USER GUIDE / SPECIFICATIONS / CIRCUIT DESIGN

6386e Tube Compressor

An original approach to tube dynamics.

6386e

The tube family.
The emulation.

Seven dials. A focused display. Independent control over compression, timing and output-stage drive.

Golden Age Series
Fletcher Audio

Version 1.0.0
24 September 2026

Inside

01 / IDENTITY

The design philosophy

Gain and character should move together.

6386e is designed as a character compressor for tracks and buses. Its identity starts with a matched, push-pull 6386-family gain cell: changing grid bias changes the modeled transfer, so compression and nonlinear behavior are connected. The letter **e** means emulation; it is our product name, not a separate hardware tube type.

Useful separation of controls.

Threshold and Ratio set compression demand; Attack and Release shape motion. Input Gain drives the main audio and the Internal detector. External detection follows a separate host-routed source. Heat adds drive after detection, Auto Gain provides reference makeup and Output Gain sets the final level.

A compact instrument.

The silver enclosure, cream and charcoal gain controls, orange Heat dial and dark LCD display give each control a clear role. The top row follows the compressor workflow; the second row groups level and color. Meter/Graph, Stereo/Dual and bypass remain inside the screen. There are no decorative selection rings or hidden Details pane.

THREE RESEARCH REFERENCES

Hardware reference	Design lesson
Fairchild 670 [1]	A 6386 push-pull gain stage and program-dependent recovery informed the interest in bias-controlled gain and a slower memory tail.
Manley Stereo Variable Mu [2]	Gentle compression, deliberate gain staging and linked stereo informed the intended workflow.
Tube-Tech CL 1B [3]	Its optical control element and tube amplification offer a useful reference for track leveling and flexible timing. 6386e does not model an optical cell.

These are complementary benchmarks, not three emulations hidden behind one faceplate. No hardware matching, endorsement or measured equivalence is claimed for 1.0.0. Sources are linked on page 12.

02 / MACOS AND WINDOWS SETUP

Install and open

> **macOS 11+:** universal Apple Silicon / Intel AU and VST3. **Windows 10+:** x64 VST3 and standalone application. Save projects and close DAWs before installation. Initial activation requires internet; see page 17.

macOS

Open the Setup DMG, launch setup, review the licence and accept if you agree, then choose Audio Unit, VST3, or both. Installation is for your Mac account. Reopen Logic and select **Audio FX > Audio Units > Fletcher Audio > 6386e Tube Compressor**. If absent, use Plug-in Manager > Reset & Rescan Selection.

Windows

Run the Setup EXE and follow its licence and component pages. Installation to Program Files requires administrator permission. Select 6386e in your DAW's VST3 list. The standalone app is available under Fletcher Audio in the Start menu; choose an audio device in its settings. For external sidechain routing, use the plug-in in a supporting DAW.

Item	Installation location
macOS Audio Unit	~/Library/Audio/Plug-Ins/Components/
macOS VST3	~/Library/Audio/Plug-Ins/VST3/
Windows VST3	C:\Program Files\Common Files\VST3
Windows app / docs	C:\Program Files\Fletcher Audio\6386e\ Guide and agreements: Documentation subfolder.

Updates and removal

On Mac, earlier owned plug-ins are backed up under **~/Library/Application Support/Fletcher Audio/**; the uninstaller, guide and terms are in **~/Applications/Fletcher Audio/**. On Windows, use **Settings > Apps** to uninstall 6386e Tube Compressor by Fletcher Audio. Both uninstallers preserve projects and unrelated plug-ins. Deactivate the computer in your AudioNode account when moving its licence.

Choose a host that supports the installed format and architecture. Runtime checks include Apple Silicon, Intel Mac and Windows x64; Windows ARM-native operation is not claimed. Package validation details accompany each release.

03 / INTERFACE

Find your way around

The display

STEREO/DUAL selects coupling. METER/GRAPH selects the view. ON processes audio; OFF bypasses it.

Compression row

Threshold, Ratio, Attack and Release determine how much reduction is requested and how it moves.

Gain and color

Input Gain, Heat and Output Gain manage drive and final level.

Utility controls

Manual/Auto selects release behavior. Auto Gain adds makeup compensation. Sidechain Source and HPF set what drives detection.

Parallel blend

Mix blends dry and compressed audio. The footer identifies the Golden Age Series.

Adjust a control

Drag a dial to adjust it. Hold **Shift** while dragging for finer control. Double-click a dial to restore its default. Click its numeric readout to type an exact value. Double-click the Mix readout for numeric entry, or its slider to restore 100%. Focused controls also support keyboard adjustment.

All 14 audio parameters are automatable and saved with the DAW project. Meter/Graph view is recalled with plug-in state. Save a DAW preset to reuse settings; external routing is managed by the host. The editor is 400 x 783 pixels, including the Licence footer.

