

AMBIENCE

Algorithmic Reverb VST3 Plugin

Official User Manual

Version 1.0.1 • OTODESK

16-ch FDN | SAPF Absorption | ISM-ER | ADAA Saturator | 44.1–192 kHz

1. Introduction

Ambience is a professional-grade algorithmic reverb VST3 plugin for Windows, built on a 16-channel Feedback Delay Network (FDN) architecture. Developed using the JUCE 8 framework and modern C++20, Ambience delivers lush, natural-sounding reverberation with the stability and precision demanded by professional audio production.

This manual covers every aspect of the plugin: installation, user interface, parameter reference, Pro Mode, preset management, signal flow, and technical DSP details.

1.1 System Requirements

Requirement	Specification
Operating System	Windows 10 / 11 (64-bit)
Plugin Format	VST3 / Standalone
CPU	AVX2-capable (Intel Haswell 2013+ / AMD Ryzen 2017+)
RAM	4 GB minimum, 8 GB recommended
Tested DAW	Ableton Live 11 / 12
Other DAWs	May work but currently unverified
JUCE Version	8.0.x

⚠ Ambience is compiled with AVX2 SIMD optimization. CPUs without AVX2 support will cause the plugin to fail to load. Verify AVX2 compatibility before installation.

1.2 Installation

Step 1 — Download the latest Ambience.vst3 from the GitHub Releases page.

Step 2 — Copy the .vst3 folder to your VST3 directory:

`C:\Program Files\Common Files\VST3\`

Step 3 — (Optional) Download the factory preset pack and extract to:

`C:\Users\<YourName>\Documents\Ambience\Presets\`

Step 4 — Open Ableton Live and rescan your plugins (Options > Plug-in Sources). Ambience will appear under VST3 instruments/effects.

2. Interface Overview

The Ambience UI is divided into five main zones:

- Header Bar — Plugin title, PRO button, ER SOLO button, VU meters
- Algorithm Selector — 7 reverb type buttons
- Parameter Panel — Normal Mode or Pro Mode knobs (toggles with PRO button)
- Visualizer Area — RT60 graph (top) and Decay Curve (bottom)
- Preset Panel — Preset browser, SAVE / LOAD / DELETE controls



2.1 Header Bar

Control	Description
AMBIENCE (title)	Plugin name — no function
PRO	Toggle button: switches Parameter Panel between Normal and Pro Mode
ER SOLO	Toggle button: mutes Late Reverb — only Early Reflections are audible. Useful for tuning ER Level and Room Size.
IN / OUT meters	Stereo VU meters for input and output signal levels
Footer text	Displays technical spec: 16ch FDN SAPF ISM-ER 44.1-192kHz

2.2 Algorithm Selector

Seven reverb topology buttons appear below the header. Selecting an algorithm resets HF Damping, LF Absorption, Diffusion, and all Pro Mode band multipliers to their defaults for that algorithm.

Algorithm	Acoustic Character	Best For
ROOM1	Small–medium rooms, ER active, Input Diffuser bypassed	Drums, percussion, tight ambience
ROOM2	Large studio / live room, ER active, Input Diffuser bypassed	Full band, orchestral tracking
HALL1	Medium concert hall, ER + Input Diffuser active	Strings, piano, vocals
HALL2	Large concert hall, ER + Input Diffuser active	Orchestral, cinematic
PLATE	EMT 140-style metal plate, ER bypassed	Vocals, snare, acoustic guitar
SPRING	Spring tank character, ER bypassed	Guitar, retro/vintage sounds
GOLDFOIL	Experimental large space, ER bypassed	Ambient, pads, special FX

3. Normal Mode Parameters

Normal Mode presents 19 knobs organized into six sections: TIME, FREQUENCY, DIFFUSION, STEREO, CHARACTER, MIX, OUT EQ, and DUCKING.

3.1 TIME Section

Parameter	Range	Default	Description
PRE-DELAY	0 – 500 ms	10 ms	Delay before the reverb onset. Useful for separating the dry signal from the reverb tail, creating a sense of space without muddying the attack.
ROOM SIZE	0.3 – 2.0	1.0	Scales all 16 FDN delay lines proportionally. Larger values create longer delays, producing a larger perceived room. Also scales Early Reflection tap timings.
DECAY	0.1 – 20.0 s	1.5 s	Sets the mid-band RT60 target (500 Hz reference). The Stage 2 GEQ solver uses this value as the anchor for all frequency bands.

