SEVERED · INC.

MODEL · AXN-1

AXION

USER MANUAL

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A GLOSSARY

00

OVERVIEW

Box contents, first connection, panel tour.

SECTION 00.0 · OVERVIEW

Getting Started

First boot, audio setup, and loading your first rig.

If you have used a guitar processor, modeler, or amp profiler before, the workflow on Axion will feel familiar: connect your instrument and monitoring, load a rig, set the output volume, and play. This page takes you through that first session. See Connections for every jack and Controls for the complete front panel tour.

Connect and start

1. Turn down your speakers, mixer channel, or headphones before making connections.
2. Connect a guitar or bass to **IN 1** with a standard instrument cable.
3. Connect **OUT L** and/or **OUT R** to powered FRFR speakers, an audio interface, an amplifier or a mixer using a standard instrument cable, or a TRS to XLR cable. You can also connect headphones to the headphone output if you prefer.
4. Connect the supplied power adapter. Axion will boot up in about ten seconds.
5. Gradually bring up the volume on your monitoring system, play, and turn knob **E** gradually to set the main output volume.

If you see input activity but hear nothing, open the menu and choose **Mixer**. Confirm that IN 1 and the output you connected are turned up. In Rig view, also check that no required devices are bypassed. See Troubleshooting for the full no audio checklist.

Load a factory rig

The current rig name and number appear at the top of the screen.

1. Tap the current rig name to open **Select Rig**.
2. Choose the factory set list in the left column.
3. Tap a rig in the right column to load it.
4. Play and adjust knob E for a comfortable output level.
5. Try other factory rigs the same way. You can also hold footswitch 4 or 5 to move to the previous or next rig.

A factory rig is read only. You can change it freely while it is loaded, then save the result as a user rig. The original factory version remains available.

Make a first change

Switch to Rig Mode if it is not already selected. Tap a device tile to open it, then turn the knobs below the screen to change the parameters shown above them. Close the device screen to return to the chain. Double tap a device tile when you want to compare it active and bypassed.

To add a device, tap a **+** position in the chain, choose a category or use search, then select the device or preset. The device is inserted at that exact point. See Building Rigs for moving, removing, and configuring the chain.

Save your first rig

1. Tap the save icon beside the rig name.
2. Choose a user set list. Select **New Setlist** if you need to create one.
3. Select the program slot where the rig should live.
4. Enter a name that will be easy to recognize on stage.
5. Tap **Save**.

Saving stores the device order, parameter values, bypass states, scenes, and control assignments with the rig. Saving over an occupied user slot replaces that slot, so choose the destination carefully.

Next steps

- Read View Modes to choose the best screen for editing or performing.
- Set up hands free control with Footswitch Assignment.
- Bind the five knobs and expression pedal with Rig Controls.
- Browse the complete Device Reference when you need a control or starting point.

Controls

The front panel: touchscreen, the A-E knobs, and the 1-5 footswitches.

Axion has almost no fixed-function controls. Where a traditional amp has a knob permanently labeled "Gain," Axion has one touchscreen, a row of five knobs, and a row of five footswitches, and what each of them does depends on what's on the screen and how the current rig is set up.

There are only three things to learn.



The Axion top panel: the touchscreen, the A-E knobs, and the 1-5 footswitches.

The touchscreen

The touchscreen is the main surface. You use it to navigate, build and edit rigs, browse devices and presets, and read meters. Most of what you do on Axion happens here; the knobs and footswitches are the fast, hands-on shortcuts to it.

Knobs (A-E)

Five knobs sit below the screen, silkscreened **A** through **E**. Their main job is to adjust whatever parameters the screen shows along its bottom edge. The on-screen label directly above each knob always tells you what that knob controls right now, so the same physical knob might be Gain on one page and Mix on another. Turn a knob, or turn its parameter on screen; they do the same thing.

One knob has a fixed default: knob **E** sets the master **output volume**, until the current screen hands it a different parameter. It is the only control that arrives with a job already assigned.

Tap a knob's on-screen control to open its **Rig Control** and change what it drives. See Rig Controls for binding knobs, the expression pedal, and MIDI.

Each knob also responds to two more gestures:

- **Press** triggers an assigned action (see Footswitch Assignment).
- **Hold** triggers a secondary set of actions. These can be reassigned, the default actions are labeled on the front of Axion such as the tuner, mode change, and previous/next rigs.

Footswitches (1-5)

Five footswitches run along the bottom, labeled **1** through **5**, for hands-free control while you play. Like the knobs, each one has a quick press and a press-and-hold:

- **Press** fires an assigned action. Typically, these can be used for stomps (toggle a device on or off), but a switch can also change scenes, step through rigs, tap tempo, trigger midi, play samples, anything that is built into Axion or that third-party plugins might provide.
- **Hold** like the knobs, switches can also be assigned secondary "hold" functions. By default they are:
  - **Hold 1**: Switch between view modes
  - **Hold 4**: Preset up
  - **Hold 5**: Preset down

Each footswitch has a light that reflects its state, so you can read the board at a glance: lit when its effect is on, dimmed when bypassed.

Assigning switches

The press and hold actions for every knob (A-E) and footswitch (1-5) are assignable, and the assignments are **saved with each rig**, so the same physical switch can do different things from one rig to the next. The panel-printed hold functions (Tuner, view mode, and preset up and down) are built-in defaults, but can be overridden and/or moved to other switches.

All switch functionality is managed in the Footswitch Assignment screen, visit this manual page for the full walkthrough.

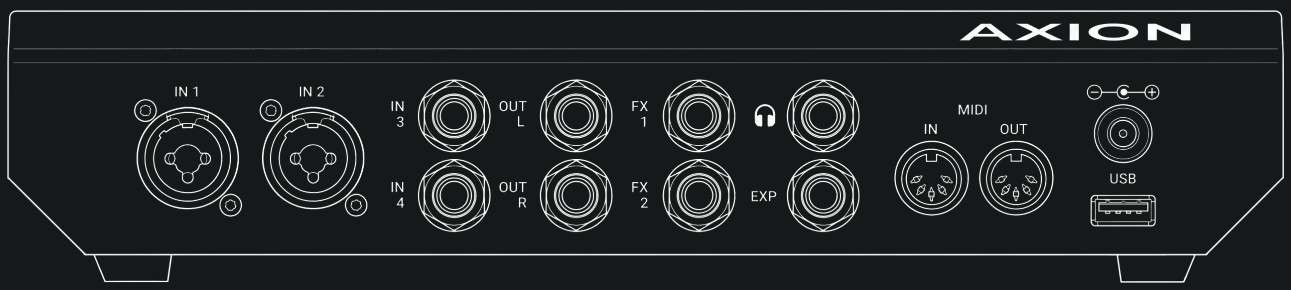
Expression pedal

An expression pedal connected to the **EXP** jack gives you continuous, foot-controlled sweep over one or more assigned parameters, such as volume or a wah. You assign it, set its travel range, and shape its response curve in Rig Controls. See Connections for the jack itself.

SECTION 00.2 · OVERVIEW

Connections

What each port on the Axion unit is for and how to connect to it.



The Axion rear panel. Each jack is described below, grouped by function.

Inputs

IN 1 & IN 2: instrument & microphone

Combo jacks that accept either a 1/4" TS plug (for unbalanced, e.g. guitar input) or an XLR for balanced input or microphone. The 1/4" input is high-impedance (Hi-Z) buffered, so it's the right input for a guitar, bass, or other instrument with passive pickups, but it will work with active pickups as well. The XLR side takes a microphone or any XLR line-level balanced input.

Unlike the rest of Axion's I/O jacks, the TRS connectors for inputs 1 and 2 are **not balanced line-level inputs**. They are special-purpose jacks for unbalanced input. To connect a keyboard, synth, or other line-level source to inputs 1 or 2, use an XLR cable. Otherwise, use input 3 or 4 for line-level TRS inputs.

IN 3 & IN 4: balanced line & FX returns

These 1/4" TRS jacks accept balanced, line-level sources such as keyboards and synths. They can also double as the **return** side of an effects loop: paired with the FX sends, they bring an external pedal or rack unit back into the signal chain. Unlike inputs 1 and 2, these two inputs are low impedance and are not ideal for directly plugging in a guitar with passive pickups, unless you use a buffer such as powered pedal or wireless guitar adapter.

CAUTION:

Never plug the output of a power amp into any of Axion's inputs. Doing so risks deep-frying your audio board, which will render your Axion useless until it is repaired or replaced.

Outputs

OUT L & OUT R: main outputs

The main stereo outputs. 1/4" TRS and balanced, but they also work unbalanced with a standard TS cable, for example straight into the front of a guitar amp. Typical destinations:

- A pair of FRFR (full-range, flat-response) powered speakers, the most common setup for amp-modeled tones
- An audio interface, mixer, or recording input
- The input to a power amp and cabinet, or the input to a traditional amp

FX 1 & FX 2: effects sends

Two additional balanced 1/4" TRS outputs, typically used as the **send** side of an effects loop, tapping the signal out to an external pedal or processor that returns through IN 3 / IN 4. They can also serve as extra auxiliary outputs, for example to stage monitors or amps, in-ear monitors, or additional mixers.

Headphones

A stereo headphone output for quiet practice and monitoring.

Control & connectivity

EXP: expression pedal

A 1/4" TRS jack for a continuous expression pedal, for real-time, foot-controlled changes to assigned parameters such as volume or wah.

MIDI In & MIDI Out

MIDI input and output for remote control and clock sync. Common uses:

- A MIDI foot controller selecting rigs by program change
- A DAW driving Axion during a recording session
- A drum machine or sequencer providing tempo to delay devices

See Rigs and Set Lists for how MIDI bank / program change maps to set lists and rigs.

USB-C

A USB-C port for connecting to a computer. Axion appears as an audio interface to your DAW, for recording its output and for re-amping, sending a recorded DI track back through the unit to capture a new tone.

Power

DC input

The barrel jack for the supplied power adapter. Use the adapter that ships with the unit. If you need a replacement power supply, go to https://severedcircuits.com.

Tuner

Tune silently or through, read pitch and cents, and set the reference frequency.



Axion's tuner listens to the instrument input and shows the nearest note, its octave, the measured frequency, and the distance from the note in cents. Open it by holding knob **E**, whose Tuner label is printed on the panel. You can also open Tuner from the navigation menu or assign its Open or Toggle command to a footswitch or knob. Close the tuner by pressing the switch again.

The output is muted when the tuner opens, so you can tune silently. The pitch display continues to work while muted. Tap **MUTED** to change the button to **THRU** and hear the instrument while tuning. Tap it again to mute the output. Closing the tuner restores normal audio automatically.

Reading the display

Play one string and let the note settle. The detected note and octave appear at the bottom left, with the measured frequency beside them. The number above the meter shows the pitch error in cents.

- Lit bars to the left of center mean the note is flat. Raise its pitch.
- Lit bars to the right of center mean the note is sharp. Lower its pitch.
- At three cents or less from the target, the center bar turns green and the display reads **IN TUNE**.

The scale covers 50 cents on either side of the target. A cent is one hundredth of the distance between two neighboring semitones, so small values near zero represent small tuning adjustments.

Setting the reference pitch

The **REF A** control sets the frequency used for A4. It starts at the standard 440 Hz. Tap the minus or plus button to adjust it one hertz at a time from 430 Hz to 450 Hz. All detected notes are calculated from this reference, which lets you match instruments or recordings that use a pitch other than A440.

Closing the tuner

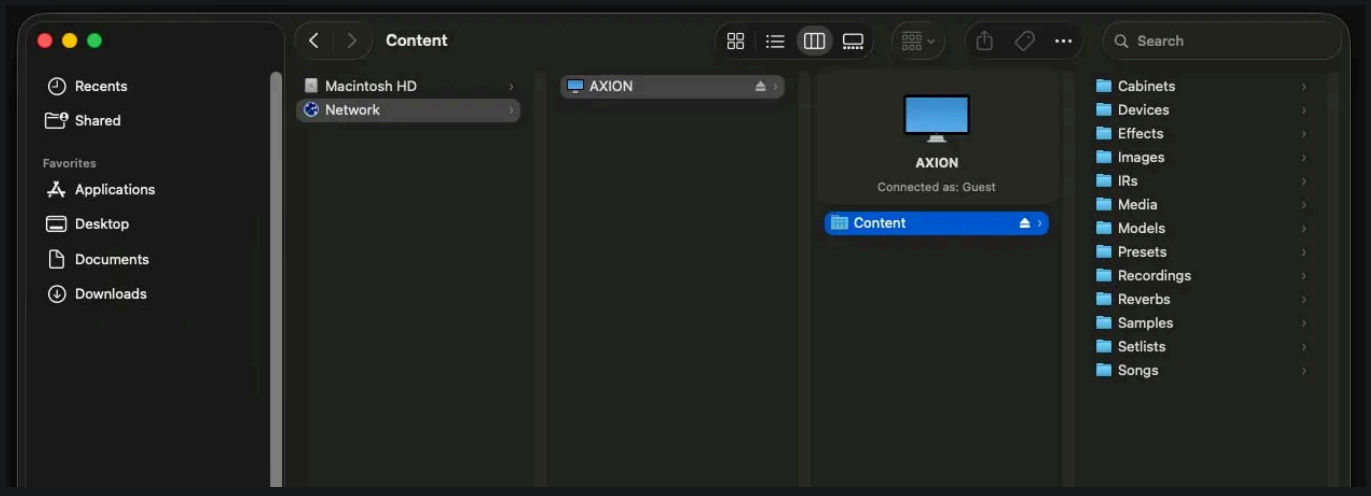
Tap the close button in the upper right, or press any footswitch. While the tuner is open, a footswitch press closes it without also triggering the switch's normal rig, scene, or stomp assignment.

Tips

- Tune one string at a time and mute the other strings to reduce sympathetic vibration.
- Use a clear, steady note rather than a hard pick attack. The display waits briefly for the pitch to settle.
- Check tuning again after large adjustments because changing one string's tension can slightly affect the others.
- Use THRU when you need to hear the instrument, and switch back to MUTED when silent tuning is more appropriate.

Axion Network Folder

Mount Axion's Content share over your network to copy plugins, presets, recordings, and other files on and off the unit.



The **Axion Network Folder** is a shared folder your computer can connect to over the network to move files on and off the unit. It exposes Axion's **Content** area, the same place your plugins, presets, recordings, and captures live, so you can drag files across in Finder or File Explorer just like any other network drive.

Behind the scenes it is a standard SMB share at `smb://axion.local/Content`. Any operating system with a built-in SMB client can connect; you do not need the companion app or any extra software.

Before you connect

- Put Axion and your computer on the **same network**.
- Know Axion's network name. By default it is `axion.local`. If you renamed the unit, use that name, or its IP address, wherever `axion.local` appears below.

The folder is open on the local network. No username or password is required in this version. Because access is unauthenticated, keep Axion on a trusted network you control rather than a public or shared one.

Connect on macOS

1. In Finder, choose **Go → Connect to Server** (or press ⌘K).
2. Enter `smb://axion.local/Content` and click **Connect**.
3. When prompted, connect as **Guest**, then click **Connect** again.

The **Content** folder opens in a Finder window. Axion also advertises itself on the network, so it usually appears on its own under **Locations** in the Finder sidebar as **Axion**. Click it and choose **Content** to connect without typing the address.

Connect on Windows

1. Open **File Explorer**.
2. Click the address bar, type `\\axion.local\Content`, and press **Enter**.

The **Content** folder opens. To keep it handy, right-click it and choose **Pin to Quick access**, or use **Map network drive** to give it a drive letter that reconnects automatically.

If Windows cannot find `axion.local`, use Axion's IP address instead (`\\<ip-address>\Content`).

Connect on Linux

Connect your file manager to `smb://axion.local/Content`, choosing anonymous or guest access when asked. From a terminal you can also mount it with `gio mount smb://axion.local/Content` or your distribution's CIFS tooling.

What's inside

The Content folder holds the files you can add to or copy from Axion:

- **Devices**: user-installed LV2 and VST3 plugins. See Third-Party Plugins.
- **Recordings**, **Captures**, and **Presets**: files Axion writes for you to copy back to your computer, plus presets you can add.

The **Factory** area holds the devices and content that ship with Axion. Add your own files under **Content**, not **Factory**.

You can leave the folder connected while you work. Axion picks up new files automatically. For example, a plugin dropped into **Devices** appears in the device browser a few seconds after the copy finishes, with no restart needed.

If Axion does not appear

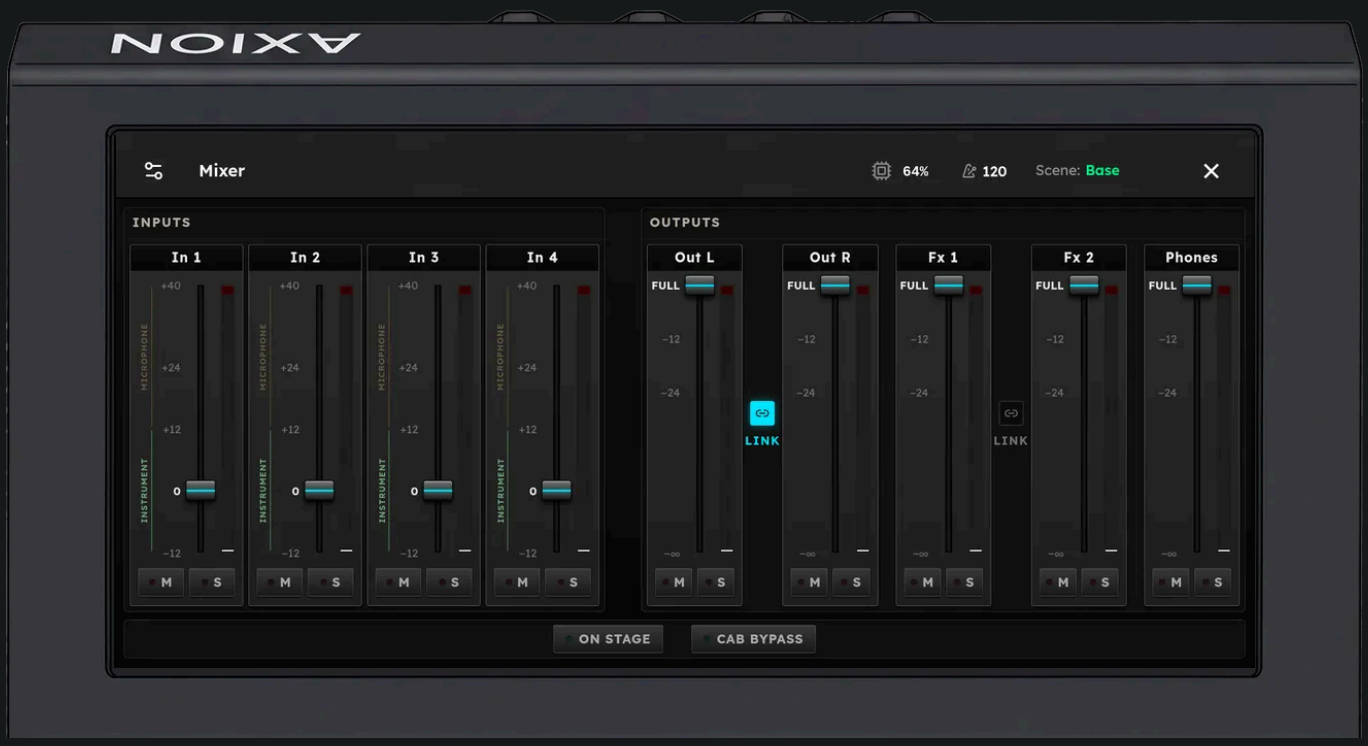
- Confirm both devices are on the same network and that Axion has finished starting up.
- Try the IP address in place of `axion.local`.

- On Windows, make sure network discovery is turned on for the network you are connected to.

For more help, see Troubleshooting.

Mixer

Set input gain for IN 1-4 and the level of every physical output, read the meters, compensate for stage volume, and bypass every cabinet at once.



The Mixer is where Axion meets the outside world. It sets how much gain each input jack gets before the signal reaches your rig, and how loud each physical output jack is after the rig has done its work. Every strip has a meter, so the Mixer is also the first place to look when you hear nothing or something sounds distorted.

To open it, tap the menu button at the top left of the screen and choose **Mixer**. Tap ✕ at the top right to go back to your rig.

Mixer settings belong to the unit, not to a rig. They stay the same when you change rigs, scenes, or set lists, and Axion keeps them when you turn it off. Set them once for your instrument and your rig, and leave them alone.

Inputs

The **INPUTS** panel has one strip for each input jack, **In 1** through **In 4**. Each fader sets the analog gain for that jack, from –12 dB to +40 dB. **0** is unity gain and is the default.

A guide next to each input fader shows the range that suits your source:

- **INSTRUMENT** (green), –12 dB to +12 dB. For guitar, bass, and other line-level or pickup sources.

- **MICROPHONE** (yellow), +12 dB to +40 dB. Use the extra gain for a microphone, which puts out a much weaker signal.

The label for the range the fader is in shows brighter than the other one.

Setting input gain

1. Turn your monitoring down.
2. Play the hardest you will play in a song, such as a full chord dug in with the pick, with your guitar's volume all the way up.
3. Watch the meter for that input. Raise the fader until the loudest peaks reach the yellow area of the meter, and stop before the red.
4. If the red clip light at the top of the meter turns on, lower the fader a little and try again.

Most passive guitars do well near **0**. Hot active pickups may need a few dB less. Input gain changes how hard the signal drives every device in your rig, so amp models and drives respond differently after you change it. Set it once and set your loudness with the output controls.

Outputs

The **OUTPUTS** panel has one strip for each output jack:

STRIP	JACK	WHAT IT SETS
Out L, Out R	OUT L, OUT R	Level of the main outputs
Fx 1, Fx 2	FX 1, FX 2	Level of the two send outputs
Phones	Headphones	Headphone level

Output faders run from **FULL** at the top down to **−∞** (silence) at the bottom. FULL means the jack puts out its full level. It does not mean the signal is clipping. Turn the fader down to match the input of whatever you connect: a powered speaker, an interface, a mixing desk, or the FX loop return of an amp.

Linking pairs

The **LINK** button between two strips makes them a stereo pair. When LINK is lit, moving either fader moves both. When you turn LINK on, the right strip jumps to match the left one (Out R matches Out L, and Fx 2 matches Fx 1). When you turn LINK off, both faders stay where they are, and you can then set them separately.

The main outputs are linked when the unit ships. The FX outputs are not, because they are often used as two separate sends.

Headphones

The headphone jack plays the same mix as the main outputs, with its own volume. The **Phones** fader sets the headphone level from **FULL** down to **–∞**, separately from the Out L and Out R faders, so you can monitor on headphones at a comfortable level without changing what goes to your speakers. The Phones meter shows whichever main output is louder.

You can also set the headphone level without opening the Mixer. Click knob **E** to switch it from **Volume** to **Phones**, turn it to set the level, and click it again to switch back.

Master volume

The Mixer's output faders are not the volume knob you use while playing. The master **Volume** (knob **E** by default, see Controls) is a separate control. It turns the main outputs and the headphones up or down before they reach these faders. It does not move the faders on this screen and does not change the FX sends.

A good way to set this up:

1. Set the master Volume near the top of its range.
2. Use the **Out L / Out R** faders to match Axion to your speakers or interface, so that full master volume is as loud as you will ever need.
3. Use the master Volume for everyday changes in loudness.

The main outputs and headphones also pass through a built-in output limiter. It does nothing at normal levels. It only stops very loud peaks from clipping the outputs.

On Stage

The **ON STAGE** button is in the row below the two panels. Turn it on when you play Axion loud: through the house PA, an amp, or a full range speaker at stage volume.

Your ears do not hear all frequencies evenly, and the balance shifts with volume. This is known as the loudness curve, or the Fletcher-Munson effect. At low volume, bass and treble are harder to hear than the midrange. As the volume goes up they come forward, so a rig you dialed in quietly at home or in the studio can sound boomy and harsh on stage, even though nothing in it changed.

On Stage compensates for this. It applies one EQ curve to the output of every rig that evens out what loud playing exaggerates, so the rig sounds much as it did at low volume.

Tap **ON STAGE** to turn it on. Its light turns green. Tap it again when you go back to quiet playing, and the light goes dark.

A few things to know:

- Your rigs are not changed. On Stage sits on top of them, so you can build and save rigs at a comfortable volume and use the same rigs live.
- Turn it off for practice, recording, and quiet headphone listening. At low volume the correction is not needed, and rigs will sound thin with it on.
- If you dial in a rig at stage volume with On Stage on, it will sound different at low volume with On Stage off. Pick one way of working: build quietly and turn On Stage on for the show.
- On Stage and Cab Bypass are separate switches. Through a power amp and a guitar cabinet at stage volume, you may want both.

Cab Bypass

The **CAB BYPASS** button is in the row next to **ON STAGE**. Turn it on when Axion feeds a power amp and a real guitar cabinet. The real speaker then does the job of the Cabinet device, and you do not want both.

Tap **CAB BYPASS** to turn it on. Its light turns green, and the Cabinet blocks in every rig are bypassed. Tap it again to turn it off. The light goes dark and each Cabinet goes back to the on or off state it has in its rig.

A few things to know:

- Your rigs are not changed. Cab Bypass sits on top of them, so you do not have to bypass or delete the Cabinet in each rig, and the same rigs sound right through headphones or full range speakers when you turn it off.
- Like every Mixer setting, it stays on when you change rigs, scenes, or set lists, and when you turn Axion off.
- It applies to every output at once, headphones included.
- Impulse IR and any other device in the **Cab & IR** category are bypassed the same way.
- In the rig editor, a Cabinet held off by Cab Bypass is dimmed and shows a red mute icon, and its own bypass does not respond until you turn Cab Bypass off. See Cab Bypass on the Cabinet page.

Meters

Every strip has a segmented level meter that shows the signal's peaks:

- **Green** is a normal playing level.
- **Yellow** (from –12 dB below full scale) is a strong, healthy level.
- **Red** (from –6 dB below full scale) is close to the limit.

A thin peak line holds at the loudest recent level for one second, then falls. The **clip light** at the top of each meter turns red when the signal reaches full scale, and stays on for two seconds so you can see it. Tap it to clear it sooner.

Input meters show the signal coming in from the jack. Output meters show the signal your rig sends to that jack.

Tips

- If a rig sounds too fuzzy or too clean compared with how it sounded elsewhere, check the input fader before changing the rig. Input gain affects every amp and drive.
- If the input meter moves but you hear nothing, check the output strip for the jack you are using, then follow Troubleshooting.
- If every rig sounds thin and fizzy through headphones or full range speakers, check that **CAB BYPASS** is off.
- If every rig sounds boomy and harsh once you turn up on stage, turn on **ON STAGE**. If every rig sounds thin at low volume, check that it is off.
- When using the FX sends for an FX Loop, set the Fx fader to suit the input of the external pedal or processor, and leave LINK off if the two sends go to different places.

01

CONCEPTS

How Axion thinks. Read this once.

Concepts

How Axion thinks about sound: signal chain, rigs, set lists, and devices.

This section explains the model behind Axion: how audio flows through the unit, how your sounds are organized, and what a device is. If you read nothing else first, read this; every other section assumes the vocabulary defined here.

The big picture



Audio enters at the instrument input, passes through each device in order, and exits at the outputs.

A **rig** is the whole picture: which devices are in the chain, in what order, with what parameter values. A **set list** is a named, ordered collection of rigs you can step through, typically one per song, gig, or scene.

Each rig can have one or more *chains*, collections of devices that modify audio from an input and deliver the modified audio to one or more outputs. We'll tackle these concepts one at a time below:

Next Topics

- The Signal Chain: how audio moves through the unit
- Devices: the building blocks of a sound
- Rigs and Set Lists: how your setups are saved, organized, and recalled
- Open Platform: third-party VST3 and LV2 support

The Signal Chain

How audio flows through Axion, from input through devices in series to outputs.

A **signal chain** is the ordered path your guitar's audio takes through the unit. Audio enters at the instrument input, passes through each device in turn, and exits at the outputs.



Order matters. Putting an overdrive before a delay is a different sound than putting a delay before an overdrive, exactly as it would be on a pedalboard. The Axion chain is a digital version of the same idea, with the same intuition.

What goes in the chain

Anything that processes audio: amp simulators, drives, cabinet simulators, EQs, compressors, modulation, delays, reverbs, and utility devices like boost or gain trim. A chain can be as short as a single device or as long as your CPU budget allows.

For the full list of what's available, see the Device Reference.

If you set up parallel chains, Axion will automatically determine if those chains can be run on parallel processor cores to increase DSP capacity, and will assign the chains to cores accordingly.

How a chain is built

You add devices to the chain one at a time, then arrange and tune them. Each device exposes a small set of parameters (knobs) you adjust to dial in the sound. The chain plus all device parameters together make up a rig, which is what you save and recall.

Mono vs stereo

Most devices in the chain operate on stereo audio internally, even though your guitar input is mono. Modulation, delays, and reverbs are where stereo width typically appears. The output stage routes audio to whichever physical outputs you have connected.

To receive stereo audio, open the chain's input node and turn **Input Source** to **IN 1+2 (Stereo)** or **IN 3+4 (Stereo)**. The first input feeds the left channel and the second feeds the right. The **IN 1–4 (Mono)** choices each feed one input to both channels. Choose **Custom** to set **Left In** and **Right In** independently. Stereo input does not change how mono devices process audio.

On the output node, **Output Destination** offers **OUT 1+2 (Stereo)** and **FX 1+2 (Stereo)** (outputs 3+4). The first output receives the left channel and the second receives the right. Mono destinations sum both channels into the selected output using the existing routing, without reducing their levels. Choose **Custom** to set **Left Out** and **Right Out** independently. Double-tap the selector's caption to restore the default stereo output on OUT 1+2.

CPU budget

The chain runs on a real-time audio thread with a fixed time budget per audio buffer. In practice this means there is a soft limit on how many heavy devices (amps, convolution cabs, reverbs) you can stack before you risk audio dropouts. Lightweight devices (boost, EQ, simple drives) cost very little. See Troubleshooting for help with audio dropouts.

Parallel chains use multiple cores

The budget above is per core, not per unit. When a rig has parallel chains, Axion analyzes the routing and automatically spreads independent chains across separate processor cores so they run at the same time rather than competing for one. Because each core carries its own share of the work, splitting a heavy rig into parallel chains can raise its total DSP capacity well beyond what a single serial chain could sustain. Chains that depend on each other's output stay on the same core, since they cannot run simultaneously anyway; you do not assign cores by hand.

Buffer size trades latency for capacity

The size of the audio buffer sets how much time the chain has to finish each block: a larger buffer gives the audio thread proportionally more time per callback, so it raises the ceiling on how many heavy devices you can run before dropouts, at the cost of higher latency. A smaller buffer lowers latency but tightens the budget. The buffer size is configurable under **Settings → Audio Engine**; the new size takes effect after the next Axion startup. If a rig you rely on is close to the edge, a larger buffer is often the simplest way to buy back headroom.

Third-party plugins are a common reason to raise the buffer. Axion supports the LV2 and VST3 specifications, but not all plugins are written for hard real-time use: some allocate memory, take locks, or do uneven amounts of work from block to block, which can cause dropouts even when the plugin's average CPU cost looks modest. A larger buffer can help alleviate this by giving these plugins more slack, though it will not always fix a plugin that is poorly behaved. If a rig only misbehaves after you add a particular plugin, test the rig without it, and try a larger buffer when you want to keep it in the chain.

