IRONHEART

Laney

User's guide

Aurora DSP

AURORA DSP

Audio Units VST AAX M1 64 bit Ableton

GETTING STARTED

Important Note 1:

The Laney Ironheart plugin comes in 3 versions which might be installed on your PC or MAC. Standalone version, VST version and AAX version. By default, all of the possible software instances are installed.

Standalone Version:

The standalone version of a guitar plugin operates as a separate application, independent of any specific digital audio workstation (DAW) or host software. It means that you can open up your Ironheart plugin from your desktop and start playing. This is beneficial for practicing, quick recording, or when not working within a larger music production setup.

VST (Virtual Studio Technology) version:

VST is a plugin format compatible with a wide range of DAWs on Windows and macOS, allowing seamless integration within these environments. This version can be loaded within a compatible DAW's plugin interface, enabling users to apply the Laney Ironheart directly to tracks within the DAW's project.

AAX (Avid Audio eXtension)

AAX plugins are optimized for use within the Pro Tools environment, offering seamless integration, stability, and specific functionalities tailored for Pro Tools users.

Important Note 2:

When you start up your **standalone** version of the Laney Ironheart plugin for the first time it will open up muted to avoid feedback resulting from the improper default audio device setting. Please make sure to choose the correct audio device type at the very beginning.

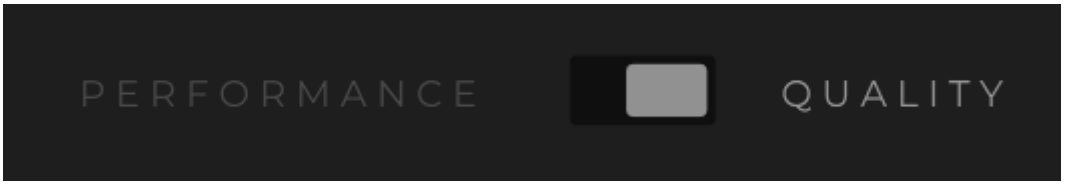
Important Note 3:

Using the plugin in DAW (Digital Audio Workstation) environment – all the audio settings are overridden by settings of the DAW software itself.

INTERFACE SIZE

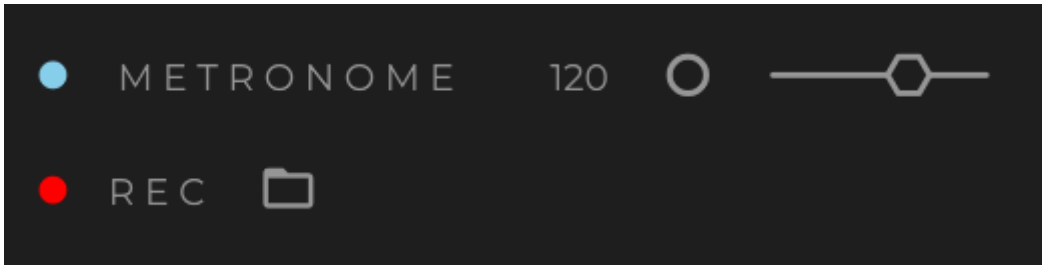
The window of the Laney Ironheart plugin is fully scalable in both VST and standalone versions. Click and drag the bottom right corner with your mouse cursor to set it to your likings.

QUALITY MODES



To increase the efficiency of CPU usage we introduced two plugin modes. Depending on your computer setup efficiency you may choose **performance** or **quality**. In **performance** mode your CPU will be less overloaded but at the cost of the slightly less accurate modeling. This setting is optimal for older hardware setups and less efficient processors. **Quality** mode is your go-to, full performance, most accurate modeling setting with the highest audio quality.

METRONOME AND RECORDING MODULES



Metronome and REC modules are additional features located in the top left corner of the interface. Those additional features are available only in the standalone version of the plugin. They facilitate the quick and convenient recording of your riff ideas and playing along with the metronome click for practice purposes.

Metronome (standalone only)

Click on the METRONOME text to toggle the metronome on or off. Blue dot indicates the module being turned on. You can adjust the tempo in several ways.

You can set tempo manually: input the desired tempo value manually by writing with your keyboard. The maximum value within the metronome is 240 BPM. Alternatively, you can click and drag the tempo value up and down with your mouse cursor to change the value on the fly.

Tap Tempo Function: By pressing the keyboard key T or clicking at the appropriate pace on the dot you can adjust the metronome tempo accordingly to the speed of your clicking. For better accuracy we recommend tapping at least 4-8 times.

Volume Control: Use the slider on the right to adjust the metronome's playback volume.

REC (standalone only)

The REC module allows you to capture your riff ideas using the currently set sound. If inspiration strikes, swiftly record your invaluable idea using the built-in module. Clicking on the "REC" label initiates recording in WAV format. A red dot appears on the left, indicating that recording has commenced.