3.2 FREQUENCY Section

Parameter	Range	Default	Description
HF DAMP	0.0 – 1.0	0.0	High-frequency absorption. At 0.0, the RT60 curve is flat. Increasing this value reduces the RT60 of high frequency bands (2k–16k Hz), simulating absorptive surfaces such as carpet or heavy drapes. Reflected in the RT60 graph (orange curve).
LF ABSORB	0.0 – 1.0	0.0	Low-frequency absorption. Increasing this value reduces the RT60 of low frequency bands (31–250 Hz), simulating rooms with less bass build-up. Reflected in the RT60 graph.

⚠ Both HF DAMP and LF ABSORB default to 0.0. When a new algorithm is loaded, these are reset to 0.0 so that the orange and gray RT60 curves are perfectly aligned.

3.3 DIFFUSION Section

Parameter	Range	Default	Description
DIFFUSION	0.0 – 1.0	0.7	Controls the density of early reflections and tail diffusion. Low values produce sparser, more distinct echoes (small room character). High values create a smooth, dense tail (large hall character). Adjusts both Input Diffuser Allpass gain and Nested Allpass gain within the FDN. Algorithm-specific sensitivity multipliers are applied.
MOD AMT	0.0 – 1.0	0.25	Depth of LFO modulation applied to FDN delay times. Higher values add pitch modulation and reduce metallic resonances, at the cost of mild pitch drift. Keep below

Parameter	Range	Default	Description
			0.3 for natural sound; higher values for chorus/shimmer effects.
MOD RATE	0.05 – 2.0 Hz	0.5 Hz	Base frequency of the BandlimitedNoise LFO. Each of the 16 FDN channels uses a unique rate multiplier (Golden Ratio Weyl sequence), ensuring no two channels are ever synchronized.

3.4 STEREO Section

Parameter	Range	Default	Description
WIDTH	0.0 – 1.0	0.8	Controls the stereo width of the reverb output. At 0.0, the reverb is mono (both channels identical). At 1.0, L/R channels are fully decorrelated. Controls three mechanisms: (1) Side-component injection strength into FDN (sideBoost = width × 1.5), (2) FDN output L/R cross-bleed (crossLeak = 1 - width), (3) ER tap left/right separation (erLeakage).

3.5 CHARACTER Section

Parameter	Range	Default	Description
ER LEVEL	0.0 – 1.0	0.6	Level of the Early Reflections path (ISM tap pattern). Blended with the Late Reverb output. Set to 0 for pure diffuse tail; increase for clarity and room definition.
SATURATE	0.0 – 1.0	0.0	Amount of ADAA saturation applied to the wet signal. Uses an algorithm-dependent multiplier (PLATE 1.00×, SPRING 1.05×, ROOM 0.90×). Drive curve: $1 + \text{amount}^2 \times 2.5$. Wet mix: $\text{amount}^2 \times 0.7$. Dry mix: $1 - \text{amount} \times 0.25$. At amount = 0, saturation is completely bypassed.

3.6 MIX Section

Parameter	Range	Default	Description
WET	-60 – 0 dB	-4 dB	Wet signal level. An internal -1 dB offset is applied, so 0 dB on the knob = -1 dB effective. Smoothed sample-accurately via SmoothedValue to prevent zipper noise under automation.
DRY	-60 – 0 dB	0 dB	Dry (unprocessed) signal level. The dry path bypasses all DSP — no EQ, saturation, or limiting is applied. Smoothed sample-accurately.

⚠ For send/return bus use in your DAW, set DRY to -60 dB (silent). The WET knob then controls the return level exclusively.