04 / CONTROL REFERENCE

The seven dials

Control	Range / default	What it changes
Threshold	-36 to 0 dBFS Default -18 dBFS	Centers the fixed 8 dB soft knee. Lower settings request more compression at the same input level.
Ratio	1.5:1 to 10:1 Default 2:1	Sets the gain-computer slope above the knee. It is nominal; the complete nonlinear signal path is not a constant-ratio hardware measurement.
Attack	0.2 to 80 ms Default 10 ms	Sets the nominal fast-envelope response to increasing demand. Faster settings catch more of an onset.
Release	20 to 2000 ms Default 300 ms	Sets fast recovery and, in Auto, also scales the slower memory tail. It remains active in both modes.
Input Gain	-18 to +18 dB Default 0 dB	Drives the main wet path and Internal detector. With External selected, it changes main drive without changing the key level.
Heat	0 to +24 dB Default 0 dB	Drives the separate fixed-bias tube transfer after detection, with inverse small-signal compensation.
Output Gain	-18 to +18 dB Default 0 dB	Trims the final dry/wet blend. It does not feed the detector. Use it to compare processed and bypassed levels.

Input, Heat and Output are different decisions.

For more compression, start with Threshold. Input Gain changes gain-cell drive and, in Internal mode, detector demand. Heat changes output-stage coloration without driving the detector. Finally, adjust Output Gain to match the comparison level.

Heat extends to +24 dB for stronger coloration after compression; 0-12 dB retains the subtler range. Heat compensation is a small-signal correction, not automatic loudness matching. Even at Heat 0 dB, the modeled stages remain in the active wet path. A high Threshold does not make that path a guaranteed transparent wire.

05 / UTILITY CONTROLS

Timing, coupling and display

Control	Behavior
Manual / Auto	Manual uses the fast reservoir. Auto combines it with a slower reservoir that builds during sustained demand. Default: Auto.
Auto Gain	Off / 0 dB / -12 dB. Default: Off. Static makeup calculated from Threshold, Ratio and the 8 dB knee at the selected reference input level. Wet path only; Output Gain remains independent.
Sidechain HPF	Off / 40 / 80 / 120 / 180 Hz. Default: 80 Hz. Filters the selected Internal or External detector source; main audio retains its bass. Source selection is explained on page 7.
STEREO / DUAL	Stereo uses the greater channel demand to drive a shared envelope. Dual uses independent envelopes with the same front-panel settings. Default: Stereo.
Mix	0-100% wet, default 100%. Linear-amplitude blend before Output Gain. At 0%, Input and Heat have no audible contribution, but Output Gain still applies.
ON / OFF	ON is active processing. OFF bypasses compression, Auto Gain and all three gain/drive controls. The matched dry filter path and reported latency remain.

Read the meter as gain-control activity.

The number reads nominal gain reduction at the end of the latest audio block; the needle adds display smoothing. These show gain-control activity, not an input/output peak difference. In Dual, the display shows the greater channel reduction.

Graph combines two views.

The **INDICATIVE** curve shows the static Threshold/Ratio calculation, excluding timing, tube coloration, Mix, Auto Gain and Output Gain. In External mode its input axis refers to the key; it cannot predict main output level. History shows about six seconds at 30 Hz, capped visually at 30 dB, and resets when the editor reopens.

Auto Gain uses 100 ms smoothing; it is not a loudness matcher, output target or limiter. At Threshold -18 dB / Ratio 2:1, its 0 dB and -12 dB modes add 9 dB and 3 dB. Fast release can add bass distortion or pumping. Auto retains a 0.8-5 s memory tail, including at 20 ms.

06 / EXTERNAL SIDECHAIN

Choose what drives compression

> **Sidechain Source:** Internal / External. Default: Internal. The key controls reduction; it is never mixed into the output.

Internal: follow the track itself.

The detector listens to the main signal after Input Gain. This is the normal self-compression mode and the default for existing sessions. Sidechain HPF reduces bass influence without filtering the main audio.

External: follow another track or bus.

Insert 6386e on the track you want to compress. Route a separate track or bus to its sidechain input in your DAW, then select **EXTERNAL** inside 6386e. In Logic Pro, choose the source from the **Side Chain** menu in the plug-in window header (Apple routing guide). Other hosts use their own sidechain routing controls.

Set the amount and timing.

Play the source and destination together. Adjust Threshold and Ratio for the required reduction, then Attack and Release for the movement. For kick-driven ducking, compare HPF Off with 80 Hz: Off lets the full kick drive detection. Input Gain changes the destination audio, not the external key level. Change the source/send level in your host to change key drive.

Routing	Behavior
Mono key	The same key controls both channels of a stereo destination.
Stereo key / stereo audio	Stereo uses the greater channel demand; Dual pairs left key with left audio and right key with right audio. The key channels are detected separately, avoiding antiphase cancellation.
Stereo key / mono audio	The greater key-channel demand controls the mono signal.
Missing or silent key	External remains selected. Existing reduction releases normally, then no new compression is requested. The main audio still passes through the active tube stages, Auto Gain, Mix and Output Gain.