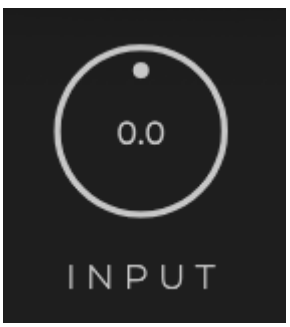
Alternatively, you can initiate recording by pressing the "R" key on your keyboard.
Clicking REC or pressing keyboard R again, stops the recording and prompts the "Save Recording" window. Before saving, customize the default file name as desired. Clicking the "SAVE" button stores the WAV file with your recording. Click "Cancel" to exit the module without saving.

All recorded files are located in the default directory, accessible by clicking the folder icon on the right side. Knowing the file locations allows playback at any time using the default system file explorer.

Input/Output & Signal Meters

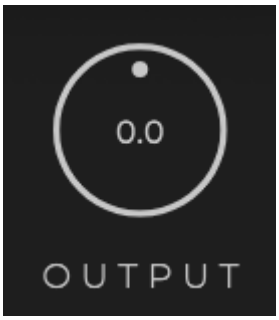
To the left and right of the centrally located Aurora logo you can see the volume meters for the input and output signals of your instrument.

Left side scale represents the current input signal of your guitar. This meter helps determine if the incorporated signal from the instrument needs any adjustment. Use the GAIN knob on your interface (where your instrument is connected) to adjust and correct the signal accordingly, to the outcome on the meter. If your signal meter is reaching the red zone of the scale (while playing loudest parts), it means that you need to back up your GAIN level on the instrument interface. Reaching red level means that your signal is too strong and it starts clipping - this should be corrected before any recording. There are various factors influencing the signal strength but the general rule of the proper recording is to avoid clipping at any cost.



This can be done using the additional **input knob**. This knob is set to 0dB by default and can help to correct the input value from the level of the plugin. Remember that various guitar pickups generate different signal strength so there is no good or bad value when it comes to the input gain. From a recording perspective it's always better to increase the volume in the post production instead of lowering the volume of tracks recorded with clipping and artificial gain created by signal being too loud at the beginning.

Right Side scale represents the current output signal of your guitar. This meter helps determine if the signal processed by the plugin needs any adjustment. There are many more factors influencing the signal output than signal input. Signal input is just your guitar direct sound being thrown out of the pickups – signal output is the same sound but being processed by all the parameters of the plugin including amplifier master volume, amount of given frequencies, overdrive boosters and even volume of the delay in the mix. The main principal remains the same – avoid reaching red zones on the signal meter.



To help you control the output signal we incorporated an additional **output**.This one is set to 0dB by default but depending on your current preset, might need slight adjustment. Illustrate it as a last volume knob of your signal chain and use it in critical situations if you really want to crank the amplifier to the maximum of its volume without clipping.

NOISE GATE

A one knob noise gate is like a virtual gatekeeper for the audio signal, allowing you to manage unwanted noise or background sounds that might accompany your guitar playing.
When you adjust the GATE knob, you're setting the threshold at which the gate opens and closes. Imagine it as a volume level: any sound below this level (threshold) will be muted or reduced, while anything above it will be allowed through unaffected. Turn the knob completely to the left to disengage the noise gate – OFF label on the knob.

Turning the knob to the right will increase the intensity with which the noise gate kicks in. Gate too sensitive might catch up even softer sounds which might trigger the gate to close, muting some of your guitar's subtler notes or nuances. Setting it more to the left might not effectively suppress unwanted noise but will act in a less aggressive way. The key is to find the sweet spot for your guitar and song you are playing where the gate is reducing the hums and unwanted noises and yet do not cut and shorten the signal in an unwanted manner – especially important in case of using clean presets with reverbs or delays which might be cut off at the end of their feedback.

Studio Environment Simulation (standalone only)

Note: This feature is designed for headphone referencing only.

The Studio Environment Simulation mode is a feature designed to replicate the acoustic conditions of a professional studio environment within your headphones. It's aimed at providing you with a reference or simulation of how your guitar would sound if it were played through studio-quality speakers within an acoustically treated studio space. This simulation can assist in fine-tuning audio content, ensuring that it sounds accurate and clear across different playback devices. This allows the user to listen critically and make adjustments or corrections as needed to achieve desired sound quality.

This feature does not translate into the sound quality of recording inside the Ironheart REC module inside the plugin. Recorded file will not be treated by the studio environment simulation – this feature is for the reference purposes only.

Please remember that this feature is dedicated for headphone referencing – using a regular audio system will not guarantee the desired reference point as it will not translate accurately.

