LYNCH STUDIO CLASSICS

AAX NATIVE | APPLE SILICON | VERSION 1.0.0

LAL76, LAL73, LAL2A, LAL4K, LAL312, LAL610, LAL6A and LAL56.

Read this first

These two AAX pages cover this installer. The complete colour control manual follows unchanged. Its AU-specific installation, Logic menu paths and format exclusions apply to the AU edition only; use these opening pages for AAX. The controls and sound engine are shared.

Requirements and tested host

An Apple Silicon Mac and an AAX Native host running natively on Apple Silicon are required. The embedded engine requires macOS 12 or later; your Pro Tools version may require a newer macOS. Actual host testing used Pro Tools Intro 2026.11. Other host releases are not independently certified by these tests. No Intel, Windows, HDX DSP, AudioSuite or standalone application is included.

1. Install

Save your sessions and quit Pro Tools. Open the signed AAX installer and follow macOS Installer. Administrator authentication may be requested. It installs only the selected product AAX bundles under /Library/Application Support/Avid/Audio/Plug-Ins/LYNCH Audio Lab/. Your installed Audio Units, sessions, presets and activation data are not replaced.

2. Load the effect

Restart Pro Tools and open an audio track or bus insert. Use insert search with the public product name, or browse the Lynch Audio Lab manufacturer menu. Choose the appropriate mono or stereo format. Radio and Thirty also offer mono/stereo: one input channel and two output channels.

3. Activate or evaluate

Open Trial & Activation in the plugin. An existing purchase key works for this product in AU and AAX on the same Mac without a separate format purchase. Studio Classics uses one shared bundle entitlement for all eight effects. A full 14-day trial starts only when you choose Start trial; reinstalling or changing format does not reset it.

Working in Pro Tools

Licensing and offline work

Purchased licences are permanent for up to three activated computers. Activation requires an internet connection; paid activated use is available offline. The developer signing iLok is not an end-user licence requirement for these effects. Pro Tools itself has separate licensing requirements. After the trial ends, the effects use latency-aligned dry bypass and retain their settings.

Presets, automation and session recall

Use the plugin user-preset controls or Pro Tools settings presets. Save the Pro Tools session to retain the plugin state. For host automation, enable the required parameters in the plugin automation window, then use the normal Pro Tools automation modes. The adapter updates automation at audio-buffer boundaries; it does not claim sample-accurate automation.

Buffer and CPU guidance

Use the lowest hardware buffer that remains stable for your session. The test system completed ten stereo inserts at 48 kHz/64 samples and 96 kHz/512 samples. A heavier session with 13 loaded instances exceeded CPU capacity at 96 kHz/64 samples; three isolated mono instances completed that setting. These are measured examples, not universal capacity guarantees. If AAE -9173 appears, raise the buffer or deactivate unused processing. A muted track is not the same as a deactivated track.

Delay compensation and monitoring

Enable Pro Tools delay compensation when mixing. The host receives the engine latency: 16 frames for Studio Classics and 256 for Radio/Thirty in the tested configurations. Hardware and host buffering add further monitoring delay. These products do not provide zero-latency hardware monitoring.

Updates, duplicates and removal

Quit Pro Tools before updating. Keep a single active AAX copy of each product; contact support if a previous private test copy is also present. To remove AAX, quit Pro Tools and move only that product .aaxplugin bundle out of the LYNCH Audio Lab scan folder to a backup folder or Trash. Retain presets, projects, licences and any AU copies you still use. Documentation is installed under /Library/Application Support/Lynch Audio Lab/Documentation/AAX/.

Support and scope

Support: info@lynchaudiolab.com. Include your product version, macOS, Pro Tools version, channel format, sample rate and buffer setting. Avid and Pro Tools are trademarks of Avid Technology, Inc. No Avid endorsement or certification is claimed.

LYNCH AUDIO LAB

STUDIO CLASSICS

Eight studio personalities.
One unmistakable character.



THE COMPLETE USER GUIDE

Version 1.0.0 / English / Mac Audio Units

4 November 2026

Inside the collection

A practical handbook for version 1.0.0. Installation, every control, gain staging, musical starting points and the thinking behind the sound.

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Character is a decision

The right imperfection can give a performance its identity.

Lynch Audio Lab develops instruments and processors with grit and character. STUDIO CLASSICS brings eight different studio personalities into one coherent collection: immediate controls, warm meters and a tactile interface that invites listening rather than menu diving.

The collection was developed by studying published operating manuals, circuit and design descriptions, and the documented equipment traditions behind the references. Those ideas informed our choices of signal path, gain staging, envelope behavior and tone controls. The DSP is written as original Lynch models: nonlinear amplifier curves, simplified magnetic memory, detector envelopes, filter sections and oversampled processing.

Why eight?

A fast FET compressor solves a different musical problem from an optical leveler. A discrete preamp and a tube-style preamp offer different drive behavior. A stereo VCA bus compressor responds differently from a broadcast tube limiter. The point is a useful range of decisions, not eight skins on the same processor.

What we mean by analog character

Level-dependent harmonics can add presence and apparent density. Compression can organize movement. Broad equalisation can put a source in the right place. None of these requires a permanently audible hiss. Our optional Analog Shimmer stage makes the extra grit, air and quiet generated noise a deliberate creative choice.

Our promise is an original musical tool

These processors are independent interpretations inspired by historical equipment. They have not been calibrated against a measured reference specimen and are not component-by-component circuit simulations. They are not endorsed by the referenced manufacturers. Audition the 14-day trial on your own material; choose the sound that helps the record.

Harmonics, saturation and modeled noise also appear in other plug-ins. What distinguishes this collection is the Lynch combination of models, tactile workflow and controllable extra texture - not a claim that nobody else models analog behavior.