Source changes use 15 ms demand smoothing. Auto Gain remains a static reference calculation; it does not follow key loudness. If ducking is absent, check EXTERNAL, the host route, source playback/send level, Threshold and bypass. Host support is required; choose track names in the DAW, not inside 6386e.

07 / PRACTICAL OPERATION

Start with the music

1. Begin at the defaults

Input, Heat and Output at 0 dB; Ratio 2:1; Attack 10 ms; Release 300 ms; Auto Release; Auto Gain Off; Internal sidechain; HPF 80 Hz; Mix 100%; Stereo; ON.

2. Find the amount

Play a representative loud passage. Lower Threshold until the meter shows the amount of reduction you want. Use 1-3 dB as an initial gentle comparison, not a target you must hit.

3. Shape the movement

Lengthen Attack if onsets feel softened. Shorten it if peaks remain too prominent. Adjust Release to recover naturally between phrases or hits; compare Auto against Manual.

4. Add character, then match level

Explore Input or Heat in small moves. Revisit Threshold if Input changes. Trim Output to match bypass, then compare ON/OFF at similar loudness.

STARTING POINTS - ADJUST THRESHOLD BY EAR

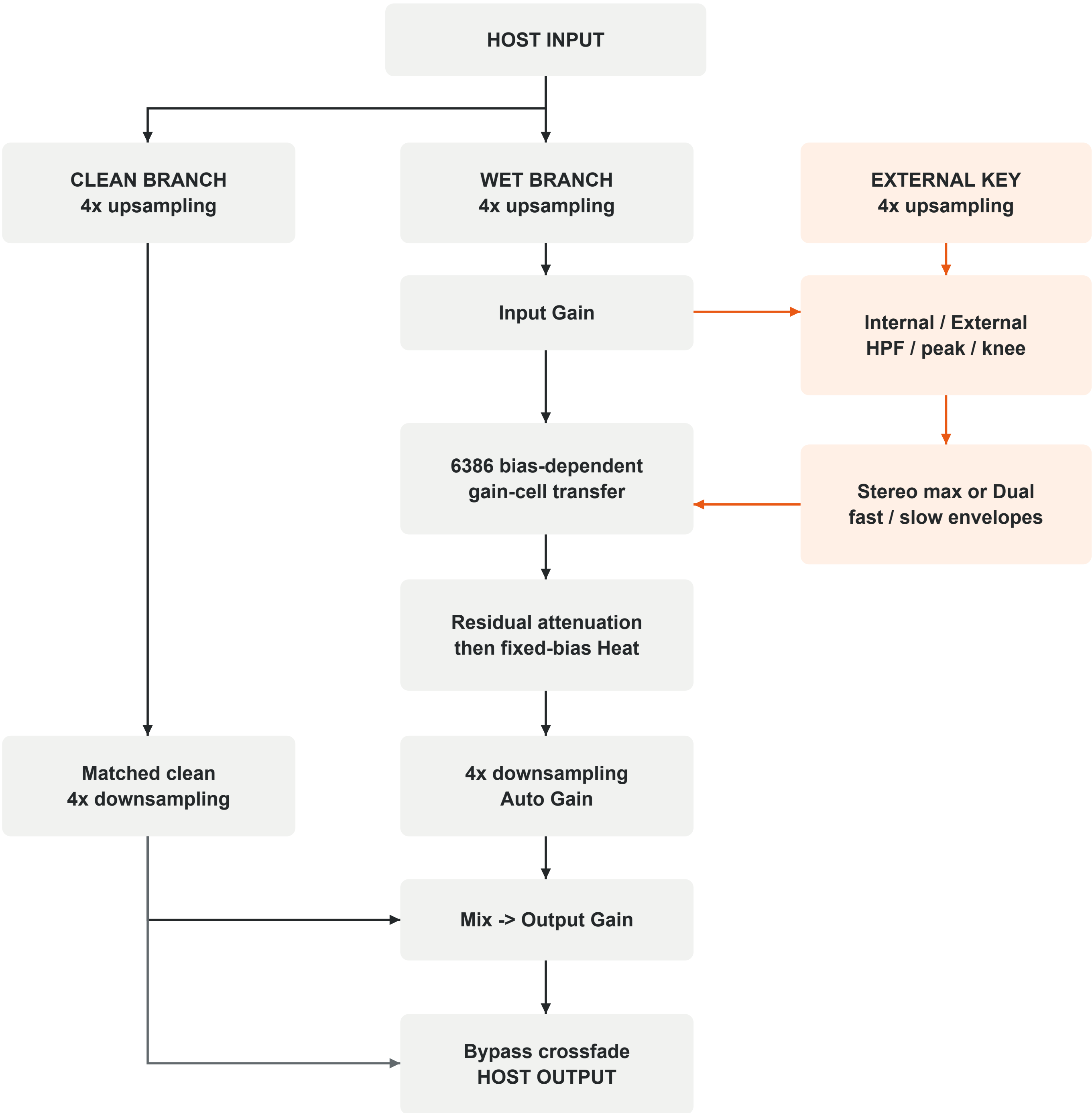
Use	Ratio	Attack	Release	Mode / Mix
Vocal leveling	2:1	10 ms	300 ms	Auto / 100%
Stereo bus	1.5:1	30 ms	500 ms	Auto / 100%
Bass control	3:1	10 ms	250 ms	Manual / 100%
Parallel drums	6:1	15 ms	120 ms	Manual / 30-60%