GLOBAL FEATURES

In this chapter we will cover the basics of the Laney Ironheart plugin interface. The main assumption of our software was to deliver the real-amp-like experience to the users of our plugin. We want you to treat it like a real amplifier, real pedals, microphones, and cabinets which are directly right in front of you so you can use it in the same intuitive way.

The Laney Ironheart plugin is divided into 4 main **modules** representing 4 different types of guitar gear that are typically used by guitarists using half-stack setups.

When starting the plugin, the amplifier (**AMP**) module is loaded by default. At the bottom center screen, you may see the docked menu consisting of four buttons responsible for switching between various modules of the Laney Ironheart plugin. To move between each of the sections, click on the corresponding button to change the screen and see the different loadouts of the plugin.

You can bypass each of the 4 modules separately by clicking the corresponding power button near the module name. It's a convenient way to quickly compare the differences of various changes made in the preset chain.

ABOUT

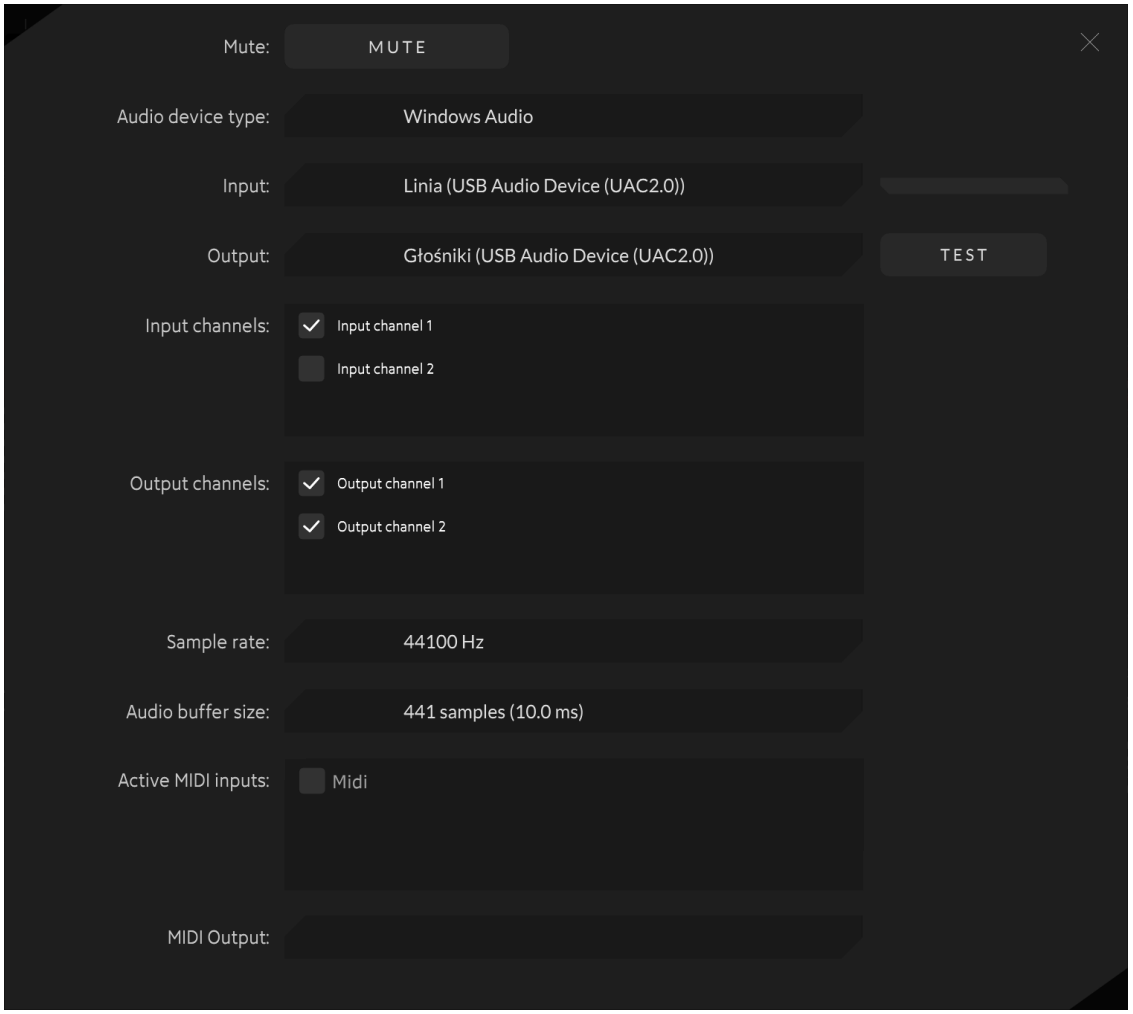
About – in this section you may easily check the information about the current version of your Laney Ironheart software. AuroraDSP is running constant improvements and updates based on the feedback of our community. We recommend our users to periodically check whether they have the latest version installed to ensure they don't miss out on any new presets and enhancements.