Install and open

One installer. Eight processors in your host.

1. Check the Mac

Use an Apple Silicon Mac (M1 or later), macOS 12 or later, and a host that supports native Audio Unit effects. This package does not include Intel binaries, Windows, VST3, AAX or standalone applications.

2. Close audio hosts

Save projects and quit Logic and other AU hosts before installation. Open the signed Mac AU installer and follow macOS Installer. Administrator approval may be required for the system-wide Audio Plug-Ins folder.

3. Let the host scan

The installer places eight components in /Library/Audio/Plug-Ins/Components/. In Logic, insert an Audio FX and select Audio Units > Lynch Audio Lab > LAL76, LAL73, LAL2A, LAL4K, LAL312, LAL610, LAL6A or LAL56.

4. Start the trial or activate

Open the editor and use Trial & Activation. Start trial enables the whole collection for 14 days on this Mac. If you purchased, paste the bundle key from your Lemon Squeezy receipt and choose Activate.

Where the manual lives

The installer also places this full-colour guide under /Library/Application Support/Lynch Audio Lab/Documentation/StudioClassics/. A freely downloadable copy is available on the product page. GUIDE in each plug-in opens the operating text above its editor.

If a model is not listed

In Logic > Settings > Plug-in Manager, search for its LAL name and run Reset & Rescan Selection. Relaunch Logic after an update. Use an audio track or bus insert, not a MIDI FX slot. Do not delete projects or presets to resolve a scan problem.

Trial and bundle activation

One Mac activation covers all eight.

14 full days	The common bundle trial starts only when you press Start trial. Installation, scanning, opening a window or loading a project does not start it. No card, subscription or automatic payment.
Shared trial	All eight use the same start date on that Mac. Starting a second model does not create a second 14-day window. Updates and reinstalls do not renew it.
When time runs out	The effect changes to latency-aligned dry bypass. Saved parameters and user presets remain intact. Activating a valid purchase key restores processing.
Permanent purchase	One licence includes all eight processors for up to three computers. It is not a subscription. Each Mac takes one activation, not one per model or per instance.
Activate	Internet is needed for activation. Paste the key for LYNCH STUDIO CLASSICS, not a key for another Lynch product. After successful activation, paid processing continues offline.
Move computers	While online, choose Deactivate this Mac in any model. It releases this bundle activation for all eight on that Mac. Settings and the original trial dates remain.
Storage and privacy	The key and trial record are kept in the login Keychain, separate from presets and DAW projects. An unlocked login Keychain and a correct system date are required. No licence key is written into an audio render.

A key is missing or an activation fails

Check the purchase receipt and spam folder. Use Refresh after unlocking the login Keychain. A network outage does not revoke a previously activated offline entitlement. If you used all three computers, deactivate one before adding another. Contact info@lynchaudiolab.com with the order number; never post the key publicly.

First session: find the sweet spot

Drive changes character. Output changes comparison.

Begin with a sensible source

Leave headroom at the audio interface and inside the DAW. A practical starting region is average recorded level around -18 dBFS with peaks well below 0 dBFS. This is a starting point, not a compulsory target. A plug-in cannot undo clipping already recorded by a converter.

Choose a role

Start with one processor. For tone choose LAL73, LAL312 or LAL610. For fast peak shaping choose LAL76; phrase leveling LAL2A or LAL56; dense broadcast movement LAL6A; linked group control LAL4K.

Set the action first

Use input, Peak Reduction or Threshold to establish the amount of change. Then adjust the output control. Do not decide that louder means better. Listen to the attack, the body of a note and the space before the next phrase.

Match the loudness

Toggle Bypass at comparable perceived level. Check both the host peak meter and the VU. A compressed or harmonically enriched signal can sound louder with lower peaks, while a bright signal can seem louder without needing much more average level.

Add texture last

Begin with Analog Shimmer at 0%. If the core sound is right, try 10-30%, then rematch output. More is useful for deliberate colour, but extra harmonics and air may also emphasize harshness.

Stop when it serves the music

Preserve expressive dynamics where they matter. Use parallel blend to keep a transient while adding body. A clean section and a gritty chorus can coexist through host automation; no permanent processing is required until you render or print it.

The shared interface

The same hand movements, eight different voices.

Knobs	Drag vertically to adjust. Hold Shift for fine movement. The pointer changes position while material lighting and shadows remain fixed. Mouse wheel and arrow keys adjust a selected control.
Double-click	Returns that control to its default value. INIT resets all sound parameters, including trims, bypass and Analog Shimmer.
DETAIL	Opens secondary controls such as parallel blend, input/output calibration, link and Shimmer. The available controls vary by model. Opening the panel alone does not change sound.
BYPASS	Compares against the dry signal with the normal reported latency. Host bypass is also supported. Saved sound settings remain available.
GUIDE and ?	GUIDE opens the operating text in a child window above the editor. Click ? to show contextual explanations, then hover a control. Click ? again to hide the help strip.
UI size	Choose 75%, 85% or 100% in DETAIL. This changes the editor size rather than the sound. Use the host screen/zoom settings if more workspace is needed.

Factory and user presets

PRESETS groups the nine factory starting points per processor by purpose. There are 72 in the complete collection. Names and available folders vary with the model. They are starting points dependent on your source level, not automatic mastering or loudness matching.

Save Preset creates a normal .aupreset file. The default folder is ~/Library/Audio/Presets/Lynch Audio Lab/[model]/User Presets/. Make subfolders there in Finder if desired. User Presets follows those folders; Show User Presets Folder opens the correct location. Open Preset loads an existing compatible file. Cross-product files are rejected before changing the current sound.

A host may also offer its own saved settings or factory preset menu. Those menus are separate views of host storage. Saving the DAW project retains the complete AU sound state. The licence is separate and is never copied into a preset.