3.7 OUT EQ Section

Parameter	Range	Default	Description
LO CUT	20 – 500 Hz	20 Hz	High-pass filter applied to the wet path only. Uses a Linkwitz-Riley topology (1st-order IIR × 2 stages = 12 dB/oct). At 20 Hz, the filter is effectively bypassed with no phase shift. Increase to remove low-frequency buildup from the reverb tail.
HI CUT	1k – 20k Hz	20 kHz	Low-pass filter applied to the wet path only. Same Linkwitz-Riley topology (12 dB/oct). At 20 kHz, effectively bypassed. Reduce to soften bright, metallic reverb tails.

3.8 DUCKING Section

Ducking reduces the wet signal level when input signal exceeds the threshold, creating space between the dry signal and the reverb tail. This is particularly effective on percussive material and vocals.

Parameter	Range	Default	Description
AMOUNT	0 – 20 dB	0 dB	Maximum gain reduction applied to the wet signal when the input exceeds the threshold. Set to 0 to disable ducking entirely.
THRESH	-60 – 0 dB	-20 dB	Input level (peak) at which ducking begins. Signals above this level trigger gain reduction.
ATTACK	0.5 – 100 ms	10 ms	Time for the ducking envelope to follow an increase in input level. Shorter values respond faster to transients.
RELEASE	10 – 2000 ms	200 ms	Time for the gain to recover after the input drops below the threshold. Longer values prevent the reverb from 'pumping' after transients.

4. Pro Mode

Press the PRO button (header bar) to reveal the Pro Mode panel. In Pro Mode, the Normal Mode parameter rows are replaced by two rows of advanced controls. Note: all Pro Mode adjustments are retained when switching back to Normal Mode — they are only reset when a different algorithm is selected.



4.1 RT60 Per Band (Row 1)

Ten knobs control a per-band RT60 multiplier for each octave band from 31 Hz to 16 kHz. These multipliers are applied on top of the base Decay time, allowing precise spectral shaping of the reverb tail.

Knob	Center Frequency	Range	Default	Effect
RT 31Hz	31 Hz	0.5 – 2.0×	1.0×	Sub-bass reverb duration
RT 63Hz	63 Hz	0.5 – 2.0×	1.0×	Bass reverb duration
RT 125Hz	125 Hz	0.5 – 2.0×	1.0×	Low-mid reverb duration
RT 250Hz	250 Hz	0.5 – 2.0×	1.0×	Mid reverb duration
RT 500Hz	500 Hz	0.5 – 2.0×	1.0×	Mid reverb (anchor band)
RT 1kHz	1 kHz	0.5 – 2.0×	1.0×	Upper-mid reverb duration
RT 2kHz	2 kHz	0.5 – 2.0×	1.0×	Presence reverb duration
RT 4kHz	4 kHz	0.5 – 2.0×	1.0×	High reverb duration
RT 8kHz	8 kHz	0.5 – 2.0×	1.0×	Treble reverb duration
RT 16kHz	16 kHz	0.5 – 2.0×	1.0×	Air reverb duration

Example: Set RT 31Hz to 1.6 and RT 16kHz to 0.6 to create a warm, concert hall-style curve with extended bass and attenuated treble.

4.2 Tilt EQ (Row 2 — left)

Three knobs provide broad RT60 tilt control across frequency groups, applied simultaneously with the per-band multipliers:

Knob	Bands Affected	Range	Description
TILT LOW	31, 63, 125 Hz	0.5 – 2.0×	Multiplies RT60 for all low-frequency bands simultaneously. Faster than adjusting the first 3 RT band knobs individually.
TILT MID	250, 500, 1k, 2k Hz	0.5 – 2.0×	Multiplies RT60 for mid-frequency bands. Useful for controlling the body/core of the reverb.
TILT HIGH	4k, 8k, 16k Hz	0.5 – 2.0×	Multiplies RT60 for high-frequency bands. Reduce for a darker reverb; increase for brighter, more airy tails.

4.3 Sat Type & Out EQ (Row 2 — right)

Control	Description
SAT TYPE	Selects the saturation character for the ADAA Saturator. Four modes: Warm, Tape, Tube, Hard (see Section 6.4 for details). This selector is only visible in Pro Mode.
LO CUT (Pro)	Same parameter as Normal Mode LO CUT — both knobs control the same value. Provided for convenience when working in Pro Mode.
HI CUT (Pro)	Same parameter as Normal Mode HI CUT — both knobs control the same value.

