not supported.

Average Fit changes model calibration, not automatic loudness compensation, and does not remove converter colour already recorded in the source. It does not establish separation of all distortion from the original recording chain.

Settings, manual and presets

12. Settings menu

Collects sound extensions and interface options.

Range: Click the upper-right three-line button. / Default: Closed / Unit: —

Effect: Click outside to close the settings panel.

Conditions: Settings are stored with the project.



Figure 7. Settings for sound, interface, details and the manual.

Option	Meaning
High precision / Low latency	High precision uses longer correction filters; Low latency reduces processing delay and cost. Switch while stopped.
Original / Average Fit	Original retains full measured left/right differences; Average Fit uses the mean response. Legacy Precision projects retain their sound, but the new menu does not offer that choice.
Hardware character	Adds baseline response and weak input-level-dependent harmonics. FLAT and working EQ retain the same FLAT harmonic kernel; this is not verified identification of every nonlinear operating state.
Measured noise	Enables statistical noise. At least Hardware character or ADDA must be on for the corresponding noise to be added. Random noise is unsuitable for deterministic sample-by-sample null comparisons.
Noise follows EQ	Changes noise distribution where identified; intermediate positions remain model estimates.
ADDA	Adds the measured linear D/A+A/D response of a high-end studio converter and its noise floor when Measured noise is on. Off does not remove converter coloration already present in the input. New instances default to Off; saved projects retain their stored setting.

Option	Meaning
Interface size	80%, 90%, 100%, 110%, 125%, 150% and 170%. Display only; stored in the project.
Language	English is the default. The English interface shows 中 to switch to Chinese; the Chinese interface shows EN to switch to English.
Details	Displays version, processing status, sample rate, current plug-in latency and quality.
Manual	Opens the installed PDF for the current interface language, available offline.
License management	Displays GrandMaster licensing status and relevant actions without changing other products' licences.

Host-managed presets

There is no internal preset bar. Use the host's VST3 preset function to load .vstpreset files; browsing and import methods vary by host. Factory files include Neutral and two entries each for Mastering, MixBus, Vocal, BV, ATG, ETG, Drums and Bass. Treat them as starting points and adjust for the source while matching loudness.

Factory files store a complete state. Loading one can change quality, mode, ADDA, routing and Trim, not just the EQ curve. Your projects restore their saved settings. Installation preserves conflicting user preset files.

12

Spectrum and EQ response display

Click the spectrum icon to the right of the upper power button to open the analysis region between the toolbar and physical panel. Click again to close it. It is closed by default. The expanded state and selected L/R or M/S side are saved with the project; hiding the editor stops capture and FFT processing. Opening the spectrum does not change audio, quality or reported plug-in latency.



Figure 8. Native capture using generated six-tone test PCM: 96 kHz processing and 48 kHz FFT analysis. Input tones are 80, 200, 1000, 3200, 9000 and 15000 Hz. Mixing EQ settings are a 100 Hz low shelf at -2 dB, a 1 kHz Bell at +3 dB / Q 1 and a 10 kHz high shelf at +1.5 dB. Trim is 0 dB; AUTO, ADDA and noise are off.

Display	Definition and unit
EQ response / left dB	The current digital EQ-engine response divided by its same-path zero-gain response. Both use the current fit, quality, enable state and control mapping. The line excludes manual Output Trim, AUTO and the separate ADDA, harmonic and noise paths.
Input spectrum / right dBFS	Calculated from actual L/R input samples. In M/S, the analysis worker applies the normalized mid/side transform for display.

Display	Definition and unit
Output spectrum / right dBFS	Independently calculated from actual complete output samples, including processing, manual Output Trim and enabled AUTO. It is not derived by adding the EQ line to the input spectrum.
Frequency / logarithmic Hz	The axis spans 10 Hz to 30 kHz. The EQ curve stops at the lower of the host and analysis Nyquist limits and 30 kHz; FFT data follows the actual analysis sample rate.

The EQ scale follows the selectable gain range, rather than the peak of the combined curve: Mastering x1 is -6 to +6 dB; an x2 setting in either stereo control bank selects -12 to +12 dB for the whole graph, including M/S. Mono checks the left bank only. Mixing is -15 to +15 dB. Overlapping bands can exceed a single-band range; the graph clips portions outside its scale. The actual spectrum has a separate visible scale of 0 to -96 dBFS on the right. Do not interpret that scale as EQ gain.