Devices

What a device is, the categories Axion ships, and how parameters work.

A **device** is a single audio processor you can place in the signal chain: an amp, a drive, a cab, a reverb, a compressor, or an EQ. On a pedalboard, each pedal is a device. Inside Axion, each block in the chain is a device.

Every device has:

- A **name** and a **category** (Drive, Modulation, Reverb, etc.)
- A small set of **parameters** (the knobs you turn)
- An internal sample-accurate audio processor that runs in real time



Categories

Axion organizes devices into categories so they are easy to find, such as:

- **Amp**: guitar amp simulators
- **Cabinet / IR**: speaker cabinet simulation, including impulse-response convolution
- **Drive**: overdrive, distortion, fuzz
- **Dynamics**: compressors, limiters, dynamic EQ
- **EQ**: equalizers

- **Modulation**: chorus, phaser, tremolo, flanger
- **Delay**: echoes
- **Reverb**: spatial / ambient
- **Utility**: boost, gain trim, splitters

Browse the Device Reference for the full list of devices that ship with Axion.

You can also add your own categories, by creating folders under Content/Devices on the Axion Network Folder. Devices placed in a top-levle folder will be listed under a category with the same name as the folder.

Parameters

A **parameter** is one adjustable value on a device. For example, the "Drive" knob on an overdrive, the "Mix" knob on a reverb, or the threshold on a compressor.

Parameters come in a few flavors:

- **Float**: a continuous value within a range (e.g. 0.0 to 1.0). Most knobs.
- **Enum**: picks from a fixed list of options (e.g. `Symmetrical` / `Asymmetrical` clip modes)
- **Boolean**: on / off

Every parameter has a default value the device ships with. Resetting a device returns all its parameters to their defaults.

First-party vs third-party

Devices that ship with Axion are **first-party**: built by the Axion team, tuned for the hardware, and guaranteed to work within the real-time budget. You can also load **third-party** devices. Axion supports the VST3 and LV2 specification, therefore many real-time audio processing plugins designed for use in a Windows or Linux DAW can also run on Axion. See Open Platform for how that works and what the trade-offs are.

Adding, reordering, removing

You can add a device to the chain, drag it earlier or later, bypass it without removing it, or delete it entirely. None of these actions are destructive to your saved rigs; they only modify the rig you're currently editing until you save it.

Rigs and Set Lists

How Axion organizes saved setups, including rigs, set lists, and MIDI program changes.



A **rig** is a complete, self-contained setup: everything Axion runs at one time, saved as a single recallable unit. A **set list** is a named, ordered collection of rigs. Together, they are how you organize and recall your work, whether at home, in the studio, or on stage.

Rig

A rig is not a single sound. It can hold as many signal chains as you need, each with its own inputs, devices, and outputs, all running at the same time. A guitar chain, a bass chain, and a vocal chain can all live in one rig. Because a rig can also carry the audio and MIDI that go with those chains, a single rig can cover an entire song, or an entire performance, rather than just one tone.

A rig captures everything Axion needs to reproduce that setup:

- One or more chains of devices, in order, each with its own inputs and outputs
- Every parameter value on every device
- Bypass states
- The routing between inputs, chains, and outputs
- Any audio clips and MIDI stored with the rig

- Any rig-level metadata (name, tags, notes)

When you load a rig, the entire rig swaps over to its saved state at once. There is no period where half the parameters have updated; the swap is atomic and click-free.

Rigs are independent of the hardware they live on. A rig made on one Axion unit will load on another, provided both units have the same first-party devices installed and the same third-party devices available.

Set List

A set list is a named collection of rigs in a specific order. Despite the name, a set list does not have to be an actual performance set. It is just as useful as a way to **group and categorize rigs** into whatever libraries make sense to you. Common uses:

- **One set list per gig**: "Tuesday Bar Show", "Studio Session 04/29"
- **One per band**: "Cover Band", "Originals"
- **One per genre or scene**: "Clean Tones", "High Gain", "Acoustic"
- **As a library or category**: "Indie Rock Rigs", "Favorites", "Work in Progress"

Each set list maps onto a **MIDI bank** (0–127). The rigs inside a set list are addressable by MIDI program change (also 0–127). This means you can step through a set list with any standard MIDI controller (a foot controller, a DAW, an iPad), and Axion will load the right rig for the right song.

```
MIDI Bank Select  →  Selects the set list
MIDI Program Change → Selects the rig within that set list
```

Bank 0 is reserved for factory rigs, the rigs that ship with Axion. Your own set lists start at bank 1.

Factory vs user rigs

- **Factory rigs** ship with Axion and live in bank 0. They are starting points (overdriven blues lead, sparkly clean, ambient pad, etc.) and are not editable in place. You can load one, modify it, and save it as a new user rig.
- **User rigs** are the ones you create. They live in user set lists (banks 1–127), can be edited freely, and travel with your data.

How rigs are stored

Rigs and set lists are stored on the unit as plain, human-readable **JSON** files, the same open, text-based format used across the web and countless applications. Axion does not use a proprietary or binary rig format. A rig file simply lists its chains, devices, parameter values, and metadata as text.

That choice has real, practical benefits:

- **Readable and editable.** You can open a rig in any text editor to inspect exactly what it contains, or make careful edits by hand. That's handy for tweaks, batch changes, or version control.
- **Easy to share.** A rig is just a file. Send one to another player over email or chat and they can drop it onto their own Axion. Because the format is text, rigs also diff and merge cleanly in tools like Git.
- **Portable across units.** JSON is not tied to a specific hardware revision, so moving rigs to a new or different Axion is straightforward, provided both units have the same devices available.
- **Durable and future-proof.** An open text format stays legible for the long term, and is simple to back up, archive, or convert with your own scripts and tools.

You do not need to touch the JSON to use Axion, because the interface manages it for you. But it is there, in the open, whenever you want it. You can copy rig files on and off the unit through the Axion Network Folder.

Open Platform

Axion is an **open platform**. Rather than a sealed appliance, it is a system you are meant to be able to open up, extend, and make your own, from the sounds it produces to the interface you touch. There is no proprietary lock-in: Axion is built on open, standard formats throughout.

In practice, "open platform" means three things.

An open, modifiable interface

The Axion touchscreen interface is a standard Javascript application, built with **Vue**, and it ships with an **SDK** for talking to the audio engine, setting parameters, LED colors, and interacting with Axion's hardware. Because it is ordinary Javascript code, anyone can modify it with industry standard tools such as a code editor or IDE, an AI coding tool such as **Claude Code**, or even emacs if, like us, you've got a point to make. ;)

That opens the door to a wide range of customization:

- **Add functionality** to the interface, such as new controls, views, or workflows.
- **Change how existing things work** to suit how you play.
- **Make Axion your own** by restyling, rearranging, or simplifying it.
- **Build entirely new modules and apps** that run on the unit alongside the built-in experience.

The SDK exposes the same control plane the built-in UI uses (reading and setting parameters, loading presets, streaming meters), so anything the stock interface can do, your own code can do too.

Standard audio plugin formats

For audio processing, Axion supports two open, widely used plugin specifications: **LV2** and **VST3**. If you can write (or obtain) a signal processing plugin in either format, built for Axion's platform, it can run in a rig alongside the devices supplied with Axion. This is the same technology used by desktop recording software, so a large world of existing plugins and plugin-development tooling applies directly.

Today the two formats cover two different kinds of plugin. **VST3** support is for **Windows** VST3 plugins, the same builds you would install in a Windows DAW, which Axion runs through a built-in compatibility layer. **LV2** support is for **native Linux** LV2 plugins built for Axion's processor. In both cases the plugin runs isolated in its own sandbox, kept separate from the main audio engine, so a poorly written or crashing plugin cannot bring Axion down with it.

Bundle custom UIs and presets with a plugin

Custom user interfaces for many audio plugins are designed for a desktop computer with a keyboard and mouse, not necessarily a touch screen on a stage-ready appliance meant for real-time processing.

To bridge this gap, Axion automatically generates a screen for modifying plugin parameters on stage. The Axion desktop software can be used to load the plugin's user interface to make detailed changes and create presets ahead of time.

However, a plugin does not have to appear as a generic list of sliders. Any plugin can be wrapped in a "device bundle" (a folder) to include its own **custom interface** and its own **presets**, exactly the way the first-party devices do:

- **Presets** are plain **JSON** files, the same format described in Rigs and Set Lists. Each file lists a device's parameter values plus metadata such as a name, author, tags, and a description. You can find working examples on the unit under `/opt/axion/Factory/Presets`, organized by device ID.
- **Custom interfaces** are **Vue single-file components** (`.vue` files) bundled with the device. Instead of controls derived automatically from the plugin's metadata, you can design a purpose-built panel (knobs, graphics, meters, whatever fits the device) that loads on the unit when the device is opened. The factory devices are laid out this way, with their UI components and presets packaged alongside the plugin; you can use the same structure for your own.

This means an LV2 or VST3 plugin can be delivered as a complete, polished device, with a bespoke interface and sensible presets, not just a raw processor.

Add your own plugins

Drag a Linux LV2 bundle or a Windows VST3 plugin into the **Devices** folder on the **Content** network share. Both formats install the same way. Axion discovers supported plugins and creates their controls and a Default preset.

The first folder inside **Devices** becomes the browser category. You can use names such as **Delay**, **Favorites**, or a plugin vendor's name. Plugins directly inside **Devices** appear under **Plugins**.

See Third-Party Plugins for connection instructions, folder examples, deletion behavior, and restoring missing plugins.

Compatibility and performance

A Linux plugin build must match Axion's processor and use host features that Axion supports. Working in a desktop recording application does not by itself guarantee that a plugin will load or meet Axion's audio deadline. Severed is working on a cloud service for browsing, downloading, and compiling LV2s for Axion, currently scheduled for release in Q1 2027. Until then, devices can be easily compiled using Docker on any desktop PC.

Devices supplied with Axion are tested on its hardware and designed for its controls. External plugins use controls derived from their metadata and can have different processing costs, latency, and resource requirements.

Use the supplied devices as a foundation, then add external plugins for sounds or features you want to explore. If a rig develops dropouts after adding a plugin, Axion will temporarily disable the offending plugin to allow your performance to continue. See Troubleshooting.

When sharing a rig that uses external plugins, the receiving Axion also needs compatible copies of those plugins. Check the vendor's license before sharing plugin files.

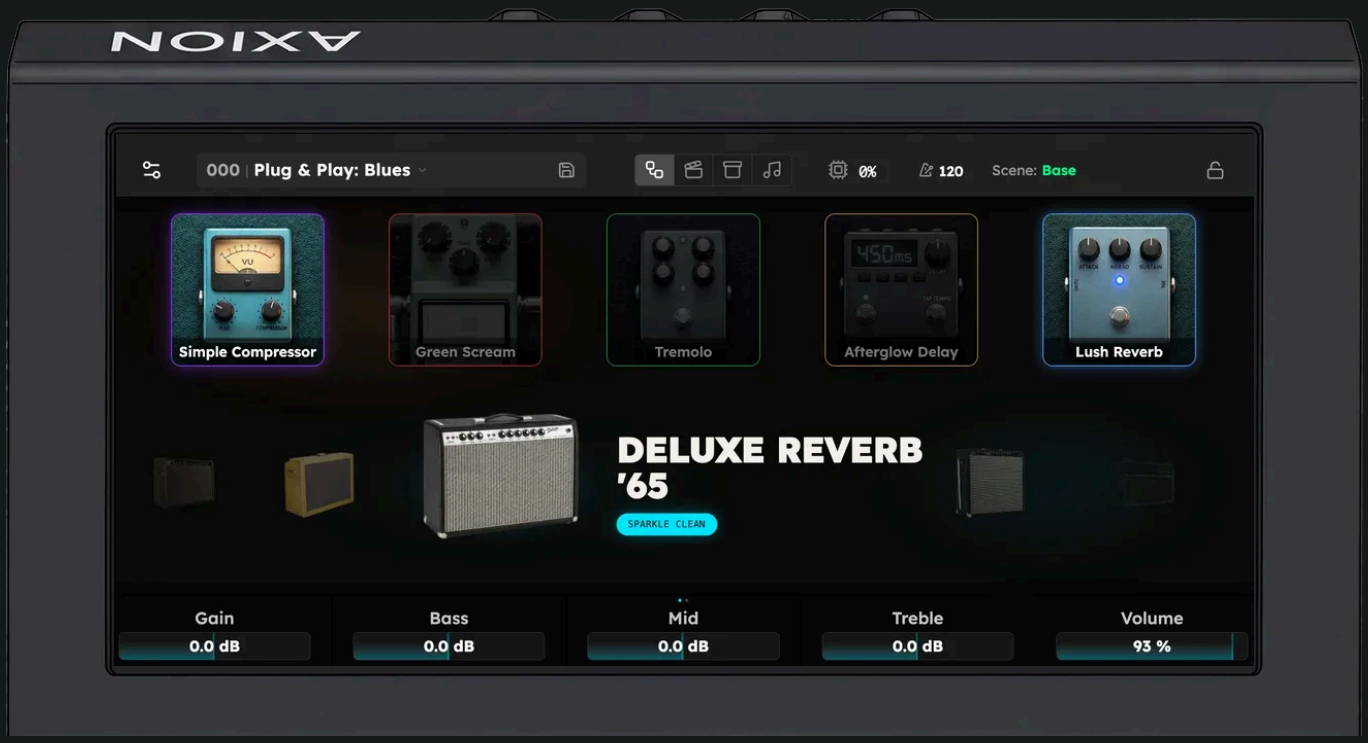
02

PLUG & PLAY

Pick a style and play. The fast path to sound.

Plug & Play

Pick a ready-made style and play. The fastest path from silence to a great tone.



Plug & Play is the fastest way to start playing guitar through Axion. Instead of building a rig device by device, you pick a ready-made style (Blues, Metal, Country, and so on) and start playing. The amp and pedals are already chosen and dialed in by hand. Tweak nothing, or tweak everything: the same style is a great tone the instant it loads and a full editing surface if you want to dig in.

Plug & Play is a good place to start on day one, and a fast way to find a starting point any time.

> **User interface is being finalized.** The Plug & Play stage described below is functional, but the exact way you open it and some on-screen labels may change before the production release. The behavior (pick a style, play, tweak) is stable.

Opening a Plug & Play style

Open the rig browser. Pinned at the very top of the list is a collection named **Plug and Play Rigs**. Each entry is a complete, professionally voiced style:

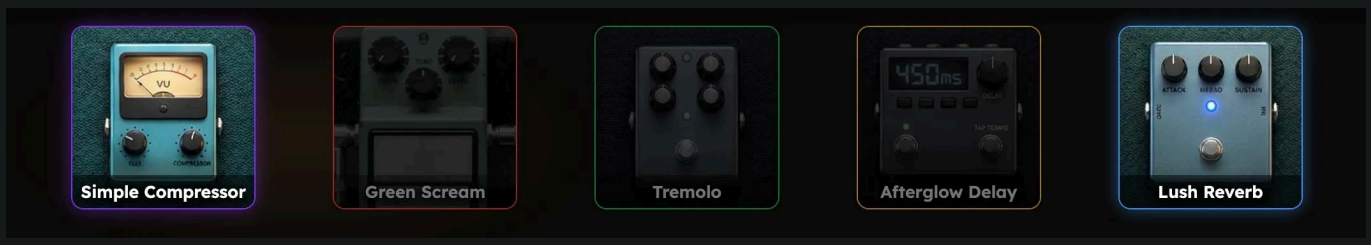
STYLE	CHARACTER
Blues	Warm breakup with just enough dirt.
Classic Rock	Cranked British crunch and sustain.
Metal	Tight, scooped, modern high gain.
Country	Sparkle-clean twang and chicken pick.
Funk	Snappy clean with a squeezed compressor.
Indie	Jangle, chime, and washy ambience.
Worship	Lush, ambient, pedal-stacked cleans.
Jazz	Round, dark, glassy clean tone.

Tap a style to load it. The screen switches to the **Plug & Play stage** and you can play immediately.

The stage

The stage is laid out in three zones, top to bottom:

Pedalboard (top)

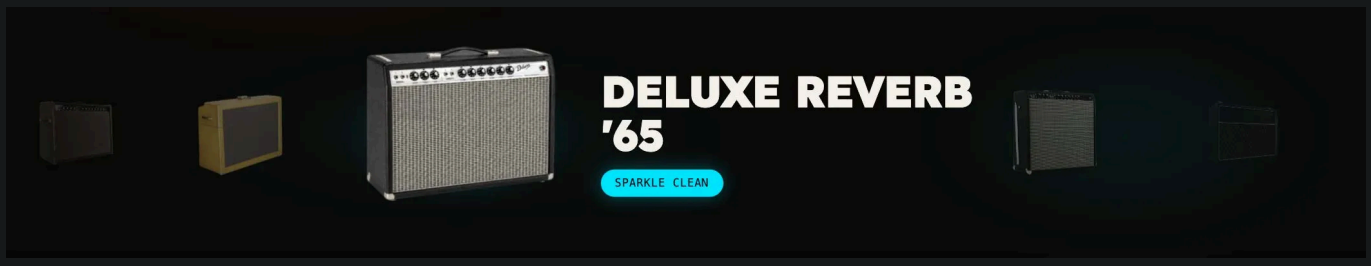


Up to five pedals sit across the top as tiles, in signal order.

- **Tap a pedal** to open its full controls, exactly as you would in the editor.
- **Double-tap a pedal** to bypass it (turn it off) or switch it back on. A bypassed pedal dims.
- The colored glow around each pedal matches its footswitch light, so you can tell at a glance which physical switch controls it.

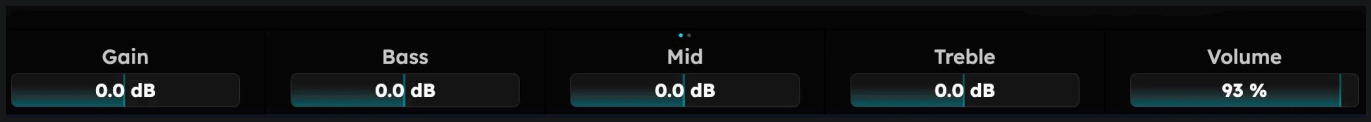
Any empty slots show an **Add** button. Tap it to drop another pedal into the chain.

Amp (middle)



The center of the stage is the amp. Swipe left or right to move through the available amp voicings; the one in the center is the one you hear. **Tap the centered amp** to open its full controls.

Parameters (bottom)



The first page keeps the familiar **Gain, Bass, Mid, Treble,** and **Volume** controls lined up directly above the five physical encoders under the screen. Swipe the dock left to reach a second page of useful controls from the pedals in the style, such as **Drive**, **Boost**, and reverb or delay **Mix**. The exact second page follows the pedals that are present.

- Gain and the three tone controls read on a familiar amp-style **0–10** scale.
- Volume reads as a percentage and shows **MUTE** at the bottom of its travel.
- Turning a physical encoder always controls the five assignments on the visible page.

Footswitches

The bottom row of five footswitches maps to the five pedals on the board: **press a footswitch to toggle its pedal on or off**, the hands-free version of double-tapping the tile. Each footswitch's light matches its pedal's color and dims when that pedal is bypassed. Footswitches under empty slots stay dark.

The performance controls you rely on elsewhere (the tuner and other press-and-hold functions) remain available while you're in Plug & Play.

Making it your own

Nothing about Plug & Play is locked. From the stage you can:

- **Swap the amp** by swiping the carousel to a different voicing.
- **Add a pedal** with the Add slot, or open any pedal and change its settings.

- **Bypass** anything you don't want, from the screen or a footswitch.
- **Dial the amp** with the five knobs or the hardware encoders.

If you open a device, you get its complete parameter set. Plug & Play is a friendlier front door to the same engine, not a walled-off "lite" mode. When you land on something you love, save it as your own rig from the top bar so you can recall it later. See Building Rigs for how saving and organizing rigs works.

Leaving Plug & Play

Load any other rig, from a set list or anywhere else in the rig browser, and Axion returns to the full experience. The Plug and Play Rigs collection stays pinned at the top of the browser, so a ready-to-play style is always one tap away.

> **You can't break the styles.** The Plug and Play Rigs are a fixed, factory collection. Reopening a style always brings it back exactly as it shipped, so experiment freely. The original is never overwritten.

03

VIEW MODES

Scenes, stomp, and lyrics.

Modes at a Glance

Axion has several main "view modes", each tailored to a specific part of performing, editing, or generally making noise in an organized fashion.

You can switch modes two ways:

From the top toolbar:



Click any icon to switch directly to that mode.

From the foot switch Press and hold "Mode" (Switch 1)** to cycle between view modes without taking your hands off the instrument.

Every mode presents a different view of the same rig, scenes, assignments, and song data. You are not loading a different version of your setup. You are simply changing how Axion presents it and what the foot controls and display are focused on.

The number and sequence of modes can be configured under Settings > Preferences.

Rig Mode

Rig Mode is where you build, inspect, and edit your signal chain.

Add devices, change their order, adjust parameters, reroute signals, monitor levels throughout, and dismantle perfectly respectable tones until they become a source of inspiration. This is the main workspace for designing and editing rigs, and understanding exactly what your audio is doing.

Scene Mode

Scene Mode is used to select, create, and edit scenes.

Scenes are snapshots of your rig that recall device settings, bypass states, routing, and other assigned changes. Use them to move between song sections, create alternate versions of a rig, or make dramatic changes without performing emergency touchscreen surgery between chords.

Stomp Mode

Stomp Mode turns the footswitches into user-assignable performance controls.

Despite the name, stomps are not limited to turning effects on and off. A stomp can load a scene, open the tuner, change a parameter, trigger an action, or perform whatever other task you assign to it. They are programmable foot controls, not tiny dictators demanding that every switch behave like a traditional pedal. You decide what each one does in Footswitch Assignment.

Song Mode

Song Mode displays lyrics and chord charts while you perform.

Use it to keep song information visible without balancing a phone on an amplifier, taping paper to the floor, or pretending you definitely remember the second verse. Songs can also be connected to rigs and scenes, keeping the things you need for a performance in one place. Charts and lyrics can optionally automatically scroll in sync with the song's tempo, so you never lose your place.

The four views are different windows into the same setup. Build in Rig View, organize changes in Scene View, control the performance in Stomp View, and remember the words in Song View. Axion handles the structure. What you do to the sound is your problem.

Once a view is set the way you want it, Screen Lock freezes the touchscreen and knobs while leaving the footswitches live, so a stray cable or a neighbour's boot can't rearrange your tone mid-song.

Rig Mode

The loaded rig as a signal chain: read it, open a device, bypass it, move it.



Rig Mode draws the loaded rig as a signal chain: an **IN** block on the left, your devices in the order they process, and an **OUT** block on the right, with the signal line running between them. A chain fits six devices per row and wraps to the next row when it needs more. If the rig has more than one chain, each chain gets its own block, stacked down the screen.

This is the mode you sit in while working on a sound, and the one that shows you what is actually happening to your signal.

Reading the chain

Each device is a tile carrying its own artwork, with a coloured border for its category: drive, reverb, dynamics, and so on. A lit tile is active. A dimmed tile is bypassed, and the signal simply passes through it.

Two things will interrupt to tell you something is wrong:

- A **flame** on a tile means that device's output has peaked above –1 dBFS and is on the edge of clipping. Tap the flame to open the **Auto Level** utility. The flame stays for ten seconds after levels come back down, so a single hot note does not vanish before you notice it.

- A **yellow warning triangle** means a problem has occurred with that device and it is **bypassed**. Tap the device to view more information about the problem.

Touch Controls

- **Tap** a device to open its own full-screen page, where all of its parameters can be modified.
- **Double-tap** a device to bypass or un-bypass it without opening it.
- **Tap and hold** a device to pick it up, then drag. Drop it on another device to swap the two, on any insert point to move it to that position, or on **Remove Device** at the bottom of the screen to take it out of the rig.
- **Tap a + dot** between two blocks to open the device browser and add a device at exactly that point in the chain.
- **Tap IN or OUT** to open the chain's input or output block. These are devices too, with their own parameters and their own power button, which is why bypassing one silences the chain.
- **Tap a +** on the left to add a new chain. Multiple chains run in parallel, and can use separate cores for increased DSP capacity and throughput.

Along the bottom of the screen, the five knobs show whatever this rig has bound to them; see Rig Controls for how that binding is set.

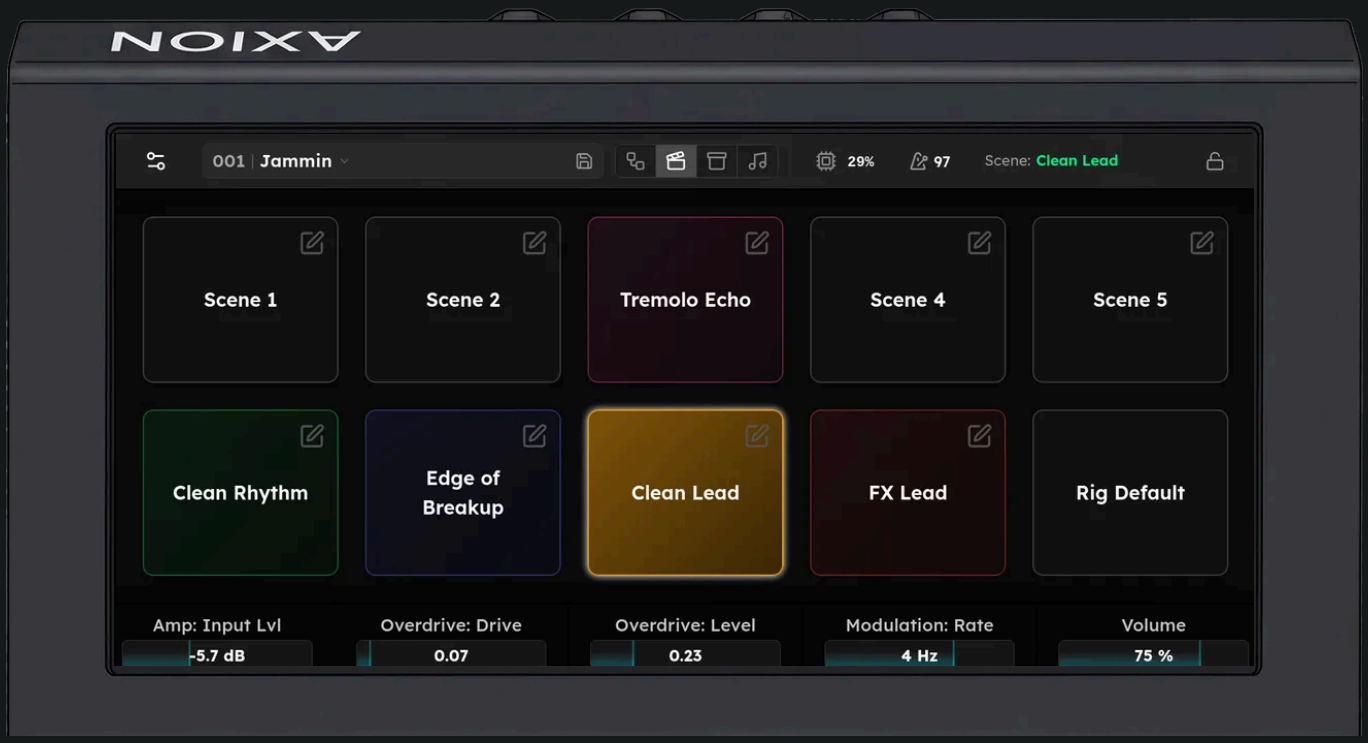
Rig Mode is the view. The work of assembling a rig (choosing devices, adding chains, saving the result) is covered in Building Rigs.

Plug & Play rigs

A Plug & Play rig is a special type of built-in rig that has no chains to edit. For these, Rig Mode shows the Plug & Play stage instead: the amp and its handful of curated knobs. If you wish to edit the routing or add more complex chains, save the Plug and Play rig to a new rig first, then reload it.

Scene Mode

Snapshot and switch between setups within a single rig.



A **scene** is a saved setup within one rig. Instead of building separate rigs for your rhythm and lead sounds, you keep a single rig and flip between scenes for instant changes without any load times or audio drop-outs. Each rig can store up to nine scenes, plus the **Base Scene**, which is the default settings for all devices in the rig. Think of it as the "foundation" that other scenes build upon.

What a scene captures

A scene stores any number of parameter **overrides**. When a scene is recalled, the changes are instantly applied: the "Base Scene" settings all all first applied, then your selected scene's **overrides** are applied over those. This allows you to fully control which parameters and settings a scene will modify, and when.

The only things a scene does not capture are settings that cannot change in real time, such as a device's loaded model file (for example a NAM capture). Those stay the same across every scene in the rig, because those settings would cause a drop in audio, which scenes are designed to avoid.

The scene grid

The Scenes mode overrides all ten footswitches, and shows a tile for each scene and a Rig Default tile. Tap a tile or its corresponding foot switch to switch to that scene instantly; the active scene is highlighted.

Naming and colour

Tap a scene's edit button to give it a name and a colour. The colour carries through to the footswitch lights and the Stomp view, so a glance tells you which scene you are on.

Switching hands-free

Assign **Change Scene** actions to footswitches (see Footswitch Assignment) to step between scenes with your feet. While the Scenes view is open, the footswitches map directly to the scenes on screen.

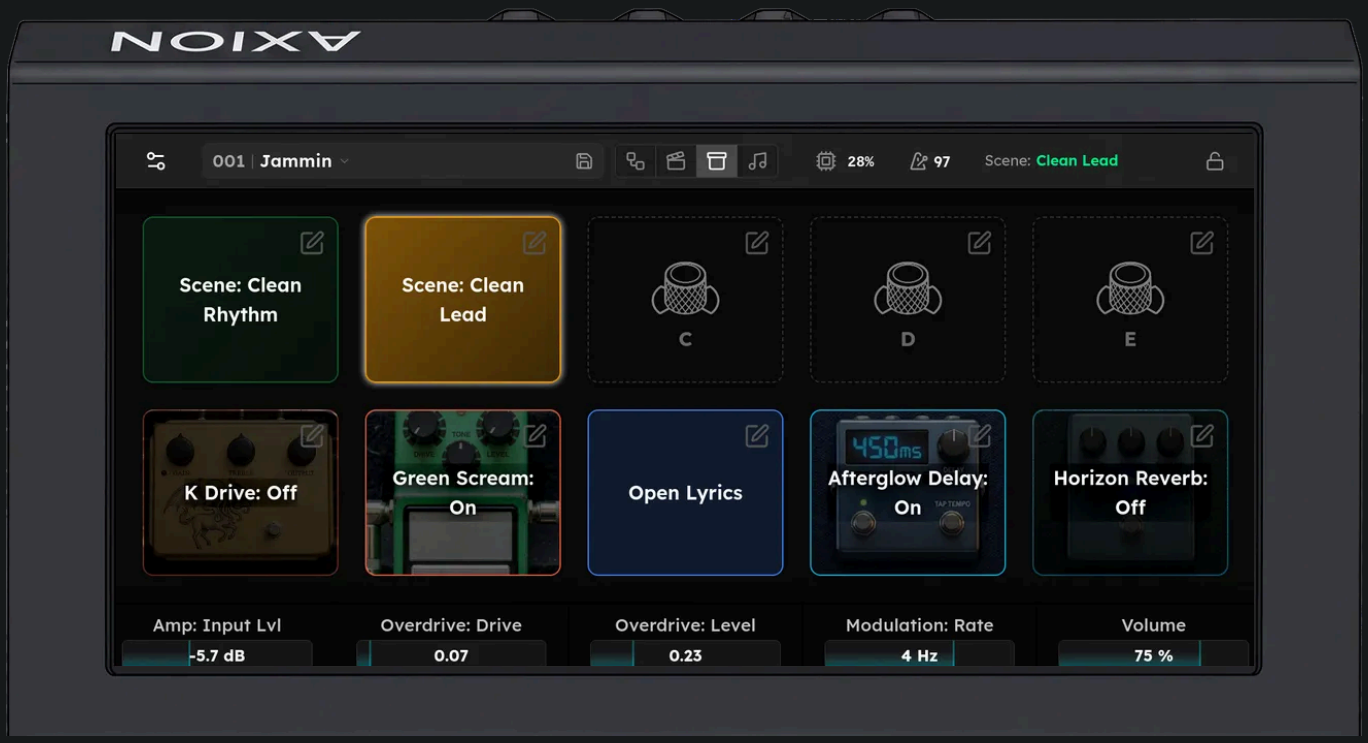
Rig Default

Rig Default is the base the scenes are built on. Returning to it drops whatever scene you were in.