SETTINGS

Important: Applies only to standalone versions. Audio settings for the VST version of the plugin are overridden by the general audio settings of your DAW (Digital Audio Workstation) software.

Note: To toggle signal mute-unmute use mute button. Do this **after** choosing the proper audio setting as described in this chapter.

Before you start and immerse yourself into the Laney Ironheart plugin, we need to adjust some settings first. In order to make the software work flawlessly and with the lowest possible latency, you need to correctly define the sound settings of your audio configuration. In the upper left corner of the Ironheart window you will find the settings button. Click it to open the pop-up window of audio settings.



Audio Device Type

Firstly choose the proper audio device type from the drop-down list of the menu. Please be noted that the audio device type is not the audio device itself. Audio Device type corresponds to the driver type which supports your interface. If your interface is using ASIO drivers, choose it as your audio device type. This will automatically search for all ASIO type devices available on your computer to be selected in the menu below.

Device

Choosing the proper audio device type will load every driver associated device installed on your computer in the next sub-menu called **device**. Now you can choose your hardware from the drop-down list. Choose your interface/audio source – the device you connected your guitar to.

If for some reason your configuration is not showing up, please make sure that drivers of your interface are installed correctly and are up to date. AuroraDSP strongly suggests using latest and original releases of the drivers originating from the hardware manufactures databases.

Note: As hardware audio settings of your computer are not defined during the first start-up, the automatic feedback protection system might mute the signal of your instrument completely. Use the **mute** button at the top of the settings window to unmute the signal.

Choosing the correct audio **device** will automatically load and display the possible **output** and **input** channels configuration of your interface.

Input Channels

Refer to the specific audio sources or paths that the plugin will utilize for processing guitar signals coming from your interface/audio source. In this case, it might

happen that you will use only one input for your guitar. You can leave all the other unused inputs unchecked since they will not be used and processed by the plugin. For example: if you got two inputs for your instruments available on the interface and you are using only one cable to connect your guitar with your interface it means that only one input is sending the signal to the plugin – this is your main source of the sound and should be selected as the input channel as the second one is not important from the plugin perspective.

Note: Each audio interface works in a slightly different way which we are unable to cover perfectly from the level of this manual, however, the general principle of choosing correct input and outputs remains the same for all the audio devices.

Output Channels

define all the available output sources of your interface/audio device. This refers to speakers, studio monitors etc.

Example: If you are using two studio monitors (left and right) connected via two separate cables this means that you got at least two channels of output. One is sending the sound from the interface to the left speaker while the second one is sending it to the right speaker. On your audio interface, locate the output ports labeled for left and right channels. These might be denoted as “Output 1/2” or “Main Out L/R.” To hear the sound properly from both of them, you need to select both left and right output channels within the plugin. If you got more speakers, monitors or other sound playback devices hooked in – choose all the output channels available for your setup.

Notes:

Correct Channel Assignment: Verify that the left and right output channels on the audio interface are connected to the corresponding left and right inputs on the studio monitors to maintain stereo imaging.

Balanced Connections: Balanced cables help reduce interference and noise, providing cleaner audio transmission. Always use quality cables for optimal performance.

Volume Matching: If you are using studio monitors - adjust the volume levels on both studio monitors to ensure balanced sound output. This prevents uneven volume distribution between the left and right channels.

Audio Interface Settings: Some audio interfaces may have external software control panels or hardware switches to configure the output settings. Ensure these settings are adjusted to enable proper routing to the studio monitors.

Sample Rate

Is the number of samples per second. It significantly influences the audio quality, both during recording and playback, within the guitar audio plugin. In the digital audio terminology, it defines the quantity of audio information which is translated into binary information data. Analog waves are divided by computer within a certain rate to reconstruct it into the digital format. The audio standard is equal to 44,1kHz. Imagine a waveform which was divided into 44100 pieces every second.

For playing back pre-recorded tracks or using the plugin solely for playback purposes without recording, the sample rate selection should align with the sample rate of the audio files you're working with. Most common rates are 44.1 kHz and 48 kHz for standard audio files. Select the same sample rate in the plugin to ensure accurate playback without any loss or distortion. When recording your guitar or any live input, it's essential to choose an appropriate sample rate. Higher sample rates, such as 48 kHz, 96 kHz, or even 192 kHz, offer higher fidelity and capture more audio detail. However, they also result in larger file sizes and may require more processing power.

For most general purposes like recording and mixing, a sample rate of 44,1kHz and 48 kHz is widely accepted and provides excellent audio quality without excessive file sizes.

Notes:

Ensure your audio interface and system support the selected sample rate for both playback and recording. Some interfaces or systems might have limitations on higher sample rates.