These are suggested experiments, not supplied factory presets or hardware-matched settings. Use modest Heat initially. For bass-driven pumping, compare HPF Off with 80 or 120 Hz. In Dual, independent reduction can change stereo position; compare against Stereo on buses. Leave output headroom: this is not a true-peak ceiling limiter.

08 / IMPLEMENTED DSP SIGNAL FLOW

The 1.0.0 signal path

This overview shows the authorised audio path. The appendix expands the tube model, detector and host routing into four implementation diagrams. The separate licence gate is shown on sheet A4.



Orange: detector/control path. Gray: matched normal bypass. Internal senses the main input after Input Gain; External senses the separate key without Input Gain. This is feed-forward control. Auto Gain follows Heat. Appendix A3 details source demand selection; A4 shows the outer licence gate.

09 / ENGINEERING NOTES

The tube model and the circuit

A device-derived transfer, with explicit limits.

The implemented gain cell uses a fitted negative-grid 6386 triode law from Raffensperger [4]. During preparation, the model solves quasi-static loaded operating points and stores a 129-bias by 1025-amplitude table. Matched sections are differenced for a push-pull transfer. During audio processing, interpolation selects the transfer for the requested bias.

Implemented model setting	Value / role
Virtual supply / loads	180 V rail; 10 kOhm plate and 100 Ohm cathode resistors; approximate 220 kOhm AC loading.
Grid bias	-7.2 to -39.2 V, controlled through a 0-32 V internal bias range.
Modeled reduction range	About 19 dB from the bias range. Additional demand uses explicitly digital attenuation.
Heat stage	A second instance of the fitted family at fixed bias. It is not a separate modeled tube type.

How the virtual circuit becomes DSP.

The tube law and resistive loads define a static transfer family. A feed-forward gain computer requests reduction; a lookup maps that demand to the cell bias. Heat reuses the fixed-bias transfer with drive and inverse gain compensation. Digital envelopes, filters and routing complete the processor. The appendix traces these implemented stages.

> **Model boundaries:** version 1.0.0 uses quasi-static tube transfers and a feed-forward detector. It does not simulate reactive transformers/capacitors, magnetic hysteresis, positive-grid conduction, supply dynamics or component tolerances. Grid-swing bounding provides overload protection, not a grid-current model.

The virtual voltages and resistances define the software transfer, not a physical hardware specification. Digital input levels are expressed in dBFS; no measured dBu calibration or hardware harmonic specification is established.

10 / RELEASE SPECIFICATIONS

Technical specifications

Item	Specification / evidence
Product / engine	6386e Tube Compressor, Golden Age Series. JUCE 8.0.15; fitted 6386-family character-compressor DSP.
Platforms / formats	macOS 11+: AU v2 and VST3, universal arm64 + x86_64. Windows 10+: x64 VST3 and standalone application.
Audio layout	Matched mono or stereo main I/O; optional mono/stereo external sidechain input. Internal/External detector selection.
Tested sample rates	44.1, 48, 88.2, 96 and 192 kHz in the DSP test suite. This is the tested set, not a hardware compatibility guarantee.
Oversampling / latency	Fixed 4x FIR oversampling. 61 samples reported in tests: 1.38 ms at 44.1 kHz, 1.27 ms at 48 kHz, 0.64 ms at 96 kHz.
Detector / knee	Feed-forward, separate channel rectification, 25 ms peak decay, fixed 8 dB soft knee.
Timing	Nominal Attack 0.2-80 ms; Release 20-2000 ms; Auto slow-tail range approximately 0.8-5 s.
Controls / state	14 host parameters, smoothed transitions, state recall and saved display mode. Float audio processing; internal model calculations use double precision.
Mix / bypass	Linear-amplitude Mix before Output Gain. Bypass is gain-independent, filter/latency matched and smoothed. No lookahead is added.
Editor	400 x 783 pixels, native controls, Meter/Graph and Stereo/Dual switching. No injected hum, hiss, drift or supply sag.