Select one L/R or M/S side to edit stereo controls. With LINK on, the selected side represents the nominal EQ curve; original calibration differences and independently measured input/output spectra remain intact. Separate host automation can still leave different raw values on the two sides; disable LINK to inspect both independent responses. Mono uses the left model only; right selection, LINK and M/S are disabled.

Band nodes and readout gestures

Action	Result
Drag a band node	Horizontal movement changes the nominal frequency; vertical movement changes effective gain. The node follows an isolated Bell peak or the half-transition of a Shelf and lies on the combined response. Its location can differ from the nominal dial frequency. Graph edits and physical controls share the current values and LINK rules.
Drag frequency / gain / Q in the readout	Adjust the corresponding value. Q defines width; Shelf on LOW or HIGH identifies the shelving shape.
Shift + drag	Reduce the change per pointer movement, including both Q handles and the Q readout. The cursor hides during the drag and returns on release.
Mouse wheel	Adjust the current band Q. Mastering uses 0.4, 1, 2, 3 and 4; Mixing adjusts continuously. LOW/HIGH can enter Shelf below minimum Q and return to Bell when moved upwards.
Double-click a node	Toggle this band's Band IN. Reactivating a Mastering OUT node returns its range to x1 and enables the band; an ordinary bypass preserves x1 or x2. Double-clicking a physical knob opens numeric entry.
Option + click a node	Restore the default band frequency and 0 dB gain. Option-clicking a readout field restores that field only; the Q field restores Q 1 and LOW/HIGH also return to Shelf.
Click outside the spectrum / Esc	End the current drag and dismiss the band readout.

Resolution and valid frequencies

FFT runs on actual captured audio in a background worker. The audio callback only writes paired input/output samples to a preallocated bounded queue. Display analysis uses an 8192-point Hann-window FFT, peak-preserving logarithmic grouping and display-only fast attack / slow decay. At higher rates, background anti-alias half-band filters reduce the analysis rate without changing the audio-processing sample rate. The spectrum is a relative analysis display, not spectral power density or integrated LUFS measurement. Below one FFT-bin spacing, existing bin powers are interpolated for display; contour segments meet smoothly without overshoot. This does not add FFT resolution or affect EQ, AUTO or the audio output.

FFT bin spacing is the actual analysis sample rate divided by 8192: approximately 5.86 Hz at 48 kHz. A denser plot does not create additional low-frequency resolution. No simulated spectrum is generated when valid samples are absent. The EQ curve stops at the lower of the host and analysis Nyquist limits and 30 kHz. A band beyond that visible boundary keeps its stored frequency, while its node stays at the last visible response coordinate. The frequency readout uses whole Hz; Q handles stay near the selected node.

Closing the spectrum or hiding the editor stops input/output capture and FFT. Enabled AUTO can still update its static estimate after EQ-state changes. Oversized blocks, a full queue or a temporarily busy worker may drop analysis data and delay the display; analysis never waits for or truncates the real audio. Legacy fixed measurement slots have no five-band zero-gain reference, so no variable-EQ curve or AUTO value is fabricated for that path; actual audio FFT remains available.

AUTO static output compensation

AUTO is to the right of OUTPUT TRIM and is off by default. When enabled, it estimates a common output correction from the current EQ response and recalculates after changes to frequency, gain, Q, Shelf, band enable, fit, control mode or sample rate. It uses a fixed reference spectrum, not continuously varying programme level; it is neither a compressor nor adaptive automatic gain control.

AUTO is an automatable plug-in parameter saved with the project. Spectrum visibility and selected side are display state, not audio-automation parameters.

Item	Specification and effect
Reference estimate	A pink reference from 20 Hz to 20 kHz assumes equal energy per octave and applies K frequency weighting. At lower rates, Nyquist also limits the upper bound.
Channels	Mono uses the left response. Stereo assumes equal reference energy in both processing channels, averages the power of their actual EQ responses and applies one common gain. This correction does not independently change left/right balance.
Independent manual trim	The OUTPUT TRIM parameter is not rewritten. Final output dB = manual Trim + AUTO compensation. Trim remains available for preference or source-dependent matching.
Bypass and disable	AUTO off removes this correction. Power off bypasses both Trim and AUTO. Without valid processing authorisation or an available EQ reference, no correction is generated.
State changes	The worker coalesces recalculation and the audio path smooths its gain. It adds no lookahead delay to the audio path.

Calculation definition

The K frequency weight uses the 48 kHz reference filter coefficients in ITU-R BS.1770-5, Annex 1, Tables 1 and 2. GrandMaster combines that frequency response with its own fixed pink-reference calculation; this is not the standard's complete gated loudness-measurement algorithm.