Read the meter correctly

A VU needle and a peak meter answer different questions.

The warm VU is a smoothed average-level or gain-reduction display. It is not a fast sample-peak or true-peak meter. A brief click can reach a high peak without moving a slow needle far. A sustained tone can move the needle strongly while staying comfortably below full scale.

Input / Output	Most models use 0 VU = -18 dBFS RMS. The display follows the stronger channel. Input is measured before the model input trims; Output follows processing, parallel blend and bypass.
LAL76 exception	Output +4 uses the -18 dBFS RMS reference. Output +8 uses -14 dBFS RMS. Meter Off darkens the display but audio keeps running.
Gain reduction	On compressors, GR is attenuation in dB. A needle at -5 means approximately 5 dB reduction. At very heavy reduction it reaches the left stop: the printed range cannot show every possible reduction value.
Check peaks in the host	Use the host peak meter for clipping. A 0 dB output knob means unity gain at that stage, not that the signal peak must reach 0 dBFS. There is no hidden brickwall ceiling in this collection.

A useful calibration example

A 1 kHz sine with sample peaks at -15 dBFS has an RMS level near -18 dBFS, so an input VU using the shared reference settles near 0. Music has a changing crest factor, so its peak and VU readings are not expected to match numerically.

Stereo link is not a width control

On compressors, Link 100% uses the stronger detector to make both channels reduce together. At 0% they act independently, which can move the stereo image. Intermediate settings blend the detector behavior. Link has no extra role in mono. One VU does not replace checking both host output channels.

Analog Shimmer: the thinking

A small texture can change how a source belongs in a mix.

Real analog equipment can change a signal even with a compressor doing little reduction: amplifiers, transformers and filters need not be perfectly linear. That idea is already present in the core models. Analog Shimmer is a separate modern extension, available in DETAIL when you want an additional controlled layer.

Harmonic grit	A small parallel contribution from a soft asymmetric nonlinear curve adds level-dependent harmonics. At low levels the response is close to linear; higher drive bends the waveform more.
A restrained air lift	A gentle high-frequency residual is mixed back in. This can bring texture and articulation forward. It can also expose an already bright source, so judge it in context.
A very quiet floor	A generated noise source is shaped to restrain deep lows and top-end hiss. It is deliberately very quiet at neutral gain. It is not a recording of a historical unit and is not hum or crackle.
0% means off	The added stage is an exact identity at 0%. The processor still has its own core colour, EQ or compression. Shimmer does not switch the whole model between analog and digital.
Where it sits	Shimmer acts on the wet path before the final wet output gain. That gain can raise its noise as well as the music. Parallel Mix and Bypass behave normally.

The theory is straightforward: modest harmonics can increase perceived presence and density without the same peak increase as clean gain. Envelope shaping can make a part feel more settled. A slight spectral change can reveal detail. These are possible musical benefits, not guaranteed improvements to every recording.

No chorus, pitch wobble, reverb, tape flutter, stereo-widening engine or automatic song analysis is hidden in Shimmer. It is intentional nonlinear colour, a small tonal lift and quiet shaped noise. Other processors also model saturation and noise; our creative contribution is the balance and integration in this collection.

Analog Shimmer: use it well

Start below the point where you can name the effect.

A controlled comparison

Set the core processor first. Loop a representative phrase at a comfortable monitor level. Compare Shimmer 0%, 15% and 30%, matching loudness after each change. Listen with the whole arrangement playing, not only in solo.

Vocal and acoustic detail

Try a low setting when a source needs more tactile presence. If consonants, pick noise or sibilance become distracting, reduce Shimmer or the high shelf. More high-frequency energy is not automatically more detail.

Bass and keys

Subtle drive may help upper harmonics translate on smaller speakers. Preserve the fundamental and avoid boosting output until the next plug-in clips. Compare mono as well as stereo.

Drum room and parallel colour

A larger setting can be deliberate on a room or parallel group. Reduce the output, then blend the processed path with the latency-aligned dry path. Do not use polarity inversion accidentally when blending a preamp.

Quiet arrangements and many instances

Noise contributions add up across a session. Listen to pauses and tails. If a quiet ending needs complete silence, reduce Shimmer, automate it to 0%, or bypass the stage as appropriate. A low noise floor is not a requirement to make a record sound analog.

Avoid the extra layer

Use 0% for transparent comparisons, clean overdub preparation, or a source that already has enough grit. Choosing less colour is part of using the collection well.

There is no automatic makeup or loudness matching in these processors. A brighter or denser result may sound louder even when its sample peaks go down. Use your ears and the host meters together.

LAL76 / FET limiting amplifier

Fast decisions. Strong personality.



Actual version 1.0.0 interface. DETAIL contains the additional model controls.

The inspiration and why we chose it

The 1176 tradition couples FET gain control with an input-driven workflow, fast timing and an amplifier path that can contribute colour even when compression is not being used. We chose this direction for deliberate transient shaping and an assertive alternative to slower levelers. LAL76 is our own feedback-inspired interpretation, not a measured model of a particular revision.

Your first setting

Choose 4:1, Attack 3, Release 5, Mix 100% and Link 100%. Increase Input until the loudest phrases show about 3-5 dB GR. Adjust Output to match bypass loudness. Attack 0 disables compression while keeping the colour stages active.

Signal path

Input gain -> nonlinear colour -> linked feedback-inspired detector and gain reduction -> output colour / Shimmer -> Output -> DC removal and downsampling -> dry/wet blend -> bypass.