Validation and compatibility

Automated checks cover finite output, reset/partition consistency, timing, linking, Mix/bypass alignment, latency, state recall and editor construction. Release validation also checks plug-in loading and installer behavior. The validation record supplied with each installer identifies the exact binaries and checks performed.

Runtime checks cover Apple Silicon locally, Intel Mac in native x86_64 CI, and Windows x64 in CI. These checks do not guarantee every DAW or computer configuration. Windows ARM-native operation is not claimed. This original tube-inspired compressor has no claimed measured hardware equivalence or specified THD, noise or aliasing limit.

11 / REFERENCES AND APPENDIX

Sources and circuit sheets

The four sheets document the **implemented 1.0.0 DSP**: audio routing, the virtual tube model, compression control and output behavior. They are vector pages for zooming and landscape printing.

Sheet	Read it for
A1 - full signal path	Wet and dry branches, detector taps, tube gain cell, Heat, Auto Gain, Mix and host output.
A2 - virtual tube model	Resistive operating point, fitted current law, push-pull transfer and table interpolation.
A3 - detector / timing	Source selection, HPF, peak sensing, soft knee, fast/slow envelopes and bias demand.
A4 - host / output routing	Stereo coupling, Auto Gain, wet/dry blend, normal bypass and the separate licence gate.

Scope: the tube equivalent circuit explains the quasi-static software model. The sheets do not claim a solved physical compressor or measured equivalence to the reference hardware.

PRIMARY REFERENCES

[1] Fairchild Model 670 instruction manual
https://fairchildusa.com/wp-content/uploads/2023/12/Fairchild-Model-670-Manual.pdf

[2] Manley Stereo Variable Mu manual
https://www.manley.com/s/VARIABLE-MU-MANUAL-WEB-REV-521.pdf

[3] Tube-Tech CL 1B owner manual
https://www.tube-tech.com/wp-content/uploads/2020/05/Tube-Tech-Manual-CL-1B-200513.pdf

[4] Raffensperger, DAFx 2012: Toward a Wave Digital Filter Model of the Fairchild 670 Limiter
https://www.dafx.de/paper-archive/2012/papers/dafx12_submission_9.pdf

Implementation authority: Source/TubeCompressor.h, Source/PluginProcessor.cpp and Source/PluginEditor.cpp in the 1.0.0 project. Hardware names identify research references and belong to their respective owners. Fletcher Audio / AudioNode: audionode.io.

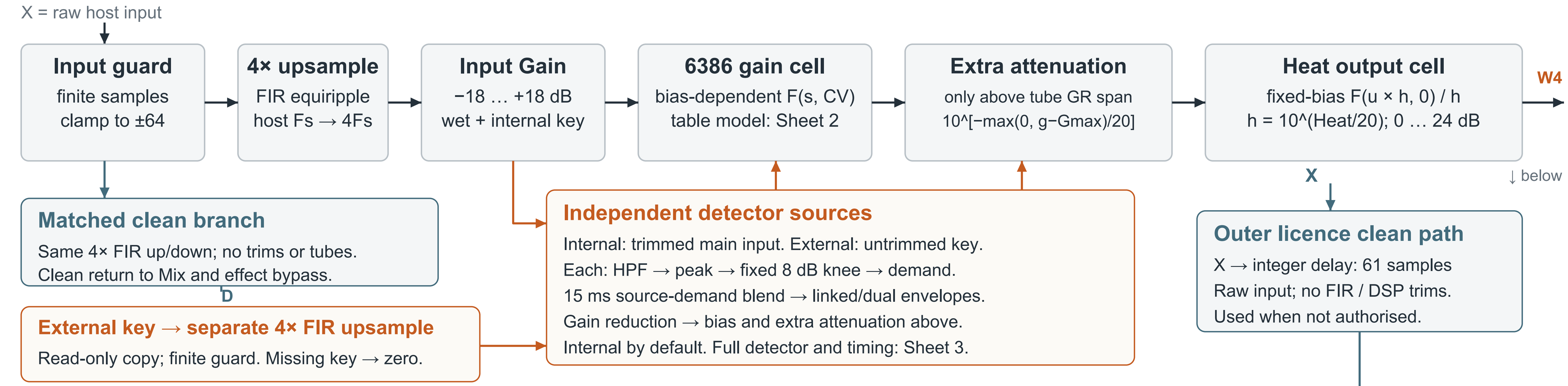
GOLDEN AGE / 6386e

IMPLEMENTED 1.0.0 DSP

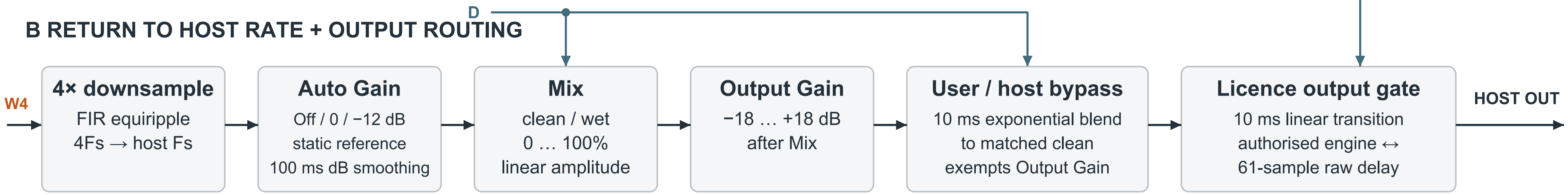
Complete implemented audio path

Per channel unless marked shared • Fs = host sample rate • all arrows represent digital sample or control paths