Maintain consistency in sample rates across your entire workflow to prevent conversion issues or audio quality degradation when combining various audio sources or tracks.

Audio Buffer Size

Is the number of samples processed by your computer. Lower buffer size will reduce the latency but will increase the usage of your computer resources. Higher buffer size will do exactly the opposite. For recording and playback purposes you want to keep your latency level as low as possible meaning setting the buffer size to the lowest value which is tolerated by your computer. The Laney Ironheart plugin displays the predicted latency in ms (milliseconds) values right next to the buffer size you can choose. Lowering the buffer size decreases latency but might cause audio glitches if your system can't handle the processing demands. Experiment with different buffer sizes to find the lowest setting without compromising performance.

Notes:

Audio latency refers to the delay between the moment a sound is created (e.g., when you play a note on your guitar) and when you hear that sound processed through the plugin. In the context of a guitar audio plugin, latency can significantly impact playing experience and recording process.

Latency values, audio buffer size and sample rate is strictly dependent on your interface and computing power of your setup.

The performance of your computer, audio interface, and the plugin itself can affect latency. A faster processor and a more capable audio interface can reduce latency.

The buffer size setting on your audio interface or within the plugin affects latency. Smaller buffer sizes reduce latency but might strain your system's performance.

LICENSE

In this menu you can activate, deactivate, or purchase license code for the Laney Ironheart plugin. If you are using a 14-day trial version of our software, you can activate it and unlock the full version by typing in license code and clicking **confirm.** By clicking **get code** you will open the product site in the default browser of your computer.

MONO / STEREO

(Vst only)Those buttons are responsible for switching your input mode between mono and stereo. One of the practical examples might be double tracking your guitars in the DAW project. You can use a single instance of Laney Ironheart plugin on the bus track within your DAW to pan two (or more) guitar tracks in stereo and control all of them from a single instance of a plugin docked in the bus track. This is an extremely powerful tool considering the possibility of balancing the microphones and cabinets signals in the **CAB** section of the plugin. Using it in standalone mode you switch between the modes by enabling second input in the settings menu. You can plug in two instruments and play them simultaneously. You can practice/jam with another guitar player using one plugin!

TUNER

Is the built-in module allowing you to quickly tune your guitar. You can change the default frequency of 440Hz or tune silently by clicking on **mute** button.

PRESETS

In the central top section of the interface, you'll find the preset drop-down menu for the Laney Ironheart plugin. Selecting a preset from this menu will modify all the currently loaded settings of the amplifier, cabinet, microphones, effects, etc., setting them to a predefined configuration stored as a preset.

When you modify any values in the saved preset, you might notice a small star icon next to the preset name. This indicates that changes have been made and need to be saved if you wish to keep them. The Undo and Redo buttons are useful when you accidentally alter something or want to swiftly compare implemented changes (e.g., during direct input playback in your DAW while refining your guitar's sound).

Whenever you are satisfied with the sound that you managed to tweak inside the software, you may easily store your idea by saving it as a User Preset. There are no quantitative restrictions - you can create as many presets as you like. To save your current loadout as your own preset, click on the "save" button.

Type in your preset name and click the green Save button to store your configuration. Your preset will be visible in the drop-down menu under the User category. To delete the preset from your library, load it in and click the Delete button.

Note: Remember that you can easily share your presets with other users. Presets are stored as an .xml file format and can be transferred between the instances of the software.

STOMP

In this section you will find two Laney guitar pedals from the world-famous Black Country Customs. Each of the pedal elements is clickable and tweakable just like with the real stomp boxes. Both pedals may be turned on simultaneously. Click on the footswitch of the pedal to toggle it on or off.

To edit any of the pedal parameters - click and hold the knobs while moving the cursor of your mouse to change the value. Alternatively double-click and type-in the desired value manually using your keyboard. To toggle the MODE switches of the pedals, click on the desired position on the pedal.

STEELPARK OVERDRIVE PEDAL

The “STEELPARK” is a versatile Boost pedal with masses of headroom designed to help you push your amp harder and cut through any mix.

Drive

Adjust the amount of compression and soft distortion.

Bass

Adjusts the low-end content post compression/ distortion. Centre knob position (5) is typically flat and serves as a solid initial reference point.

Volume

Adjusts the output level to balance sounds or push an amplifier harder into natural distortion.

Mode

STEELPARK can be configured to run in 1 of 3 modes available
Mode D1 - Orange LED Pre MID BOOST – good for thin sounding pickups .
Mode D2 - Blue LED Pre LO MID BOOST. Gives a fatter sound.
Mode O/D - Purple LED. No pre boost. Good for pushing clean sounds.