Scenes are **saved with the rig**.

Stomp Mode

A pedalboard-style grid of your assigned switches.



The **Stomp** view is a pedalboard-style grid of all ten switches, the five knobs and the five footswitches, laid out the way they sit on the unit. Each tile shows what its switch is assigned to and its current state:

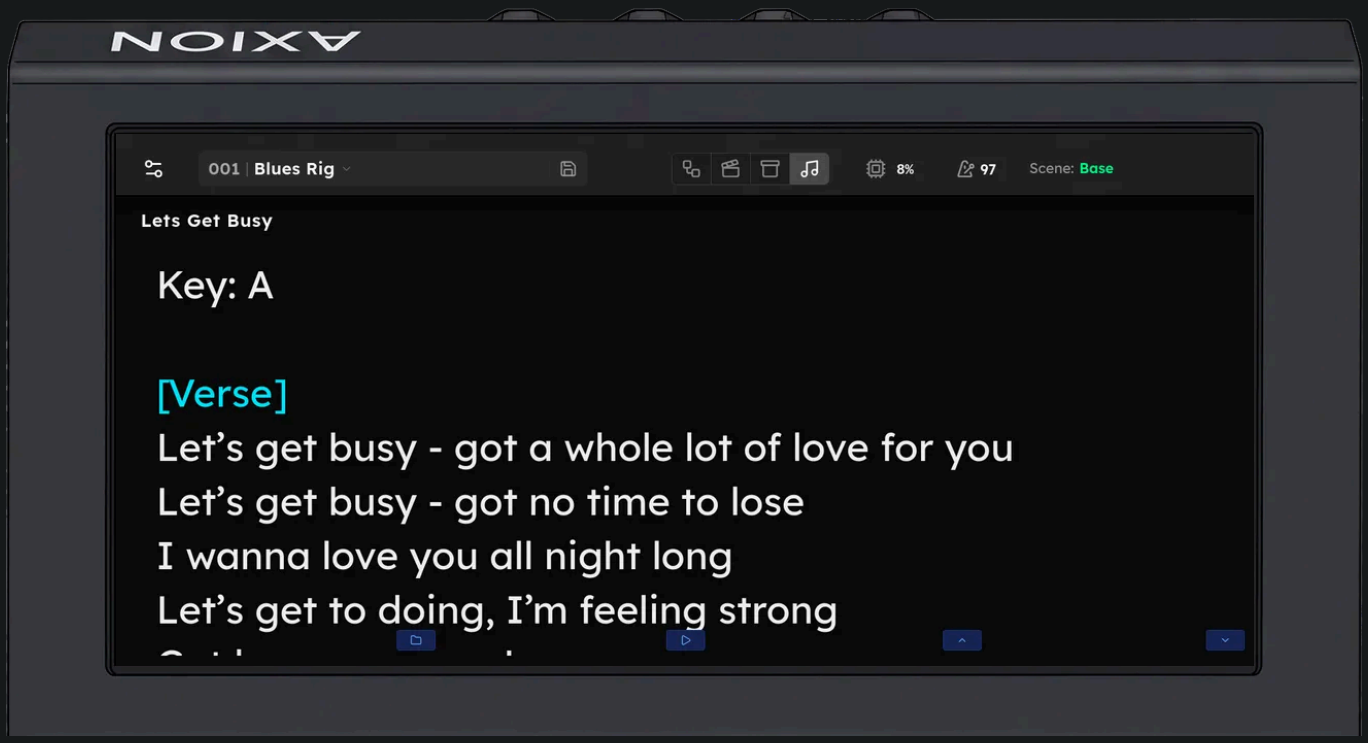
- A **bypass toggle** shows the device name and On or Off, lit when the effect is on and dimmed when it is bypassed, over the device's artwork.
- A **scene change** shows the scene name in its colour, highlighted when that scene is active.
- An **unassigned** switch shows a dashed placeholder tile.

Tap a tile to fire it, exactly as pressing the switch would: toggle the effect, change the scene, and so on. Tap a tile's edit button, or tap an unassigned tile, to jump straight to that switch in Footswitch Assignment.

The Stomp view is a live picture of your switch assignments; it stores nothing of its own. Change what the switches do in Footswitch Assignment.

Song Mode

Scrolling lyrics and charts for the current song, synced to tempo.



The **Lyrics** view shows a large, scrolling lyric or chord sheet for the current song, sized to be easily read while performing on stage. This screen can be set to scroll automatically in time with the song's tempo to avoid the need for manual scrolling.

Loading a song

Press and hold footswitch 2, or tap **Load Song / Lyrics** (or the song name at the top) to browse the Songs folder and pick a text file. Lines wrapped in `[brackets]` are treated as section headings and coloured. The chosen song is **saved with the rig**, so loading the rig brings its lyrics back.

Scrolling

- **Autoscroll**: the play/pause control scrolls the sheet automatically, paced to the rig's tempo. It pauses while you touch the screen and resumes shortly after.
- **Manual**: the up and down controls nudge the sheet, or scroll continuously while held. Press and hold footswitch 4 or 5 to scroll up or down.

Hands-free

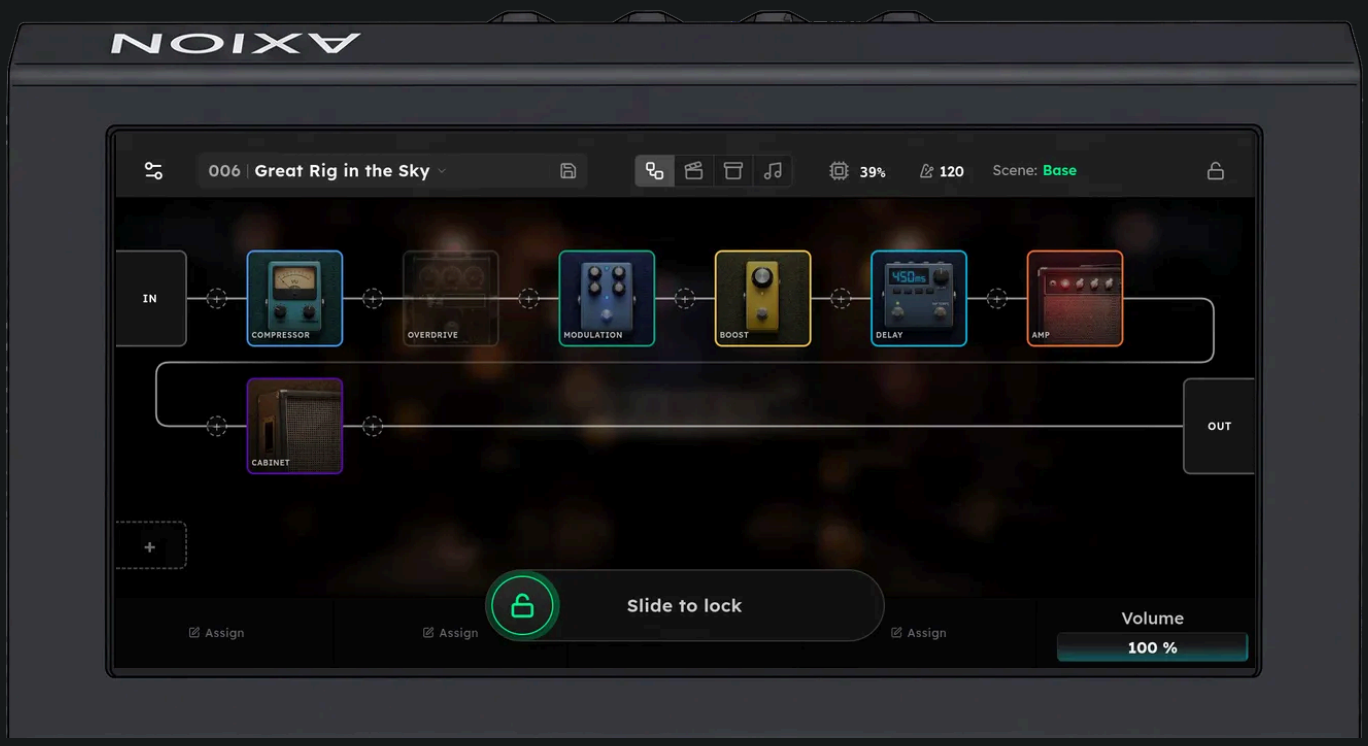
While Lyrics is the active view, the **hold** functions on footswitches 2 to 5 are taken over for lyrics control. They **override** those switches' normal hold functions for as long as you are on this screen:

- **Hold 2**: Open the song browser
- **Hold 3**: Start/pause autoscrolling
- **Hold 4**: Scroll up
- **Hold 5**: Scroll down

Footswitch 1's hold (change view mode) is not changed, so you can still switch away. Leaving the Lyrics view restores every switch's normal hold functions.

Screen Lock

Lock the touchscreen and knobs so a stray cable, a bumped panel, or someone else's boot can't change your tone mid-song. Footswitches keep working.



Stages are hostile places for a touchscreen. Cables swing, people lean, and a neighbouring musician's enthusiastic footwork does not always respect your carefully dialled-in rig. **Screen Lock** freezes the front panel so none of that can touch your sound, while leaving the footswitches fully live, because you still have a show to play.

Screen Lock is a performance safeguard, so it only applies to the four main view modes: Rig, Scene, Stomp, and Song. Step into the menu, the tuner, settings, or any other screen and the lock steps aside. Return to a main view and it re-engages exactly as you left it.

What it blocks

When the screen is locked:

- **The touchscreen is inert.** Taps, drags, and presses do nothing.
- **The knobs are inert.** Turning an encoder changes nothing.
- **The footswitches keep working.** Scenes, stomps, tuner, and rig changes all behave exactly as they normally would. Locking the screen never costs you a switch.

In other words, the two things a flailing cable or a wayward boot can knock, the screen and the knobs, are exactly the two things that stop responding.

Locking the screen

1. Tap the **padlock icon** in the top-right corner.
2. A **Slide to lock** control appears along the bottom of the screen.
3. Slide it all the way to the right.

The control confirms with a red **Locked**, then disappears. The padlock in the corner turns into a red **Locked** badge so there is never any doubt about which state you are in. If you tap the icon and then think better of it, just wait. The slider dismisses itself after five seconds of inactivity.

Unlocking the screen

1. Tap anywhere on the screen, or turn any knob.
2. A **Slide to unlock** control appears along the bottom.
3. Slide it all the way to the right.

The control confirms with a green **Unlocked** and the corner returns to the plain padlock. The slider is red while the screen is locked and flips to green the moment you unlock, so the colour always tells you where you stand.

A locked screen ignores taps and knobs everywhere *except* that unlock slider. An accidental brush of the screen simply offers you the way out rather than doing anything to your rig.

Screen Lock deliberately does nothing to your footswitch assignments, scenes, or audio. It is purely a guard over the front panel. Lock it at soundcheck, forget it exists, and let the cables misbehave all they like.

04

BUILDING RIGS

Chain devices, save presets, assign footswitches.

Building Rigs

Chain devices, save presets, assign footswitches.



Rig Mode is the workspace for building a complete sound. A rig stores the signal chain, every device setting, bypass states, scenes, and the assignments for the physical controls. Start from a factory rig when you want a known working signal path, or clear and reshape a user rig for a new sound.

- **Footswitch Assignment** sets what each knob-press and footswitch does.
- **Rig Controls** binds the knobs and expression pedal to the parameters you want under your hands.

Both are saved with the rig, so each rig carries its own switch assignments and knob bindings.

Add a device

In Rig Mode, the signal flows from **IN** on the left through each device to **OUT** on the right. Tap a **+** dot at the point where you want a new device. The device browser opens with category and tag filters, search, and available presets. Tap a device or one of its presets to insert it at the selected position.

To add your own plugins to the browser, see Third-Party Plugins. It covers copying LV2 and VST3 plugins onto Axion, organizing browser categories with folders, and removing plugins.

Order has an audible effect. Drives usually sit before the amp, a cabinet usually follows the amp, and delay and reverb commonly sit near the end. These are starting conventions, not restrictions. See The Signal Chain for examples and the reason order matters.



Edit a device

Tap a device tile to open its full screen controls. The labels above knobs A through E show which parameters are currently assigned to the hardware row. Turn a knob or use the touchscreen control. Additional parameters appear on the device screen when a device has more than five.

Use the power control on the device screen to bypass it, or double tap its tile in Rig Mode for a quick comparison. A dim tile is bypassed. If a flame appears, that device has produced a peak above minus 1 dBFS. Tap the flame to open Auto Level and correct the gain through the chain.

Save a device preset

A device preset stores that device's parameter values without storing the rest of the rig. On the device screen, tap the save icon beside the preset name, enter a name, and confirm. Factory presets remain unchanged; edited versions are saved as user presets and appear in the device browser.

Move or remove a device

Touch and hold a tile for about a third of a second, then drag it.

- Drop it on another device to swap their positions.
- Drop it on a **+** dot to move it between devices.
- Drop it on **Remove Device** at the bottom of the screen to delete it from the rig.

Removing a device also removes its sound and any control assignment that targets that instance. Save a device preset first if you want to reuse its settings elsewhere.

Save the rig

Tap the save icon beside the rig name. In the save dialog:

1. Choose a user set list in the left column, or select **New Setlist**.
2. Choose the program slot in the right column.
3. Tap **Save** on that slot. To rename the rig, tap the pencil beside its name instead, enter the name, and tap **Save** in the name dialog.

An occupied user slot is replaced when you save to it. Factory rigs cannot be overwritten, so editing one and saving creates a user copy. Saving captures the base device settings as well as scenes and hardware assignments.

Recall a rig

Tap the current rig name to open **Select Rig**, choose a set list, and tap the desired rig. Loading replaces the complete active signal path in one operation. Unsaved edits to the previous rig are not carried forward, so save before loading another rig when you want to keep the changes.

On stage, use assigned next and previous rig actions, the built in preset hold actions, or MIDI bank and program messages. See Rigs and Set Lists for addressing and organization.

Export and share

The Axion desktop companion can connect to a unit on the same network, browse its set lists, and export a rig as a JSON file. Select the set list, find the rig, and choose **Export** on its row. Keep exported files as backups or send them to another Axion owner.

The current companion app exports rigs but does not expose an import action. Keep the exported JSON unchanged so it remains suitable for restore and import support that accepts Axion rig files.

Stacks

Group devices into one block, or run several chains in parallel, with a Stack.



A **chain** is a row of blocks in series. A **Stack** is a single block in a chain that holds a stack of chains of its own.

- **One chain:** the Stack works as a composite block. For example, Noise Gate → Compressor → NAM can sit in the main chain as one block named “Great Marshall Stack”.
- **Several chains:** each chain receives the same input and runs in parallel. Their outputs are summed and returned at the Stack block.

Add a Stack

Tap a **+** dot in a chain and choose **Stack** from the Utility category of the device browser. It is added immediately and named **Stack 1**, **Stack 2**, and so on. The Stack’s first chain is created with it.

The Stack’s chains appear in the rig editor below the main chain. Add devices to them as you would to any chain.

The inputs and outputs of a chain inside a Stack show **Stack in** and **Stack out**. They have no jack selection; they take their audio from, and return it to, the Stack block.

Example: Parallel NAM (amp) models

1. Add a Stack where you want your two amps to go. Optionally, open the Stack to rename it “Amps”.
2. Click “+” to add a NAM preset to the chain in the stack.
3. Click “+” on the left to add a new parallel chain. Set “Stack: Amps” as the source. Choose a second NAM preset.

The main chain will now show one block, and the “Amps” Stack’s chains will show two NAM devices running in parallel.



When a Stack holds several chains, each one receives the Stack’s input and the chains’ outputs are summed at the Stack block.

Parallel chains are summed without automatic level normalization: two equally loud chains are louder than one. Processing latency is aligned automatically between the chains and the dry path; differences in the phase of the amp models themselves are not corrected.

Bypass and Mix

Tap the Stack block to open its device page.

- **Bypass** passes the original signal around the Stack.
- **Mix** blends around the whole round trip. **0%** is the original signal; **100%** is the Stack’s combined return. The default is 100%.

Save and reuse a Stack

A Stack can be saved as a preset and added to any rig.

- **Save:** open the Stack's device page and tap **Save**. Name the preset and choose its category. The preset holds every chain in the Stack, the devices in them with their settings, and any Stacks nested inside.
- **Add to a rig:** tap a **+** dot and pick the saved Stack in the device browser. It is listed under the category it was saved in, and the new block takes the preset's name.
- **Load into an existing Stack:** open the Stack's device page and tap the folder button. Loading replaces the Stack's chains and everything in them, including nested Stacks.

A loaded Stack is a copy. Editing it does not change the preset, and saving the preset again does not change rigs that already use it.

Rig Controls, footswitch assignments and scenes belong to the rig, not to the Stack, so a Stack preset does not carry them.

If a saved Stack uses a plugin or model that is not installed on this Axion, the Stack still loads and that block is shown as unavailable.

Nesting and deleting

A Stack chain can contain another Stack. A Stack cannot contain itself.

Deleting a Stack block deletes its chains and everything in them, including any Stacks nested inside.

External effects with the FX Loop

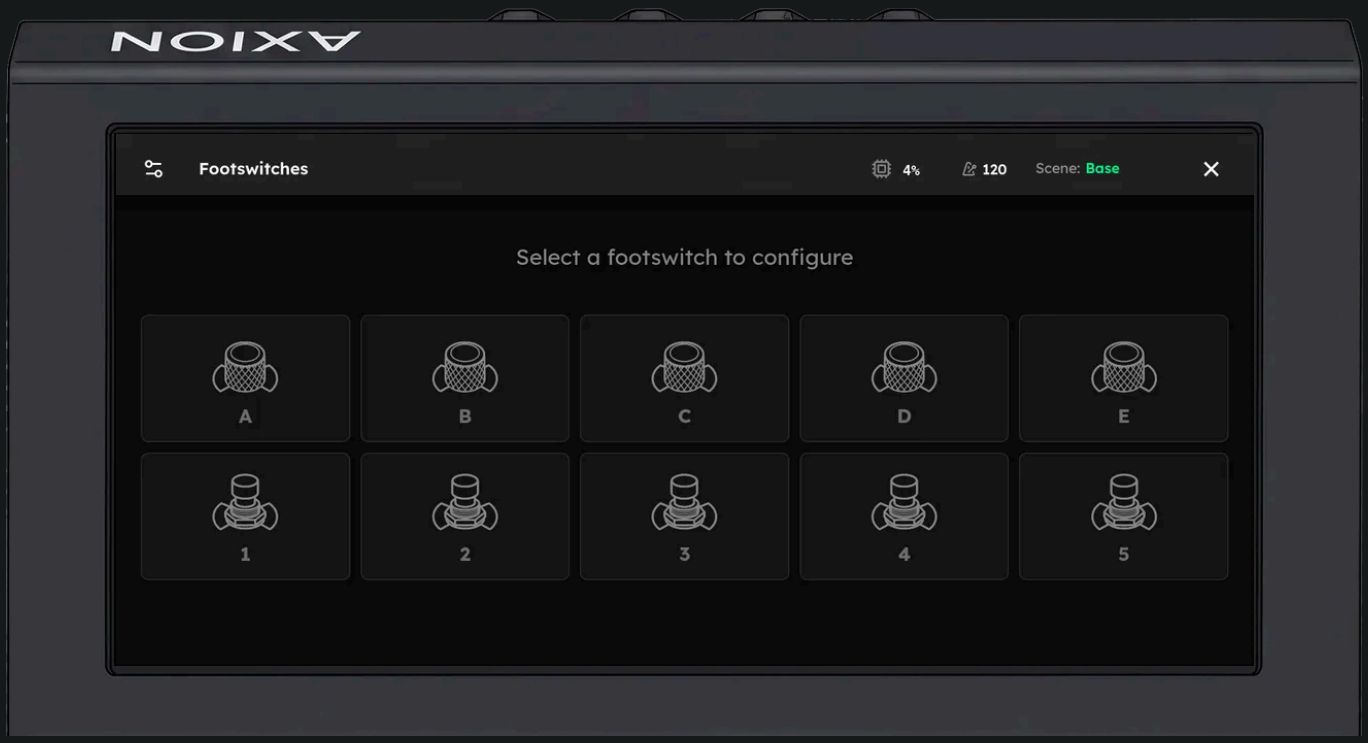
The physical **FX Loop** is an ordinary block. Configure its Send and Return jacks and levels on its device page.

- **Serial loop:** put the FX Loop in the main chain where the pedals belong.
- **Parallel with other chains:** put the FX Loop in a Stack chain. For example, chain 1 runs your pedalboard through the FX Loop and chain 2 holds a NAM amp.

The FX Loop's own Mix stays at 100%. The physical round trip through your pedals is not yet latency-compensated. If an FX Loop chain runs in parallel with a chain carrying a similar signal, the combination can sound phasey or comb-filtered. Parallel chains with clearly different sounds are less affected.

Footswitch Assignment

Assign an action to every knob-press and footswitch, saved per rig.



Every knob and footswitch can trigger an action, and you decide what. Assignments are **saved with the rig**, so the same physical switch can do different things from one rig to the next. The Controls page covers the physical knobs and switches; this is where you decide what they do.

Opening the screen

From the main menu, choose "Switches" to open the **Footswitches** screen. It shows all ten switches in two rows: the top row is the five knobs, **A-E**, and the bottom row is the five footswitches, **1-5**. A small badge on a switch shows how many actions it already has.

Pick a switch to configure by tapping it on screen, or by pressing the physical switch.

Assigning a new device quickly

Right after you add an effect to the signal chain, every switch that is free to take a press action pulses in the accent color, and the new device's tile shows a matching **Assign** badge. Press any pulsing switch and the device's bypass toggle lands on it: the badge disappears, and that switch's LED and the device's tile flash three

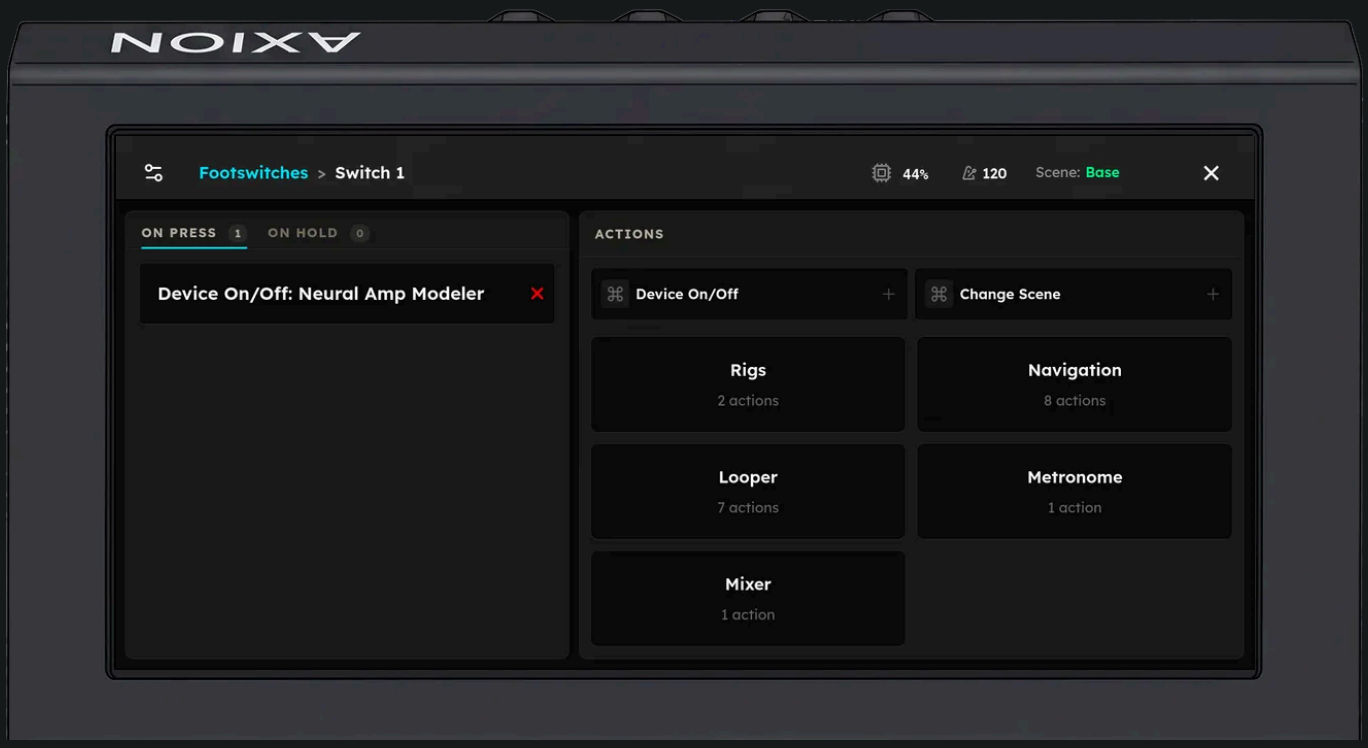
times in the device's color before settling.

Tap anywhere on screen, press another switch, or just wait a few seconds and the invitation quietly goes away with nothing assigned. Tapping the device itself opens it as usual. Amps and cabinets are not offered. Turn off **Auto-Assign New Devices** under **Settings → Preferences** if you would rather assign everything by hand.

Press and hold

Each switch has two separate action lists: **On Press** and **On Hold**. A quick tap runs the On Press list; holding the switch runs On Hold. Add actions to whichever you want.

Choosing an action



Actions come from a browser grouped by category:

- **Signal Chain**: toggle a device's bypass, the classic stomp
- **Scenes**: change to a scene
- **Rigs**: step or select rigs
- **Navigation**: switch views and open screens
- **Looper**
- **Metronome**: tap tempo, and so on

The two most common, **Toggle Device Bypass** and **Change Scene**, are pinned at the top. Some actions take a parameter, such as which device to bypass or which scene to select; you set it as you add the action.

Stacking actions

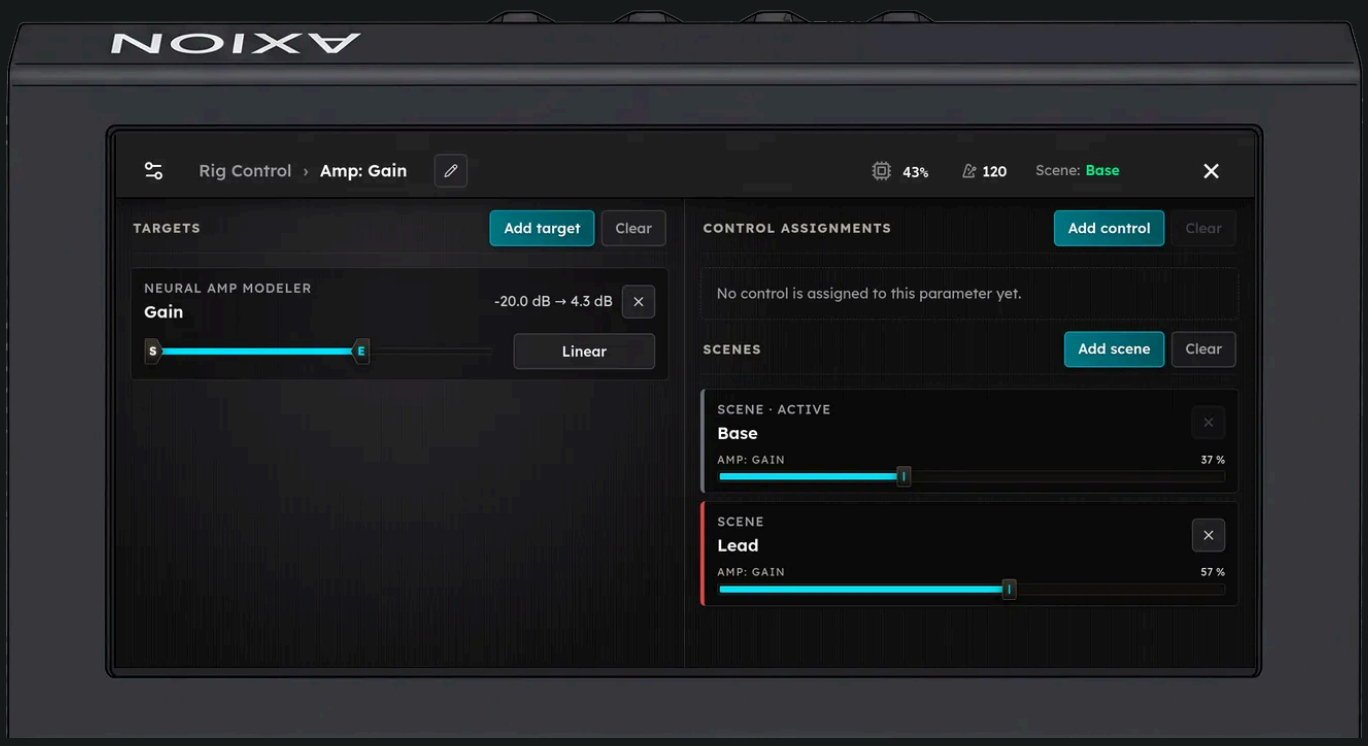
A switch can carry several actions that fire in order. Drag to reorder them. If one action stops the sequence, any actions below it are marked **unreachable** so you can see they will not run.

What is not assigned here

The hold utilities printed on the panel (Tuner, view mode, and preset up and down) are built-in and always available, unless you re-assign them. To see your assignments as a pedalboard, and to tap them from the screen, use the Stomp view.

Rig Controls

Bind the knobs, expression pedal, and MIDI to the parameters you want under your hands and feet, with per-target ranges and curves.



A **Rig Control** enables you to assign one or more device parameters to a single "macro" control, which appears on the main rig screen. Bind a knob to the amp's gain, and that knob is gain whenever this rig is loaded. The dock holds five knobs per page and adds more pages as you assign more than five controls, reachable by swiping the dock left or right. Rig Controls, along with the expression pedal and MIDI assignments alongside them, are **saved with the rig**.

Creating a Rig Control from the Dock

On the parameter dock, tap any empty slot to add a new rig control. Choose a **device** from the current rig's chain, or **System** for the main output, then pick the parameter you want. The picker lists the device on the left and its parameters on the right. The knob now turns that parameter, labelled with the device and parameter, for example "Amp: Gain".