A AUDIO ENGINE: 4× NONLINEAR PROCESSING



B RETURN TO HOST RATE + OUTPUT ROUTING



Wet audio: black • Clean routes: blue • Detector / reduction control: orange

14 native automatable controls

Input Gain, Output Gain, Threshold, Ratio, Attack, Release, Heat,
Mix, Auto Release, Sidechain HPF, Stereo Link, Bypass, Auto Gain.
Sidechain Source: Internal / External; default Internal.
Meter/Graph is saved UI state, not an audio parameter.
Mono / stereo main input; mono / stereo external key. See Sheet 3.
61 host samples of latency; full output routing on Sheet 4.

Implementation boundary

Both normal and host-bypassed callbacks pass through the licence gate.
When fully unauthorised, the compressor is skipped; raw dry is delayed.
Input and Heat can change coloration; Auto Gain is only static makeup.
No transformer network, positive-grid current, or analog feedback solve.
4× oversampling is a design choice, not proof of inaudible aliasing.

Implementation appendix • source: TubeCompressor.h / PluginProcessor.cpp / AudioNodeLicensing.h

Digital behavioral model • not a physical manufacturing circuit or a hardware-matching claim

SHEET 1 / 4

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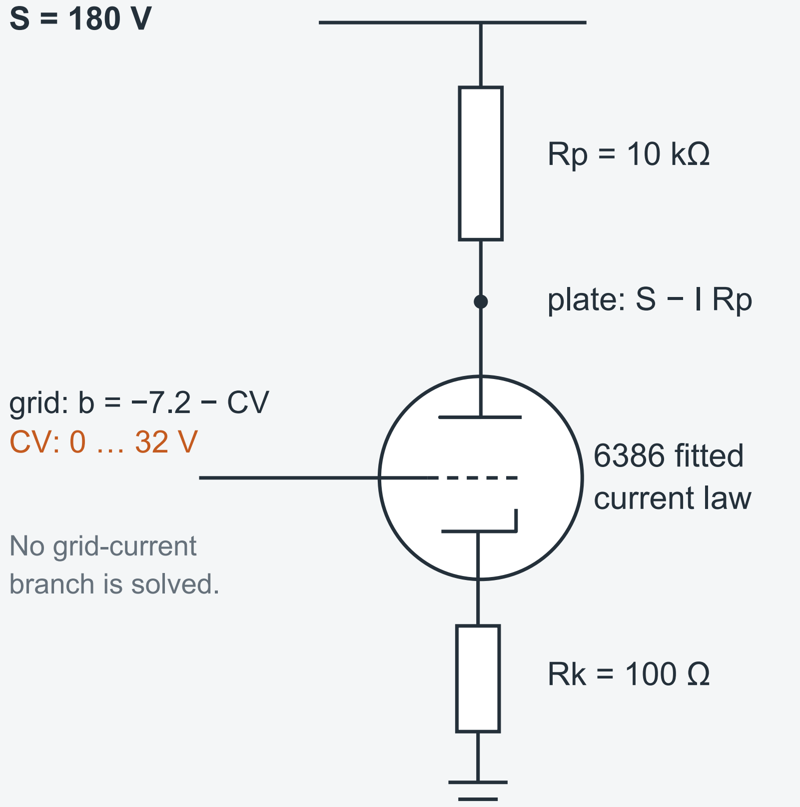
GOLDEN AGE / 6386e

IMPLEMENTED 1.0.0 DSP

Quasi-static 6386 tube model

What the software actually calculates • virtual voltages and resistances set a bounded table-based nonlinearity

A DC BIAS SOLVE: ONE VIRTUAL SECTION



Both pair halves use independent cathode feedback.

B NEGATIVE-GRID CURRENT LAW + SOLVER

```
I = 3.981e−8 × Va^2.383 /
[(0.5 − 0.1Vg)^1.8 × (0.5 + exp(−0.03922Va − 0.2Vg))]
```

Va and Vg are plate/grid relative to cathode, in volts; I is amperes.

```
solve(b, S, R): I = f(S − I(R + 100), b − 100I)
```

46 bisection steps; $0 \le I \le S/(R + 100)$. For $Va \le 0$, $f = 0$.