At frequency point k , $R_c(k) = H_c(\text{current}) / H_c(\text{zero gain})$, where H is the same-path actual digital EQ response. Mono uses $E(k) = |R_L(k)|^2$; stereo uses $E(k) = (|R_L(k)|^2 + |R_R(k)|^2) / 2$. Compensation $C = -10 \log_{10}[\text{sum}(w(k) E(k)) / \text{sum}(w(k))]$, in dB. Weight w includes K-weighted power and logarithmic interval width. Only the valid range from 20 Hz to the lower of 20 kHz and actual Nyquist contributes.

Equal nominal gains at different frequencies or bandwidths can therefore produce different estimates. Overlapping bands use their combined response, not a simple average of the dial gains. An unchanged EQ response relative to zero gives 0 dB compensation. Manual Trim is excluded from the estimate.

Scope and limits

K weighting assigns frequency contributions to an assumed reference spectrum; AUTO does not measure the current source's LUFs. Source spectrum, listening sound pressure and masking change perceived loudness. AUTO cannot guarantee exact bypass loudness for every source and is not a strict equal-loudness-contour reconstruction. The separate ADDA, harmonic and noise paths are outside the incremental EQ reference. Play representative music, then use manual Trim to correct remaining differences.

Compensation is not a safety limiter. Instantaneous peaks can rise even when reference energy falls. Overlapping boosts, phase relationships and manual Trim all affect actual peaks. Check peak headroom when comparing processed and bypassed sound.

Linear response and harmonics

Component	What changes	Level behaviour
Linear EQ and baseline	Magnitude, phase and frequency balance	At fixed parameters, amplitude scales proportionally
Weak harmonic increment	Filtered 2nd- and 3rd-order signal-dependent components	Harmonic ratios depend on input level
ADDA	Adds the measured forward linear D/A + A/D response	The current ADDA response itself is linear filtering. New instances default to Off; saved projects retain their stored setting.
Measured noise	Random noise generated from an estimated spectrum	Not guaranteed to repeat sample for sample

Fixed coefficients do not imply linear processing. The weak kernel retains its calibration while operating on input squares, cubes and related time-dependent filtering. Its result changes with the input and remains nonlinear; one coefficient set is not one fixed harmonic recording.

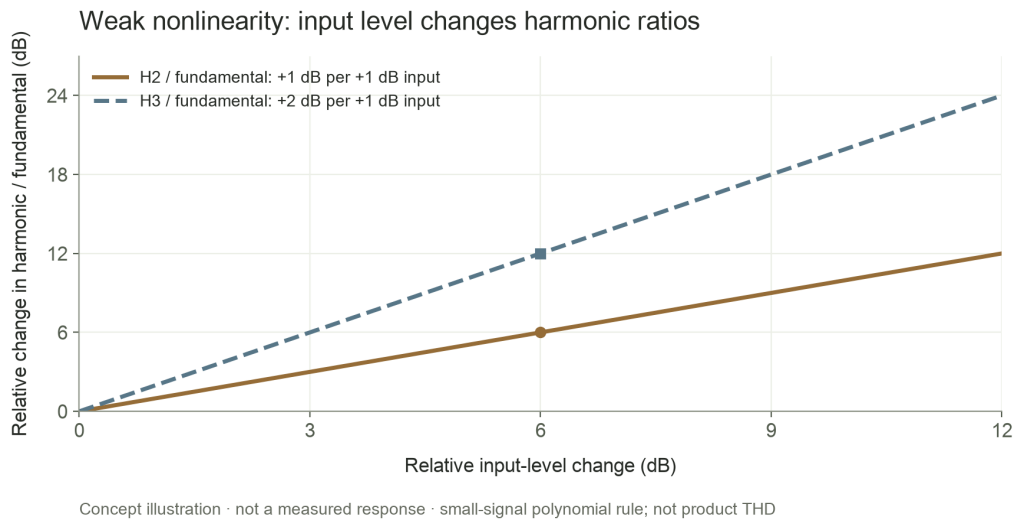


Figure 9. Small-signal polynomial scaling: +6 dB input can raise H2/fundamental by about 6 dB and H3/fundamental by about 12 dB. Concept illustration, not product THD or measured distortion levels.