LAL76 / controls and practice

Input	0-48; default 24. Each unit is 1 dB relative to unity at 24. More input drives both colour and gain reduction.
Output	0-48; default 24, also unity. Makeup after compression; changing it should not increase the GR reading.
Attack	0-7; default 3. At 1 the nominal timing is slower, towards 800 microseconds; 7 is faster, towards 20 microseconds. 0 switches the reduction cell off. These are model timing controls, not measured hardware response times.
Release	1-7; default 5. 1 is slower, towards 1.1 s; 7 is faster, towards 50 ms. Recent reduction also affects recovery.
Ratio	4:1, 8:1, 12:1, 20:1 or All Buttons. ALL adds a broader knee, altered threshold and stronger colour. Shift-click a ratio also selects ALL.
Meter	GR, Output +8, Output +4 or Off. +4 uses 0 VU = -18 dBFS RMS; +8 uses -14 dBFS RMS. Off darkens the meter without muting audio.
Detail	Mix 0-100% (100); Stereo Link 0-100% (100); Analog Shimmer 0-100% (0); UI size 75/85/100%.

Lead vocal

Use 4:1 and moderate input. Back off the attack if words lose their leading edge. Keep reduction musical rather than chasing an unmoving needle.

Parallel drums

Select ALL, drive the input and reduce output. Start Mix around 15-30%. Raise the wet contribution only until the room feels more urgent.

Colour only

Set Attack to 0, match output and compare. Shimmer can then add a separate restrained layer without switching compression back on.

Keep in mind

Input and Output are relative dial scales, not absolute dBFS readouts. High ratios do not make this a true-peak mastering limiter. The attack markings describe the control law; feedback and detector memory change the resulting envelope.

Shared Bypass, INIT, presets, contextual help, trial and automation behavior are covered earlier in this guide. Default values are shown in parentheses where useful.

LAL73 / Discrete preamp and equaliser

Weight, presence and a beautiful broad stroke.



Actual version 1.0.0 interface. DETAIL contains the additional model controls.

The inspiration and why we chose it

The 1073 family combines discrete Class A amplification, transformer-coupled stages and a deliberately concise three-band equaliser. Its stepped gain and broad tonal choices make it a natural recording and mix-shaping reference. LAL73 brings that operating philosophy into an original nonlinear preamp and digital filter design.

Your first setting

Start at LINE 0 with EQ In on, the shelves and mid gain at 0 dB, High Pass Off and Shimmer 0%. Add a small amount of high shelf or mid cut for the source. Explore input drive only after checking the output level. Use Detail / Output Trim to compensate.

Signal path

Stepped input gain -> transformer-inspired nonlinear stage -> shelves / mid / high-pass with EQ In blend -> output-stage colour / Shimmer -> Output Trim and polarity -> DC removal / downsampling -> parallel blend -> bypass.

LAL73 / controls and practice

Input Gain	LINE -10 to +20 dB in 5 dB steps, OFF, then MIC +20 to +80 dB in 5 dB steps. Default LINE 0. OFF mutes only the processed path.
High Shelf	12 kHz shelf, -16 to +16 dB; default 0 dB.
Mid	Inner knob: -18 to +18 dB; default 0. Outer ring: 360, 700 Hz, 1.6, 3.2, 4.8 or 7.2 kHz; default 1.6 kHz.
Low	Inner knob: -16 to +16 dB; default 0. Outer ring: 35, 60, 110 or 220 Hz; default 110 Hz.
High Pass	Off, 50, 80, 160 or 300 Hz. An 18 dB/octave design. EQ In bypasses this section as well as the three tone bands.
EQ In / polarity	EQ In defaults on; its off state leaves preamp colour active. The polarity button reverses the wet signal and may cause cancellation at partial Mix.
Meter	IN / OUT selects average input or output display. 0 VU = -18 dBFS RMS. The display uses the stronger channel.
Detail	Output Trim -60 to +12 dB (0); Mix 0-100% (100); Shimmer 0-100% (0); UI size 75/85/100%.

Vocal body

Try the low shelf at 110 Hz and a gentle mid cut where the voice sounds boxy. Avoid boosting every band; preserve the consonants and output headroom.

Guitar presence

Compare 1.6 and 3.2 kHz with a small gain change. Set High Pass only high enough to remove unwanted rumble.

Driven line sound

Stay in the LINE range, raise input and lower Output Trim. The MIC range expects a much lower digital input and can overload an ordinary recorded track immediately.

Keep in mind

These gain choices operate on audio already converted by your interface. They do not control a real microphone preamp, phantom power or impedance. The filter model does not claim measured inductor or transformer interaction.

Shared Bypass, INIT, presets, contextual help, trial and automation behavior are covered earlier in this guide. Default values are shown in parentheses where useful.

LAL2A / Optical leveling amplifier

Let the performance breathe.



Actual version 1.0.0 interface. DETAIL contains the additional model controls.

The inspiration and why we chose it

The LA-2A tradition uses a light-dependent gain element and tube amplification. Its appeal includes a recovery that depends on the preceding performance, and a simple separation between reduction and makeup gain. We included LAL2A for smooth phrase leveling that complements the faster LAL76.

Your first setting

Choose Compress, Gain 50 and Mix 100%. Raise Peak Reduction until louder phrases show 2-5 dB GR. Listen through the end of a sustained note to hear the recovery. Set Gain afterwards for equal apparent loudness; it is normal for Gain to leave the GR reading unchanged.

Signal path

Input Trim -> amplifier colour -> linked optical-like envelope / reduction -> output colour / Shimmer -> Gain and Output Trim -> DC removal / downsampling -> parallel blend -> bypass.