The screen for a single knob is titled **Rig Control**. The section that lists what a control drives is **Control Assignments**.

To change a slot later, open it and tap **Change**. To empty it, use **Clear** or **Delete Rig Control**.

Creating a Rig Control from the Parameter Screen

Any parameter can be turned into a rig control. From the Parameter Editor, choose **Add control** > **Rig Control** to add the parameter to the rig controls and the main dock.

Macros: more than one target

A Rig Control can adjust multiple parameters at once, for seamless blending between sounds. On a control's screen, tap **Add target** to add a second parameter, then a third, and so on. Every target has its own range and curve: tap the sliders icon on a target to set where the knob's sweep starts and ends and how it gets there. By default a target covers its whole range. A control with one target reads that parameter's own value; with more than one target it becomes a **macro**, the knob itself reads 0 to 100 %, and each target follows its own range and curve as the knob sweeps.

This is how you build a single performance knob that for example, opens up a reverb's mix while backing off a delay's feedback at the same time.

Ranges and curves

Every target has a **Start** and an **End** value, so you decide how far along the parameter the control travels and in which direction. Set Start above End for an inverted sweep, or set them equal to pin the parameter to a fixed value.

Each target also has a response **Curve** that shapes how knob or pedal travel maps to the parameter:

- **Linear**: even travel across the range.
- **Logarithmic**: finer control low, faster up top, useful for a smoother volume swell.
- **Exponential**: the opposite weighting.

The System output

Choose **System** in the target picker to reach the unit's main **Volume**, the overall output level. Knob **E** is bound to it out of the box; see Controls.

Control sources: the expression pedal and MIDI

Beyond the knobs, a parameter can be driven by the expression pedal or by a MIDI control change. On a parameter's Control Assignments, tap **Add control** and choose a tile:

- **Rig Control**: an encoder dock slot, as above.
- **Exp. Pedal**: a pedal on the **EXP** jack, sweeping the parameter from heel to toe. Its range endpoints are labelled **Heel** and **Toe** rather than Start and End.
- **MIDI**: a control change message. Tap **MIDI Learn** and move a knob or pedal on your controller to capture its port, channel, and CC, or enter the **CH** and **CC** by hand and tap **Use**.

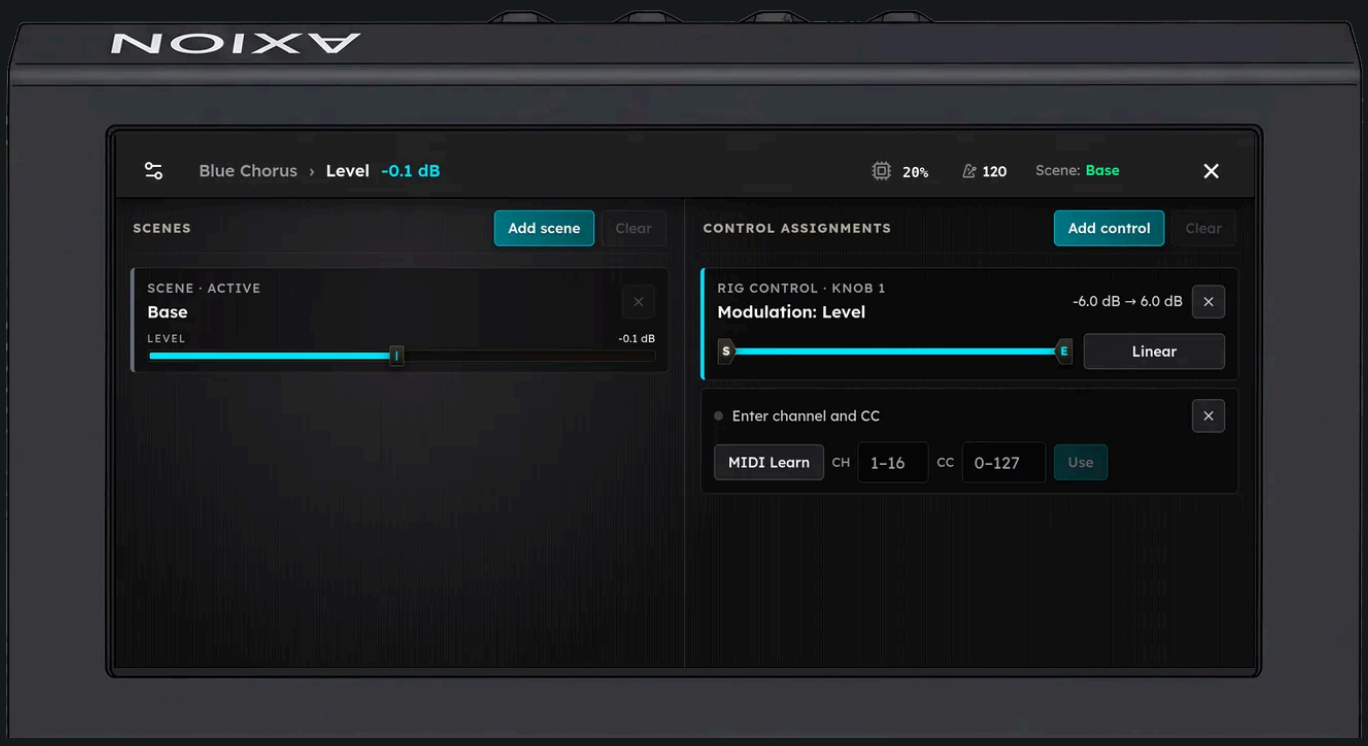
One source can drive several parameters. Each mapping it holds keeps its own range and curve, so one pedal or one CC can move a handful of parameters differently at the same time. See Connections for the EXP jack.

Control Mappings

For the whole picture in one place, open **Control Mappings** from the main menu. It lists every Rig Control, every **Expression** mapping, and every **MIDI** mapping the rig carries, so you can review and edit them without hunting across device screens. Tapping a knob or dock slot on the rig screen jumps you to that one control; Control Mappings shows them all together.

Parameter Editor

One screen per parameter that shows every scene that changes it and every knob, pedal, and MIDI control that drives it, with per-control ranges and curves.



The **Parameter Editor** is the one place to see everything that can move a single parameter. Open it on the amp's Gain, for example, and you can see which scenes change the gain, which knob, pedal, or MIDI control is wired to it, and how far each one sweeps it.

Opening the Parameter Editor

Tap any parameter to open its editor:

- **A knob on a device screen.** Tap the knob itself rather than dragging it.
- **A dock slot while a device is open.** Tap an assigned slot on the parameter dock.
- **A target on a Rig Control, pedal, or MIDI screen.** Tap the parameter's name in the list of what that control drives.

The top bar shows the device, the parameter, and its current value, for example **Blue Chorus › Level -0.1 dB**. Tap the **×** to close the editor and return to the screen you came from, even if you hopped between several parameters and controls on the way.

Small dots under a knob tell you there is something to see before you open it. A dot in the accent colour means a control drives the parameter. A light grey dot means at least one scene stores its own value for it.

The Scenes column

The left column lists every scene that stores a value for this parameter, in scene order. Each row carries the scene's name and colour, and the scene you are currently on is marked **Active**. **Base** always appears first, because the Base Scene holds the starting value for every parameter.

Each row has a fader for that scene's value. You don't need to switch scenes to use it. Drag the fader on "Lead" and you have set the Lead value, while you keep playing whatever scene you are on. Editing the active scene's row changes the sound immediately, exactly as turning the knob would.

- **Add scene** opens the scene grid. Pick a scene and it stores the parameter's current value as its own override. Then fine-tune it with that row's fader.
- **×** on a row removes that scene's override, so the scene goes back to using the Base value. Base cannot be removed.
- **Clear** removes every scene override for this parameter and leaves only Base.

Scene values are part of the rig. **Save the rig** to keep them, just as you would after turning a knob.

The System **Volume** has no Scenes column entries. Scenes store device parameters only.

The Control Assignments column

The right column lists every control that drives this parameter. Each row shows what kind of control it is and which one, for example **Rig Control · Knob 1**, **Expression**, or **MIDI · CH 1 · CC 11**. The row also shows the range the control sweeps, such as **-6.0 dB → 6.0 dB**.

Adding a control

Tap **Add control** and choose one of three tiles:

- **Rig Control** puts the parameter on the next free slot of the rig's knob dock, so a knob turns it whenever this rig is loaded. See Rig Controls.
- **Exp. Pedal** lets a pedal on the **EXP** jack sweep the parameter from heel to toe. A parameter can take only one expression mapping, so the tile reads **Already assigned** once it has one.
- **MIDI** adds an **Enter channel and CC** row for a MIDI control change message. Tap **MIDI Learn** and move a knob or pedal on your controller to capture its port, channel, and CC. You can also enter the **CH** and **CC** by hand and tap **Use**. A CC that already drives this parameter is refused, so pick another.

Parameters with a number value can be driven by all three. A switch or a selector, such as a mode with a fixed list of choices, can be driven too. The control's travel is divided into equal slices, one per choice, and the parameter steps through them as you sweep. Only a parameter without a number behind it, such as a loaded model file, cannot be assigned. The **Exp. Pedal** and **MIDI** tiles show **Not a numeric value** for those.

Ranges and curves

Every row has its own range slider and curve button, so the same parameter can respond differently to each control. For example, the pedal can sweep the whole range while a knob covers only the useful middle.

- **Range.** Drag the **S** (start) and **E** (end) handles to set where the sweep starts and ends. Put the start above the end for an inverted sweep, so that pressing the toe down turns the parameter down. Put both handles in the same place to pin the parameter to a fixed value.
- **Curve.** Tap the curve button to cycle between **Linear**, **Logarithmic**, and **Exponential**. Linear spreads the change evenly across the travel. Logarithmic gives finer control at the bottom and moves faster near the top, which suits volume swells. Exponential is the opposite.

Changes to ranges and curves are stored in the rig straight away. There's nothing extra to save.

Following a control to its other end

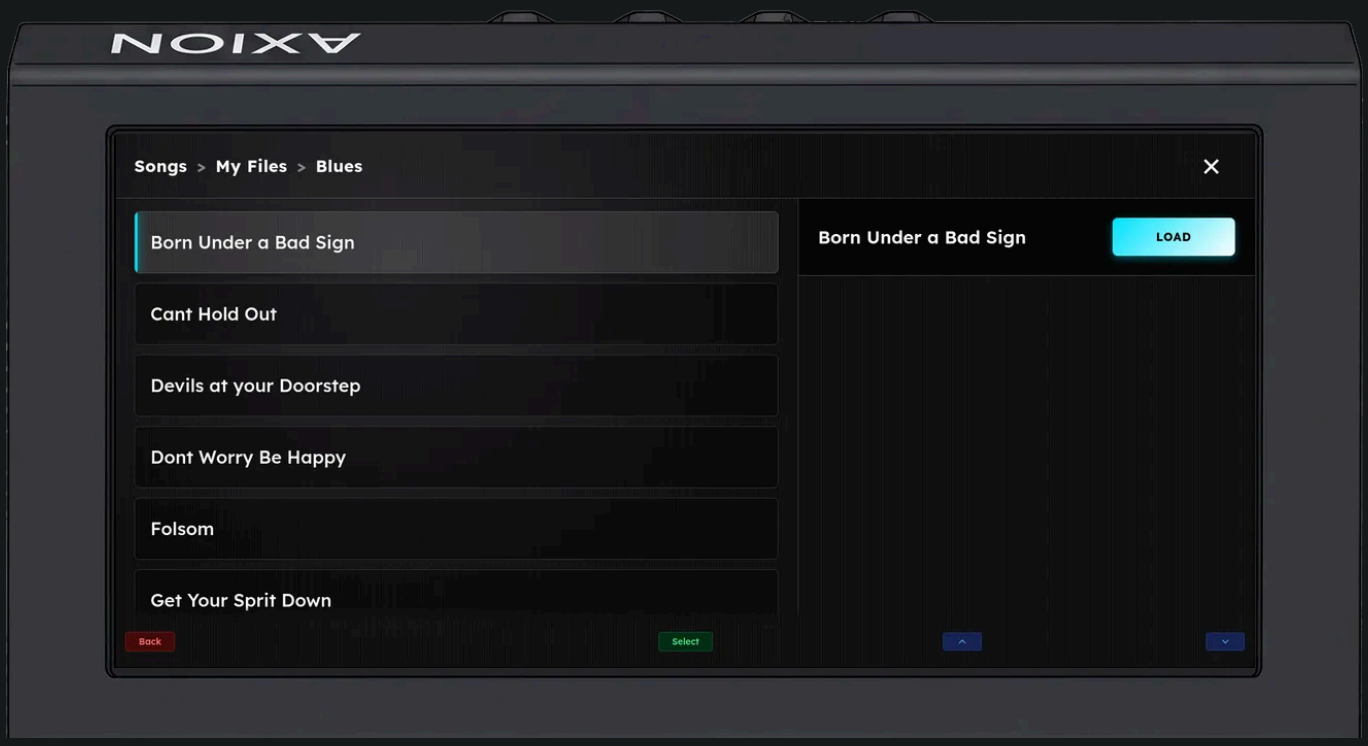
Tap a row's name to open the control it belongs to. A Rig Control row opens that knob's Rig Control screen, where you can see anything else the knob drives. A pedal or MIDI row opens its entry in **Control Mappings**. From there, tapping a parameter brings you back to its editor.

Removing controls

Tap **×** on a row to disconnect that control from this parameter. Other parameters the control drives are unaffected. **Clear** disconnects every expression pedal and MIDI control from this parameter at once. Rig Control rows stay put, so remove those with their own **×**.

Focus Mode

Navigate lists and browser screens entirely from the footswitches, with color-coded labels along the bottom of the screen.



Some screens can be driven entirely from the footswitches, so you can browse and load without reaching for the touchscreen. When one of these screens is open, a row of color-coded labels appears along the bottom of the display. Each label sits directly above the footswitch it belongs to, and the footswitch itself lights in the same color, so you can find the right switch without looking down.

HOLD to activate focus mode functions!

To avoid conflicting with your normal stomps and footswitch assignments, focus mode actions require you to **hold the switch** to activate them. Be aware that single-tap on footswitches will still activate their assigned stomps, even on browser and list screens!

Where Focus Mode works

Focus Mode appears on the screens where you pick something from a list:

- **Select Rig**, where you choose a set list and load a rig.

- **The file browser**, where you move through folders and load a file such as an impulse response, a capture, or a lyrics file.

Any screen that opens with the label row along the bottom is fully foot navigable. Screens that do not show the row are touchscreen only.

The labels and their colors

Each label is color coded by what it does, and its footswitch LED matches that color:

- **Red** is **Cancel** or **Back**. In the file browser it steps up one folder, and at the top level it becomes Cancel and closes the screen. In the rig browser it closes the screen.
- **Green** is **Select**. It acts on the highlighted item: it opens the folder you are on, previews or loads a file, or loads the highlighted rig.
- **Blue** is **Up** and **Down**, shown as arrow labels. These move the highlight through the list.

A footswitch with no job on the current screen stays dark and shows no label.

Moving through a list

Use the blue **Up** and **Down** switches to move the highlight one item at a time. The list scrolls to keep the highlighted item in view, so a long set list or a full folder stays readable as you go. Hold either switch to repeat and travel quickly through a long list.

Selecting and going back

Press the green **Select** switch to act on the highlighted item.

- On a folder, Select opens it and the browser shows what is inside.
- On a file, the first Select previews it and a second Select on the same file loads it.
- On a rig, Select loads it and closes the browser.

Press the red switch to go the other way. In the file browser it steps back to the folder above the one you are in. When there is nowhere further up to go, that same switch cancels and closes the screen. In the rig browser the red switch closes the screen without loading anything.

With these four actions, move, select, and back, you can open a browser, walk into the folder you want, and load a file or a rig from across the room, hands free.

Third-Party Plugins

Copy LV2 and VST3 plugins onto Axion, organize them into browser categories, and remove or restore them.

You can add plugins by dragging them onto Axion over your network. Axion supports two plugin formats, and installs both the same way:

- **LV2** plugins built for Linux on Axion's processor.
- **VST3** plugins built for Windows, the same builds you would install in a Windows DAW.

Axion discovers plugins automatically and adds them to the device browser. No separate installer or device manifest is needed.

Before you start

Connect Axion and your computer to the same network, and get the right build of the plugin:

- An **LV2 bundle** is a folder that usually ends in `.lv2`. It contains the plugin binary, metadata, and sometimes additional resources. You need a Linux build for Axion's processor.
- A **VST3 plugin** ends in `.vst3`. It is either a folder or a single file. You need the Windows build. On a Windows computer, installed VST3 plugins are usually in `C:\Program Files\Common Files\VST3`.

macOS builds of plugins do not currently work on Axion. If your download is an archive, extract it first. Keep the complete `.lv2` or `.vst3` together rather than copying only part of it.

Windows Installers

If your plugin comes with an .exe or .msi installer, you will need to install it through the Axion desktop companion app. Instructions on how to install VST3 plugins this way are coming soon!

Copy your plugin onto Axion

1. Connect the Axion Network Folder from your computer.
2. Open the **Devices** folder inside **Content**. Create it if it does not already exist.
3. Drag the complete `.lv2` or `.vst3` into **Devices**, or into a sub-folder beneath **Devices** (see below).

4. Wait for the transfer to finish. Allow a few more seconds for Axion to automatically scan the plugin and import it into the device browser. A VST3 takes longer the first time Axion sees it.
5. In Rig Mode, tap a **+** insertion point, find the plugin in its category, and select it to add it to your rig.

Axion creates controls and a **Default** preset from the plugin's metadata. The controls may look different from the plugin's desktop interface. You do not need to restart Axion after each upload.

Use the **Content** share for your plugins. The **Factory** directory is for devices supplied with Axion.

Organize plugins with folders

The first folder directly inside **Devices** supplies the category shown in the device browser. You can use existing categories such as **Delay** and **Reverb**, or create your own, such as **Favorites** or **Spacey Effects**.

LOCATION INSIDE THE CONTENT SHARE	BROWSER CATEGORY
`Devices/regrader.lv2`	Plugins
`Devices/Delay/regrader.lv2`	Delay
`Devices/Favorites/regrader.lv2`	Favorites
`Devices/Native Instruments/Effects/Raum.vst3`	Native Instruments
`Devices/Bogus/Folder/Structure/regrader.lv2`	Bogus

Plugins can sit at any depth beneath **Devices**, and a folder can contain multiple plugins of either format. Only the first folder sets the category. The folders below it are for your own organization.

Custom categories appear automatically when Axion discovers plugins in them. An empty folder does not create an empty browser category. Plugins placed directly in **Devices** appear under **Plugins**. Existing category names are recognized regardless of capitalization.

Moving a plugin between folders changes its category without changing its plugin identity or saved rig references. Renaming a category folder updates the category after the next scan. Avoid keeping several copies of the same plugin in different folders, since Axion registers only one copy of a plugin identity.

Some plugins declare their own category, and that category takes precedence over the folder name:

- A VST3 that describes itself as a reverb, delay, modulation, distortion, dynamics, EQ, filter, or pitch-shift effect appears in that category. VST3 plugins that do not describe their kind use the folder name.
- A device with an explicit category in an Axion `manifest.json` uses that category.

Hidden folders and directory shortcuts made with symbolic links are not scanned.

Delete a plugin

Delete the plugin's `.lv2` or `.vst3` from the **Content** share to uninstall it. Once the deletion settles, it disappears from the device browser. A custom category also disappears when it no longer contains any discovered devices.

Deleting a plugin does not immediately stop an instance that is already playing. Axion keeps a temporary copy of a loaded plugin so the active instance can continue using its files. When you reload the rig or make an edit that rebuilds its signal chain, the deleted plugin cannot load and its block is bypassed.

Restarting Axion clears the temporary copies. Axion discovers installed plugins from **Content/Devices** again. You do not need to find or delete the temporary copies yourself.

Saved rigs retain the missing plugin's block, settings, and connections. The rig editor shows **Missing plugin** on its block. Tapping it opens this notice:

> This plugin was removed from the device and is bypassed. Copy the plugin back onto Axion and reload the rig to restore it.

The device view shows the same explanation. Plug & Play adds **Missing plugin** beside the pedal's name. Removing the plugin from the share does not remove the block from your saved rig.

Restore a missing plugin

Copy the same plugin, or a compatible version with the same plugin identity, back into **Devices** or any folder beneath it. Wait for it to appear in the device browser, then reload the rig. Its saved settings and connections are available again. The folder does not need to have its original name.

If a plugin does not appear or load

- Confirm that the upload finished and the complete `.lv2` or `.vst3` was copied.
- Check the build: a Linux LV2 or a Windows VST3, not a macOS build, an archive, or an installer.
- For a VST3 you just copied, allow a little longer. Axion checks each new VST3 before it appears in the browser.
- Keep an LV2's metadata and resources inside the bundle. Symbolic links inside a bundle are not supported by the runtime loader.
- A bare LV2 bundle must describe exactly one plugin. Bundles containing several plugins need an Axion device manifest that selects one of them. A VST3 that contains several effects adds each one to the browser.

- Some plugins require host features that Axion does not support. A rejected plugin does not become an available device.

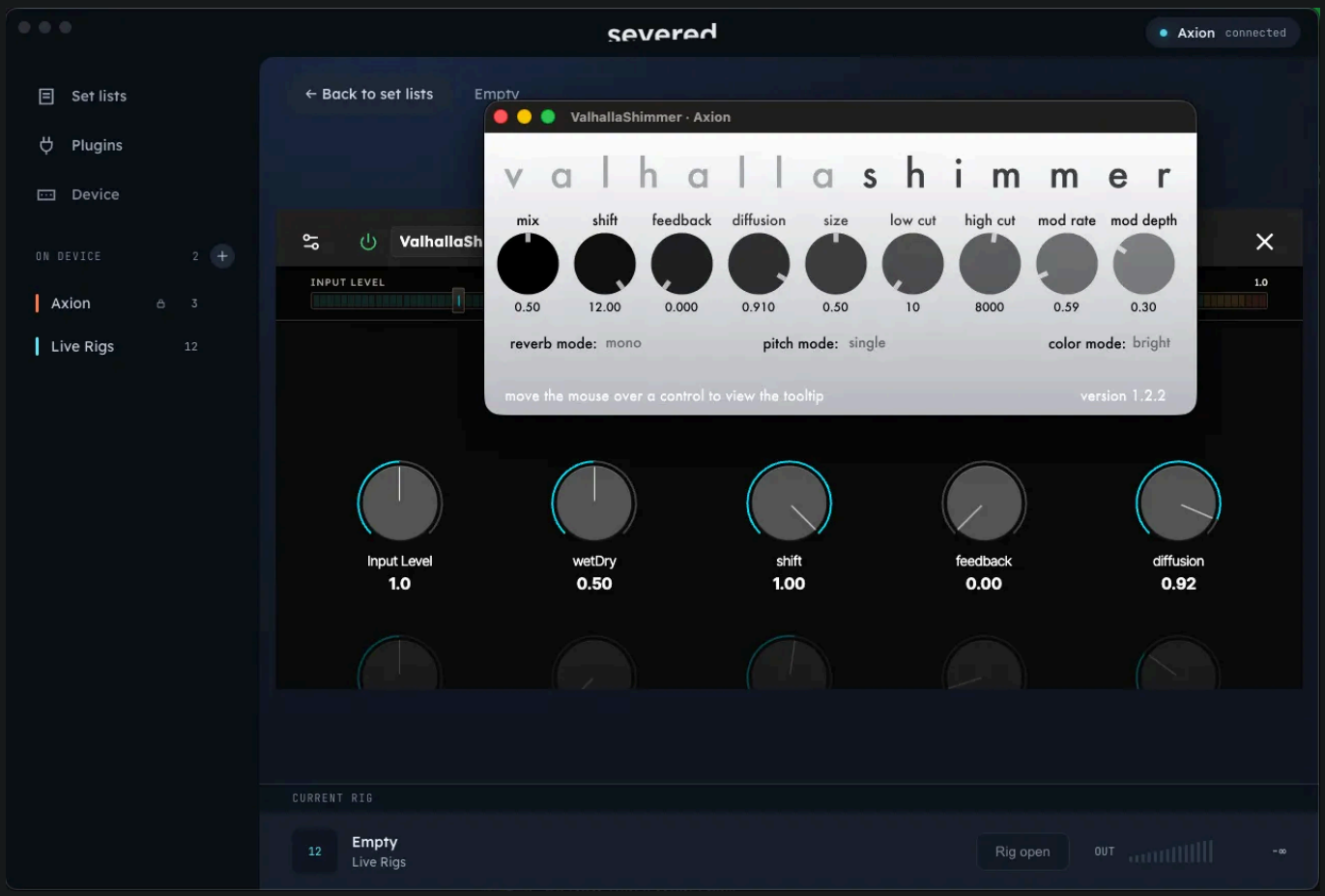
For audio dropouts after adding a plugin, see Troubleshooting. For background on plugin compatibility, see Open Platform.

Plugin Editors

Many plugins come with their own editor, the same interface you would see in a desktop DAW. Axion shows plugin editors on your computer, where you have a mouse and keyboard, rather than on the touchscreen.

1. Open the Axion desktop companion app on your computer.
2. Pair and connect it to Axion.
3. Open the plugin's device page, either on Axion's touchscreen or in the companion app.
4. Tap or click the **Desktop** icon at the top of the device page.

The plugin's editor opens in a new window on the connected computer. Changes you make in the editor take effect immediately, and can be saved in Axion as a preset.



The **Desktop** icon appears only when a computer is connected and the plugin provides its own editor. Plugins without an editor use the controls Axion creates for them. If several computers are connected, an editor opened from the touchscreen appears on the one that connected most recently.

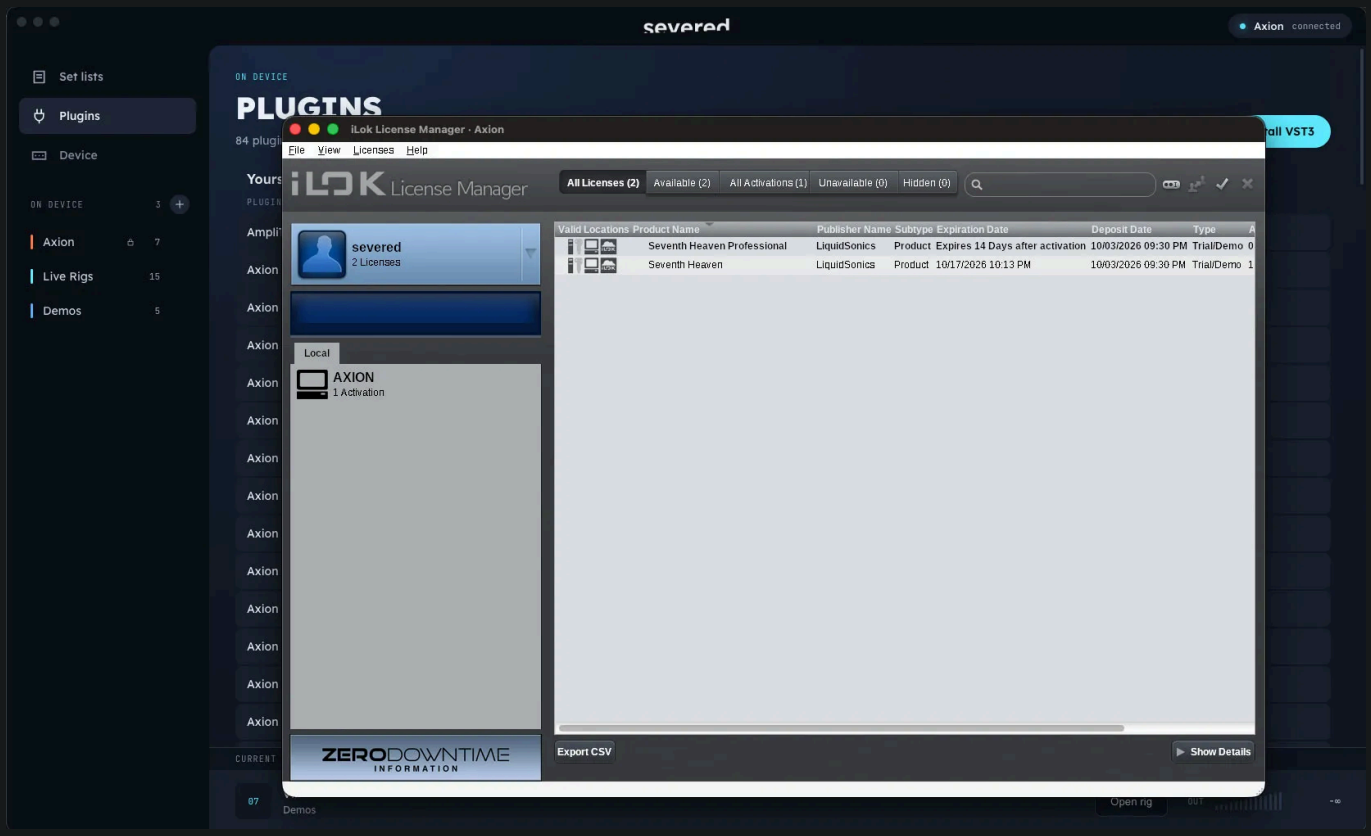
If the icon does not appear, check the connection first. See WiFi or companion app connection problems.

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iLok License Manager

Axion supports iLok License Manager for plugins that are licensed through iLok. iLok License Manager runs on Axion alongside your Windows VST3 plugins, and its window opens on your connected computer through the companion app, the same way plugin editors do.

Sign in with your iLok account to see your licenses. Axion appears under **Local** as a computer named **AXION**. Activate a plugin's license on **AXION**, either by dragging it there from your license list or when the plugin asks you to activate, and the plugin is authorized on Axion.



05

SETTINGS

Global EQ, audio engine, and unit-wide setup.

Settings

Settings that belong to the unit, not to a rig: the Global EQ and the audio engine.

Most of what you change on Axion is saved in a rig. The settings in this chapter are different: they belong to the unit. They apply to every rig, stay the same when you change rigs, scenes, or set lists, and Axion keeps them when you turn it off.

To open the settings, tap the menu button at the top left of the screen and choose **Settings**. Pick a page from the list on the left. Tap ✕ at the top right to go back to your rig.

- Global EQ: one equalizer on the output of every rig, for matching Axion to your speakers, room, or PA.
- Audio Engine: the sample rate and buffer size, which set how much delay there is between your pick and your speaker, and how much processing your rigs can use.

The Mixer is also a unit-wide setting. It has its own page in the Overview chapter.

Global EQ

One eight-band equalizer on the output of every rig. Match Axion to your speakers, your room, or the PA without editing a single rig.



The Global EQ is an equalizer on the output of every rig. It sits after your rig and before the master volume, so whatever rig is loaded, its sound passes through the Global EQ on the way to your speakers.

Use it to fix the speaker, not the rig. Your rigs should sound right on their own. The Global EQ corrects what the playback system adds on top: a boomy room, a bright PA, a powered speaker with a bump in the low mids, or headphones that hide the bass. Set it once for where you are playing, and every rig benefits.

To open it, tap the menu button at the top left of the screen and choose **Global EQ**. You can also open it from **Settings**. Tap ✕ at the top right to go back to your rig.

Like the Mixer, the Global EQ belongs to the unit, not to a rig. It stays the same when you change rigs, scenes, or set lists, and Axion keeps it when you turn it off.

What it affects

- **Main outputs** (OUT L and OUT R): yes.
- **Headphones**: yes. The headphones play the same mix as the main outputs.