MONOLITH DISTORTION PEDAL

Designed to give the player total command and control over a of spectrum different gains and harmonic distortions from classic fat rock tones to jangly indie lines and everything in between. The MONOLITH delivers expressive gain even at low gain levels. With a very pleasing dynamic response to pick attack the harder you dig, the more expressive it gets. Even at high gain levels, when you hit It hard you can still hear every note of big chords and the nuances of intricate lead lines.

Distortion

Sets the amount of distortion and sustain. As you dial in more gain you get more compression and sustain.

Range

Adjust the low-end content before distortion is added to the signal. Using the **range** control allows you to tighten up the bottom end of your tone and clean up muddy sounding pickups at high gain settings. 0 – Full range 10 - Maximum low-end reduction

Volume

Adjust the output level to balance sounds or push an amplifier harder into the natural distortion with a lower level of distortion dialed in.

Mode

MONOLITH can be configured to run in 1 of 3 modes available. Mode D1 - Orange LED – Heavily compressed distortion, smooth sounding without loss of definition.
Mode D2 - Blue LED – Soft compressed distortion. A more expressive sounding distortion.
Mode O/D - Purple LED. Traditional overdrive distortion. Good for boosting your sound level

or pushing the front end of tube amps harder. There's minimal compression, meaning your amplifier will primarily handle the tonal coloring.

AMP SECTION

PRE-BOOST SWITCH

Switch the input boost circuit on or off. The Pre-Boost circuit increases the input signal to the preamp tubes, just like placing a boost pedal in your signal path. This drives the preamp tubes harder, resulting in more distortion. Work for both channels. Turning it on will be indicated by the LED illumination.

PRE-BOOST CONTROL

Controls the level of boost applied to the guitar signal.

LEAD GAIN

Controls the level of preamp gain on the lead channel. Turning this knob clockwise will add more distortion to your guitar signal, ranging from light overdrive to full on metal sounds. Use this in conjunction with **LEAD VOLUME** to achieve the correct volume and distortion level you require.

LEAD EQ CONTROLS

These are traditional sets of passive tone controls. Passive controls have the advantage of always sounding musical at any of their settings, due to their unique interactive nature. Set these to midway (0) as a good starting point for further tone shaping.

LEAD EQ PULL SWITCHES

Note: use the right click of your mouse to activate the pull-switch knob mode. Controls with activated pull-switch will be indicated by the red backlight.

Pulling each of the EQ control knobs will shift the response of each knob as follows:

BASS: DEEP

This extends the low-end frequency response, resulting in fuller, heavier sound of lower notes.

MIDDLE:SHIFT

lowers the frequency range of the **MIDDLE** control to give a tighter sound.

TREBLE:SHIFT

broadens the **TREBLE** control frequency response, to give a rounder sound to higher notes, especially when with thin sounding pickups.

LEAD VOLUME

Controls the **lead channel** volume. Experiment with different combinations of **GAIN** and **VOLUME** controls to achieve various sound results. Reducing the GAIN while increasing the VOLUME will result in warm, open, overdriven sound as the power amp is driven harder, while reducing the VOLUME and increasing the GAIN will give a tighter, more modern sound with more distortion.

CHANNEL SWITCH

Switches between the LEAD and CLEAN/RHYTHM channels.

CHANNEL LEDS

Indicates which channel is currently selected by the **CHANNEL SWITCH**

CLEAN/RHYTHM SWITCH

This switch activates the **CLEAN** mode on the **RHYTHM** channel. When operated, the **CLEAN VOLUME** control becomes active, while **RHYTHM GAIN** and **RHYTHM VOLUME** are removed from the signal path. When using the clean mode, the preamp gain is lowered, resulting in cleaner tone.

CLEAN VOLUME

use this knob to control the volume of the amplifier on the clean mode. The amp can still be driven to overdrive with the knob turned fully clockwise and can be driven harder by using the **PRE-BOOST**.

CLEAN/RHYTHM

Works just like controls of the **LEAD CHANNEL.**

RHYTHM GAIN, EQ CONTROLS, PULL-SWITCHES, RHYTHM VOLUME

Works just like describe for **LEAD GAIN**

DYNAMICS

Allows you to control the response of the amplifier at lower frequencies. Turning the knob clockwise gives a looser low end, while lower settings provide a tighter response. The optimum setting is dependent on the speaker cabinet or IR used.

TONE

This control works in a similar fashion to the tone control you probably have on your guitar except that it uniquely works at the other end of the amplification chain. This has the ability to not only control the overall top end response but also reduce upper harmonics on the output stage and preamplifier overdrive sounds. This will give you bright cutting sounds at high settings and smooth rounded sounds at lower settings. Midway (0) is a good starting point. Both the **TONE** and **DYNAMICS** controls depend greatly on the speaker cabinet set.

REVERB

Control the level of the built-in Laney designed digital reverb.