LAL2A / controls and practice

Gain	0-100; default 50. Output makeup after reduction. 50 is unity in LAL2A; 0 mutes the wet path. The scale is deliberately nonlinear and reaches about +30 dB at 100.
Peak Reduction	0-100; default 25. Raises the amount of compression by changing detector sensitivity. At 0 the reduction is disabled, while amplifier colour remains.
Limit	Off = Compress (default), on = firmer Limit. The model uses approximately 3:1 versus 15:1 behavior with different knees.
Meter	GR or Output; default GR. GR displays attenuation, Output displays RMS at the shared -18 dBFS reference.
Input Trim / Output Trim	Detail: input -24 to +24 dB, output -24 to +12 dB; both start at 0. Input affects detector drive, output does not.
Mix / Link / Shimmer	Detail: Mix and Stereo Link 0-100%, initially 100%; Shimmer 0-100%, initially 0%. UI size 75/85/100%.

Even lead

Use 2-4 dB GR over whole phrases. If the next quiet word stays too far down, reduce Peak Reduction instead of adding makeup.

Bass sustain

Choose Compress, then raise reduction until the sustain sits in the mix. Compare with LAL56 if you want a different long recovery.

Serial control

Place LAL76 first to catch short peaks, then ask LAL2A for only gentle phrase leveling. Match levels after each stage.

Keep in mind

Gain is not an input-drive knob. The GR needle should follow incoming performance and Peak Reduction, not output makeup. The nominal 10 ms attack and combined fast/slow release are an independent optical-like model, not a measured T4 cell.

Shared Bypass, INIT, presets, contextual help, trial and automation behavior are covered earlier in this guide. Default values are shown in parentheses where useful.

LAL4K / Stereo VCA bus compressor

Bring the group together.



Actual version 1.0.0 interface. DETAIL contains the additional model controls.

The inspiration and why we chose it

The console bus-compression tradition joins predictable VCA control, selectable timing and linked stereo action. We chose it for rhythmic cohesion across groups and mixes. LAL4K uses an original feed-forward envelope and restrained colour, with a detector high-pass and parallel blend for modern routing.

Your first setting

On a drum or mix bus choose 2:1, Attack 10 or 30 ms, Auto Release, Link 100% and Mix 100%. Lower Threshold for 1-3 dB GR on louder sections. Match the level with Makeup. If the kick pulls the whole mix down, try the sidechain high-pass.

Signal path

Input Trim -> subtle colour -> linked feed-forward detector / VCA-style gain control -> output colour / Shimmer -> Makeup -> downsampling -> parallel blend -> bypass.

LAL4K / controls and practice

Threshold	-40 to +20 dB; default -12. Lower values give more reduction at the same source level.
Makeup	0 to +24 dB; default 0. Manual output gain after reduction. There is no automatic gain compensation.
Attack	0.1, 0.3, 1, 3, 10 or 30 ms; default 10 ms. Faster settings catch more of the initial transient.
Release	100 ms, 300 ms, 600 ms, 1.2 s or Auto; default Auto. Auto combines fast recovery with longer program memory.
Ratio	2:1, 4:1 or 10:1; default 4:1. The model also varies the knee with the selected ratio.
Sidechain HP	Off, 60, 90, 150 or 300 Hz; default Off. Changes detector sensitivity, not the audio bass response directly.
Meter	GR or Output; default GR. The printed negative VU scale is used for reduction.
Detail	Input Trim -24 to +24 dB (0); Mix and Link 0-100% (100); Shimmer 0-100% (0); UI size 75/85/100%.

Mix cohesion

Start at 2:1 with only 1-2 dB GR. If the mix sounds smaller, increase attack time or reduce reduction before reaching for Makeup.

Room excitement

Use 4:1 or 10:1 on a drum room, then restore attack with the parallel blend. Keep the wet level under control.

Bass-heavy music

Compare sidechain HP Off and 90 Hz at the same threshold. The filter lets bass trigger less reduction; it is not a low-cut on the track.

Keep in mind

The slow part of Auto can retain gain-reduction memory. Wait through several phrases when comparing. This is a bus compressor rather than a loudness normaliser or final true-peak limiter.

Shared Bypass, INIT, presets, contextual help, trial and automation behavior are covered earlier in this guide. Default values are shown in parentheses where useful.

LAL312 / Discrete transformer preamplifier

Make the front edge count.



Actual version 1.0.0 interface. DETAIL contains the additional model controls.

The inspiration and why we chose it

The API 312 reference combines discrete amplification and transformer coupling in a direct preamp workflow. Its practical gain range, pad, polarity and output tap provide a focused way to explore level and drive. We included LAL312 as the compact, punch-oriented preamp in the collection.

Your first setting

Use Gain 34 dB, Input Trim -34 dB, Output Trim 0, Pad off and Tap off. This digital calibration starts close to unity for recorded line-level audio. Raise Gain for more drive and lower Output Trim to compare fairly. Turn on 3:1 Tap if you want an additional output reduction.

Signal path

Input Trim + Gain - Pad -> discrete-style nonlinear stage with gentle magnetic memory -> output-stage colour / Shimmer -> Output Trim / Tap / polarity -> DC removal / downsampling -> parallel blend -> bypass.

LAL312 / controls and practice

Gain	34-69 dB; default 34. Changes modeled gain and the input-stage drive law. It is applied after the digital input calibration.
Output Trim	-36 to +12 dB; default 0. Final wet-path output control; does not restore lost transients from an overdriven input.
Pad	Off by default; on reduces signal before the model by 20 dB. This is a software pad, not your interface pad.
Polarity	Off by default; on reverses the wet signal. Check cancellation if Mix is less than 100%.
3:1 Tap	Off by default; on reduces wet output by 9.54 dB. This is the level ratio inspired by the reference tap, not a simulation of both physical transformer loads.
Meter	Input or Output; default Output. Input reads incoming audio before Input Trim / Gain. 0 VU = -18 dBFS RMS.
Detail	Input Trim -60 to 0 dB (default -34); Mix 0-100% (100); Shimmer 0-100% (0); UI size 75/85/100%.