- **FX sends** (FX 1 and FX 2): no. An FX Loop sends the signal from its place in your rig, untouched by the Global EQ.

The graph

The graph shows frequency from 20 Hz on the left to 20 kHz on the right, and gain in dB up the side. The line marked **0** is flat: no boost and no cut.

- The **white curve** is the combined effect of every band that is on. Where it is above 0, those frequencies are boosted. Where it is below, they are cut.
- The **colored spectrum** behind it is a live analyzer of Axion's output. Play and watch it to see where the energy in your sound is.
- Each band that is on has a **node** on the curve: **H** for the high-pass filter, **1** to **6** for the six main bands, and **L** for the low-pass filter.

Drag a node to move its band. Left and right changes the frequency. Up and down changes the gain. The **H** and **L** nodes only move left and right.

The bands

The row under the graph has one switch for each band. Tap a band's name to select it, so the controls below show its settings. Tap its switch to turn it on or off. A band that is off does nothing, and its node disappears from the graph.

BAND	WHAT IT DOES
HPF	High-pass filter. Cuts everything below its frequency. Use it to remove rumble and boom.
1 to 6	Main bands. Each one boosts or cuts around its frequency.
LPF	Low-pass filter. Cuts everything above its frequency. Use it to remove fizz and hiss.

To take the Global EQ out completely, turn every band off.

Controls

The bottom row shows the settings for the selected band. Knobs **A**, **B**, and **C** below the screen turn **Frequency**, **Gain**, and **Q**, in that order. You can also turn the controls on screen.

- **Frequency**: where the band works, from 20 Hz to 20 kHz.
- **Gain**: how much the band boosts or cuts, from –12 dB to +12 dB.

- **Q**: how wide the band is, from 0.1 to 10. A low Q affects a broad range of frequencies around the band's frequency. A high Q affects a narrow range.
- **Filter type**: the shape of the band.
  - **Low shelf** boosts or cuts everything below the frequency.
  - **Bell** boosts or cuts a range centered on the frequency.
  - **High shelf** boosts or cuts everything above the frequency.

HPF and **LPF** only use **Frequency**. When one of them is selected, **Gain** and **Q** are dimmed and the filter type choices are not shown.

Setting it up

1. Load a rig you know well, one that sounds right on a system you trust.
2. Play it through the speakers, room, or PA you want to correct, at the volume you will play at.
3. Listen for what is wrong compared with how the rig should sound, and use the analyzer to find where it is.
4. Fix it with the fewest bands and the smallest moves you can. Cut what is too loud before you boost what is missing.
5. Switch to a few other rigs and check that they all sound better, not just the one you started with.

Some common fixes:

- **Boomy room or stage:** turn on **HPF** and raise it until the boom clears, often somewhere around 80 to 120 Hz.
- **Harsh or fizzy through full range speakers:** turn on **LPF** and bring it down until the fizz goes, or use a **High shelf** with a small cut.
- **One note booms louder than the rest:** use a **Bell** with a high **Q** and a cut, and move it until that note evens out.
- **Muddy:** try a broad **Bell** cut somewhere between 200 and 400 Hz.

Tips

- If you find yourself boosting a lot, turn down the boost and check the Mixer output levels and the master volume instead. Big boosts push the output toward its limit.
- If every rig needs the same change at stage volume, try **ON STAGE** in the Mixer first. It corrects for how your ears hear loud sound, and you may need less Global EQ with it on.
- If only one rig needs a change, put a Signal 8-Band EQ in that rig instead. The Global EQ is for problems every rig shares.
- When you move to a different room or playback system, revisit the Global EQ. A correction that helped one room can hurt in another.

Audio Engine

Choose the sample rate and buffer size: read the latency grid, understand the trade-off, and pick the setting that suits your rigs.



The Audio Engine page sets how Axion processes audio: the **sample rate** and the **buffer size**. Together they decide two things that pull against each other:

- **Latency**: how long it takes for a note you pick to come out of your speakers.
- **Processing capacity**: how much work your rigs can do before the audio starts to crackle.

To open it, tap the menu button at the top left of the screen, choose **Settings**, then choose **Audio Engine** from the list on the left.

The grid

Every cell in the grid is one combination of sample rate and buffer size.

- **Columns** are buffer sizes, from **32** to **128** samples. Axion processes audio in blocks of this many samples at a time.
- **Rows** are sample rates: **44.1 kHz**, **48 kHz**, and **96 kHz**. This is how many samples Axion processes every second.

- The number in each cell is the **estimated round-trip latency** in milliseconds: the time from your guitar signal arriving at the input jack to the processed sound leaving the output jack. These figures were measured on Axion hardware with an analog loopback, not worked out on paper.

The cell marked **Running** is the setting Axion is using right now. A hatched cell is a combination Axion does not support. If a cell shows **—** instead of a number, its latency has not been measured for your unit's current configuration. You can still choose it.

The shading tells you how the latency will feel to play:

SHADE	LATENCY	FEEL
Tight (dark)	8 ms or less	Feels immediate, like plugging into an amp.
Good (amber)	Over 8 ms, up to 10 ms	Fine for most players. Fast, percussive playing may start to feel a touch soft.
Loose (red)	Over 10 ms	Many players start to feel a lag between pick and sound.

For a sense of scale, sound travels about 34 cm (a little over a foot) in one millisecond. Standing 2 m from a guitar amp puts about 6 ms between your pick and your ears, and nobody calls that laggy. Every figure in the grid is in the range of standing a few meters from an amp.

The trade-off

Smaller buffers mean lower latency, but less time to process each block. At 32 samples Axion has to finish your whole rig in a fraction of a millisecond, over and over. If a block is not finished in time, you hear a click or crackle. Larger buffers give your rig more breathing room, at the cost of more latency.

Higher sample rates mean lower latency for the same buffer, but more work. A 64-sample block is half as long in time at 96 kHz as at 48 kHz, so latency drops. But every device in your rig has twice as many samples to process each second, so a rig uses roughly twice the processing at 96 kHz. A higher sample rate also leaves more room above the range of hearing, which can help high-gain amps and drives sound smoother.

In practice:

- Moving **left** in the grid (smaller buffer) or **down** (higher sample rate) lowers latency and uses more of Axion's capacity.
- Moving **right** (larger buffer) or **up** (lower sample rate) raises latency and frees up capacity.

Choosing a setting

The best setting is the lowest latency at which your heaviest rig plays without a single click.

1. Load the most demanding rig you play: the one with the most amps, cabs, reverbs, and third-party plugins. Turn on every device you would use in a song.
2. Open **Settings**, then **Audio Engine**, and tap a cell in the **Tight** area.
3. Tap **Apply**. Audio stops for a few seconds while the engine restarts with the new setting, then your rig comes back.
4. Play hard for a few minutes, through every scene of the rig. Watch the CPU meter in the top bar (the chip icon with a percentage) and listen for clicks, crackles, or dropouts.
5. If you hear any, or the CPU meter sits near the top, move one cell to the **right** (a larger buffer) or **up** (a lower sample rate), apply, and test again.
6. Stop at the first setting that stays clean. Leave some headroom: a rig that only just plays cleanly at home may not on a hot stage, or after you add one more device.

If you are not sure, a setting in the **Tight** area at **96 kHz** or **48 kHz** with a buffer of **64** is a good place to start.

Applying a change

Tapping a cell only selects it. Nothing changes until you tap **Apply**.

- Applying restarts the audio engine, so **audio stops for a few seconds**. Do not apply a change in the middle of a song.
- Your rig, scene, and settings are restored when audio comes back.
- If the Looper holds a loop, Axion will not apply the change, because restarting the audio would erase the loop. Clear the looper first.
- If the new setting cannot start, Axion goes back to the setting that was running before and shows a message under the grid.
- The setting is saved. Axion uses it every time you turn it on, until you apply a different one.

Tips

- Crackles that come and go only on one rig usually mean that rig is too heavy for the current setting. Try a larger buffer before removing devices.
- Third-party plugins vary a lot in how much processing they need. If adding one makes a rig crackle, a larger buffer is often the fix.
- If you change to a lower sample rate and a rig sounds harsher or grainier at high gain, try 96 kHz with a larger buffer instead. The latency may be about the same.
- Latency adds up. If you play through other gear that adds its own delay, such as a digital mixer or wireless system, choose a tighter setting on Axion to make up for it.

06

DEVICES

Every amp, drive, cab, and effect in the unit.

Device Reference

Every device that ships with Axion: parameters, defaults, and use.

Reference for every device that ships with Axion. The list below is generated from each device's `manifest.json`.

Amp

- Neural Amp Modeler: Neural amp modeler with integrated bass/mid/treble tone stack, HPF/LPF and pick-attack shaping. Native device backed directly by the NeuralAudio inference library.
- Valve: Circuit-modeled two-triode valve preamp with passive tone stack, compiled ahead of time from its schematic by the Valve circuit compiler. Preamp only: follow it with a cab/IR. Experimental.

Boost

- Boost: Simple gain boost for testing
- Copper Crest: Circuit model of a vintage single-germanium-transistor treble booster.

Cab

- Cabinet: Dual cabinet simulator with microphone modeling and position control
- Impulse IR: Impulse response loader with start/end trim and low/high cut

Delay

- Afterglow Delay: Stereo dual delay with independent tempo sync, saturation, and LFO modulation
- Double Take: MN3005-style clocked analog echo with companding and circuit-derived filters
- Tape Delay: Stereo tape echo with tempo sync, repeat saturation, age-dependent bandwidth, wow, flutter, and stereo head spread

Distortion

- Bloom: Envelope-driven harmonic generator. Pick attacks bloom into bright, complex harmonics; sustained notes settle into smooth, rounded saturation. Reacts to playing dynamics instead of clipping statically.
- Enforcer: Hard-clipping distortion modelled on a classic orange-box pedal. Diode-to-ground clipping with pre-emphasis filtering and a passive blend tone stack.
- Pug Blast: Aggressive op-amp distortion with symmetrical clipping and variable post-distortion filter
- Slaghammer: Heavy distortion with a coupled transistor preamp, asymmetric clipping, germanium crossover gate and interacting three-gyrator Color Mix EQ.

- The Regent: Amp-in-a-box distortion modelled on the Marshall Guv'nor. Op-amp soft clipping into silicon diode shunt with a passive Marshall-style 3-band tone stack.

Dynamics

- Dynamic Compressor: 4-band dynamic compressor / dynamic EQ. Each band is a peaking filter that ducks only when the band exceeds its threshold. Useful for de-essing, pick/string-noise reduction, and surgical dynamics control.
- Kompressor: Circuit-derived OTA pedal compressor with parallel blend, treble recovery, and pickup-tuned release
- Limiter: Transparent stereo true-peak limiter for low-latency output protection
- Multiband Compressor: 3-band multiband compressor with Linkwitz-Riley crossovers, per-band compression controls, and stereo linking
- Simple Compressor: Stereo compressor with RMS/Peak detection, soft knee, and linked/unlinked stereo operation
- Swell: Automatic volume swell: suppresses the pick transient and blooms each note in. Works like an inverse noise gate, giving hands-free violining without a volume pedal.

EQ

- Axion 8-Band EQ: 8-band parametric equalizer with soft clipping

Fuzz

- Big Buff: Circuit-derived 1976-77 V3 four-transistor fuzz with cascaded 1N914 feedback clipping and the component-matched passive tone stack
- Facemelt: Circuit model of the original positive-ground PNP germanium two-transistor fuzz, including the selected AC128 pair, feedback interaction and loaded two-knob controls
- Octave Forge: Transformer octave fuzz with a coupled three-transistor gain circuit and germanium full-wave rectifier.

Modulation

- Blue Chorus: Low-latency analog-inspired chorus with rounded modulation, BBD-style bandwidth, stereo width, and adjustable age
- Chorus: Stereo chorus effect with LFO modulation, feedback, and multiple waveforms
- Miss Conduct: Vintage 15 V flanger circuit model with short clock-controlled delay, resonant feedback, and stationary Filter Matrix
- Night Vibe: Four unequal optical phase stages with incandescent lamp dynamics and chorus/vibrato routing.
- Phaser: Stereo phaser effect with cascaded all-pass filters, multiple waveforms, and variable stage count

- Quack: Circuit-derived vintage optical envelope filter with three physical filter taps and up/down sweep.
- Tremolo: Amplitude modulation effect with LFO, multiple waveforms, stereo phase offset, and tempo sync
- Wah: Crybaby-style resonant-bandpass wah with pedal or envelope (auto-wah) control, adjustable heel/toe range, resonance, and a toe-switch cut

Overdrive

- Blues Rider: Circuit-derived dual-stage overdrive with the original BD-2 gain, clipping, and tone networks
- Crystal Pistol: Transparent low-to-medium gain overdrive with post-clipping tone shaping
- Green Scream: Tube Screamer TS808/TS9 style overdrive with implicit diode solver and 4x oversampling
- K Drive: Circuit-derived Klon Centaur overdrive with coupled dual-gang gain paths, germanium clipping and component-matched active treble control
- Plague OD: Circuit-derived vintage RAT distortion with LM308 dynamics, 1N914 hard clipping, component-matched filter and 2N5458 output stage
- SuperDrive: Asymmetric feedback clipping overdrive with mid-forward voicing and even-order harmonics
- Velvet Break: Circuit model of the original pre-1995 two-stage Bluesbreaker overdrive.
- WideDrive: Circuit-derived OD-3-style dual-stage overdrive with discrete gain cells and component-matched filtering

Pitch

- Low Divide: Analog-style octave divider with peak tracking, germanium clamping and cascaded polarity-switch voices
- Octaver: Four-voice monophonic octaver (-2, -1, +1, +2) with an independent level per voice. Each voice is a tape-style pitch shift: the input is replayed at the octave ratio and spliced by whole periods of the tracked note, so the result sounds like the same guitar tuned an octave away with its dynamics intact. Tracks single notes and bass lines; on chords the tracker loses lock and the voices gate off. Octave voices sit a little behind the dry signal (a few ms on high notes, more on low E). Dry/wet mix and output volume are the engine's global device parameters.
- Transpose: Polyphonic pitch shifter: moves everything you play, chords included, up or down by up to two octaves. With Step on it retunes the guitar (Eb or D standard, an octave down for bass lines); with Step off the shift is continuous and sweeps like a whammy from an expression pedal. Shifting down, pick attacks arrive about 2-4 ms behind the dry signal. Shifting up, they arrive later, and more so with a larger Buffer: about 7-12 ms on 20 ms, up to about 20 ms on 60 ms. A larger Buffer splices less often and handles lower notes cleanly (20 ms for guitar, 30 ms and up for bass). Tonality lets the dry signal through unshifted above its frequency, keeping pick and string noise natural; at 20 kHz it is off. Dry/wet mix and output volume are the engine's global device parameters.

Reverb

- Aether: Live spectral clouds: capture a note, chord or whisper and let it bloom into an evolving cinematic texture.
- Halflife: Convolution reverb
- Horizon Reverb: Algorithmic reverb with diffusion, FDN tank, modulation, and shimmer

Utility

- FX Loop: Serial send/return insert: routes the signal out a physical output jack to external gear and returns it from a physical input jack (100% wet). Defaults to FX Send 1 / FX Return 1. Parallel/blended mix is disabled until return-latency calibration exists.
- Looper: Records and plays back a stereo loop. Position in the chain determines what's captured and where it's mixed back in.
- Stack: A stack allows you to group devices and chains into one reusable block. Add chains in parallel, then use Mix to blend their combined return with the original signal.

Aether

Live spectral clouds: capture a note, chord or whisper and let it bloom into an evolving cinematic texture.



Parameters

PARAMETER	RANGE	DEFAULT
Bloom	0 – 4000 ms	180 ms
Decay	1 – 60 s	12 s
Diffuse	0 – 100 %	85 %
Tone	0 – 100 %	55 %
Width	0 – 100 %	85 %
Freeze	Off, On	Off
Smooth	40 – 680 ms	340 ms

Overview

Live spectral clouds: capture a note, chord or whisper and let it bloom into an evolving cinematic texture.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Afterglow Delay

Stereo dual delay with independent tempo sync, saturation, and LFO modulation



Parameters

PARAMETER	RANGE	DEFAULT
Delay 1 Time	10 – 2000	500
Delay 1 Sync	`Off`, `On`	`Off`
Delay 1 Division	`1/1`, `1/2`, `1/4`, `1/8`, `1/16`, `1/4 D`, `1/8 D`, `1/16 D`, `1/4 T`, `1/8 T`, `1/16 T`	`1/4`
Delay 1 Pan	0 – 1	0
Delay 1 Level	0 – 1	0.5
Delay 1 Feedback	0 – 0.95	0.4
Delay 1 Saturation	0 – 1	0
Delay 1 LFO Depth	0 – 1	0
Delay 2 Time	10 – 2000	375

PARAMETER	RANGE	DEFAULT
Delay 2 Sync	`Off`, `On`	`Off`
Delay 2 Division	`1/1`, `1/2`, `1/4`, `1/8`, `1/16`, `1/4 D`, `1/8 D`, `1/16 D`, `1/4 T`, `1/8 T`, `1/16 T`	`1/8 D`
Delay 2 Pan	0 – 1	1
Delay 2 Level	0 – 1	0.5
Delay 2 Feedback	0 – 0.95	0.4
Delay 2 Saturation	0 – 1	0
Delay 2 LFO Depth	0 – 1	0
LFO Rate	0.01 – 10	0.5

Overview

Afterglow Delay provides two independent stereo delay lines. Each line has its own time, pan, level, feedback, saturation, and modulation depth, while both share one LFO rate. Use the two lines as a left and right stereo pair, as two rhythmic taps, or as contrasting clean and dark repeats.

Time sets the spacing of each echo in milliseconds when **Sync** is Off. With Sync On, **Division** follows the engine tempo using straight, dotted (D), or triplet (T) notes. Each line can use a different mode and division. Turning Sync off restores that line's stored millisecond time. Synced times reach 12 seconds, with extra buffer room for modulation. Time changes glide briefly in pitch as the delay tap moves.

Pan places that line in the stereo field and **Level** sets its contribution. **Feedback** controls the number of repeats. **Saturation** colors the feedback path, so successive repeats become warmer and more compressed. **LFO Depth** varies delay time for pitch movement, and the global **LFO Rate** sets the speed of that movement.

Tips

- For a wide rhythmic delay that follows tempo, turn both Sync controls On, use `1/4` for Delay 1 and `1/8 D` for Delay 2, and pan them left and right. At 120 BPM these give 500 ms and 375 ms. Keep both Levels near 40%.
- Add a small LFO Depth with a slow LFO Rate for tape like movement. Large Depth values create obvious chorus and pitch effects.
- Raise Saturation to make repeats soften as they decay. Keep it low when the delayed attack must remain precise.
- Feedback values near the top of the range can build very quickly. Increase them while playing and reduce them if the repeats continue to grow.

- For a centered mono style delay, set both Pan controls to the center and use matching times, or lower one line's Level and use it as a quieter second tap.

Big Buff

Circuit-derived 1976-77 V3 four-transistor fuzz with cascaded 1N914 feedback clipping and the component-matched passive tone stack

Parameters

PARAMETER	RANGE	DEFAULT
Sustain	0 – 100 %	50 %
Tone	0 – 100 %	50 %
Level	-20 – 12 dB	0 dB

Overview

Circuit-derived 1976-77 V3 four-transistor fuzz with cascaded 1N914 feedback clipping and the component-matched passive tone stack

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Bloom

Envelope-driven harmonic generator. Pick attacks bloom into bright, complex harmonics; sustained notes settle into smooth, rounded saturation. Reacts to playing dynamics instead of clipping statically.

Parameters

PARAMETER	RANGE	DEFAULT
Sensitivity	0 – 1	0.5
Response	0 – 1	0.5
Smooth	0 – 1	0.4
Harmonics	0 – 1	0.5
Mix	0 – 1	1
Level	0 – 1	0.5

Overview

Bloom is a dynamics sensitive harmonic generator. It adds a brighter, more complex edge to the start of a note, then settles into smoother saturation as the note sustains. The result follows picking strength instead of applying the same clipping curve to every note. It works well for touch responsive lead tones, lively clean sounds, and adding definition before an amp.

Sensitivity sets how readily the effect reacts to the input. **Response** sets how strongly the detected attack changes the harmonic character. **Smooth** moves the steady sound from a sharper shape toward a rounder one, although strong attacks still lean brighter. **Harmonics** controls the overall amount of saturation. **Mix** blends the processed and dry paths, and **Level** sets the output.

Tips

- Begin with Sensitivity and Response at 50%, Smooth near 40%, and Harmonics around 35%. Raise Sensitivity until soft and hard picking feel distinct.
- For an articulate clean enhancer, keep Harmonics below 30% and Mix between 30% and 50%.

- For a more aggressive lead sound, raise Harmonics and Response, then use Smooth to take the edge off sustained notes.
- Place Bloom early in the chain so it can respond directly to playing dynamics. Heavy compression before it reduces the difference between soft and hard attacks.
- Match Level with bypass before judging a setting. Extra loudness can make a harmonic setting seem better than it is.

Blue Chorus

Low-latency analog-inspired chorus with rounded modulation, BBD-style bandwidth, stereo width, and adjustable age

Parameters

PARAMETER	RANGE	DEFAULT
Rate	0.2 – 5 Hz	0.9 Hz
Depth	0 – 100 %	55 %
Width	0 – 100 %	75 %
Color	0 – 100 %	55 %
Age	0 – 100 %	35 %
Level	-6 – 6 dB	0 dB

Overview

Low-latency analog-inspired chorus with rounded modulation, BBD-style bandwidth, stereo width, and adjustable age

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Blues Rider

Bluesy overdrive with shunt-style clipping and a BD-2-style tilt tone control



Parameters

PARAMETER	RANGE	DEFAULT
Level	-24 – 6 dB	0 dB
Tone	0 – 100 %	50 %
Gain	0 – 100 %	50 %
Mode	Stock, Mod	Stock

Overview

Blues Rider is a Blues Driver style overdrive with an open frequency response, a crisp attack, and a broad gain range. It responds well to picking strength and guitar volume, moving from a lightly colored boost to grainy blues and rock crunch. At high Drive settings, it becomes more compressed and saturated while retaining note

definition.

Compared with a mid-focused overdrive, Blues Rider preserves more low and high frequency detail. This makes it useful as a standalone drive into a clean amp, a dynamic rhythm sound, or a boost into an amp that is already beginning to break up.

Controls

- **Drive** sets the gain feeding the clipping stage. Low settings keep more of the natural pick attack. Higher settings add saturation, compression, and sustain.
- **Tone** tilts the frequency response around a central point. At 50%, the response is flat. Lower settings boost bass and reduce treble. Higher settings reduce bass and boost treble.
- **Level** sets the output volume from -24 dB to +6 dB. It does not change the amount of clipping.
- **Mode** selects the voicing. **Stock** provides the classic Blues Driver style response. **Mod** changes the clipping and filter voicing for slightly more low-end extension, a smoother top end, and a subtly different Tone response.

Tips

For a touch-sensitive blues sound, keep Drive below halfway and Tone near 50%. Pick lightly for a cleaner response, then dig in for more grit.

For a cleanish boost, set Drive low and raise Level. Keep Tone at 50% for the most neutral response, or move it slightly higher to tighten the bass before a driven amp.

For classic rock rhythm, raise Drive past halfway and adjust Tone for the amp or cabinet. Bright systems often benefit from a setting below 50%, which adds body while softening the top end.

Use Mod mode when Stock feels too lean or sharp. Because Mode changes both the clipping and filtering, recheck Tone and Level after switching.

At high Drive settings, lowering Tone can keep the extra harmonics from sounding brittle. Raising Level by a few decibels can also help compensate for the added compression.

Starting Points

SETTING	DRIVE	TONE	LEVEL	MODE
Cleanish Boost	10 %	50 %	4 dB	Stock
Touch-Sensitive Blues	30 %	45 %	0 dB	Stock
Classic Rock Crunch	55 %	50 %	1 dB	Stock

SETTING	DRIVE	TONE	LEVEL	MODE
Cutting Lead	65 %	60 %	3 dB	Stock
Smooth, Full Drive	70 %	45 %	3 dB	Mod

Treat the Level values as starting points. Pickup output, other devices, and amp gain all affect the final volume.

Boost

Simple gain boost for testing

Parameters

PARAMETER	RANGE	DEFAULT
Boost	0–24 dB	0 dB

Overview

Boost is a clean level control that adds up to 24 dB without intentionally changing the tone. Use it to make a solo louder, compensate for a quiet device or pickup, or drive the input of a later amp or distortion harder.

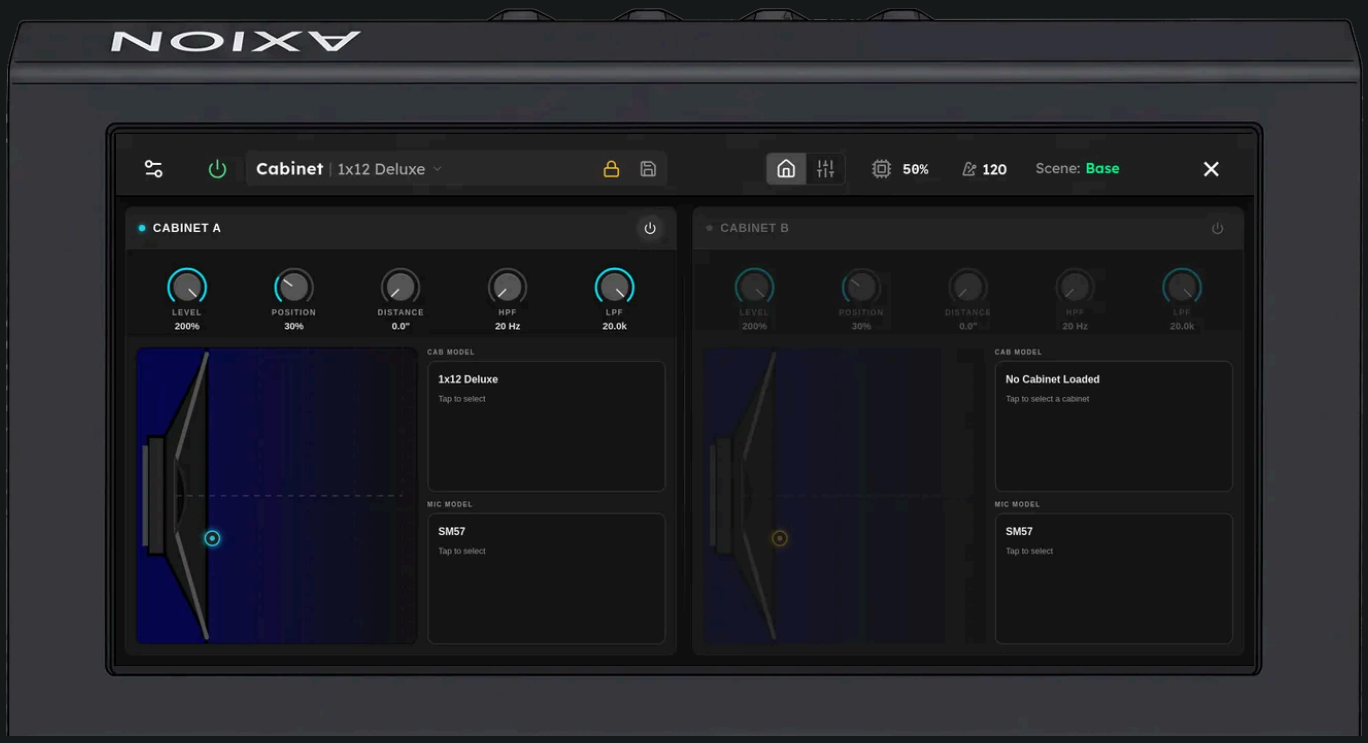
Placement determines the result. Before a nonlinear device, extra Boost level usually creates more saturation and compression with only a modest change in final volume. Near the end of the chain, it acts more like a straightforward volume lift, provided the following devices and outputs still have headroom.

Tips

- Start around 3 to 6 dB for a solo lift, then compare against bypass in the context of the full band.
- Place Boost before an amp for more gain and sustain. Place it after the amp and cabinet for a cleaner level increase.
- Watch downstream meters when using large boosts. If a later device clips, reduce Boost or lower that device's input or output level.
- Assign Boost bypass to a footswitch for a simple lead switch.

Cabinet

Dual cabinet simulator with microphone modeling and position control



Parameters

PARAMETER	RANGE	DEFAULT
Microphone A	SM57, MD421, Royer 121, U87, C414, e906	SM57
Distance A	0 – 100 %	0 %
Position A	0 – 100 %	30 %
Reverse A	Off, On	Off
Head Cut A	0 – 100 %	0 %
Tail Cut A	0 – 100 %	0 %
Pre-Delay L A	0 – 200 ms	0 ms
Pre-Delay R A	0 – 200 ms	0 ms
Trim Threshold A	-80 – 0 dB	0 dB

PARAMETER	RANGE	DEFAULT
High Pass A	20 – 20000 Hz	20 Hz
Low Pass A	20 – 20000 Hz	20000 Hz
Level A	0 – 200 %	200 %
Microphone B	SM57, MD421, Royer 121, U87, C414, e906	SM57
Distance B	0 – 100 %	0 %
Position B	0 – 100 %	30 %
Reverse B	Off, On	Off
Head Cut B	0 – 100 %	0 %
Tail Cut B	0 – 100 %	0 %
Pre-Delay L B	0 – 200 ms	0 ms
Pre-Delay R B	0 – 200 ms	0 ms
Trim Threshold B	-80 – 0 dB	0 dB
High Pass B	20 – 20000 Hz	20 Hz
Low Pass B	20 – 20000 Hz	20000 Hz
Level B	0 – 200 %	200 %
Enable A	Off, On	On
Enable B	Off, On	On
Mix	0 – 100 %	50 %

Overview

Cabinet reproduces the speaker and microphone stage of a guitar rig. Place it after an amp model to turn the amp's direct output into a finished miked cabinet sound. Two independent slots, A and B, can be enabled separately and blended with **Mix**. At 0% the output is all A, at 100% it is all B, and 50% gives equal amounts of both.

Each slot offers six microphone models. **Position** moves from the center of the speaker at 0% toward the edge at 100%. Center positions are usually brighter and more direct. Edge positions are usually warmer. **Distance** moves from near to far, adding propagation delay and reducing some high frequency energy. High Pass and Low Pass provide final cleanup, while Level balances the two slots.

Reverse, Head Cut, Tail Cut, the two Pre Delay controls, and Trim Threshold reshape the slot's impulse response. These are primarily sound design and alignment tools. A Trim Threshold of 0 dB disables automatic tail trimming.

Following the amp

Amp presets know whether their sound already includes a speaker cabinet. A preset for an amp head on its own does not, and it needs a Cabinet after it. When you choose an amp preset, the Cabinet after that amp follows the choice:

- **Amp head preset, no Cabinet after the amp.** A Cabinet is added right after the amp, loaded with the cabinet preset suggested for that amp.
- **Amp head preset, Cabinet already in place.** The Cabinet loads the suggested cabinet preset, and is switched back on if it was bypassed.
- **Amp preset that includes its cabinet.** The Cabinet is bypassed so you do not hear two cabinets in a row. It stays in the chain and comes back when you return to an amp head preset.

Each amp head preset can suggest the cabinet that suits it, for example a 4x12 for a high gain head. When a preset suggests none, Axion loads the 1x12 Deluxe.