C BALANCED PAIR + QUASI-STATIC AC LOAD

```
b = −7.2 − CV; I0 = solve(b, 180, 10000)
Rac = 10000 || 220000 = 9565.217 Ω
Sac = 180 − I0(10000 − Rac)
w = (−b − 0.05) × tanh(3x / (−b − 0.05))
P(x,CV) = Rac × [solve(b+w, Sac, Rac)
− solve(b−w, Sac, Rac)] / 2
A0 = P(0.0001, 0) / 0.0001; F(x,CV) = P(x,CV) / A0
```

D TABLES USED ON THE AUDIO THREAD

129 bias points: CV = 32b/128. 1025 amplitudes: $x = 32(i/1024)^2$.
Tables are generated during preparation, using the two branch solves.
Audio interpolates in signal amplitude and bias; odd symmetry restores sign.
|x| is bounded to 32. The tanh grid bound keeps both grid drives negative.
Small-signal GR is derived from P(0.0001,CV), relative to CV = 0.
A 1025-point inverse map converts requested GR to bias (span ≈ 19.2 dB).
Reduction above that span uses clean attenuation before the Heat stage.

E MODEL LIMITS

220 kΩ affects the incremental load through Rac / Sac; it is absent from the DC bias solve. The table is quasi-static: no transformer, capacitor memory, hysteresis, or tube noise. The Heat cell uses the same F at CV = 0. This drawing explains the implemented math, not a complete or buildable hardware circuit.

Current-law source: Raffensperger, DAFx-12, Eq. 1 / Table 1 • www.dafx.de/paper-archive/2012/papers/dafx12_submission_9.pdf

Implementation appendix • source: TubeCompressor.h / PluginProcessor.cpp / AudioNodeLicensing.h
Digital behavioral model • not a physical manufacturing circuit or a hardware-matching claim

SHEET 2 / 4
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GOLDEN AGE / 6386e

IMPLEMENTED 1.0.0 DSP

Feed-forward detector and gain reduction

Control path at 4 × host rate • independent internal / external detector histories • no tube-stage audio feedback

A PARALLEL DETECTORS - EACH SOURCE, EACH CHANNEL

Internal source

4× main input × Input Gain
Before gain cell; default source

HPF → peak → dB

Selectable HPF; own LP/peak state
25 ms peak decay; level in dBFS

Gain computer

Threshold: −36…0 dBFS
Ratio: 1.5:1…10:1

External source

Sanitized copy → own 4× FIR up
No Input Gain; missing key → 0
Mono key duplicated; stereo paired

HPF → peak → dB

Same settings; separate LP/peak state
Both sources run across switches
No audio summing or cancellation

Gain computer

Same fixed 8 dB soft knee
Requested reduction in dB
Independent D_int / D_ext

Source-demand blend

$D_c = (1-s)\,D_{int,c} + s\,D_{ext,c}$
s = 0: Internal (default)
s = 1: External
15 ms exponential source blend
Key audio never reaches output.
Missing key supplies zero samples;
filter and envelope tails decay.

HPF: Off / 40 / 80 / 120 / 180 Hz

$LP \mathrel{+}= c(x-LP)$; $h = x - hpBlend \cdot LP$
$c = 1-\exp(-2\pi f/Fi)$; $Fi = 4Fs$
15 ms HPF coefficient / bypass blend

Peak and level

$p = \max(|h|,\ p\cdot\exp[-1/(0.025\ Fi)])$
$L = 20\log10[\max(p, 10^{-12})]$
Rectify before stereo / source selection.

$D = \mathrm{clamp}[(1-1/R)\,Q(L-T), 0, 60]$

$Q(u) = 0$ for $u \le -4$; $Q(u) = u$ for $u \ge 4$
Inside knee: $Q(u) = (u+4)^2 / 16$
Threshold, Ratio and timing: 15 ms smoothing.

B LINKING AND ENVELOPE MEMORY

Demand routing

Shared: max(D_L, D_R)
Dual Mono: D_L and D_R
In stereo, one shared + two individual pairs always run.
Mono uses the shared maximum, including stereo external keys.
No L + R audio summation.

Envelope pair - states measured in dB

Fast state F

Target: D; attack: 0.2-80 ms
Release: 20-2000 ms
$F \leftarrow D + a(F-D)$
$a = \exp[-1/(\tau\ Fi)]$

Slow state S

Target: 0.4 D; attack: 150 ms
Release: 0.8-5 s
$S \leftarrow 0.4D + a(S-0.4D)$
$a = \exp[-1/(\tau\ Fi)]$

Choose attack while target > state; otherwise release.
Manual: E = F. Auto: E = max(F,S). 15 ms mode blend.
Both states approach their current targets.