Drum attack

Increase Gain in small steps, compensate output and listen to the first few milliseconds of the snare. Stop before the attack becomes blunt.

Bass definition

Try modest drive with Shimmer off first. Add only a little Shimmer if the bass needs more upper harmonic presence.

Cleaner level

Engage Pad or lower Input Trim to retain more headroom. Raising Output afterwards can recover level but does not add input saturation.

Keep in mind

Do not remove the -34 dB input calibration and feed full recorded audio into 34-69 dB gain without expecting extreme drive. This effect cannot recreate microphone loading or recover clipping already captured by the converter.

Shared Bypass, INIT, presets, contextual help, trial and automation behavior are covered earlier in this guide. Default values are shown in parentheses where useful.

LAL610 / Dual-stage tube preamp and EQ

Round the edges. Keep the soul.



Actual version 1.0.0 interface. DETAIL contains the additional model controls.

The inspiration and why we chose it

The 610 lineage offers a tube preamp workflow with separate stage gain and output level, plus broad shelf equalisation. We chose it for a softer, more rounded alternative to the discrete preamps. LAL610 uses independent asymmetric nonlinear stages and digital shelving filters informed by the published operating choices.

Your first setting

Begin at Gain 0 dB, Level 5 and both shelves at 0 dB. Raise Gain one step, then lower Level or Output Trim for equal loudness. Add low shelf for body or high shelf for air in small increments. The effect of a frequency selector becomes audible only when its shelf gain is nonzero.

Signal path

Input Trim + stepped Gain - Pad -> first asymmetric tube-like stage -> low / high shelves -> second colour stage / Shimmer -> Level / Output Trim / polarity -> DC removal / downsampling -> parallel blend -> bypass.

LAL610 / controls and practice

Gain	Five settings: -10, -5, 0, +5 or +10 dB; default 0. Changes level into the model and its drive law.
Level	0-10; default 5, unity. A relative linear-amplitude output control after the tube stages. 0 mutes the processed path.
Low / High EQ	Each uses 11 gain steps: -9, -6, -4.5, -3, -1.5, 0, +1.5, +3, +4.5, +6, +9 dB. Default 0.
Low Frequency	70, 100 or 200 Hz; default 100 Hz. Broad shelving corner, not a narrow resonant bass peak.
High Frequency	4.5, 7 or 10 kHz; default 7 kHz. Compare a presence lift with a higher air lift.
Pad / polarity	Pad reduces input by 15 dB, initially off. Polarity reverses the wet output, initially off.
Meter	Input or Output; default Output. Incoming Input display precedes digital trim; Output follows blend / bypass.
Detail	Input Trim -24 to +24 dB (0); Output Trim -36 to +12 dB (0); Mix 0-100% (100); Shimmer 0-100% (0); UI size 75/85/100%.

Intimate voice

Try Gain +5, then reduce Level. A small shelf move can add weight without needing more drive. Use a separate compressor for phrase control.

Keys and bass

Compare 70 and 200 Hz with a small positive low shelf. One reaches deep weight, the other a wider body region.

Too much softness

Reduce Gain or input, then match output. Shimmer is an optional extra layer rather than a substitute for clear transients.

Keep in mind

LAL610 does not have a dedicated compressor cell. Saturation can reduce crest factor, but it is not the same as setting a threshold and ratio. Level and Output Trim are output controls, not physical tube bias controls.

Shared Bypass, INIT, presets, contextual help, trial and automation behavior are covered earlier in this guide. Default values are shown in parentheses where useful.

LAL6A / Variable-mu broadcast limiter

Broadcast authority, with movement.



Actual version 1.0.0 interface. DETAIL contains the additional model controls.

The inspiration and why we chose it

The BA-6A direction brings balanced tube amplification, input-driven limiting and a recovery that can remember sustained peaks. We included this less everyday broadcast reference for dense, purposeful control. LAL6A interprets the idea with an increasing-ratio feedback model and Single / Dual recovery choices.

Your first setting

Choose Dual, Mix 100%, Link 100% and GR display. Raise Input Trim for about 3-6 dB reduction on stronger phrases, then reduce Output Trim. Compare Single for quicker recovery. If low notes dominate the action, try the detector high-pass in Detail.

Signal path

Input Trim -> transformer-inspired colour -> linked feedback variable-ratio detector / reduction with optional dual memory -> output colour / Shimmer -> Output Trim -> DC removal / downsampling -> blend -> bypass.

LAL6A / controls and practice

Input Trim	0 to +36 dB; default +10. Drives the colour and feedback-inspired reduction. There is no separate threshold control.
Output Trim	-36 to +12 dB; default 0. Adjust after setting reduction and recovery.
Mode	Single or Dual; default Dual. Single uses the quicker recovery path. Dual adds sustained-signal memory with a slower component around two seconds.
Meter	GR or Output; default GR. The printed scale shows only part of the possible reduction range.
Detail	Sidechain HP Off / 60 / 120 / 200 Hz (Off); Mix and Link 0-100% (100); Shimmer 0-100% (0); UI size 75/85/100%.

Dense vocal

Use Dual and moderate reduction; listen to the recovery between lines. Compare against LAL2A at equal loudness for a different envelope personality.

Parallel room

Raise input for deliberate density, lower output and blend in a little processed signal. Use Link 100% on stereo rooms to keep the action together.

Sustained bass

Try a detector high-pass and Single if low notes hold the reduction down too long. Change one control at a time.

Keep in mind

Variable ratio here means an original level-dependent feedback law, not a component-by-component variable-mu tube calculation. It is not a calibrated BA-6A restoration or an absolute broadcast compliance limiter.

Shared Bypass, INIT, presets, contextual help, trial and automation behavior are covered earlier in this guide. Default values are shown in parentheses where useful.