A locked Cabinet is the exception to all of this: it is never added to, reloaded, or bypassed by an amp preset.

A few things to know:

- A Cabinet only follows the amp when you choose an amp preset, or add or swap an amp. Loading a rig or a scene never changes it.
- If you delete the Cabinet after an amp, Axion takes that as your decision and does not add another one when you change presets on that amp.
- An Impulse IR after the amp counts as its cabinet. It is switched on or bypassed to follow the amp, but the IR you loaded is never replaced.

Cab Bypass

If Axion feeds a power amp and a real guitar cabinet, the real speaker does the Cabinet's job and you do not want both. Rather than bypassing or deleting the Cabinet in each rig, turn on **CAB BYPASS** on the Mixer screen. This one switch bypasses the Cabinet blocks in every rig, without changing the rigs themselves. Turn it off again and the same rigs sound right through headphones or full range speakers.



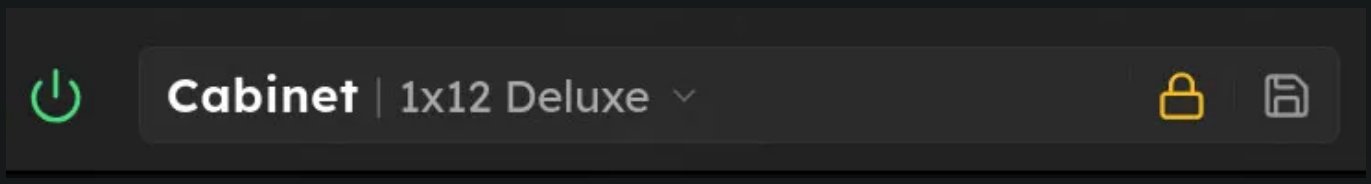
While Cab Bypass is on, every Cabinet block in the rig editor is dimmed and shows a red mute icon in its center. The icon tells it apart from a Cabinet you bypassed yourself, which is dimmed with no icon.

A few things to know:

- Cab Bypass is a Mixer setting, so it belongs to the unit and not to a rig. It stays on when you change rigs, scenes, or set lists, and when you turn Axion off, until you switch it off yourself.
- It applies to every output at once, headphones included.
- The Cabinet's own bypass does not respond while Cab Bypass is on. Double tapping the block, the power button on its device screen (which shows the same mute icon), and a footswitch assigned to it all do nothing. The Cabinet keeps the on or off state it had, and goes back to it when you turn Cab Bypass off.
- Amp presets keep pairing with their cabinets in the background, so each rig has the right Cabinet waiting when you turn Cab Bypass off.
- A locked Cabinet is bypassed too. The lock only protects it from amp presets.
- Impulse IR and any other device in the **Cab & IR** category are bypassed the same way.
- Cab Bypass cannot remove a cabinet that is part of an amp preset's own sound. For a power amp and guitar cabinet, choose amp head presets.

Cabinet lock

A locked Cabinet keeps its preset, its settings, and its bypass state no matter which amp preset you choose. Use the lock when you have dialed in a cabinet sound you like and want to try different amps through it.



To lock a Cabinet:

1. Tap the Cabinet block in the rig editor to open its device screen.
2. Tap the padlock button to the right of the preset name. The padlock closes and turns amber.

3. Tap it again to unlock. An open gray padlock means the Cabinet follows the amp.

In the rig editor, a locked Cabinet shows a small amber padlock in the top left corner of its block, so you can see at a glance which cabinets are held.

A few things to know:

- The lock belongs to that one Cabinet block and is saved with the rig.
- Impulse IR has the same lock.
- The padlock in the top right corner of Rig view is Screen Lock, which is a different feature.

Tips

- Begin with one slot enabled, Position around 25% to 40%, Distance at 0%, and the filters fully open. Choose the microphone before fine tuning placement.
- Move Position toward the edge when a close sound is too sharp. Increase Distance when it feels too immediate or bright.
- For a common two microphone blend, use a dynamic microphone in slot A for attack and a ribbon microphone in slot B for body. Balance them with Level, then use Mix for the final proportion.
- Raise High Pass to remove rumble and excess low end. Lower Low Pass to reduce fizz after the microphone blend is established.
- Different Pre Delay values can align or separate the two slots. Make small changes and check the result in mono because timing offsets can cause phase cancellation.

Chorus

Stereo chorus effect with LFO modulation, feedback, and multiple waveforms

Parameters

PARAMETER	RANGE	DEFAULT
Rate	0.05 – 8 Hz	0.85 Hz
Depth	0 – 100 %	40 %
Delay	5 – 25 ms	12 ms
Stereo Phase	0 – 180 deg	90 deg
Feedback	0 – 100 %	34 %
Waveform	`Sine`, `Triangle`	`Sine`
High Cut	2000 – 16000 Hz	5656.854 Hz

Overview

Chorus mixes the dry signal with a short delay whose time moves continuously. The small pitch and timing differences make one part sound like several performances, adding width and motion to clean chords, bass, and sustained lead tones.

Rate sets the modulation speed. **Depth** sets how far the delay time moves, and **Delay** sets its center value. Shorter Delay values sound tighter, while longer values make the doubled character more obvious. **Stereo Phase** offsets the modulation between left and right. **Feedback** returns part of the delayed signal for a stronger, more resonant effect. **Waveform** selects smooth sine movement or more linear triangle movement. **High Cut** darkens the feedback path and can keep a deep setting from sounding brittle.

Tips

- For a subtle stereo clean sound, try Rate near 0.6 Hz, Depth around 2 ms, Delay near 10 ms, Stereo Phase at 90 degrees, and low Feedback.
- For an obvious slow chorus, lower Rate, raise Depth, and use a longer Delay.

- For faster vibrato like movement, raise Rate and Depth while keeping Feedback modest.
- Set Stereo Phase to 0 degrees for matching movement in both channels. Raise it for increasing stereo width.
- Lower High Cut when high Feedback creates sharp or metallic repeats.

Copper Crest

Circuit model of a vintage single-germanium-transistor treble booster.

Parameters

PARAMETER	RANGE	DEFAULT
Boost	0 – 100 %	75 %

Overview

Circuit model of a vintage single-germanium-transistor treble booster.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

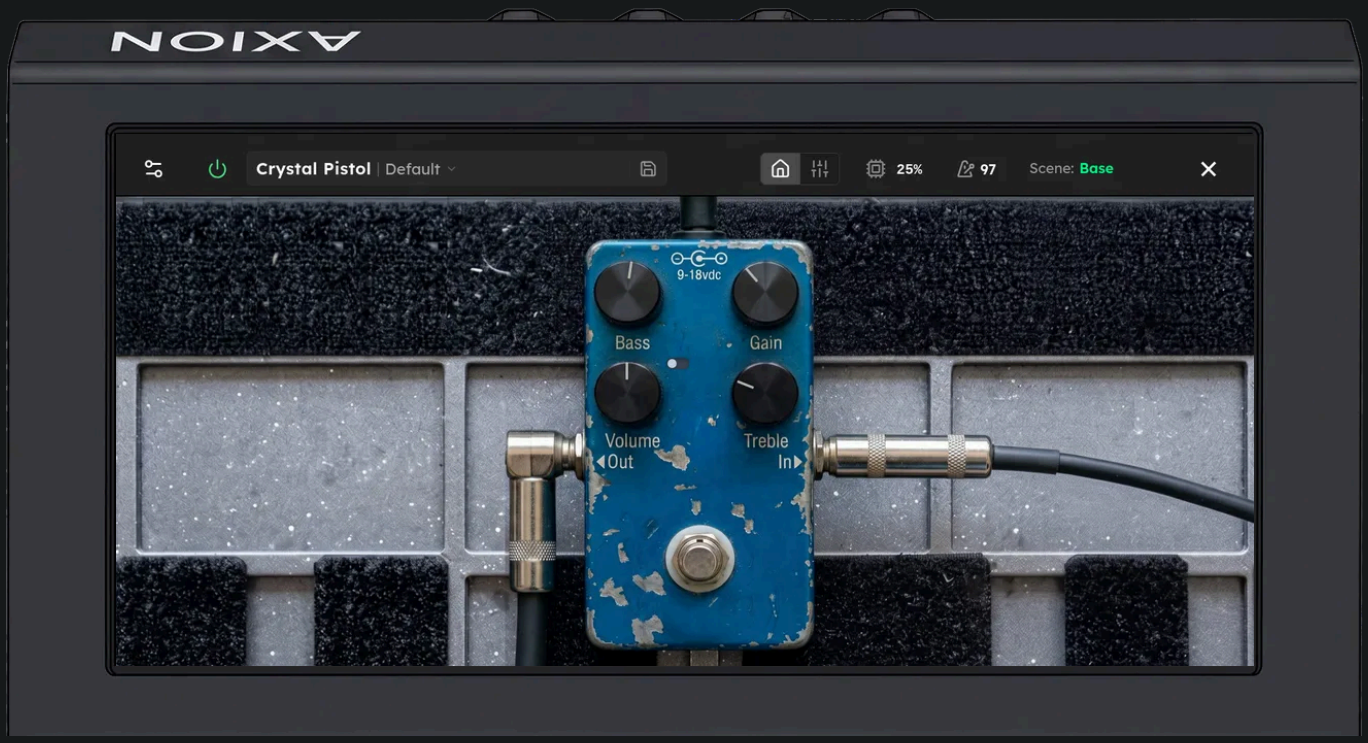
Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

SECTION 06.11 · DEVICES

Crystal Pistol

Transparent low-to-medium gain overdrive with post-clipping tone shaping



Parameters

PARAMETER	RANGE	DEFAULT
Bass	60 – 500 Hz	294.2831 Hz
Gain	0 – 100 %	35 %
Treble	3000 – 12000 Hz	5223.3034 Hz
Clip Mode	`Symmetrical`, `Asymmetrical`	`Symmetrical`

Overview

Crystal Pistol is a clear, low to medium gain overdrive designed to preserve pick attack and chord detail. It works as a light standalone drive, an always on source of edge, or a boost that adds clipping before an amp without a strong fixed midrange voice.

Drive sets the clipping intensity. **Bass** is the cutoff of a high pass filter after clipping, so higher frequency values remove more low end and tighten the sound. **Treble** is the cutoff of a low pass filter. Higher values retain more brightness, while lower values make the drive warmer. `Symmetrical` clipping has a balanced, smooth character. `Asymmetrical` clipping adds a less even waveform and more harmonic texture.

Tips

- For an open rhythm sound, start with Drive around 25%, Bass near 100 Hz, Treble around 8 kHz, and Symmetrical clipping.
- To tighten a driven amp, keep Drive low and raise the Bass cutoff until palm muted notes stay controlled.
- For a warmer lead, lower the Treble cutoff and try Asymmetrical clipping for added texture.
- Use the lowest Bass cutoff that keeps the rig clear. Excessive bass removal can make solo guitar sound impressive but thin out the full band.
- Compare both Clip Modes at the same loudness. The extra harmonics of Asymmetrical mode can change perceived level.

Double Take

MN3005-style clocked analog echo with companding and circuit-derived filters

Parameters

PARAMETER	RANGE	DEFAULT
Repeat Rate	0 – 100 %	46 %
Intensity	0 – 100 %	25 %
Echo	0 – 100 %	70 %

Overview

MN3005-style clocked analog echo with companding and circuit-derived filters

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Dynamic Compressor

4-band dynamic compressor / dynamic EQ. Each band is a peaking filter that ducks only when the band exceeds its threshold. Useful for de-essing, pick/string-noise reduction, and surgical dynamics control.

Parameters

PARAMETER	RANGE	DEFAULT
Band 1 Enable	Off, On	On
Band 1 Frequency	20 – 20000 Hz	225 Hz
Band 1 Q	0.5 – 10	1.4
Band 1 Threshold	-60 – 0 dB	-24 dB
Band 1 Ratio	1 – 20	4
Band 1 Attack	0.1 – 100 ms	5 ms
Band 1 Release	10 – 1000 ms	100 ms
Band 1 Knee	0 – 24 dB	6 dB
Band 1 Range	0 – 24 dB	12 dB
Band 2 Enable	Off, On	On
Band 2 Frequency	20 – 20000 Hz	915 Hz
Band 2 Q	0.5 – 10	1.4
Band 2 Threshold	-60 – 0 dB	-24 dB
Band 2 Ratio	1 – 20	4
Band 2 Attack	0.1 – 100 ms	5 ms
Band 2 Release	10 – 1000 ms	100 ms
Band 2 Knee	0 – 24 dB	6 dB
Band 2 Range	0 – 24 dB	12 dB
Band 3 Enable	Off, On	On

PARAMETER	RANGE	DEFAULT
Band 3 Frequency	20 – 20000 Hz	3000 Hz
Band 3 Q	0.5 – 10	2.2
Band 3 Threshold	-60 – 0 dB	-24 dB
Band 3 Ratio	1 – 20	5
Band 3 Attack	0.1 – 100 ms	2 ms
Band 3 Release	10 – 1000 ms	100 ms
Band 3 Knee	0 – 24 dB	6 dB
Band 3 Range	0 – 24 dB	12 dB
Band 4 Enable	`Off`, `On`	`On`
Band 4 Frequency	20 – 20000 Hz	8500 Hz
Band 4 Q	0.5 – 10	2.2
Band 4 Threshold	-60 – 0 dB	-24 dB
Band 4 Ratio	1 – 20	5
Band 4 Attack	0.1 – 100 ms	2 ms
Band 4 Release	10 – 1000 ms	100 ms
Band 4 Knee	0 – 24 dB	6 dB
Band 4 Range	0 – 24 dB	12 dB
Detector Mode	`RMS`, `Peak`	`Peak`
Stereo Link	`Unlinked`, `Linked`	`Linked`
Output Gain	-24 – 24 dB	0 dB

Overview

Dynamic Compressor is a four band dynamic equalizer. Each enabled band listens around its chosen **Frequency** and turns that region down only when it exceeds the band's **Threshold**. Unlike a conventional multiband compressor, it does not divide the full signal into fixed ranges. This makes it useful for precise deessing, taming pick click or string noise, controlling a resonant note, and reducing low end buildup only on the loudest attacks.

Q controls how wide an area each band affects. Ratio, Attack, Release, and Knee set the compression behavior. **Range** limits the maximum cut, so a band cannot remove more than the chosen number of decibels. **Output Gain** trims the complete result. `Peak` Detector Mode catches short events, while `RMS` responds more smoothly. `Linked` Stereo Link keeps the image stable; `Unlinked` lets each channel react on its own.

Tips

- Work on one band at a time. Disable the others, find the problem frequency, then lower Threshold until only the unwanted moments trigger reduction.
- For pick or string noise, try 3 to 5 kHz, a Q around 2, fast Attack, and a Range of 3 to 6 dB.
- For harsh upper harmonics, try 7 to 10 kHz with a moderate Q and Release near 100 ms.
- For a boomy note, use a narrower Q at its frequency and a modest Range. Excessive Ratio and Range can make the correction obvious.
- Use Linked mode for stereo material. Compare with bypass at matched level, and use Output Gain only after all active bands are set.

Enforcer

Hard-clipping distortion modelled on a classic orange-box pedal. Diode-to-ground clipping with pre-emphasis filtering and a passive blend tone stack.

Parameters

PARAMETER	RANGE	DEFAULT
Distortion	0 – 1	0.5
Tone	0 – 1	0.5
Level	0 – 1	0.5

Overview

Enforcer is a hard clipping distortion with the tight bass, sharp attack, and scooped tone associated with a classic orange distortion pedal. Its fixed input gain means it remains clearly distorted even at low **Distortion** settings. Higher settings add sustain and compression while rolling off some high frequency content before the clipping stage.

Tone blends dark and bright filtered paths. Low values are warmer, the middle range has the strongest scoop, and high values add bite. **Level** sets the final output and can be used to match bypass or push the next device. Enforcer suits cutting rhythm parts, saturated alternative rock sounds, and focused leads that need a firm edge.

Tips

- Start with Distortion around 35%, Tone near 50%, and Level matched to bypass for the core sound.
- Raise Tone for dark pickups or cabinets. Lower it when high gain becomes fizzy or tiring.
- For a tighter high gain sound, place Enforcer before the amp and keep its Distortion below halfway.
- For a more saturated pedal sound into a clean amp, raise Distortion above 60% and adjust Level after comparing with bypass.
- If the centered Tone setting feels too scooped, move the control slightly toward either side and let the downstream amp supply the final EQ.

Facemelt

Circuit model of the original positive-ground PNP germanium two-transistor fuzz, including the selected AC128 pair, feedback interaction and loaded two-knob controls

Parameters

PARAMETER	RANGE	DEFAULT
Fuzz	0 – 100 %	75 %
Volume	0 – 100 %	80 %

Overview

Circuit model of the original positive-ground PNP germanium two-transistor fuzz, including the selected AC128 pair, feedback interaction and loaded two-knob controls

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

FX Loop

Serial send/return insert: routes the signal out a physical output jack to external gear and returns it from a physical input jack (100% wet). Defaults to FX Send 1 / FX Return 1. Parallel/blended mix is disabled until return-latency calibration exists.

Parameters

PARAMETER	RANGE	DEFAULT
Send	0 – 3	2
Return	0 – 3	2
Send Level	0 – 2 x	1x
Return Level	0 – 2 x	1x

Overview

FX Loop inserts external hardware at a specific point in the Axion signal chain. Audio at the device is sent to the selected physical output, passes through the external pedal or processor, and returns through the selected input. The returned signal then continues through the remaining devices.

Send chooses OUT 1, OUT 2, FX Send 1, or FX Send 2. **Return** chooses IN 1, IN 2, FX Return 1, or FX Return 2. The default value 2 selects FX Send 1 and FX Return 1. **Send Level** and **Return Level** are linear trims. A value of 1 is unity, values below 1 attenuate, and values above 1 add gain.

The loop is serial and fully returned. There is no dry blend, so an unplugged return or an external device that is off without bypass will silence the chain after the FX Loop.

Tips

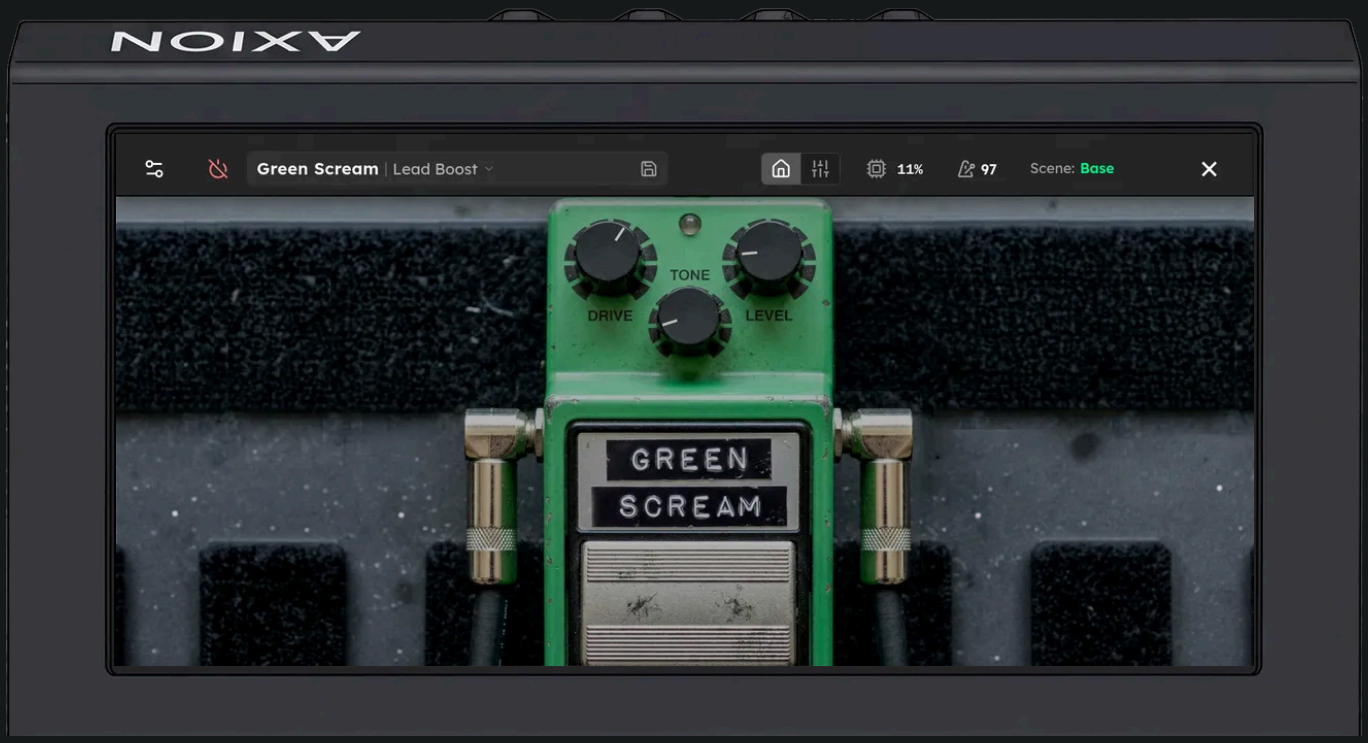
- Connect the selected send to the external device input and its output to the selected return before enabling the loop.
- Start with both Level controls at 1. Reduce Send Level if the external input clips, then use Return Level to restore the volume.

- Place the loop before an amp device for external drives, fuzzes, and boosts. Place it after the amp for delays, reverbs, and studio processors.
- Prefer the dedicated FX Send and FX Return jacks unless the rig specifically needs a main output or instrument input.
- Bypass the FX Loop and compare levels. Large mismatches can create feedback, noise, or clipping in external equipment.

SECTION 06.17 · DEVICES

Green Scream

Tube Screamer TS808/TS9 style overdrive with implicit diode solver and 4x oversampling



Parameters

PARAMETER	RANGE	DEFAULT
Drive	0 – 1	0.5
Tone	0 – 1	0.5
Level	0 – 1	0.5
Mode	TS808, TS9	TS808

Overview

Green Scream is a classic Tube Screamer style overdrive with a smooth, mid-focused tone and light to moderate gain. It adds warmth, sustain, and a tighter low end while preserving note clarity.

It works well as a standalone blues or rock overdrive, but it is especially effective for pushing an already driven amp into more saturation. Guitarists also use it as a boost for solos, to tighten high-gain rhythm tones, or to help a guitar cut through a dense mix.

Tips

For a clean boost, keep Drive low, Level high, and Tone near noon. This pushes the amp harder while tightening the bass and adding midrange presence.

For classic blues overdrive, set Drive around noon, Tone slightly below noon, and Level to match or slightly exceed your clean volume.

To tighten a high-gain amp, set Drive nearly off, Level high, and Tone to taste. This reduces low-end flub and helps palm-muted notes sound sharper and more focused.

Starting Points

SETTING	DRIVE	TONE	LEVEL
Clean Boost	8:00	12:00	3:00
Blues Drive	12:00	11:00	1:00
Lead Boost	10:00	1:00	2:00
High-Gain Tightener	Minimum	12:00	Maximum

Halflife

Convolution reverb



Parameters

PARAMETER	RANGE	DEFAULT
IR File	File	None selected
Reverse	`Off`, `On`	`Off`
Start	0 – 10000	0
Length	0 – 20000	20000
Fade In	0 – 1	0
Fade Out	0 – 1	0
Loudness	-100 – 100	0
Pre-Delay	0 – 200	0
Low Cut	20 – 1000	20

PARAMETER	RANGE	DEFAULT
High Cut	1000 – 20000	20000
Decay Curve	1 – 5	2.5
Trim Threshold	-80 – 0	0
Normalize IR Level	`Off`, `On`	`On`

Overview

Convolution reverb

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Horizon Reverb

Algorithmic reverb with diffusion, FDN tank, modulation, and shimmer



Parameters

PARAMETER	RANGE	DEFAULT
Decay	200 – 30000 ms	7650 ms
Pre-Delay	0 – 200 ms	20 ms
Diffusion	0 – 100 %	70 %
Room Size	0 – 100 %	50 %
Damping	0 – 100 %	50 %
Low Cut	20 – 200 Hz	20 Hz
High Cut	2000 – 16000 Hz	16000 Hz
Mod Rate	0 – 100 %	30 %
Mod Depth	0 – 100 %	30 %

PARAMETER	RANGE	DEFAULT
Shimmer	0 – 100 %	0 %

Pedal controls

The pedal face has five knobs:

- **Pre-Delay**: delay before the reverb starts, from 0 to 200 ms.
- **Size**: room size, from 0 to 100%; scales the reflection timing and spacing.
- **Decay**: tail duration, from 0.2 to 30 seconds.
- **Mod Depth**: how much the tail is modulated, from 0 to 100%.
- **Mix**: the engine's dry/wet balance, from 0 to 100%.

All ten reverb parameters are also available in the device parameter panel, including Diffusion, Damping, filters, Mod Rate, and Shimmer.

Overview

Horizon Reverb is an algorithmic stereo reverb for rooms, long ambient tails, and modulated spaces. **Decay** sets tail length, while **Pre Delay** leaves space between the dry attack and the start of the reverb. **Diffusion** controls how quickly individual reflections merge into a dense tail, and **Room Size** changes the scale and spacing of the reflection network.

Damping, **Low Cut**, and **High Cut** shape the tone of the decaying signal. Mod Rate and Mod Depth add movement to the tail, helping long settings sound less static. **Shimmer** introduces a pitched upper layer for bright, expansive textures.

Tips

- For a natural room, use a short Decay, modest Room Size, high Diffusion, and little modulation or Shimmer.
- For a clear lead reverb, try 20 to 50 ms of Pre Delay. This lets the pick attack speak before the reverb fills in behind it.
- For an ambient pad, lengthen Decay, raise Room Size and Diffusion, then add Mod Depth gradually. Use Shimmer sparingly at first.
- Raise Low Cut to keep bass from accumulating in long tails. Lower High Cut and increase Damping when the space sounds brittle.

- Put Horizon Reverb near the end of the chain so drives and amp models do not distort the reverb tail unless that effect is intentional.

Compatibility

Horizon Reverb was previously a convolution reverb. It now uses an algorithmic engine and needs no IR file. Old Horizon settings and IR selections do not carry over, so revisit saved Horizon instances to dial in their sound. For impulse-response reverb, use Halflife.

Impulse IR

Impulse response loader with start/end trim and low/high cut



Parameters

PARAMETER	RANGE	DEFAULT
IR File	File	None selected
Start	0 – 10000	0
End	0 – 30000	30000
Low Cut	20 – 1000	20
High Cut	1000 – 20000	20000
Master	-24 – 12	0
Normalize	`Off`, `Energy`, `Response peak`	`Response peak`

Mix is the standard device mix (not listed above).

Overview

Impulse IR convolves the input with an impulse response audio file. An IR can capture a speaker cabinet, microphone chain, room, plate, or other linear system, so the sound depends mostly on the selected **IR File**. Impulse IR reads WAV, FLAC, and AIFF files, mono or stereo. A stereo file keeps its separate left and right responses. Files at a different sample rate than the system are resampled when they load. The convolution adds no latency.

Choosing an IR

Tap **LOAD IR** (or the IR name at the top of the screen) to browse the IR folder. The LOAD IR light is on while an IR is loaded. **PREV IR** and **NEXT IR** step through the other IRs in the same folder in name order, and wrap around at either end.

The top of the screen shows the loaded file's name, sample rate, mono or stereo, and length.

Trimming

The screen draws the IR's waveform on a time axis in seconds. **Start** and **End** set the part of the file that is used, in milliseconds from the beginning of the file. Everything before Start and after End is dropped, and the dropped parts are dimmed on the screen. Turn the knobs, or touch the waveform and drag. A touch grabs whichever marker is nearer.

Once the file is analysed, the Start and End knobs span the length of that file. End cannot go past the end of the file, and Start always stays before End. Choosing a different IR file resets the trim to the whole file. Loading a preset keeps the preset's trim.

Tone and Level

Low Cut and **High Cut** filter the convolved signal. They turn on a log scale, so each octave gets the same knob travel. At their extremes (20 Hz and 20 kHz) they are off, and the screen reads **Off**.

IRs are captured at very different levels, so Impulse IR evens them out as they load. Each IR is scaled so its loudest frequency between 80 Hz and 8 kHz sits at 0 dB: no IR boosts any part of the guitar range above the level it went in at, and switching IRs does not jump the volume. Trimming with Start or End re-levels the IR the same way. An IR with a darker or more scooped response can still sound a little quieter than a bright one.

Master sets the IR's level after that, from -24 dB to +12 dB, with 0 dB at 12 o'clock. Use it when an IR still sounds louder or softer than the others. It changes only the convolved (wet) signal, so with Mix below 100 % it also shifts the balance against the dry signal.

Mix blends the convolved signal with the dry input. The **INPUT** and **OUTPUT** meters show the level going into and coming out of the device.

BYPASS turns the device off. Its light is on and the screen dims while the device is bypassed.

On the encoder dock, the first page holds Start, End, Low Cut, High Cut, and Mix, in panel order. The second page holds Master, and the standard device Gain and Output.

Tips

- Place a cabinet IR after an amp model and before time based effects unless a different order is part of the sound.
- Many cabinet IRs begin with a few milliseconds of silence. Move Start up to the first peak to remove the delay it adds.
- Shorten End to tighten a roomy cabinet capture, or to cut the tail of a room IR.
- Use Low Cut to remove boom and High Cut to tame fizz without changing the IR file.
- Level matching is automatic, so IR files can be compared by stepping through them with PREV IR and NEXT IR. Nudge Master for any that still stand out.

K Drive

Transparent Klon-style overdrive with parallel clean/driven paths and gain-dependent blend



Parameters

PARAMETER	RANGE	DEFAULT
Gain	0 – 100 %	50 %
Treble	0 – 100 %	50 %
Channel Mode	Mono, Stereo	Mono
High Fidelity	Off, On	Off
Level	-20 – 12 dB	0 dB

Overview

K Drive is a Klon-style overdrive that combines parallel clean and driven signal paths. At low Gain settings, the clean path dominates and the device behaves like a clear, lightly colored boost. As Gain rises, the driven path becomes more prominent while some clean signal remains in the mix, preserving attack and definition.

The driven path limits low frequencies before soft germanium-style clipping, while the full-range clean path carries the body of the guitar. This produces the tight low end, focused midrange, and touch-sensitive compression associated with this type of circuit. K Drive works well as an always-on tone enhancer, a boost into a driven amp, or a medium-gain overdrive into a clean amp.

Controls

- **Gain** sets both the gain of the clipping stage and the balance between the clean and driven paths. Low settings are mostly clean. High settings add more saturation and make the driven path dominant, but never remove the clean path completely.
- **Treble** applies a high-frequency shelf after the two paths are blended. Lower settings darken the sound, while higher settings add presence and bite. Around 25% is closest to a flat shelf. The default 50% setting adds a modest treble lift.
- **Level** sets the output gain from -20 dB to +12 dB. Use lower values to match bypass volume at higher Gain settings, or positive values to push the next device or amp.
- **Mode** selects the overall voicing. **Classic** provides the warmer standard response. **Bright** tightens the driven low end and extends its upper-frequency response, with a slightly stronger and higher Treble shelf.

Tips

For a clean boost, keep Gain very low and raise Level. This emphasizes the clean path while adding a small amount of soft compression and midrange focus.

For an always-on sound, set Gain between 10% and 25%, then use Treble to suit the guitar and amp. Single-coil pickups may benefit from a lower Treble setting, while dark humbuckers can often take more.