5. Visualizers

5.1 RT60 Graph

The RT60 graph displays the frequency-dependent reverberation time across 10 octave bands (31 Hz – 16 kHz) on a logarithmic vertical scale.

Element	Description
Gray curve	The reference RT60 of the current algorithm preset. This curve is the 'ideal' before any HF Damping, LF Absorption, or Pro Mode adjustments.
Orange curve	The actual effective RT60 after all adjustments are applied, including HF Damping, LF Absorption, and Pro Mode per-band multipliers. This is what you actually hear.
Dynamic Y axis	The vertical scale adjusts automatically to the current maximum RT60 value, with smooth exponential tracking. Grid lines appear at standard RT60 values.
Frequency labels	X-axis shows 31, 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16k Hz.

When an algorithm is first selected and HF/LF are at default (0.0), the orange and gray curves will perfectly overlap, confirming the preset is unmodified.

5.2 Decay Curve Visualizer

The Decay Curve visualizer provides a time-domain view of the reverb structure, with a split time axis for enhanced readability:

Zone	Time Range	Color	Description
ER Zone (left 30%)	0 – 200 ms	Blue	Expanded 2× zoom showing Early Reflection taps. Each tap is rendered as a 5 px bar with gradient fill and a diamond-shaped marker at the peak. The ER envelope (connecting all tap peaks) is drawn as a blue line.
Late Zone (right 70%)	200 ms – max	Orange	Shows the theoretical exponential decay curve based on mid-band RT60. A 2D gradient fill (orange, top-left to transparent, bottom-right) represents the Late Reverb energy envelope.
Boundary line	200 ms	Gray	Vertical separator between ER and Late zones.

⚠ The visualizer updates at 15 Hz to keep CPU usage minimal. RT60 values shown are based on the 500 Hz mid-band (cachedRT60Mid).

5.3 Acoustic Metrics

Four acoustic metrics are displayed in the lower-right of the Decay Curve panel, computed from the actual reverb output signal in real time:

Metric	Full Name	Formula	Interpretation
D50	Definition	$E(0-50\text{ms}) / E(\text{total})$	Ratio of early energy to total energy. Values > 0.5 indicate high clarity. Low in large halls, high in small rooms.
C50	Clarity (speech)	$10 \log_{10}[E(0-50\text{ms}) / E(50\text{ms}+)]$	Positive values = more early energy. > 0 dB recommended for speech intelligibility.
C80	Clarity (music)	$10 \log_{10}[E(0-80\text{ms}) / E(80\text{ms}+)]$	Positive values favor direct sound. -2 to +4 dB is typical for music spaces.
EDT	Early Decay Time	RT60 estimated from first 10 dB	Perceptually dominant measure of reverb character. Correlates with subjective 'liveness' of a space.

6. Preset Management

The Preset Panel is always visible in the lower-right area of the parameter section, regardless of Normal/Pro Mode state.

6.1 Preset UI Controls

Control	Description
< (PREV)	Load the previous preset in alphabetical order. Wraps from first to last.
Preset Name (combo box)	Drop-down list of all available presets. Select to preview; click LOAD to apply.
> (NEXT)	Load the next preset in alphabetical order.
SAVE	Opens a dialog to enter a name and save the current plugin state as a .ambpreset file. If a preset with the same name exists, it will be overwritten.
LOAD	Loads the currently selected preset from the combo box. Use this when you have changed algorithms and want to restore a preset without changing the combo selection.
DELETE	Deletes the currently selected preset after confirmation. This action cannot be undone.