The figure illustrates $y=x+a2x^2+a3x^3$ in a small-signal region. The implemented model adds filtering, calibration, bounded increments and local corrections, so arbitrary signals, frequencies and high levels need not follow these lines exactly. It does not specify total THD, a clipping threshold or maximum drive.

Character and the limits of identification

With Hardware character on, FLAT and working EQ retain the same baseline weak kernel. EQ precedes that kernel, so gain or spectrum changes can alter the signal it receives and the resulting harmonics. This does not mean every frequency, Q, gain or multi-band combination has its own independently measured nonlinear kernel.

Validated local working states use their corresponding corrections; unmeasured states still involve inference and interpolation. A closer linear curve, a smaller music null or a heavier impression alone cannot prove complete reproduction of hardware character.

ADDA On adds the measured forward response of a studio converter; Off omits that optional path. It does not inversely cancel D/A, A/D or other recording-chain colour already present in the audio. The weak harmonic calibration does not prove unique separation of converter nonlinearity.

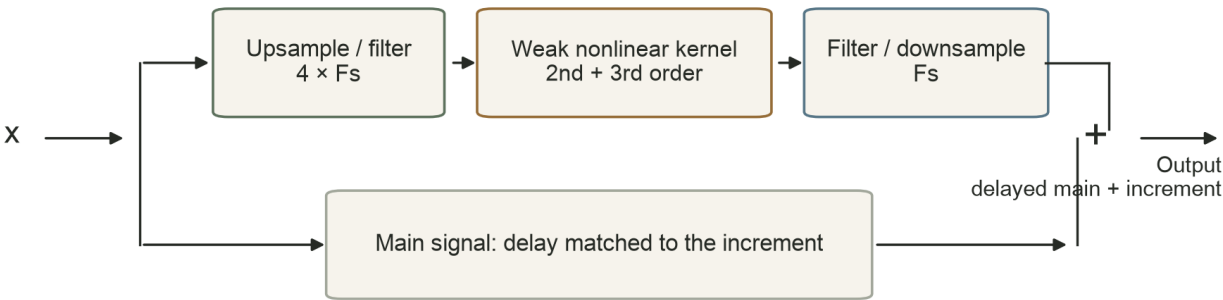
Make a repeatable comparison

1. Use the same source and fixed input level, sample rate, routing, control style, quality and ADDA state.
2. Turn off random noise and align samples using reported plug-in latency. Match one fixed output gain rather than changing the result with continuous automatic gain.
3. Inspect response, phase, transients and real music together. Judge the null with source peaks, repeated-measurement drift and loudness differences.
4. Restore the production settings and save. One null value is not a similarity score for all audio.

Oversampling and latency

Both quality settings calculate the weak nonlinear increment with 4x oversampling. Upsampling and downsampling filters help reduce the folding of generated harmonics into the audible band. The main signal is delay-matched and added to the increment. This does not make the full EQ a 4x path.

4x processes the harmonic increment only



Concept illustration · not a measured response · simplified module; the full EQ is not 4x

Figure 10. Main and increment paths within the weak harmonic module. Both qualities use 4x; High precision has longer filters. Concept illustration; other EQ, ADDA and noise modules are omitted.

Current full-model component	High precision	Low latency
EQ correction FIR / delay	513 taps / 256 samples	129 taps / 64 samples
Weak harmonic filter / delay	257 taps / 65 samples	129 taps / 33 samples
ADDA FIR / delay	257 taps / 128 samples	129 taps / 64 samples

Taps count filter coefficients; samples here count delay at the host sample rate. Harmonic filters operate at a 4x clock, so their tap count cannot be converted using a base-rate FIR half-length. Enabled modules and the currently applied processing state determine the total delay.

Use the current Details readout

The examples below are derived from the unchanged processing structure at 96 kHz, with Power, Hardware character and ADDA on and L/R routing. They are not a new-version host measurement and do not replace your host's current readout.

Configuration	Delay samples	Approx. time
Mastering / High precision	449	4.68 ms
Original + Mixing / High precision	513	5.34 ms
Current full model / Low latency	161	1.68 ms

Original + Mixing + High precision uses reference-clock increment transport at 44.1, 48, 88.2, 96, 176.4 and 192 kHz, which can add delay. Do not reuse the 96 kHz sample count at another rate. Quality, ADDA or control-style changes can alter filter histories and total latency; switch while playback is stopped.

Milliseconds = $1000 \times \text{delay samples} / \text{sample rate}$. For example, 449 samples is about 9.35 ms at 48 kHz or 4.68 ms at 96 kHz. Low latency still has filter delay; host buffers, interface I/O and other plug-ins add their own latency.