LAL56 / Remote-cutoff tube leveler

Long notes. Quiet confidence.



Actual version 1.0.0 interface. DETAIL contains the additional model controls.

The inspiration and why we chose it

The Gates Sta-Level tradition is known for program-controlled tube leveling. Later interpretations broadened the recovery choices and modes. We included LAL56 for sustained vocal and bass movement, with a longer, adjustable sense of recovery. Its six recovery settings and Single / Double / Fast modes are Lynch extensions rather than a replica of the 1956 front panel.

Your first setting

Choose Double, Recovery 3, Mix 100%, Link 100% and GR display. Raise input until the loudest phrases show about 3-6 dB reduction. Lower output to compare. Listen over several bars: a long recovery can keep influencing the next phrase after the loud note has stopped.

Signal path

Input Trim -> tube-like colour -> linked remote-cutoff-inspired feedback leveling / recovery memory -> output colour / Shimmer -> Output Trim -> DC removal / downsampling -> blend -> bypass.

LAL56 / controls and practice

Input Trim	0 to +36 dB; default +10. Drives the colour and gain reduction. Reduce this first if the track loses expression.
Output Trim	-36 to +12 dB; default 0. Makeup after the modeled leveling path.
Recovery	1-6; default 3. Nominal single-path values are 0.3, 0.6, 1.2, 2.4, 4.8 and 9.6 s. Mode and performance history change the practical recovery.
Mode	Single, Double or Fast; default Double. Single uses one smoother path. Double adds slow program memory. Fast uses faster attack and a shorter release multiplier.
Meter	GR or Output; default GR. Heavy reduction can park the needle at the left stop.
Detail	Sidechain HP Off / 60 / 120 / 200 Hz (Off); Mix and Link 0-100% (100); Shimmer 0-100% (0); UI size 75/85/100%.

Singing bass

Try Single and Recovery 3-4. Raise input only enough to stabilize sustained notes; retain the player's articulation.

Held vocal notes

Double can keep a long note settled, but leave enough recovery for the next quiet word. Compare at phrase level, not a single peak.

Faster parallel texture

Fast with a lower Mix gives a different rhythmic density. It is a creative option, not a claim to reproduce an original vintage mode.

Keep in mind

Fast is a Lynch mode; do not confuse it with Retro Instruments' named Triple mode. Recovery numbers are not seconds on the faceplate, and doubling a setting does not guarantee twice the audible recovery on changing material.

Shared Bypass, INIT, presets, contextual help, trial and automation behavior are covered earlier in this guide. Default values are shown in parentheses where useful.

Build a useful chain

Small moves in series often beat one stage working too hard.

Vocal: LAL73 > LAL76 > LAL2A

Use LAL73 for broad tone and sensible rumble removal. Let LAL76 catch short peaks. Use LAL2A for a small amount of phrase leveling. Set and level-match each stage in turn. Do not aim for a fixed total GR; judge diction, emotion and breathing.

Drums: LAL312 > LAL4K

Use the calibrated LAL312 input as the starting point for discrete-style colour. In LAL4K try a slower attack, a modest ratio and a small amount of reduction. Adjust the detector high-pass if the kick drives too much pumping.

Bass: LAL610 > LAL56

Use a restrained low shelf and stage drive in LAL610. LAL56 can hold sustained notes while leaving the articulation usable. Listen to the quiet note after a loud one when choosing recovery.

Alternative: LAL6A on a parallel group

Make a denser version, reduce its output and blend it underneath the original. Keep routing simple: do not combine a host parallel bus and internal Mix without understanding both contributions.

Mix bus: keep headroom

LAL4K can supply gentle movement; a preamp can add subtle colour if desired. Use a separate final limiter and loudness meter when delivery requires a true-peak ceiling or integrated LUFS target. Studio Classics does not automatically master or normalise a mix.

Automation, printing and export

Keep the performance. Make the commitment deliberately.

Automation

Sound controls are exposed as AU parameters. Use the host automation lanes to change drive, tone, reduction or Mix. Continuous controls are smoothed; stepped gain, ratio, mode and frequency choices remain discrete musical decisions. Avoid rapid discontinuous switches just because they can be drawn.

Project recall

The host saves the sound settings with the project. User presets are reusable snapshots; they are not backups of the audio project. Keep original recordings and project backups under your normal studio policy.

Print or keep editable

An insert affects the monitored or rendered sound according to the host routing. Ordinary playback inserts do not automatically rewrite an existing audio file. To commit the sound, bounce, freeze or record a routed output in the host. Check the result before removing the source.

Render a short test first

Compare a short realtime and offline export with the same settings, sample rate and routing. Program-dependent compressors need the preceding musical context; starting halfway through a held note may give a different initial envelope.

Changing sample rate

The plug-ins recalculate timing and filters for the host rate. They do not claim sample-identical renders between rates. At rates with a low Nyquist limit, requested EQ frequencies are constrained to safe realizable values.

Sharing a project

The recipient needs compatible plug-ins and a valid licence or active trial for processing. A preset does not transfer an activation. For dependable handover, also provide printed audio using the host export tools.

Troubleshooting

Follow the signal before changing the sound.