To push an already driven amp, use low Gain and positive Level. The filtered driven path helps keep the bass tight, while the retained clean signal preserves pick attack.

For a standalone overdrive, raise Gain above halfway. As the clean-to-driven blend shifts, the sound becomes thicker and more compressed without losing all of the original note definition.

Bright mode is useful with dark guitars, cabinets, or downstream drives. If it becomes too sharp, reduce Treble toward 25% before switching back to Classic.

Starting Points

SETTING	GAIN	TREBLE	LEVEL	MODE
Clean Boost	8 %	50 %	4 dB	Classic
Always On	15 %	55 %	0.8 dB	Classic
Classic Drive	65 %	50 %	-2.4 dB	Classic
Warm Bright Mode	50 %	30 %	0.8 dB	Bright
Singing Lead	75 %	55 %	2 dB	Classic

Treat the Level values as starting points. Pickup output, other devices, and amp gain all affect the final volume.

Kompressor

Circuit-derived OTA pedal compressor with parallel blend, treble recovery, and pickup-tuned release



Parameters

PARAMETER	RANGE	DEFAULT
Sustain	0 – 100 %	25 %
Blend	0 – 100 %	75 %
Tone	0 – 100 %	58 %
Level	0 – 100 %	58 %
Pickup Mode	`Single Coil`, `Humbucker`	`Single Coil`

Overview

Kompressor models the analog feedback circuit of a Keeley Compressor Plus, including its LM13700 gain cell, transistor envelope detector, parallel dry path, active treble-recovery circuit, and audio-taper controls. It preserves the characteristic fast squeeze, long sustain, and program-dependent recovery of the pedal rather than translating the controls into a conventional digital threshold and ratio.

Sustain changes the OTA's available bias current, so it affects gain, compression depth, and detector behavior together. Blend moves between the uncompressed and compressed paths. Tone restores upper harmonics on the wet path, while Level controls the final analog-style output attenuation.

Pickup Mode selects the two recovery networks in the original circuit. Single Coil uses the slower recovery; Humbucker uses the quick-release path to retain definition with a hotter input.

Tips

- For a subtle always-on sound, keep Sustain below noon and use Blend to bring back the original pick transient.
- Raise Tone when heavy compression makes the attack feel too dark.
- Full Blend exposes the classic OTA squash and sustain most clearly.
- The control loop responds to input voltage like the pedal. Hot pickups will compress sooner and can produce some of the original circuit's coloration.

Limiter

Transparent stereo true-peak limiter for low-latency output protection

Parameters

PARAMETER	RANGE	DEFAULT
Ceiling	-18.0 – 0.0 dBTP	-1.0 dBTP
Release	20 – 1000 ms	120 ms

Overview

Limiter is a transparent final peak ceiling for a live rig. Place it after the amp, cabinet, compressor, and every other effect that can change level, with only the Output node after it. Any gain added downstream can exceed the chosen ceiling.

The device looks ahead by 1 ms and detects reconstructed stereo peaks at 4x resolution. It applies one shared gain envelope to both channels, preserving the stereo image. It never adds makeup gain: below Ceiling it is a unity-gain path whose only change is the reported 1 ms latency.

Ceiling sets the maximum intended true peak. Lower values cause more limiting. Release controls how quickly attenuation returns to unity after a peak. Very short releases recover quickly but may sound gritty on bass or low guitar notes.

Mix, Input, and Output controls are intentionally unavailable. A dry blend or gain after the limiter would defeat its protection. Ordinary bypass remains available and is latency-aligned by the audio engine.

Limiter or compressor?

A post-amp compressor controls perceived differences between clean, crunch, and driven sounds. Limiter has a narrower job: catch exceptional attacks, boosts, feedback onset, and scene changes before they reach the output. Use a compressor for musical level balance and Limiter as the final safety ceiling.

Watch Gain Reduction while playing. Occasional 1–3 dB movement is normal and should be difficult to hear. It should usually return to 0 dB; sustained deep reduction means the signal before Limiter is too hot or Ceiling is set too low.

Parameters

PARAMETER	RANGE	DEFAULT
Ceiling	-18 – 0 dBTP	-1 dBTP
Release	20 – 1000 ms	120 ms

Looper

Records and plays back a stereo loop. Position in the chain determines what's captured and where it's mixed back in.



Parameters

This device has no parameters.

Overview

Looper records a stereo performance and repeats it for practice, arrangement, and live layering. Tap the Looper device in Rig Mode to open its dedicated screen. The timeline shows loop length, current position, remaining time, and a separate waveform row for each active layer.

Press **Record** to arm or begin the first pass. When recording arming is enabled in Settings, the looper waits for a note above the chosen threshold so the loop starts with your playing. A second press while armed starts recording immediately. Press **Stop** to set the loop length. Use **Play** to start the loop, then **Overdub** to add another layer. A short press on **Undo** removes the latest overdub. Hold Undo for two seconds to clear the whole loop.

The Looper's chain position matters. Devices before it are recorded into the loop. Devices after it process both the loop and live playing. Only one Looper can be added to a rig.

Tips

- Place Looper near the end of the chain to capture the complete sound. Place delay or reverb after it when you want to change ambience while the loop continues.
- Let the first beat establish the beginning clearly, and stop after an exact number of bars. A clean first pass makes every overdub easier.
- Use recording arming for hands free starts. Adjust its threshold in Settings if background noise triggers recording or soft notes fail to start it.
- Keep each overdub simple and leave level space for later layers. Undo the latest layer if the arrangement becomes crowded.
- Assign looper actions to footswitches when recording while playing. The dedicated screen also maps its first switches to Record, Play, and Undo.

Low Divide

Analog-style octave divider with peak tracking, germanium clamping and cascaded polarity-switch voices

Parameters

PARAMETER	RANGE	DEFAULT
Direct Level	0 – 100 %	100 %
Oct 1	0 – 100 %	100 %
Oct 2	0 – 100 %	0 %

Overview

Analog-style octave divider with peak tracking, germanium clamping and cascaded polarity-switch voices

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Miss Conduct

Vintage 15 V flanger circuit model with short clock-controlled delay, resonant feedback, and stationary Filter Matrix



Parameters

PARAMETER	RANGE	DEFAULT
Rate	0.05 – 5 Hz	0.25 Hz
Range	0 – 100 %	85 %
Color	0 – 100 %	45 %
Level	-12 – 6 dB	0 dB
Filter Matrix	Off, On	Off

Overview

Miss Conduct is a circuit-derived model of the Electro-Harmonix Electric Mistress flanger: the vintage 15 V circuit with its short clock-controlled delay, resonant feedback, and stationary Filter Matrix mode.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Multiband Compressor

3-band multiband compressor with Linkwitz-Riley crossovers, per-band compression controls, and stereo linking

Parameters

PARAMETER	RANGE	DEFAULT
Low/Mid Crossover	50 – 500 Hz	200 Hz
Mid/High Crossover	500 – 8000 Hz	2000 Hz
Low Threshold	-60 – 0 dB	-18 dB
Low Ratio	1 – 20	4
Low Attack	0.1 – 100 ms	10 ms
Low Release	10 – 1000 ms	150 ms
Low Knee	0 – 24 dB	6 dB
Low Makeup	-12 – 24 dB	0 dB
Mid Threshold	-60 – 0 dB	-18 dB
Mid Ratio	1 – 20	4
Mid Attack	0.1 – 100 ms	10 ms
Mid Release	10 – 1000 ms	150 ms
Mid Knee	0 – 24 dB	6 dB
Mid Makeup	-12 – 24 dB	0 dB
High Threshold	-60 – 0 dB	-18 dB
High Ratio	1 – 20	4
High Attack	0.1 – 100 ms	10 ms
High Release	10 – 1000 ms	150 ms
High Knee	0 – 24 dB	6 dB

PARAMETER	RANGE	DEFAULT
High Makeup	-12 – 24 dB	0 dB
Detector Mode	`RMS`, `Peak`	`RMS`
Stereo Link	`Unlinked`, `Linked`	`Linked`

Overview

Multiband Compressor divides the signal into low, mid, and high bands, then compresses each band independently. It can control palm muted bass without dulling the entire guitar, smooth an aggressive upper midrange, or contain bright attacks while leaving the body of the note alone.

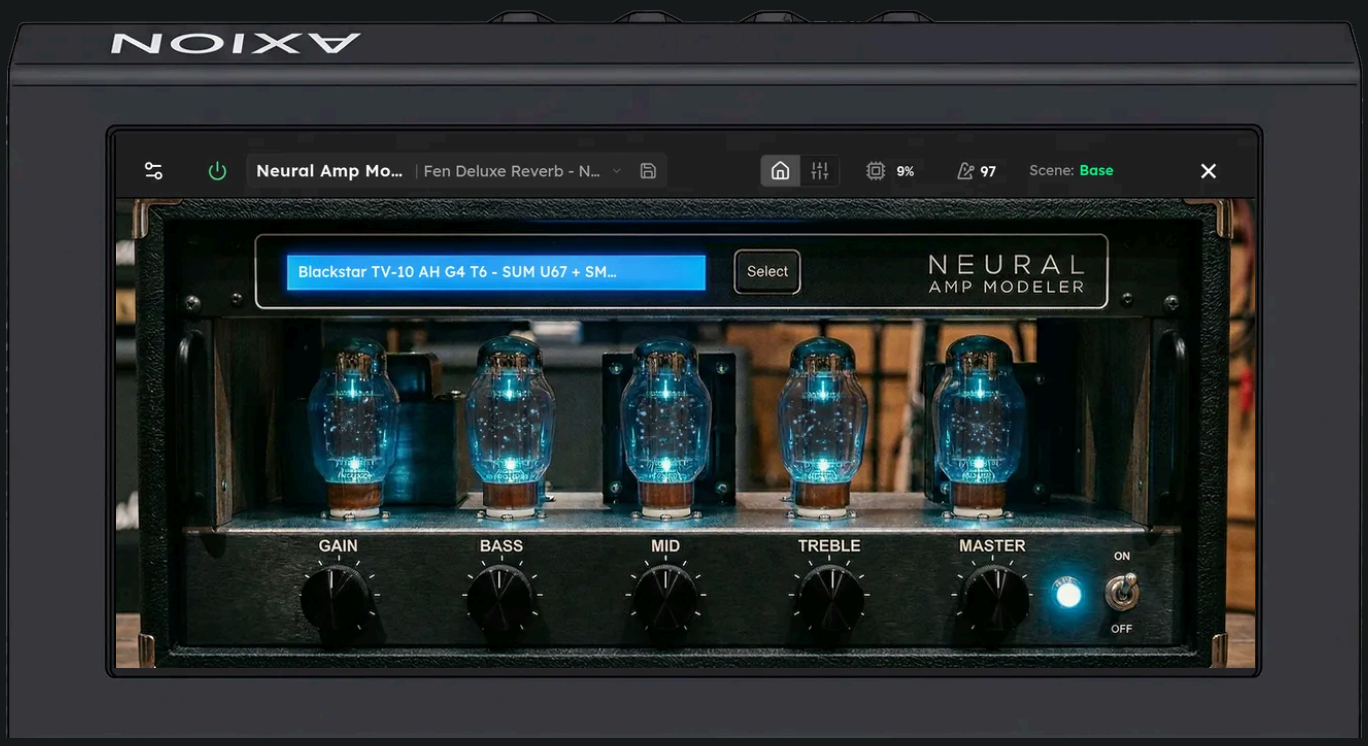
The two **Crossover** controls define the three frequency regions. Each band has its own Threshold, Ratio, Attack, Release, Knee, and Makeup controls. These work like a conventional compressor but affect only that region. `RMS` detection gives a smoother response, while `Peak` catches fast transients. `Linked` Stereo Link applies the same gain change to both channels and preserves stereo position. `Unlinked` allows independent compression.

Tips

- Start with the default crossovers, moderate ratios around 2:1 or 3:1, and no Makeup. Lower one Threshold at a time while listening to that band's effect.
- To control low end thump, lower the Low Threshold and use an Attack around 10 to 25 ms. Avoid excessive Low Makeup, which can undo the control.
- To soften pick noise, use a faster High Attack and lower High Threshold.
- Keep crossover points far enough apart to leave a useful mid band. A low crossover around 150 to 300 Hz and high crossover around 2 to 4 kHz is a practical guitar starting range.
- Use Linked mode for ordinary stereo rigs. Match output loudness with the three Makeup controls before judging whether the result is better.

Neural Amp Modeler

Neural amp modeler with integrated bass/mid/treble tone stack, HPF/LPF and pick-attack shaping. Native device backed directly by the NeuralAudio inference library.



Parameters

PARAMETER	RANGE	DEFAULT
Neural Model	File	None selected
Input Level	-20 – 20 dB	0 dB
Bass	-20 – 20 dB	0 dB
Mid	-20 – 20 dB	0 dB
Treble	-20 – 20 dB	0 dB
Bass Freq	60 – 500 Hz	250 Hz
Mid Freq	200 – 2000 Hz	800 Hz
Treble Freq	1000 – 8000 Hz	2500 Hz

PARAMETER	RANGE	DEFAULT
Pick Attack	-1 – 1	0
HPF	20 – 500 Hz	20 Hz
LPF	1000 – 20000 Hz	20000 Hz
Squish	0 – 100 %	0 %
Model Knob 1	0 – 1	0.5
Model Knob 2	0 – 1	0.5
Model Knob 3	0 – 1	0.5
Model Knob 4	0 – 1	0.5
Model Knob 5	0 – 1	0.5
Model Knob 6	0 – 1	0.5
Model Knob 7	0 – 1	0.5
Model Knob 8	0 – 1	0.5
Model Knob 9	0 – 1	0.5
Model Knob 10	0 – 1	0.5

Overview

Neural Amp Modeler loads a compatible neural model and recreates the nonlinear response captured in that file. Depending on the model, it may represent an amplifier, a drive pedal, or a complete recorded chain. Select **Neural Model** to browse the Models library, then use **Input Lvl** to hit the model at the intended strength. More input can produce more drive on models that respond dynamically.

The integrated Bass, Mid, and Treble bands provide gain and an adjustable frequency for each region. **HPF** removes low frequencies below its cutoff, and **LPF** removes highs above its cutoff. **Pick Attack** uses positive values to emphasize the transient and negative values to soften it. **Squish** evens playing dynamics while preserving the way pick strength drives the model. At 0% it is inactive.

Deep NAM models

Most models are a snapshot of an amp at one setting of its controls. A Deep NAM model is trained across the travel of the amp's own controls, so the knobs come with the model. Turning **Gain** or **Presence** on a Deep NAM model changes the sound the way that control does on the real amp, which an EQ placed after a snapshot cannot do.

You load a Deep NAM model like any other. Axion recognizes it from the file, and its presets carry the **Deep NAM** tag in the device browser. When one is loaded:

- **The power light turns purple**, so you can tell at a glance that the knobs belong to the model.
- **The knobs on the device screen change to the model's own.** They carry the model's names, for example Gain, Low, Mid, High, and Presence, and how many there are depends on the model. Bass, Mid, and Treble join them on the panel when the model has no tone controls of its own. Volume is always last.
- **The dock follows.** The model's knobs come first, so the encoders under the screen reach them.

The device's own tone and dynamics controls (Bass, Mid, Treble and their frequencies, HPF, LPF, Pick Attack, and Squish) keep working with a Deep NAM model. Any that are not on the panel are on the dock.

In the Parameters table above, **Model Knob 1** to **Model Knob 10** are the slots these knobs use. A Deep NAM model takes as many as it has knobs and shows them under its own names and scales. With a standard model they do nothing.

Choosing a model file sets its knobs to the model's own defaults. Presets and rigs restore the knob positions they were saved with.

Amp heads and cabinets

Amp presets know whether their sound already includes a speaker cabinet. A preset for an amp head on its own does not, and it needs a Cabinet after it to sound like a miked amp. When you choose an amp head preset, Axion sets that up for you:

- **No Cabinet after the amp.** A Cabinet is added right after the amp, loaded with the cabinet preset suggested for that amp.
- **Cabinet already in place.** It loads the suggested cabinet preset, and is switched back on if it was bypassed.
- **Locked Cabinet.** It is left exactly as it is. See Cabinet lock.

Each amp head preset can suggest the cabinet that suits it, for example a 4x12 for a high gain head. When a preset suggests none, Axion loads the 1x12 Deluxe.

When you choose a preset that already includes its cabinet, an unlocked Cabinet after the amp is bypassed so you do not hear two cabinets in a row. It stays in the chain and comes back when you return to an amp head preset.

Playing through a power amp and guitar cabinet

If Axion feeds a power amp and a real guitar cabinet, the real speaker does the Cabinet's job and you do not want both. Rather than bypassing or deleting the Cabinet in each rig, turn on **CAB BYPASS** on the Mixer screen. This one switch bypasses the Cabinet blocks in every rig, without changing the rigs themselves. Amp presets keep pairing with their cabinets, and the same rigs sound right through headphones or full range speakers as soon as you turn it off again.

Cab Bypass applies to every output at once, headphones included. It cannot remove a cabinet that is part of an amp preset's own sound, so use amp head presets with a power amp and guitar cabinet. See Cab Bypass for how it shows in the rig editor.

Tips

- Load the model first and begin with Input Lvl at 0 dB and all tone gains at 0 dB. Make level changes before compensating with EQ.
- If a model is too clean or too driven, adjust Input Lvl. This changes how the model responds, not just the final volume.
- Use HPF to remove rumble and LPF to tame cabinet fizz. Leave both open when the selected model already contains suitable filtering.
- Use a small positive Pick Attack value for more definition or a small negative value for softer leads. Extreme values are better suited to special effects.
- Raise Squish gradually for more consistent notes. Compare at equal loudness and stop when the playing response begins to feel too restrained.

Night Vibe

Four unequal optical phase stages with incandescent lamp dynamics and chorus/vibrato routing.

Parameters

PARAMETER	RANGE	DEFAULT
Speed	0.1 – 7 Hz	1.5 Hz
Intensity	0 – 100 %	70 %
Mode	`Chorus`, `Vibrato`	`Chorus`
Level	-24 – 12 dB	0 dB
Lamp Bias	0 – 100 %	50 %
Lamp	`Run`, `Cancel`	`Run`

Overview

Four unequal optical phase stages with incandescent lamp dynamics and chorus/vibrato routing.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Octave Forge

Transformer octave fuzz with a coupled three-transistor gain circuit and germanium full-wave rectifier.

Parameters

PARAMETER	RANGE	DEFAULT
Drive	0 – 100 %	55.00000000000001 %
Volume	0 – 100 %	65 %

Overview

Transformer octave fuzz with a coupled three-transistor gain circuit and germanium full-wave rectifier.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Octaver

Four-voice monophonic octaver (-2, -1, +1, +2) with an independent level per voice. Each voice is a tape-style pitch shift: the input is replayed at the octave ratio and spliced by whole periods of the tracked note, so the result sounds like the same guitar tuned an octave away with its dynamics intact. Tracks single notes and bass lines; on chords the tracker loses lock and the voices gate off. Octave voices sit a little behind the dry signal (a few ms on high notes, more on low E). Dry/wet mix and output volume are the engine's global device parameters.

Parameters

PARAMETER	RANGE	DEFAULT
-2 Oct	0 – 100 %	0 %
-1 Oct	0 – 100 %	0 %
+1 Oct	0 – 100 %	50 %
+2 Oct	0 – 100 %	0 %

Overview

Four-voice monophonic octaver (-2, -1, +1, +2) with an independent level per voice. Each voice is a tape-style pitch shift: the input is replayed at the octave ratio and spliced by whole periods of the tracked note, so the result sounds like the same guitar tuned an octave away with its dynamics intact. Tracks single notes and bass lines; on chords the tracker loses lock and the voices gate off. Octave voices sit a little behind the dry signal (a few ms on high notes, more on low E). Dry/wet mix and output volume are the engine's global device parameters.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Phaser

Stereo phaser effect with cascaded all-pass filters, multiple waveforms, and variable stage count

Parameters

PARAMETER	RANGE	DEFAULT
Rate	0.05 – 8 Hz	0.5 Hz
Depth	0 – 100 %	50 %
Stages	4, 6, 8, 12	4
Center Freq	300 – 3000 Hz	948.683 Hz
Feedback	0 – 100 %	50 %
Waveform	Sine, Triangle, Square, Saw Up, Saw Down	Sine
Stereo Phase	0 – 180 deg	90 deg
Mix	0 – 100 %	50 %

Overview

Phaser creates moving notches in the frequency response by combining the dry signal with a bank of modulated all pass filters. The result ranges from a gentle swirl behind clean chords to a deep, resonant sweep for funk rhythm, psychedelic leads, and animated synth textures.

Rate sets sweep speed, **Depth** sets sweep width, and **Center Freq** chooses the region around which the motion occurs. **Stages** controls the number of filter sections. More stages produce more notches and a more complex sound. **Feedback** emphasizes the notches, **Waveform** changes the motion, and **Mix** balances dry and phased signals. **Stereo Phase** offsets the right channel sweep for width.

Tips

- For a classic gentle sweep, start with 4 stages, Rate near 0.4 Hz, Depth at 50%, Feedback around 30%, and Mix near 50%.

- Add stages and Feedback for a deeper, more synthetic sound. Reduce Mix if the result overwhelms the original note.
- Lower Center Freq for a darker sweep that favors bass and rhythm parts. Raise it for more presence on guitar leads.
- Sine is the smoothest waveform. Square and saw shapes create abrupt or one direction movement that works well for special effects.
- Use a nonzero Stereo Phase for width. Return it to 0 degrees when you need a stable mono compatible image.

Plague OD

Aggressive op-amp distortion with symmetrical clipping and variable post-distortion filter



Parameters

PARAMETER	RANGE	DEFAULT
Drive	0 – 100 %	50 %
Filter	0 – 100 %	50 %
Volume	0 – 25 dB	12.5 dB

Overview

Plague OD is a faithful RAT-style distortion with a tight low end, a biting upper-mid voice, and hard symmetrical clipping. Its wide gain range moves from gritty boost and crunchy rhythm sounds to thick, compressed distortion with a fuzz-like edge.

The Filter control sits after the clipping stage, so it shapes the harmonics without changing how hard the signal is distorted. It works in the traditional direction for this circuit: low settings are bright and open, while higher settings roll off more treble for a darker, smoother sound.

Controls

- **Distortion** sets the gain feeding the clipping stage. Low settings add light grit and presence. High settings produce more compression, sustain, and saturation.
- **Filter** controls a post-distortion low-pass filter. Turn it down for more bite and harmonic detail, or turn it up to tame fizz and soften the attack.
- **Level** sets the output gain without changing the clipping character. Use it to match the bypassed volume or to push the next device or amp harder.

Tips

For a gritty boost, keep Distortion low, set Filter below halfway, and raise Level to taste. This keeps the attack clear while adding the pedal's characteristic upper-mid bite.

For classic hard rock or punk rhythm tones, start with Distortion and Filter near halfway. Lower Filter for more pick attack and cut, or raise it if the sound becomes harsh through a bright amp or cabinet.

At high Distortion settings, the response becomes more compressed and fuzz-like. Raise Filter to smooth the extra harmonics, then increase Level as needed to maintain volume.

When placing Plague OD before an already driven amp, reduce Distortion and use Level for the push. Into a clean amp, higher Distortion settings provide more of the complete distortion sound.

Starting Points

SETTING	DISTORTION	FILTER	LEVEL
Gritty Boost	10 %	20 %	4 dB
Classic Crunch	45 %	45 %	8 dB
Punk Rhythm	60 %	30 %	10 dB
Thick Lead	70 %	65 %	12 dB
Fuzz-Like Sustain	90 %	55 %	18 dB

Treat the Level values as starting points. Pickup output, other devices, and amp gain all affect the final volume.

Pug Blast

Aggressive op-amp distortion with symmetrical clipping and variable post-distortion filter



Parameters

PARAMETER	RANGE	DEFAULT
Distortion	0 – 100 %	50 %
Filter	0 – 100 %	50 %
Level	-12 – 12 dB	0 dB

Overview

Pug Blast is an aggressive hard clipping distortion with a tight input and a variable filter after the clipping stage. **Distortion** moves from gritty drive to dense compression and sustain. **Filter** removes high frequencies after distortion, with higher values producing a darker sound. **Level** sets the output from 12 dB of cut to 12 dB of boost.

Because the filter follows the clipping stage, it changes the amount of edge without changing how hard the signal clips. This makes Pug Blast useful for raspy garage rhythm, singing leads, and heavy sounds that need a simple way to control fizz.

Tips

- For a balanced starting point, set Distortion to 45%, Filter to 50%, and Level to 0 dB.
- For articulate rhythm, keep Distortion below halfway and use a lower Filter value for more bite.
- For a thick lead, raise Distortion above 65% and turn Filter upward until the top end sits behind the note.
- Set the Filter while listening through the final cabinet or impulse response. Cabinet choice strongly affects how much high frequency filtering is useful.
- Use Level to match bypass before comparing tones, or add a few decibels when the device is assigned as a solo sound.

Quack

Circuit-derived vintage optical envelope filter with three physical filter taps and up/down sweep.

Parameters

PARAMETER	RANGE	DEFAULT
Gain	0 – 100 %	50 %
Peak	0 – 100 %	65 %
Filter	`Low Pass`, `Band Pass`, `High Pass`	`Low Pass`
Sweep	`Up`, `Down`	`Up`
Range	`Low`, `High`	`Low`
Level	-24 – 6 dB	-6 dB

Overview

Circuit-derived vintage optical envelope filter with three physical filter taps and up/down sweep.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Signal 8-Band EQ

8-band parametric equalizer with soft clipping



Parameters

PARAMETER	RANGE	DEFAULT
Band 1 Frequency	20 – 20000	100
Band 1 Q	0.1 – 10	0.707
Band 1 Gain	-12 – 12	0
Band 1 Enable	0 – 1	0
Band 0 Type	0 – 4	0
Band 2 Frequency	20 – 20000	250
Band 2 Q	0.1 – 10	0.707
Band 2 Gain	-12 – 12	0
Band 2 Enable	0 – 1	0

PARAMETER	RANGE	DEFAULT
Band 1 Type	0 – 4	0
Band 3 Frequency	20 – 20000	500
Band 3 Q	0.1 – 10	0.707
Band 3 Gain	-12 – 12	0
Band 3 Enable	0 – 1	0
Band 2 Type	0 – 4	0
Band 4 Frequency	20 – 20000	1000
Band 4 Q	0.1 – 10	0.707
Band 4 Gain	-12 – 12	0
Band 4 Enable	0 – 1	0
Band 3 Type	0 – 4	0
Band 5 Frequency	20 – 20000	2000
Band 5 Q	0.1 – 10	0.707
Band 5 Gain	-12 – 12	0
Band 5 Enable	0 – 1	0
Band 4 Type	0 – 4	0
Band 6 Frequency	20 – 20000	4000
Band 6 Q	0.1 – 10	0.707
Band 6 Gain	-12 – 12	0
Band 6 Enable	0 – 1	0
Band 5 Type	0 – 4	0
Band 7 Frequency	20 – 20000	8000
Band 7 Q	0.1 – 10	0.707
Band 7 Gain	-12 – 12	0
Band 7 Enable	0 – 1	0
Band 6 Type	0 – 4	0

PARAMETER	RANGE	DEFAULT
Band 8 Frequency	20 – 20000	16000
Band 8 Q	0.1 – 10	0.707
Band 8 Gain	-12 – 12	0
Band 8 Enable	0 – 1	0
Band 7 Type	0 – 4	0
Volume	0 – 2	1
Clipper Enable	0 – 1	0
Clipper Amount	0 – 1	0

Overview

Signal 8 Band EQ provides eight independently enabled filters for corrective or creative tone shaping. Each band can be a peaking filter, low pass, high pass, low shelf, or high shelf. **Frequency** chooses the center or cutoff, **Q** sets the width or resonance, and **Gain** sets boost or cut for peaking and shelf types.

Use broad, small adjustments to balance a complete rig, or a narrow Q to find and reduce a resonance. **Volume** sets the output after all eight bands. **Clipper Enable** turns on a soft output clipper, and **Clipper Amount** sets its drive. The clipper can round occasional peaks, but it is also a nonlinear effect and should not replace sensible gain staging.

Tips

- Enable only the bands you need. Begin with Q around 0.7 to 1.4 and make small cuts or boosts of 1 to 3 dB.
- To find an unpleasant resonance, temporarily boost a narrow band and sweep Frequency. Once found, change the boost to a modest cut and widen Q if the result sounds unnatural.
- Use a high pass filter to remove rumble and a low pass filter to tame excess fizz. Set both while listening through the final output system.
- Use shelves for broad tonal balance and peaking bands for localized changes.
- Leave the clipper off while setting the EQ. Enable it afterward and raise Clipper Amount only until occasional peaks are controlled.

Simple Compressor

Stereo compressor with RMS/Peak detection, soft knee, and linked/unlinked stereo operation

Parameters

PARAMETER	RANGE	DEFAULT
Threshold	-60 – 0 dB	-18 dB
Ratio	1 – 20	4
Attack	0.1 – 100 ms	10 ms
Release	10 – 1000 ms	150 ms
Knee	0 – 24 dB	6 dB
Makeup Gain	-12 – 24 dB	0 dB
Detector Mode	`RMS`, `Peak`	`RMS`
Stereo Link	`Unlinked`, `Linked`	`Linked`

Overview

Simple Compressor reduces level when the signal rises above **Threshold**. It evens out clean playing, adds sustain, controls sharp peaks, and can make a rig sit at a more consistent level. **Ratio** controls how strongly signals above the threshold are reduced. **Attack** decides how much of the initial pick can pass before compression, while **Release** controls how quickly gain recovers.

Knee softens the transition into compression. A wider knee sounds more gradual, while 0 dB gives a firm transition at the threshold. **Makeup Gain** restores level after reduction. `RMS` Detector Mode follows average energy for a smoother musical response. `Peak` reacts more directly to transients. With Stereo Link set to `Linked`, both channels receive the same gain change and the stereo image remains steady. `Unlinked` lets each channel react independently.

Tips

- For transparent leveling, start at a 3:1 ratio, 15 ms Attack, 150 ms Release, and 6 dB Knee. Lower Threshold until louder notes compress gently.
- For more pick definition, lengthen Attack. For tighter peak control, shorten it.
- If the level pumps after each chord, lengthen Release or reduce Ratio. If the compressor stays clamped between notes, shorten Release or raise Threshold.
- Use Linked mode on stereo rigs unless independent channel movement is an intentional effect.
- Add Makeup Gain only after setting Threshold and Ratio, and compare against bypass at similar loudness.

Slaghammer

Heavy distortion with a coupled transistor preamp, asymmetric clipping, germanium crossover gate and interacting three-gyrator Color Mix EQ.



Parameters

PARAMETER	RANGE	DEFAULT
Level	0 – 100 %	60 %
Low	0 – 100 %	50 %
High	0 – 100 %	50 %
Distortion	0 – 100 %	75 %

Overview

Heavy distortion with a coupled transistor preamp, asymmetric clipping, germanium crossover gate and interacting three-gyrator Color Mix EQ.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Song Input

Play song tracks through devices at any point in a chain.

Song Input adds audio from the song view at its position in a rig's chain. Incoming chain audio passes through, and devices after Song Input process both signals.

Route a track

1. Add **Song Input** from the **Utility** device category to a chain. Each new instance receives a numbered label, such as **Song Input 1**.
2. Open a song in the song view.
3. Use the track's **Output** selector to choose that Song Input.