The plug-in reports processing latency; the host manages compensation. Power Off internally bypasses the processing chain and Trim, while host external bypass is separate. For live recording, check Details and host compensation settings. An aligned offline render does not establish low monitoring latency.

High precision uses longer correction filters for the current response fit. Higher rates and more instances still require computation. This guide does not claim oversampling is audible on every source, cost-free, or a complete reconstruction of unknown high-frequency harmonics.

Troubleshooting and support

Symptom	Check and action
Plug-in not listed	Confirm the supported VST3, AU or AAX format, installation location and macOS 11+ support, then use the host's normal rescan procedure.
No audio	Check licensing, first-use confirmation and input/output routing. Unlicensed internal bypass is silent; with a valid licence, Power Off should pass the input.
Band edits do nothing	Check band IN, channel EQ IN and Mastering OUT. A legacy fixed-state project needs an explicit current-model selection.
Typed value differs	Enter a finite number; frequency accepts Hz, k or kHz. Mastering snaps to detents and out-of-range entries are limited to the allowed range.
Activation fails	Use a GrandMaster code and check system time and helper network permissions. Do not use another product's code.
Agent startup timed out	Reopen the licence manager and retry. If the local agent remains unavailable or cannot be verified, run the complete official installer. Do not delete protected licence files.
First-use identity preparation timed out	Wait briefly and retry. The first request prepares this computer's protected identity; subsequent attempts reuse the existing identity.
Licensing server timed out	Check Internet access and retry. A server timeout differs from a local agent startup failure. Include the displayed error code when contacting support.
Trial service unavailable	Retry later or contact support with the error code. Starting or restarting the manager does not reset an already issued seven-day trial.
Trial fails offline	Each new host process needs online trial validation. Perpetual licensing has a separate offline policy.
Update appears unchanged	Restart the DAW; a running host may retain its previous loaded version.
Higher CPU or latency	Switch to Low latency while stopped and check instance count, sample rate and buffer. High precision is not zero-cost processing.
Manual does not open	Check that current-version local documentation is installed, or use the installer's PDF. Rerun the current installer if files are missing.