6.2 Factory Presets

Ambience ships with 21 factory presets modeled after real-world acoustic spaces:

Algorithm	Preset Name	Reference Space
ROOM1	Abbey Road Studio2	Abbey Road Studio 2, London
ROOM1	Drums in a Box	Small dead studio room
ROOM1	Tracking Room	Medium tracking room
ROOM2	Abbey Road Studio1	Abbey Road Studio 1 (orchestral hall)
ROOM2	Capitol Studio A	Capitol Studios Studio A, Los Angeles
ROOM2	Skywalker Sound	Skywalker Sound, California
HALL1	Carnegie Hall	Carnegie Hall, New York
HALL1	Tokyo Opera City	Tokyo Opera City Concert Hall
HALL1	Berlin Konzerthaus	Konzerthaus Berlin
HALL2	Vienna Musikverein	Wiener Musikverein — Goldener Saal
HALL2	Boston Symphony	Boston Symphony Hall
HALL2	Concertgebouw	Amsterdam Concertgebouw
PLATE	EMT140 Vocal	EMT 140 plate — vocal setting
PLATE	EMT140 Snare	EMT 140 plate — snare setting
PLATE	Dark Plate	Dark plate, enhanced low end

Algorithm	Preset Name	Reference Space
SPRING	Surf Guitar	Guitar amp spring tank
SPRING	Vintage Studio	Fender-style spring tank
SPRING	Deep Tank	AKG BX-20 style deep spring
GOLDFOIL	Gothic Cathedral	Cologne Cathedral acoustic reference
GOLDFOIL	Stone Chamber	Stone underground chamber
GOLDFOIL	Infinite Space	Synthetic infinite ambient space

6.3 Preset File Location

All presets — both factory and user-created — are stored as .ambpreset files at:

`C:\Users\<YourName>\Documents\Ambience\Presets\`

This folder is created automatically on first use. Each .ambpreset file is a standard JUCE binary XML state file containing all plugin parameters.

6.4 Sharing Presets

To share individual presets, simply send the .ambpreset file. The recipient places it in their own Documents\Ambience\Presets\ folder. The preset will appear in the browser on next plugin load.

To share your complete preset library, ZIP the entire Presets folder and distribute the archive.

⚠ Preset files are version-forward compatible. When opening a preset created with an older version of Ambience, any unknown parameters are silently ignored and fall back to their default values.

7. Signal Flow

The complete signal path through Ambience from input to output:

Stage	Module	Description
1	Input Split	Stereo L/R converted to Mid/Side. Mid feeds FDN; Side injects stereo character.
2	Input Diffuser	4-stage series Allpass filter (Hall, Goldfoil only). Diffusion gain controlled by DIFFUSION knob × algorithm sensitivity (1.0 for Hall, 0.5 for Spring, etc.).
3	Early Reflections	ISM-based tapped delay line. Algorithm-specific tap patterns. ER Level and Room Size control output. Bypassed for Plate, Spring, Goldfoil.
4	FDN (16 ch)	16-channel Feedback Delay Network with FWHT feedback matrix and sign-flip decorrelation.
4a	LFO Modulation	Per-channel BandlimitedNoise LFO modulates delay times. MOD AMT × rate multiplier.
4b	Stage 2 Absorption	10-band biquad GEQ cascade per channel. WLS-solved to match target RT60 per band.
4c	Micro-Saturation	Padé $x(27+x^2)/(27+9x^2)$ limiter in feedback loop. klnScale=0.15, nearly linear.
4d	Nested Allpass	Per-channel Allpass filter for additional diffusion. Gain = apfGain × (0.6 + diffusion × 0.4).
5	Mix	ER × erLevel + Late × lateMakeupGain blended to stereo. Ducking envelope applied.
6	ADAA Saturator	Anti-Derivative Anti-Aliasing saturation (Warm/Tape/Tube/Hard). Wet path only.
7	Output EQ	LO CUT HPF + HI CUT LPF, 12 dB/oct Linkwitz-Riley. Wet path only.
8	Output Limiter	Brick-wall limiter at -0.5 dBFS. Attack 0.5 ms, Release 50 ms. Wet path only.
9	Final Mix	Wet × SmoothedWetGain + Dry × SmoothedDryGain. Sample-accurate smoothing.

8. DSP Technical Reference

8.1 FDN Architecture

Ambience uses a 16-channel Feedback Delay Network (FDN). The feedback matrix is a normalized Fast Walsh-Hadamard Transform (FWHT) with sign flipping applied to ensure energy preservation and modal density.