For example, placing Song Input before an amp and reverb lets a recorded guitar DI pass through both. Moving Song Input after the amp sends the recording through just the reverb. Multiple tracks can use the same Song Input; their audio is mixed before downstream processing. Multiple Song Input devices can be placed in the rig.

Main plays a track directly through the song playback mix. Choosing a Song Input removes that track from the direct mix. Track gain, mute and solo apply before the audio reaches Song Input. Mono recordings feed both sides; stereo recordings retain their left and right channels.

Bypass and missing destinations

Bypassing Song Input stops its playback contribution while passing incoming chain audio through. Stopping the song stops its feed, allowing downstream delay and reverb tails to finish.

Track destinations are saved in `song.json`; Song Input devices are saved with the rig. Save the rig as well as the song settings. Moving a device within the rig preserves its routing. If the current rig does not contain the assigned device, the track shows **Output: Unavailable** and its playback is silent. Load the matching rig or select another destination. Audio never switches to Main automatically.

Song Input uses the song transport and adds no extra playback block of latency. Effects after it can introduce their own latency; this feature does not add compensation between the direct song mix and processed tracks.

Stack

A stack of chains in one block. Add chains in parallel, then use Mix to blend their combined return with the original signal.

Parameters

This device has no parameters.

Overview

A stack of chains in one block. Add chains in parallel, then use Mix to blend their combined return with the original signal.

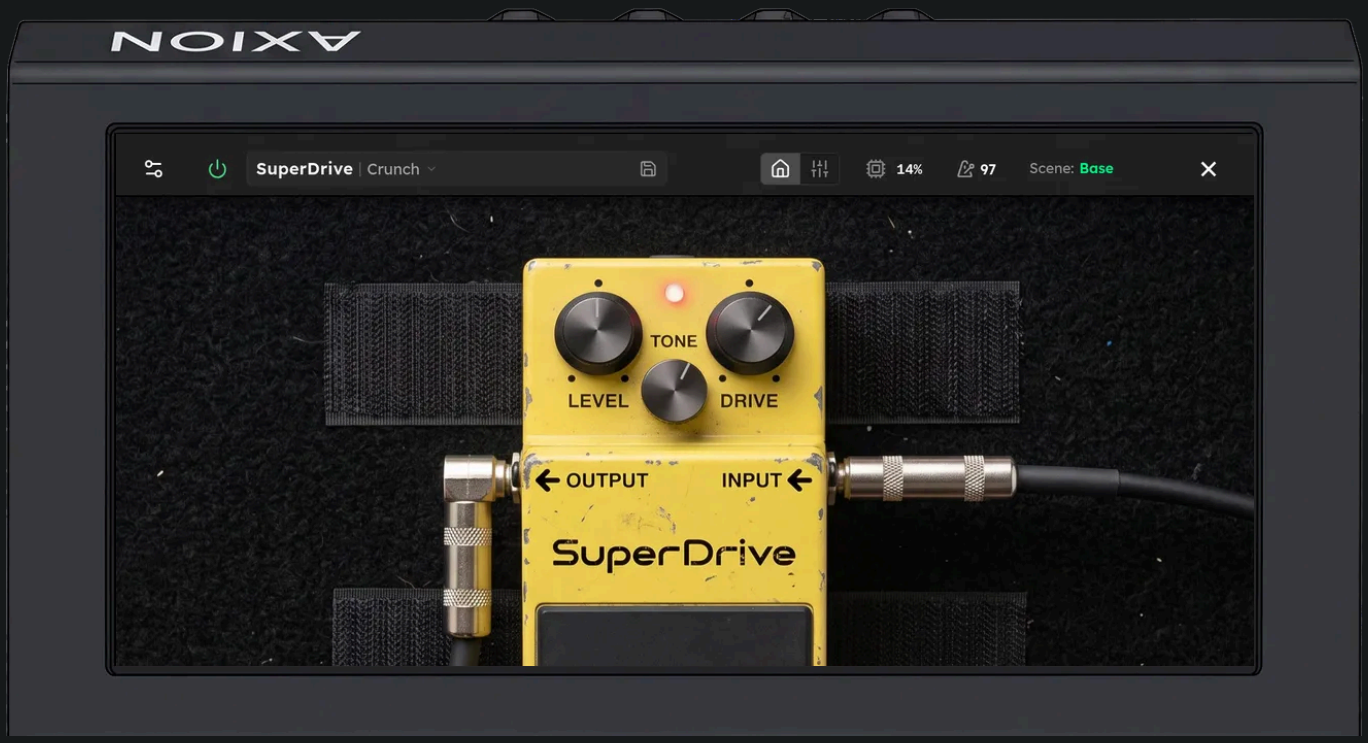
Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

SuperDrive

Asymmetric feedback clipping overdrive with mid-forward voicing and even-order harmonics



Parameters

PARAMETER	RANGE	DEFAULT
Level	0 – 12 dB	6 dB
Tone	0 – 100 %	50 %
Drive	0 – 100 %	50 %

Overview

SuperDrive is a mid focused overdrive with asymmetric feedback clipping. One side of the waveform clips sooner than the other, adding even harmonics and a slightly rougher texture than a symmetrical overdrive. The filtered low end keeps the sound focused, making the device useful for rock rhythm, singing leads, and tightening an already driven amp.

Drive increases feedback gain and clipping. **Tone** controls a filter after the distortion, moving from dark at low values to bright at high values. **Level** adds up to 12 dB at the output for volume matching or for pushing the next stage.

Tips

- For a focused boost, keep Drive below 20%, set Tone near 50%, and raise Level until the amp responds with the desired extra gain.
- For classic mid focused rhythm, start with Drive around 45% and adjust Tone with the full cabinet sound active.
- For a lead sound, raise Drive above 60% and add a modest Level boost. Lower Tone if the upper harmonics become sharp.
- Place SuperDrive before an amp to shape and tighten its distortion. Place it after a clean amp model when you want more of the pedal's own clipping character.
- Match loudness before comparing Drive settings because the Level control can otherwise mask changes in compression and texture.

Swell

Automatic volume swell: suppresses the pick transient and blooms each note in. Works like an inverse noise gate, giving hands-free violining without a volume pedal.

Parameters

PARAMETER	RANGE	DEFAULT
Sensitivity	0 – 100 %	55 %
Attack	0 – 4000 ms	1000 ms

Overview

Swell automatically fades in each new note by suppressing its pick transient. It creates bowed or volume pedal style attacks without requiring a pedal, and is useful for ambient pads, soft chord entrances, and layered lead lines.

Sensitivity determines how easily a new note triggers the envelope. Raise it for quieter pickups or gentle playing. Lower it if noise or ringing strings retrigger the swell. **Attack** sets the fade in time. Short values soften the pick while keeping the phrase immediate. Long values create a pronounced, slow bloom.

Tips

- For a gentle attack softener, try an Attack of 150 to 350 ms and set Sensitivity just high enough to catch normal picking.
- For violin style swells, start around 800 to 1500 ms. Play cleanly and mute unused strings so each intended note starts a fresh envelope.
- Put delay and reverb after Swell so the faded note feeds clear repeats and a continuous tail.
- Compression before Swell can make triggering more consistent, but too much compression may cause quiet noise to retrigger the envelope.
- If notes fail to open, raise Sensitivity. If the swell starts before each new pick, let the previous note decay further or reduce Sensitivity.

Tape Delay

Stereo tape echo with tempo sync, repeat saturation, age-dependent bandwidth, wow, flutter, and stereo head spread

Parameters

PARAMETER	RANGE	DEFAULT
Time	40 – 2000 ms	420 ms
Sync	Off, On	Off
Division	1/1, 1/2, 1/4, 1/8, 1/16, 1/4 D, 1/8 D, 1/16 D, 1/4 T, 1/8 T, 1/16 T	1/4
Feedback	0 – 100 %	42 %
Wow	0 – 100 %	28 %
Flutter	0 – 100 %	12 %
Color	0 – 100 %	55 %
Age	0 – 100 %	35 %
Saturation	0 – 100 %	30 %
Stereo	0 – 100 %	60 %
Level	-12 – 6 dB	0 dB

Overview

Stereo tape echo with tempo sync, repeat saturation, age-dependent bandwidth, wow, flutter, and stereo head spread

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

The Regent

Amp-in-a-box distortion modelled on the Marshall Guv'nor. Op-amp soft clipping into silicon diode shunt with a passive Marshall-style 3-band tone stack.

Parameters

PARAMETER	RANGE	DEFAULT
Gain	0 – 1	0.5
Bass	0 – 1	0.5
Middle	0 – 1	0.5
Treble	0 – 1	0.5
Level	0 – 1	0.5

Overview

The Regent is an amp style distortion with a responsive gain stage and a three band passive tone stack. It covers low gain crunch, classic rock rhythm, and saturated lead sounds while offering more tonal control than a typical three knob drive pedal.

Gain sets the amount of clipping and sustain. **Bass**, **Middle**, and **Treble** interact like the controls on an amplifier, so changing one band can affect the apparent balance of the others. **Level** sets the final output. Use The Regent into a clean amp model as the main source of drive, or before an already driven amp for another layer of gain and tone shaping.

Tips

- For open crunch, start with Gain around 35%, Bass at 45%, Middle at 60%, and Treble at 55%.
- For a thicker lead, raise Gain and Middle, then trim Bass if the low strings lose definition.
- For a scooped heavy rhythm sound, lower Middle modestly rather than removing it completely. Keep enough midrange for the guitar to remain audible in a mix.
- Set the tone stack with the final cabinet active because cabinet voicing can change the useful range of Bass and Treble.

- Match Level with bypass while dialing the sound, then add only the boost needed for the part.

Transpose

Polyphonic pitch shifter: moves everything you play, chords included, up or down by up to two octaves. With Step on it retunes the guitar (Eb or D standard, an octave down for bass lines); with Step off the shift is continuous and sweeps like a whammy from an expression pedal. Shifting down, pick attacks arrive about 2-4 ms behind the dry signal. Shifting up, they arrive later, and more so with a larger Buffer: about 7-12 ms on 20 ms, up to about 20 ms on 60 ms. A larger Buffer splices less often and handles lower notes cleanly (20 ms for guitar, 30 ms and up for bass). Tonality lets the dry signal through unshifted above its frequency, keeping pick and string noise natural; at 20 kHz it is off. Dry/wet mix and output volume are the engine's global device parameters.

Parameters

PARAMETER	RANGE	DEFAULT
Pitch	-24 – 24 st	-2 st
Step	Off, On	On
Tonality	1000 – 20000 Hz	20000 Hz
Buffer	20 ms, 30 ms, 40 ms, 60 ms	30 ms

Overview

Polyphonic pitch shifter: moves everything you play, chords included, up or down by up to two octaves. With Step on it retunes the guitar (Eb or D standard, an octave down for bass lines); with Step off the shift is continuous and sweeps like a whammy from an expression pedal. Shifting down, pick attacks arrive about 2-4 ms behind the dry signal. Shifting up, they arrive later, and more so with a larger Buffer: about 7-12 ms on 20 ms, up to about 20 ms on 60 ms. A larger Buffer splices less often and handles lower notes cleanly (20 ms for guitar, 30 ms and up for bass). Tonality lets the dry signal through unshifted above its frequency, keeping pick and string noise natural; at 20 kHz it is off. Dry/wet mix and output volume are the engine's global device parameters.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Tremolo

Amplitude modulation effect with LFO, multiple waveforms, stereo phase offset, and tempo sync

Parameters

PARAMETER	RANGE	DEFAULT
Rate	0.1 – 20 Hz	5 Hz
Depth	0 – 100 %	60 %
Waveform	`Sine`, `Triangle`, `Square`, `Saw Up`, `Saw Down`	`Sine`
Stereo Mode	`Mono`, `Phase Offset`	`Mono`
Stereo Phase	0 – 180 deg	90 deg
Tempo Sync	`Free Run`, `Tempo Sync`	`Free Run`
Division	`Whole`, `Half`, `Quarter`, `Eighth`, `Sixteenth`, `Half Dotted`, `Quarter Dotted`, `Eighth Dotted`, `Half Triplet`, `Quarter Triplet`, `Eighth Triplet`	`Quarter`
Output Level	-12 – 12 dB	0 dB

Overview

Tremolo changes volume in a repeating pattern. It can add a gentle pulse to clean chords, create a hard rhythmic chop, or move the level differently in the left and right channels for a wide stereo effect.

Depth controls the amount of volume change and **Waveform** controls its shape. With **Tempo Sync** set to `Free Run`, **Rate** sets speed in hertz. Set Tempo Sync to `Tempo Sync` and **Division** replaces Rate as the timing control, locking the pulse to Axion's current tempo with straight, dotted, or triplet values. In `Mono` Stereo Mode both channels use the same modulation. In `Phase Offset` mode, **Stereo Phase** delays the right channel's cycle relative to the left. **Output Level** compensates for perceived volume loss or gain.

Tips

- For a traditional amp tremolo, use Sine, Mono mode, Depth around 50%, and a Rate between 3 and 5 Hz.

- For a tempo locked pulse, enable Tempo Sync and begin with Quarter or Eighth Division. Dotted and triplet divisions create less obvious repetition.
- For a hard chop, select Square and raise Depth. Use Output Level to restore the average volume if needed.
- For stereo movement, select Phase Offset and start at 90 degrees. Higher values make the channels alternate more strongly.
- Saw Up and Saw Down create directional ramps. They are useful for rhythmic effects that should feel as if they rise or fall into each beat.

Valve

Circuit-modeled two-triode valve preamp with passive tone stack, compiled ahead of time from its schematic by the Valve circuit compiler. Preamp only: follow it with a cab/IR. Experimental.

Parameters

PARAMETER	RANGE	DEFAULT
Drive	0 – 1	0.5
Bass	0 – 1	0.5
Mid	0 – 1	0.5
Treble	0 – 1	0.5

Overview

Valve is a tube preamp built from a schematic instead of a capture. The circuit is two 12AX7 triode gain stages with a Bassman-style passive tone stack between them: first triode, then the tone stack, then Drive, then the second triode. Axion solves that circuit while you play, so the controls behave like the pots on a real amp. They interact with each other and with how hard you hit the strings.

Valve is a preamp only. It has no power amp, output transformer or speaker. Put a Cabinet or Impulse IR after it, or the sound will be thin and fizzy.

Valve is **experimental**. Its sound may change in future updates, and rigs saved with it may not sound exactly the same after an update.

Controls

- **Drive** sets how hard the tone stack's output drives the second triode. It is modeled as an audio-taper pot, as on a real amp. Low settings give a clean tone with a little warmth. As you turn it up, the second triode starts to compress and then clips.
- **Bass**, **Mid** and **Treble** are the three pots of a passive tone stack. Like a real passive stack, they cut rather than boost, and they interact. Changing Bass or Treble also changes how much mid you hear. With all three at 0 the stack is close to silent. Because the stack comes before the second triode, the tone controls also shape what gets distorted, not just the final EQ.

Every control runs from 0 to 1, the pot's position from fully left to fully right.

Presets

PRESET	DRIVE	BASS	MID	TREBLE	SOUND
Warm Preamp (default)	0.25	0.5	0.5	0.5	Near clean, with soft compression and a rounded top end
High Drive	1.0	0.7	0.3	0.8	Thick, saturated crunch with scooped mids, a tight low end and a cutting top

When you add Valve to a chain it loads **Warm Preamp**, so Drive starts at 0.25 rather than the 0.5 shown in the table above.

Tips

- Set the cabinet first. Valve sounds harsh straight into the outputs, and most tone decisions depend on the cab or IR after it.
- For a clean tone that still feels like a tube amp, keep Drive below about 0.3 and dig in with your pick. The circuit responds to playing dynamics.
- Use an overdrive pedal before Valve, with its own gain low and its level up, to push the triodes harder and tighten the low end.
- If the tone gets muddy at high Drive, lower Bass. Less bass going into the second triode tightens the distortion, which an EQ after the amp cannot do.
- Drive changes the output level a lot. Use the device's output volume to level-match when you compare settings or switch between rigs.

Velvet Break

Circuit model of the original pre-1995 two-stage Bluesbreaker overdrive.

Parameters

PARAMETER	RANGE	DEFAULT
Drive	0 – 100 %	50 %
Tone	0 – 100 %	50 %
Volume	0 – 100 %	75 %

Overview

Circuit model of the original pre-1995 two-stage Bluesbreaker overdrive.

Use the parameter table above to see the available controls, ranges, and defaults. Device behavior and useful settings depend on its position in the signal chain and on the devices before and after it.

Tips

- Begin with the default settings and adjust one control at a time.
- Compare the device with bypass at a similar output level.
- Save a device preset when you find a setting you want to reuse.

Wah

Crybaby-style resonant-bandpass wah with adjustable heel/toe range and resonance

Parameters

PARAMETER	RANGE	DEFAULT
Position	0 – 100 %	50 %
Heel	200 – 800 Hz	400 Hz
Toe	1000 – 3500 Hz	2200 Hz
Resonance	0 – 100 %	50 %
Level	-12 – 12 dB	0 dB
Mode	`Pedal`, `Auto`	`Pedal`
Sensitivity	0 – 100 %	50 %
Attack	1 – 100 ms	10 ms
Release	20 – 1000 ms	120 ms
Toe Off	`Off`, `On`	`Off`

Overview

Wah is a resonant filter whose center frequency moves between the **Heel** and **Toe** frequencies. The moving peak creates the familiar vocal sweep used for funk rhythm, expressive lead lines, and filtered clean textures. **Resonance** sets how broad or sharp that peak sounds, while **Level** trims the output.

Pedal and Auto modes

Set **Mode** to `Pedal` for direct control. **Position** moves from heel at 0% to toe at 100%, and is the parameter to assign to an expression pedal. Adjust Heel and Toe to keep the sweep in the useful range for your guitar and rig.

Set **Mode** to `Auto` to enable auto wah. In this mode the input envelope controls the sweep, so harder notes move toward Toe and decaying notes return toward Heel. Position is ignored. **Sensitivity** determines how strongly the wah reacts to input level. **Attack** sets how quickly it opens, and **Release** sets how quickly it closes.

Increase Sensitivity for quiet pickups or a wider sweep. Reduce it if every note jumps immediately to Toe.

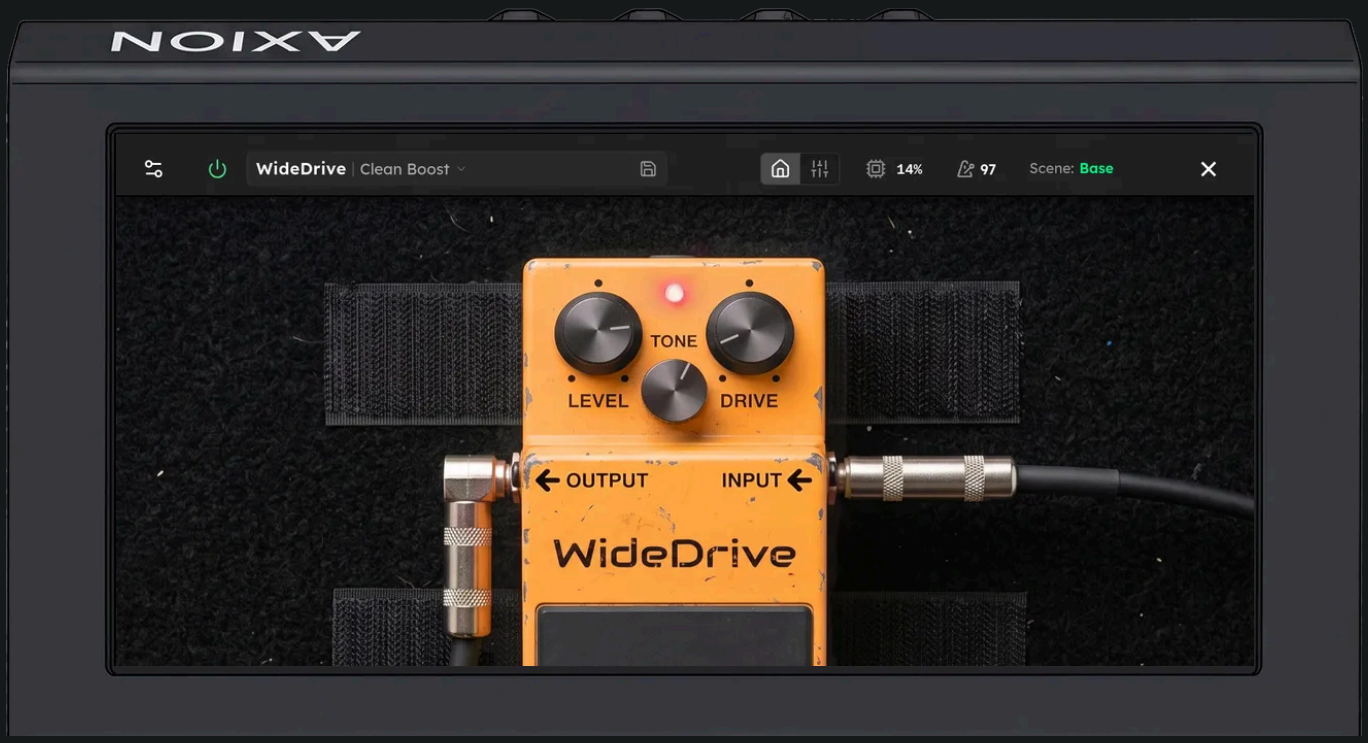
Toe Off applies only in Pedal mode. When it is `On`, the useful sweep is compressed into the first 80% of pedal travel. Holding Position at the toe for half a second cuts the wah to the dry signal, like the switch under a physical wah treadle. Rock the pedal back to engage the wah again. Leave Toe Off `Off` when you want the full expression pedal travel to control the sweep without the toe switch behavior.

Tips

- For a traditional pedal wah, start with Heel at 400 Hz, Toe at 2200 Hz, Resonance at 50%, and assign the expression pedal to Position.
- For funk rhythm, use Auto mode with a fast Attack and a Release around 100 to 180 ms. Set Sensitivity so ordinary accents open most of the sweep.
- For slower synth like filtering, lengthen Release to 300 ms or more and lower Resonance for a rounder movement.
- Put Wah before distortion for the classic focused sweep. Put it after distortion for a broader and more pronounced filter effect.
- Match Level with bypass after setting Resonance. A narrow peak can sound louder even when the measured output is similar.

WideDrive

Dual-stage wide-range overdrive with extended low-end response and smooth symmetrical clipping



Parameters

PARAMETER	RANGE	DEFAULT
Level	-20 – 12 dB	6 dB
Tone	0 – 100 %	50 %
Drive	0 – 100 %	50 %

Overview

WideDrive is a broad frequency overdrive built around two clipping stages. It keeps more low end than a strongly mid focused drive while producing smooth, symmetrical saturation. That combination works well for full chords, thick single note lines, and bass friendly drive sounds.

Drive controls the gain and distortion intensity of the clipping circuit. **Tone** moves from warmer to brighter voicing. **Level** spans substantial cut and boost, which makes it possible to match a dense high Drive sound or use a lower Drive setting to push the next device.

Tips

- For a broad rhythm sound, start with Drive around 40%, Tone near 50%, and adjust Level to match bypass.
- For a cleanish boost with body, keep Drive below 20% and raise Level by a few decibels.
- For a thick lead, set Drive between 60% and 75%, then raise Tone only as much as needed to restore articulation.
- If the low end becomes crowded, reduce bass in the amp or add an EQ after WideDrive. The device is intentionally fuller than many mid focused drives.
- Set Tone while monitoring through the final cabinet or impulse response, since that stage determines much of the audible top end.

07

TROUBLESHOOTING

When something is wrong.

Troubleshooting

Audio dropouts, MIDI, networking, recovery.

If something is not working as expected, start with the checks in the relevant section. Change one thing at a time and retest with a simple factory rig. That makes it easier to separate a connection problem from a device or rig problem.

No audio or no output signal

1. Turn the monitoring level down, then confirm the instrument, output, and power connections. Passive guitar and bass pickups belong on the instrument input side of IN 1 or IN 2. See Connections.
2. Open the menu and choose **Mixer**. Play and watch the meter for the input you connected. Raise that input gain only as needed. If there is no input activity, try another cable and instrument before changing the rig.
3. Check the meter for the physical output you connected. Raise its level and the external speaker, mixer, or interface level gradually.
4. In Rig Mode, confirm the chain's IN and OUT blocks are active. A bypassed endpoint dims the whole chain and shows a warning triangle.
5. Confirm that required devices are not bypassed. For an FX Loop, also confirm that the selected return is connected and producing signal.
6. Load a factory rig. If the factory rig works, the problem is in the original rig. Bypass or remove its devices one at a time until audio returns.

If the input meter moves but the output meter does not on a simple factory rig, note the rig name and the last action you took before reporting the problem.

Clicks, pops, or dropouts

The CPU meter at the top of the screen shows how much audio processing time is in use. A dropout occurs when the audio thread misses its deadline.

- Remove or bypass the most demanding devices first. Amp models, convolution, cabinets, and long reverbs generally cost more than boosts and simple EQ.
- If the problem began after adding a third party plugin, test the same rig without it. Third party plugins do not all meet Axion's audio deadline.
- Close unnecessary editing activity and test without rapidly moving several parameters at once.

- Open **Settings**, choose **Audio Engine**, and select a larger supported buffer size when the rig must remain complex. The new size takes effect after the next Axion startup and increases latency.
- Test a factory rig. Dropouts on every factory rig point to a system or audio hardware problem rather than an overloaded user rig.

MIDI does not select the expected rig

- Confirm the controller's MIDI Out is connected to Axion MIDI In and that both devices are powered.
- Send Bank Select before Program Change. The bank chooses the set list and the program number chooses the rig within it.
- Remember that bank 0 is reserved for factory rigs. User set lists begin at bank 1.
- Check whether the controller displays MIDI values from 1 to 128 while sending protocol values from 0 to 127. An apparent one slot offset usually comes from that display convention.
- Test with one direct controller connection and a simple program change before adding MIDI routing through a DAW or another device.

For the addressing model, see Rigs and Set Lists.

WiFi or companion app connection problems

Open **Settings** and select **General**. The page shows the current Device IP and WiFi state.

1. Confirm WiFi is on and select the intended network. Secured networks prompt for their password. A saved network shows a **Forget** button; use it to drop a stale or mistyped password and join again from scratch.
2. Confirm the computer and Axion are on the same local network. Guest networks often prevent devices from talking to one another.
3. Enter the Device IP shown by Axion in the companion app. If discovery by name fails, the numeric IP avoids local name resolution problems.
4. If the IP changes after reconnecting or restarting the router, use the new value shown in Settings.
5. Temporarily disable a computer VPN or restrictive firewall only long enough to determine whether it blocks local connections, then restore the security setting and add a narrow local network exception if needed.

Software update and recovery

Use **Settings**, then **Software Update** to check for and install an update. Keep Axion connected to power while it downloads and restarts. The update system returns to the current working version automatically if installation does not complete successfully.

If the interface stops responding but audio continues, avoid repeated control changes and allow any visible update or reset operation to finish. If Axion does not return to its normal screen after a restart, record what appears on the display and report it. There is no user data recovery operation that requires opening the unit.

Factory reset

Factory reset is a last resort for damaged user content or configuration. It does not remove the factory amps, cabinets, impulse responses, or effects.

Open **Settings**, choose **System**, select **Factory Reset**, read the warning, and confirm. The audio engine restarts while the interface remains open.

Factory reset permanently erases all user set lists, rigs, saved device presets, captures, and songs. It cannot be undone. Export anything you need before resetting.

Report a bug

Post it in Bug Reports on Severed Backstage and include:

- The build number from **Settings**, **System**, **About**
- Clear steps that reproduce the problem
- What you expected and what happened instead
- The rig, device, or plugin involved, including whether it is first party or third party
- Photos, a short recording, or exported rig data when they help demonstrate the issue and contain no private information

For audio problems, say whether input and output meters moved, whether a factory rig worked, and whether the CPU meter was near its limit. For network problems, include the connection method and error message, but never post a WiFi password or other credential.

If the problem involves anything private, such as details of your network, contact support privately instead.

APPENDIX A

Glossary

Terms used throughout this manual.

Terms used throughout this manual. Cross-references to the longer explanations are linked.

A

Amp simulator: A device that emulates the sound of a guitar amplifier. See Devices.

B

Bank: A MIDI bank number (0–127). Each set list maps to one bank. Bank 0 is reserved for factory rigs.

Buffer: A small chunk of audio samples processed at once. See Troubleshooting.

Bypass: Disable a device without removing it from the chain. The device passes audio through unchanged.

C

Cabinet (cab): A device that simulates a guitar speaker cabinet, typically the last stage in an amp-style chain.

Category: A grouping of devices (Drive, Reverb, etc.). See Devices.

Chain: See Signal chain.

Control Mappings: The screen that lists a rig's Rig Controls, expression pedal mappings, and MIDI mappings in one place. See Rig Controls.

Control source: Anything that drives a parameter: a knob (a Rig Control), the expression pedal, or a MIDI control change.

D

Device: A single audio processor in the signal chain, such as an amp, drive, cab, EQ, modulation, delay, or reverb. See Devices.

Dropout: A click, pop, or brief silence in the audio caused by the audio thread missing its deadline. See Troubleshooting.

E

Encoder: A rotary control on the front panel used to adjust parameters. Axion has 5 encoders.

Enum parameter: A parameter that picks from a fixed list of options (e.g. `Symmetrical` / `Asymmetrical`).

F

Factory rig: A rig that ships with Axion, in bank 0. Not editable in place; load and save-as to modify.

Footswitch: A foot-operated switch on the unit, typically used to load rigs, toggle device bypass, or trigger tap tempo. Axion has 5 footswitches.

I

IR (Impulse Response): A short audio recording that captures the acoustic signature of a speaker cabinet, room, or other space. Used by convolution cab devices to model real cabinets.

L

Latency: The time between picking a note and hearing it from the output. See Troubleshooting.

LV2: An open standard for audio plugins on Linux. Axion supports LV2 plugins as third-party devices.

M

Macro: A Rig Control that turns more than one parameter at once. The knob reads 0 to 100 %, and each target keeps its own range and curve. See Rig Controls.

MIDI: Musical Instrument Digital Interface. Used to remotely control Axion, typically with Bank Select to choose a set list and Program Change to choose a rig within it.

P

Parameter: One adjustable value on a device, such as a knob, switch, or selector.

Preset: In some contexts, the same as a rig. The term "rig" is preferred in this manual for the saved sound as a whole; "preset" sometimes refers to a saved parameter snapshot for a single device.

Program Change: A MIDI message (0–127) that selects a rig within the active set list. See Rigs and Set Lists.

R

Real-time: Audio processing that meets a strict per-buffer deadline. See Troubleshooting.

Rig Control: A binding from a dock knob to a device or system parameter, saved with the rig. Assign several parameters to one knob to make a macro. See Rig Controls.

Rig: One complete saved sound: the full chain of devices and all their parameter values. See Rigs and Set Lists.

S

Sample rate: The number of audio samples processed per second. Axion runs at 48 kHz.

Set list: A named, ordered collection of rigs, mapped to a MIDI bank. See Rigs and Set Lists.

Signal chain: The ordered series of devices that audio passes through, from input to output. See The Signal Chain.

V

VST3: A widely-used cross-platform audio plugin format. Axion supports VST3 plugins as third-party devices.