WATTS

This knob adjusts the signal level within the power amplifier, allowing it to be driven harder at lower volume levels. For full output-power, running the power tubes at maximum levels, turn this control fully clockwise. To reduce the output volume, turn this control to the left.

POWER SWITCH

Main power switch for the unit. You can turn off and on the amplifier within the plugin using the power switch.

CAB MODULE

IMPULSE RESPONSE (IR)

In the context of guitar audio plugins, IR refers to the digital representation of how a particular speaker cabinet, microphone, and room combination sound when a guitar signal is played through them. It captures the tonal nuances, frequency response, and spatial characteristics of that setup.
Musicians and audio engineers use IRs in guitar plugins to emulate the sound of different speaker cabinets, mic placements, and environments without needing to physically set up those specific rigs. This flexibility allows users to experiment with various tones and create music without the limitations of physical gear.
The process involves capturing the response of a speaker cabinet to a short audio burst (an impulse) and then using this data within the plugin to shape the guitar signal to replicate the characteristics of that particular setup.

The **CAB** module of the Laney Ironheart plugin is your virtual playground composed of two original cabinets designed by Laney Amplification.

GS 4x12 - The Laney GS412IS is a monster cabinet loaded with 4x12” HH Custom designed drivers. This closed-back speaker cabinet utilizes time tested construction methods to deliver the bass, mid response, and great sound you would expect from a Laney 4x12 cab. The HH 12" drivers (16 ohms) deliver a strong, aggressive low-end and

powerful low-mids along with an attacking upper mid-range and crystal-clear top- end. The GS412IS delivers a clean warm tone, beautiful crunch tones and all out aggression without unwanted coloration.

LA 4x12 – Laney Black Country LA412 loaded with Celestion G12H speakers. The LA12 is part of the Black Country Customs series, aiming to capture the classic British rock tones while offering modern versatility.

In this section you will shape your sound by changing speakers, cabinets, and microphone settings to achieve the desired sound. We know that adjusting those components give guitarists the strongest headaches, so our goal was to make it as simple as possible. A slight microphone adjustment may drastically change the character of your final sound. Fortunately, all the tedious steps you would have to do to check the possibilities of the speakers, microphones and cabinets are already done for you. We spend tons of hours in a studio-controlled environment to capture every sound detail in a hi-res 192KhZ. Effect of our work is within the reach of your mouse cursor.

Cabinet Selection - at the top of the module loadout you may see two cabinets to choose from. You can switch between two original Laney cabinets.

Laney Ironheart plugin gives you an opportunity to choose 2 independent microphones and speakers setups simultaneously, over the same or different cabinets. You can create your sound by blending two various microphones and speakers' configurations or use a single microphone/speaker setup. On the left and right side of the cabinet you may see the same options relating to those settings.

MIC

Mic menu allows you to switch between one of the four highest quality studio standard microphones. You can choose between: DYN57, TUBE47, DYN421 & RIBBON 121.

POSITION

Position slider adjusts the horizontal distance of the microphone against the center of the speaker cone. You can manually change the microphone placement by clicking and dragging your mouse cursor left and right.

ANGLE

Angle slider adjusts the microphone angle against the speaker cone. Each angle variation can capture distinct nuances of the speaker's sound, allowing for different textures and tonal qualities in the recorded guitar signal. Experimenting with these positions helps in finding the desired sound for a particular recording or style. You can change the angle using the mouse wheel while having your cursor on the microphone.

DISTANCE

Distance slider adjusts the position of the speaker against the cabinet. Moving the slider to the right increases the distance, while moving the slider to the left moves the microphone closer to the cabinet. You can manually change the distance of the microphone by clicking and dragging the microphone with your mouse cursor. You can manually change the microphone placement by clicking and dragging your mouse cursor up and down.

CUSTOM

This button gives you endless sound shaping possibilities by allowing you to choose any of your favorite impulse responses. Click the button and choose the location of your IR file to load it up in the Laney Ironheart plugin.

Note: please be advised that loading your own IR file will block the other functions of microphone and speakers in the CAB section of the software.

IR ON

This button engages and disengages the given microphone/speaker combination.

Slider at the bottom center is made for a microphone blending purpose. At the center position, both of the engaged microphone/speaker configurations are blended together in the 1:1 ratio. Remember that blending does not mean that the volume share is at the same levels. Various speakers and microphones are characterized by a different output level. By moving the slider to the left and right you can adjust which of the configurations will dominate in your final sound. Remember that both IR's need to be turned on.

AIR

Air knob adjusts the high-frequency content in the sense of space and openness in the sound. boosts the higher frequencies in the simulated cabinet response. This can add brightness and sparkle to the sound, making it more present and detailed.