I hear no change	Check Bypass and the Trial & Activation state, then increase a relevant control. A subtle starting preset may deliberately be close to clean. Confirm the plug-in is on the track you are hearing.
The effect is dry	An expired trial, unavailable licence store or clock error uses dry bypass. Open Trial & Activation and inspect the status. Do not reinstall repeatedly to try to renew the trial.
The sound is too loud	Reduce output / makeup and check downstream headroom. On LAL312 keep the digital Input Trim calibration in mind. On LAL73 the MIC range can be much too hot for a recorded line-level signal.
The sound disappears	Check input gain OFF, Gain or Level at 0, the host mute and routing. At partial Mix, reversed wet polarity can cancel the dry path.
The needle hardly moves	Select the correct meter mode and allow average-level ballistics to settle. On a compressor, raise input / reduction or lower threshold. Changing output makeup alone should not increase GR.
The meter stays left	A quiet source may be below its printed VU range, or very heavy GR may exceed it. Check the host peaks and chosen VU mode before adding gain.
A selector seems inactive	Shelf frequency changes need nonzero shelf gain. Stereo Link needs unequal stereo detector levels to show a difference. A bypassed EQ section will not respond until enabled.
A preset is missing	Open PRESETS > Show User Presets Folder. Use the model-specific User Presets directory. Host-native saved settings may live in a separate location. A different model's file is intentionally rejected.
Crackles or slow response	Increase the host buffer for a dense mix, reduce unneeded active instances and check device/sample-rate settings. Do not confuse interface roundtrip latency with the 16-sample latency reported by each processor.
An update looks unchanged	Quit the host fully and rescan that AU. Check the version displayed in the editor and Plug-in Manager. Duplicate user/system components can confuse diagnosis; contact support before deleting files.

Under the hood

Original models, built for real-time work.

Processing format	Native arm64 AUv2 audio effects. Mono-to-mono and stereo-to-stereo operation. No instrument, MIDI output, external sidechain input or standalone recorder is supplied.
Sample rates	8-192 kHz processing range. Common music-session rates are 44.1, 48, 88.2 and 96 kHz. Very low rates necessarily constrain high EQ bands.
Oversampling	The nonlinear path runs at four times the host rate with fixed FIR reconstruction filters. Oversampling reduces some aliasing; it does not make all nonlinear artifacts impossible.
Latency	16 host samples per instance, reported to the host: approximately 0.36 ms at 44.1 kHz or 0.33 ms at 48 kHz. The internal dry path is aligned to it. Multiple serial instances add latency.
Core models	Simplified nonlinear amplifier stages, gentle low-frequency memory terms, independent digital EQ filters and original detector / envelope laws. No reference-unit neural capture or measured hardware transfer matching is claimed.
Real-time design	The processing path uses fixed storage. Licence I/O, preset file access and interface work are kept away from the audio callback.
Output headroom	Floating-point output can exceed 0 dBFS internally. There is no final brickwall limiter, automatic loudness target or true-peak measurement. Use downstream protection when required.
Compatibility testing	Local functional, state, automation, sample-rate, block-size, coexistence and native AU host tests were performed. They do not certify equivalence to hardware or compatibility with every host and Mac configuration. Try your own host before purchase.

Reference character versus added features

The historic references guide the kind of work each model invites. Parallel Mix, some stereo-link facilities, contextual help, preset folders and Analog Shimmer are software conveniences or Lynch extensions. LAL56's extended modes and LAL312's simplified tap behavior are described explicitly so you know what you are adjusting.

References and further reading

Study the originals. Listen to the result.

These manufacturer documents and historical sources informed the design references. Controls and numerical ranges in this guide describe the Lynch plug-ins; they are not an assertion that every reference feature is simulated. No manufacturer text or schematic is reproduced in the interface.

LAL76 - UREI / Universal Audio 1176

https://media.uaudio.com/assetlibrary/2/-/2-1176_manual.pdf

LAL73 - Neve 1073 preamp / equaliser tradition

https://www.ams-neve.com/outboard/1073spx/

LAL2A - Teletronix / Universal Audio LA-2A tradition

https://media.uaudio.com/assetlibrary/2/-/2-la-2-manual.pdf

LAL4K - SSL centre-section bus compressor tradition

https://support.solidstatelogic.com/hc/en-gb/articles/4408670926109-SSL-Bus-Compressor-2-Plug-in-User-Guide

LAL312 - API 312 / discrete op-amp preamp tradition

https://apiaudio.com/docs/manuals/312_user_2020-06-22.pdf

LAL610 - Universal Audio 610 / 2-610 tradition

https://media.uaudio.com/support/downloads/2-610-manual.pdf

LAL6A - RCA BA-6A broadcast limiting tradition

https://thehistoryofrecording.com/Literature/RCA/RCA-Broadcast-Audio-Equipment-Catalog-1955.pdf

LAL56 - Gates Sta-Level tradition and later extended-recovery designs

https://retroinstruments.com/product.php?product_id=stalevel

The 2-1176 and 2-LA-2 manuals describe later dual-channel hardware in their respective traditions. They are reference literature, not a claim that LAL76 or LAL2A is a reproduction of those exact dual units. Retro Instruments documents a later Sta-Level interpretation; its extended facilities are distinguished from the 1956 design.

All third-party names and trademarks identify historical design references only. They belong to their respective owners. Lynch Audio Lab is independent and is not affiliated with, sponsored by or endorsed by those companies.

Make it yours

Grit when you want it. Space when you need it.

The collection is designed to be played as part of a mix. Choose a processor for its role, move one control, match the output, and listen to the next phrase. A beautiful analog-style interface is an invitation to work with your ears; the result still belongs to you.

Support and downloads

Product, trial and manual: https://www.lynchaudiolab.com/studio-classics
Support: info@lynchaudiolab.com

What to send with a support request

Include the model and version, macOS version, host and version, sample rate, mono/stereo format, steps to reproduce, and a screenshot of the relevant settings. If possible, attach a short audio example you are allowed to share. For activation help, include the order number; do not send passwords or post licence keys publicly.

Version 1.0.0 / 4 November 2026

Eight processors, one shared 14-day trial and bundle activation; 72 factory starting points; model-specific user preset folders; contextual help; optional Analog Shimmer; neutral black hardware-style controls and warm VU displays. Existing component identities and parameter IDs are retained for project recall.

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