Delay times are assigned using the Nearest-Prime logarithmic distribution method. For each FDN channel, a target delay time is computed by logarithmically interpolating between the minimum and maximum delay (determined by Room Size). The nearest unused prime number to each target is selected. This guarantees:

- All 16 delay lines are mutually coprime (no shared factors)
- Delay times are distributed logarithmically (perceptually uniform spacing)
- No comb-filter clustering at any Room Size setting

8.2 Stage 2 GEQ Absorption (SAPF)

Each of the 16 FDN channels has an independent 10-stage biquad cascade designed by the Välimäki-Liski Accurate Cascade Graphic Equalizer method. A Weighted Least Squares system is solved to find biquad coefficients that best approximate the target attenuation per band:

```
targetDb[b] = -60 * delaySamples / (fs * targetRT60[b])
```

HF Damping and LF Absorption modify the target RT60 before the WLS solve, creating perceptually natural roll-off at the spectral extremes.

8.3 BandlimitedNoise LFO

Each FDN channel has an independent LFO driven by a white noise source (XOR shift PRNG) filtered by a 1st-order IIR low-pass filter. The IIR cutoff frequency is set to MOD RATE × rateMultiplier for each channel. Rate multipliers are initialized using the Golden Ratio Weyl sequence:

```
angle = channel_index * 1.6180339887
rateMultiplier = 0.80 + frac(angle) * 0.40    [range: 0.80 - 1.20]
```

This guarantees that no two channels share the same LFO rate, preventing periodic modulation patterns that would create audible beating.

8.4 ADAA Saturator

Anti-Derivative Anti-Aliasing (ADAA) is applied to three of the four saturation modes to prevent aliasing artifacts when the saturator introduces nonlinear distortion. ADAA computes the output as the finite difference of the antiderivative rather than directly evaluating the nonlinear function:

```
y[n] = (F(x[n]) - F(x[n-1])) / (x[n] - x[n-1])
```

A fallback to direct evaluation is used when $|x[n] - x[n-1]| < 1e-5$ (near-constant input) or when the sign of x changes (zero crossing).

Mode	Function f(x)	Antiderivative F(x)	Character
Warm	$x / \sqrt{1+x^2}$	$\sqrt{1+x^2}$	Symmetric soft clip. Vicanek-style. Warm, rounded odd harmonics.

Mode	Function f(x)	Antiderivative F(x)	Character
Tape	$x(27+x^2) / (27+9x^2)$	(No ADAA — Padé is inherently smooth)	Padé rational approximation. Tape-like transient rounding. Lightest CPU.
Tube	Asymmetric: pos: $x/\sqrt{1+x^2}$, neg: $x/\sqrt{1+(2x)^2}$	pos: $\sqrt{1+x^2}$, neg: $(1/4)\sqrt{1+4x^2}+0.75$	Asymmetric — generates even-order harmonics (2nd, 4th). Analog warmth.
Hard	$\text{clip}(x, -1, 1)$	$x^2/2$ ($ x \leq 1$), $x-0.5$ ($x>1$), $-x-0.5$ ($x<-1$)	Hard clip with ADAA. Bright, aggressive. C ¹ continuous antiderivative.

8.5 Linkwitz-Riley Output EQ

The Output EQ uses a Linkwitz-Riley topology: two cascaded 1st-order IIR filters with identical cutoff frequencies, yielding a 12 dB/oct roll-off with flat magnitude response in the passband.

$$\begin{aligned}
 \text{HPF: } y[n] &= R \times (y[n-1] + x[n] - x[n-1]) & \text{where } R &= \exp(-2\pi \times f_c / f_s) \\
 \text{LPF: } y[n] &= (1-R) \times x[n] + R \times y[n-1]
 \end{aligned}$$

Both L and R channels are processed independently through 2 cascaded stages each (4 filter states total per filter type). At bypass frequencies (LO CUT ≤ 20 Hz, HI CUT ≥ 20 kHz), the filter computation is skipped entirely — zero CPU cost and zero phase shift.

9. Real-Time Safety & Ableton Live Compatibility

Ambience is engineered to the strictest real-time audio standards. The following measures ensure zero dropouts and stable operation, with specific hardening for Ableton Live edge cases.