EXPORT IR

If you are satisfied with the sound achieved from combining all of the edited parameters you can export it as a single IR file in .wav format to implement in any other plugin. Click the **EXPORT IR** button, choose the file name and save location to export the file.

EQ AIR

When turned up, it increases the presence and brightness of the high-frequency content (7 kHz and above) in the audio signal. This is particularly applied after the simulated cabinet section, enhancing the overall tonal characteristics as if the sound were coming from a specific type of speaker cabinet. Adjusting this knob allows users to tailor the sound to their preferences, achieving a desired level of clarity and brilliance in the high-frequency range.

IRDX

(Impulse Response Dynamix) technology breathes life into the guitar sound by adding dynamic and true-to-life speaker behavior to static impulse responses. IRDX technology, developed by producer Jens Bogren, uses advanced machine learning methods to reconstruct the difference between a real cab and an impulse response. The end result is a guitar amp sim that reacts like a real amp, easily fits in a mix without sounding stale, and feels great when playing through.

FX MODULE

The FX module section consists of four additional effects based on the original Aurora DSP algorithms that give your sound a final color. Each of the effects might be turned on and off by clicking on the red LED light over the given component.

A 10-band equalizer adjusts the share of the given frequencies in the range of -12 / +12 dB. To cut or boost the given frequency, move the corresponding slider to the left or right. To access the manual window, double-click the slider, enabling you to input the desired value using the keyboard. HP and LP stands for High Pass and Low Pass filter and may be adjusted by turning the potentiometer knob respectively. The HP filter removes low frequencies, allowing higher frequencies to pass through, while the LP filter removes high frequencies, letting lower frequencies pass. They help shape the tone by cutting unwanted frequencies at the extremes, allowing you to fine-tune the sound of your guitar.

LIMITER

Serves as a basic compression tool that helps establish the highest signal level. The Input knob manages the incoming signal, while the Ceiling knob sets the maximum dB point for the limiter to activate. This indicates the threshold beyond which the limiter compresses your sound. The Low-End control knob fine-tunes the limiter's impact specifically within the 100-250 Hz range.

The delay module offers three distinct delay types: Stereo, Slap, and Analog, each with unique characteristics. It includes five control knobs:

Mix:

Adjusts the balance between the original signal and the delayed signal.

Feedback:

Controls the number of repetitions or echoes of the delayed signal.

HP/LP

(High Pass) and (Low Pass): Tailors the frequency range of the delayed signal.

Time:

Sets the duration or interval between each delay repetition.

Moreover, there's a switch to toggle between BPM (tempo-based) and MS (milliseconds) settings, allowing you to synchronize the delay time either to the project tempo or set it manually in milliseconds.

REVERB

The reverb module enriches your sound with various reverberation effects. It offers several controls:

Size:

Adjusts the simulated room size or the length of the reverberation decay.

Pre Delay:

Sets a delay before the onset of the reverb effect.

Mix:

Determines the balance between the original signal and the reverberated signal.

Width:

Controls the stereo spread or spatial width of the reverb.

HP/LP

(High Pass) and LP (Low Pass): Tailors the frequency range of the reverberated signal.

These controls empower you to shape the reverberation characteristics, from room size to the timing and tonal qualities of the reverb, enabling you to create a rich and immersive sonic environment for your guitar sound.

Midi

These steps guide you through connecting your MIDI controller, configuring settings, assigning functions, and understanding how to control various aspects using MIDI commands.

- Connect Your MIDI Controller:
- Plug in your MIDI controller to your computer using a USB cable.
- Connect MIDI Cables:
- Use a MIDI cable to connect the "Output" of your MIDI controller to the "Input" of your audio interface.
- Open Standalone App:
- Launch the standalone application for your MIDI controller on your computer.
- Access Settings:
- Within the application, find and open the settings menu.
- Locate MIDI Inputs and Outputs:
- Look for options labeled "MIDI IN" and "MIDI OUT" at the bottom of the settings.
- Select MIDI Input:
- Choose your MIDI controller's input from the list (for example, "ESI MIDI INPUT 1").
- Assign Functions to Controller Buttons:

- To assign functions like volume control or effects to buttons on your controller, right-click the selected MIDI input, choose "Assign," and press the desired button on your controller.
- Control Knobs and Sliders:
- Continuous Controller (CC) functions, such as volume pedals and knobs, can be assigned to manipulate various knobs and sliders within the plugin.
- Use C0 Commands for Presets and Modes:
- For tasks like changing presets, channels, or modes on pedals, utilize C0 commands.
- Check Assigned MIDI Values:
- In the top menu of the plugin, click on the "MIDI" button. This will display the MIDI values assigned to specific features.
- Configure MIDI Thru and External Connections:
- To enable MIDI thru and connect the standalone app with your external equipment, select the appropriate MIDI output in the settings.