Link blend and GR

Stereo: $G_c = E_c + link\cdot(E_{shared} - E_c)$
link = 0: Dual Mono; 1: linked. 15 ms blend.
Mono: G = E_shared.
Bias maps min(G, ≈19.2 dB); extra clean attenuation supplies more GR. See Sheet 2.
Meter: max channel GR × (1−bypass)
Last internal sample / end-of-block bypass.

Auto slow-release map: $\tau = 0.8 + 4.2 \cdot \ln[\max(Release_{ms}, 80)/80] / \ln(25)$ seconds

20-80 ms keeps the 0.8 s slow tail. These are state time constants; the peak detector also shapes observed recovery.
Gain reduction is control demand, not measured input/output level loss. Internal and external HPFs affect detection only.

Implementation appendix • source: TubeCompressor.h / PluginProcessor.cpp

Digital behavioral model • not a physical manufacturing circuit or a hardware-matching claim

SHEET 3 / 4

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GOLDEN AGE / 6386e

IMPLEMENTED 1.0.0 DSP

Stereo output, Auto Gain and licence gate

Mono or stereo • 61-sample reported latency • matching wet/dry filters inside the effect; separate raw dry path outside it

01 / EFFECT PROCESSING - MATCHED WET AND DRY PATHS

Core input x
Finite-input guard
Clamp to ±64
Before Input Gain

Wet processing
4× upsample → Input Gain
Gain cell → Heat 0-24 dB
4× downsample; 61-sample latency

Auto Gain
Static reference makeup
Off / 0 dB ref / −12 dB ref
100 ms smoothing in dB

Mix
(1−m)·dry + m·wet
m = 0…1
10 ms exp smoothing

Output Gain
−18…+18 dB
After wet/dry mix
10 ms dB smoothing

Matched clean branch
Identical 4× FIR up/down and integer-latency correction
No Input Gain, Heat or Auto Gain

Static makeup - not an output normalizer
M = gainComputerDb(reference, Threshold, Ratio); same 8 dB knee as sheet 3.
Default T = −18 dBFS, R = 2:1: Off = 0 dB; 0 dB ref = 9 dB; −12 dB ref = 3 dB.
Target ignores Input Gain, Heat, HPF, sidechain audio, envelopes and output.
Shared L/R makeup; wet only, after Heat and downsampling. Default: Off.

Effect bypass
y = (1−b)·processed + b·filteredDry
b follows host / parameter bypass with a
10 ms exponential blend at host rate.
Dry excludes Auto Gain and Output Gain.

02 / OUTER LICENCE GATE - BOTH HOST CALLBACKS

Core output above

Unauthorised dry path
Raw host input → integer delay: 61
Captured before core processing
No effect gain or oversampling filters

AudioGate blend
out = a·core + (1−a)·rawDelayed
a ramps LINEARLY over 10 ms
0 = unauthorised; 1 = authorised

Host output
61-sample reported latency
processBlock and bypassed
callback share this gate

Authorization is read through atomics; network, signature checks and storage stay outside the audio callback.
After the unauthorised fade, both callbacks skip the core and return delayed dry. DAW parameter state cannot authorize.
The licence dry path is an integer delay. The effect's dry/bypass path includes oversampling filters; these are distinct paths.

Implemented 1.0.0 DSP • source-aligned documentation

Sources: TubeCompressor.h • PluginProcessor.cpp • AudioNodeLicensing.h • behavioral software model

SHEET 4 / 4

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12 / LICENSING AND OFFLINE USE

Activate your licence

> One licence holder. Up to three computers. Up to 30 days offline per computer after successful validation.

First activation

Open 6386e in your host. Enter your AudioNode key, or choose Import licence file with the file recovered from AudioNode History, then select Activate. Initial activation needs internet access. Find my licence opens your account. Once activated, Licence reopens the panel, where Manage my licence opens audionode.io/licences and Return to plugin closes the panel.

Before going offline

Open the plug-in while online before an extended offline session to allow due validation. Each computer has up to 30 days from its last successful validation. Automatic renewal is normally due after one day; merely connecting or seeing Activated does not reset the offline period. Keep your system clock accurate.

If processing is unavailable

Expired or invalid authorisation makes the compressor unavailable; audio passes through without compressor processing. Saved projects and settings remain. Reconnect and allow renewal. If prompted, enter or import your key and select Activate; contact hello@audionode.io if you need help. Check authorisation before final playback or export because the unprocessed result can sound different.

Moving computers and sharing work

Deactivate an old computer in your AudioNode account before activating a fourth. The allowance is for one licence holder, not three different users. Keep the plug-in, installer, key and activation data private. You can share your own presets, projects, stems and finished audio; collaborators need their own licence to run the plug-in.

Keep your agreement

The installer includes the complete 6386e Licence Terms as PDF and text, plus third-party notices, beside this guide in the locations on page 3. Those terms cover commercial use, backups, activation data, updates, refunds, termination and statutory rights. This page is a practical summary; the full agreement and applicable mandatory law govern.