Measure	Details
No heap allocation on audio thread	All buffers (FDN delays, wet buffer, absorption coefficients) are pre-allocated in prepareToPlay(). No new / malloc / std::vector::resize() calls occur in processBlock().
Dirty-flag parameter dispatch	A DSPParams equality check (operator==) prevents setParams() from being called when nothing has changed. The expensive Stage 2 WLS matrix solve (16ch × 10-band LDLT) is skipped when idle.
SmoothedValue gain changes	WET and DRY gain changes are interpolated at sample rate over 50 ms to eliminate zipper noise under automation. No step changes in gain.
ScopedNoDenormals	Applied at the start of every processBlock() call to flush denormal floating-point values to zero, preventing CPU spikes in DSP feedback loops.
Ableton Live sample-rate guard	A check at the start of processBlock() compares the current sample rate against the stored rate. Mismatch triggers a safe internal reset to prevent corrupted state.
Safe editor destruction	The editor destructor explicitly calls stopTimer(), setLookAndFeel(nullptr), and resets all APVTS attachments before the component is destroyed, preventing Ableton Live's non-standard cleanup-order crashes.
VST3 parameter sync	All internal parameter changes use setValueNotifyingHost() to keep Ableton Live's automation state synchronized and prevent value rollback during automation recording.

10. Tips & Workflow

10.1 Send vs. Insert Usage

Ambience works in both configurations:

- Insert (on a track): Keep DRY at 0 dB. Adjust WET to taste. The dry signal passes through unchanged.
- Send/Return (on a bus): Set DRY to -60 dB (or -inf). Set WET to 0 dB. Control reverb level from the send amount.
- Parallel compression style: Use multiple instances with different algorithms at low WET levels blended together.

10.2 Using ER SOLO

Enable ER SOLO to hear only the Early Reflections. This allows you to:

- Fine-tune ROOM SIZE — listen for natural pre-echo spacing
- Adjust ER LEVEL — hear how much definition the early reflections add
- Verify the algorithm's ER pattern matches the intended space character

Disable ER SOLO to return to full reverb output.

10.3 RT60 Graph Reading

The gray and orange curves will perfectly overlap when HF DAMP = 0 and LF ABSORB = 0 (the default state after loading a preset). Divergence indicates active spectral shaping:

- Orange below gray at high frequencies = HF DAMP is active
- Orange below gray at low frequencies = LF ABSORB is active
- Orange above gray = Pro Mode RT band multiplier > 1.0 for those bands

10.4 Plate Reverb Tips

The PLATE algorithm has inverted RT60 frequency behavior compared to room reverb — high frequencies decay longer than low frequencies (matching the physical behavior of the EMT 140 metal plate). For best results:

- Use HI CUT to tame brightness if the plate sounds too metallic
- Try Tape or Warm saturation mode for classic EMT 140 coloration
- Lower DIFFUSION for a more 'ringy' plate character; raise for smoother tails

10.5 Spring Reverb Tips

The SPRING algorithm uses a 0.5 diffusion sensitivity multiplier — the DIFFUSION knob has a softer effect compared to Hall or Room modes. To get the characteristic spring 'boing':

- Use Tube saturation mode (kNeg = 2.0 adds harmonic richness)
- Increase MOD AMT to add the characteristic pitch instability of a physical spring
- Try DECAY between 1.5 – 3.5 s for vintage amp spring tank simulation

10.6 Creating Large Ambient Pads

For infinite, evolving ambient pads using the GOLDFOIL algorithm:

- Set DECAY to 8–12 s
- Set WIDTH to 0.95, MOD AMT to 0.45, MOD RATE to 0.20 Hz
- Set DIFFUSION to 0.90 for maximum diffusion
- Use LO CUT at 60–80 Hz to remove sub-bass buildup
- In Pro Mode, reduce RT 8kHz and RT 16kHz to 0.6 for a warmer, less harsh tail

11. Factory Preset Parameter Details

The following table lists all 21 factory presets with their key parameter values. RT60 Curve shows the per-band multipliers (31 Hz / 63 Hz / 125 Hz / 250 Hz / 500 Hz / 1k / 2k / 4k / 8k / 16k Hz).