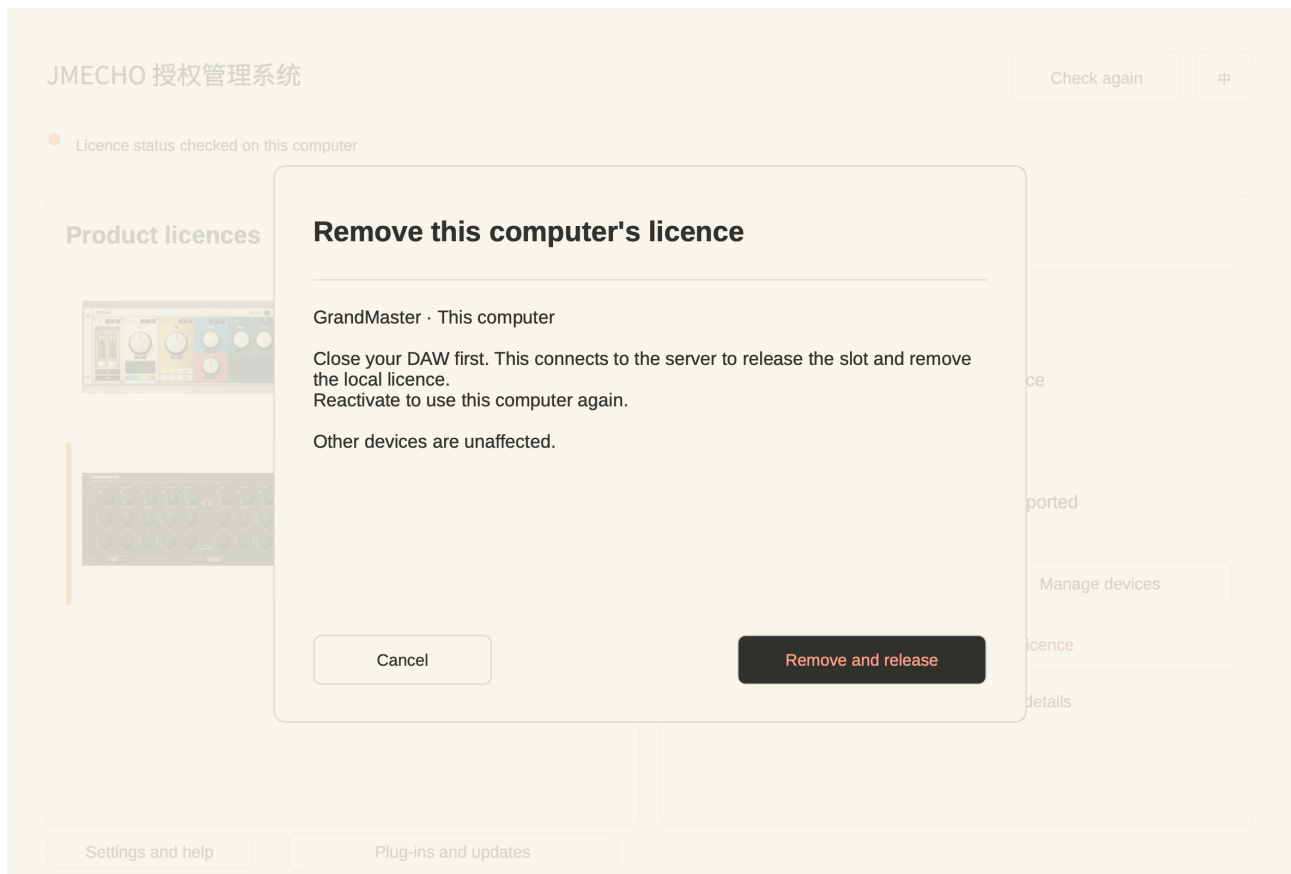


Figure 11. Licence removal confirmation. Check the product and device; Cancel keeps the licence.

Perpetual licensing provides 2 slots for the same user and up to 3 self-service migrations in any rolling 365-day period. Reinstalling the system or deleting protected storage does not guarantee preserved device identity. Contact support for recovery instead of manually removing key files.

To remove a perpetual licence from this computer, select GrandMaster in the licence manager and choose Remove this computer's licence. Review and confirm the selected product. An Internet connection is required to release this device's slot. Cancellation keeps the licence. An uncertain response pauses local processing while the same request is reconciled; reopen or check again rather than deleting protected files. Other devices and products are unaffected, and trial eligibility is not reset. After confirmed removal, activate again to use this computer.

Support: the feedback entry at <https://www.jmechoaudio.com/> or lijabin124@outlook.com. Include the product, version, macOS, DAW, sample rate, buffer and reproduction steps. Do not send passwords, verification codes or device private keys.

Licence and intellectual property

The EULA, licensing privacy notice and third-party licences are installed with the versioned documentation. Format and platform references imply no endorsement or affiliation.

JMECHO | Intellectual Property and Anti-Piracy Notice

Rights in the legally protected code, interface, graphics, resources and accompanying documentation belong to their respective lawful rights holders. A licence to use the software does not transfer those intellectual property rights. This notice concerns only rights lawfully held by the relevant rights holders and does not claim ownership of music that users create with the software.

Except as authorized by the relevant rights holder or expressly permitted by applicable law, the following are strictly prohibited:

1. Cracking, circumventing, defeating or tampering with licensing, activation, encryption or other technical protection measures; forging, misappropriating or unlawfully distributing licences, activation credentials or decryption keys.
2. Unauthorized reverse engineering, decompilation or disassembly of the software, or extraction, reproduction, adaptation or exploitation of its protected code and resources.
3. Creating, copying, selling, renting, uploading, sharing or distributing pirated or cracked versions, or tools, patches or services for unlawfully circumventing this software's protections.
4. Unlawfully removing, falsifying or altering copyright notices or rights-management information, or presenting unauthorized modified versions as official products.

JMECHO takes infringement seriously. Where infringement is substantiated, the relevant rights holders will promptly pursue appropriate lawful measures, which may include demands to cease infringement, platform complaints, preservation of evidence, claims for damages and legal proceedings. Suspected administrative or criminal violations may be reported to the competent authorities. Liability and its extent depend on applicable law, the evidence and determinations by the competent authorities.

This notice does not exclude rights that cannot be restricted by contract, statutory exceptions, or security testing and research permitted by applicable law. It does not restrict rights granted directly under third-party component licences, which continue to govern those components. This notice supplements rather than replaces the product's End User Licence Agreement (EULA). Providing software free of charge does not waive intellectual property rights or introduce payment or activation obligations absent from its existing licence.

For additional permissions, security reports or infringement reports, contact JMECHO through its official support channels. Avoid publicly disclosing customer data or secret material, or distributing unauthorized software copies.