11.1 ROOM Presets

Preset	Decay	Size	Diff	Mod	Width	ER	Sat	Type
Abbey Road Std2	0.45s	0.75	0.55	0.15	0.75	0.70	0.08	Warm
Drums in a Box	0.30s	0.55	0.40	0.10	0.65	0.80	0.05	Warm
Tracking Room	0.60s	0.90	0.65	0.20	0.80	0.65	0.10	Tape
Abbey Road Std1	1.80s	1.60	0.70	0.25	0.85	0.60	0.08	Warm
Capitol Studio A	1.40s	1.35	0.65	0.20	0.80	0.65	0.10	Tape
Skywalker Sound	2.00s	1.70	0.75	0.30	0.90	0.55	0.06	Warm

11.2 HALL Presets

Preset	Decay	Size	Diff	Mod	Width	ER	Sat	Type
Carnegie Hall	1.70s	1.50	0.72	0.20	0.88	0.55	0.08	Warm
Tokyo Opera City	2.00s	1.65	0.75	0.25	0.88	0.50	0.06	Warm
Berlin Konzerthaus	2.00s	1.60	0.73	0.22	0.87	0.52	0.07	Warm
Vienna Musikverein	2.05s	1.80	0.80	0.25	0.90	0.50	0.07	Tube
Boston Symphony	1.85s	1.70	0.78	0.22	0.88	0.52	0.06	Warm
Concertgebouw	2.05s	1.85	0.82	0.28	0.92	0.48	0.06	Warm

11.3 PLATE / SPRING / GOLDFOIL Presets

Preset	Algorithm	Decay	Size	Diff	Width	Sat	Type
EMT140 Vocal	PLATE	1.80s	1.00	0.85	0.85	0.12	Tape
EMT140 Snare	PLATE	1.00s	0.80	0.80	0.80	0.15	Tape
Dark Plate	PLATE	2.50s	1.20	0.88	0.88	0.10	Warm
Surf Guitar	SPRING	1.50s	0.70	0.45	0.70	0.20	Tube
Vintage Studio	SPRING	2.20s	0.85	0.50	0.75	0.18	Tube
Deep Tank	SPRING	3.50s	1.10	0.55	0.78	0.15	Tube

Preset	Algorithm	Decay	Size	Diff	Width	Sat	Type
Gothic Cathedral	GOLDFOIL	8.00s	2.00	0.90	0.92	0.05	Warm
Stone Chamber	GOLDFOIL	4.00s	1.80	0.85	0.88	0.08	Warm
Infinite Space	GOLDFOIL	12.0s	2.00	0.95	0.95	0.04	Warm

12. Troubleshooting

Problem	Likely Cause	Solution
Plugin does not appear in DAW	AVX2 not supported, or VST3 not in correct folder	Verify CPU supports AVX2. Confirm file is in C:\Program Files\Common Files\VST3\ and rescan.
No sound output	DRY and WET both at -60 dB, or audio routing issue	Check WET level. In Ableton Live, confirm the plugin is on an audio track or return track with signal.
Reverb sounds metallic / ringy	DIFFUSION too low, or MOD AMT too low for PLATE/SPRING	Increase DIFFUSION. For PLATE, increase MOD AMT slightly (0.2–0.3).
Reverb is too short despite high Decay setting	HF DAMP or LF ABSORB may be reducing some bands	Set HF DAMP and LF ABSORB to 0.0. Verify orange curve on RT60 graph.
Orange and gray RT60 curves are misaligned at startup	HF DAMP / LF ABSORB are not at 0	Reset HF DAMP and LF ABSORB to 0.0, or load a fresh preset.
Preset not appearing in browser	File placed in wrong folder	Confirm .ambpreset file is in Documents\Ambience\Presets\. Restart Ambience.
Audio dropout / CPU spike	Block size too small, or too many simultaneous instances	Increase audio buffer size in DAW settings. Ambience requires AVX2 for normal CPU usage.
Ableton Live crash on plugin delete	This is a known Ableton edge case — Ambience has specific fix	Ensure you are using v1.0 or later. The editor destructor includes Ableton-specific cleanup ordering.

13. Credits & License

Developer

@kijyoumusic (OTODESK) — Electronic Music, Sound Design, DSP Engineering

Framework

JUCE 8.0.x — ROLI / Raw Material Software

<https://juce.com>

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