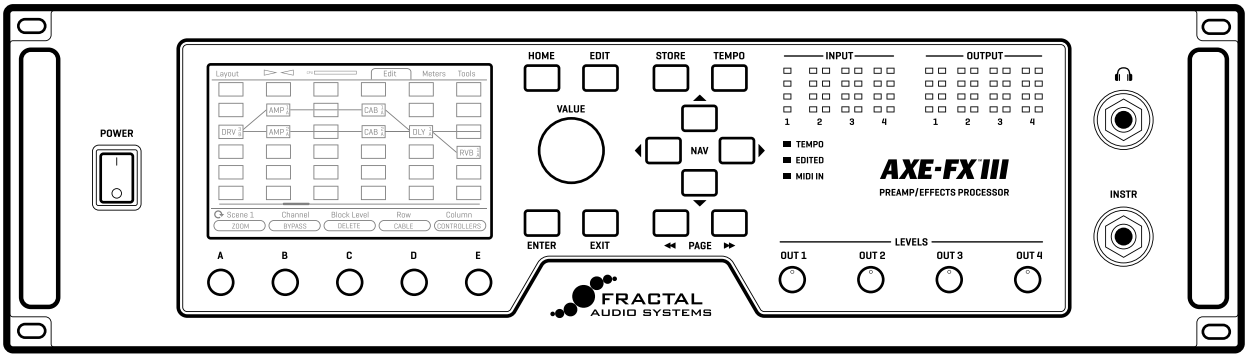
FRACTAL AUDIO SYSTEMS

AXE-FX™ III

PREAMP/EFFECTS PROCESSOR



OWNER'S MANUAL

Manual Version 2 for Axe-Fx III Firmware 2.x — December 2018

Inside Front Cover

Declaration of Conformity

Manufacturer's Name: **Fractal Audio Systems, LLC**
Manufacturer's Address: **4 Wilder Drive, Plaistow, NH 03865 USA**

Declares that the product:
Product name: **Axe-Fx III** Product option: **None**

Conforms to the following Product Specifications:

Safety: EN60065:2014
EMC: EN55013:2013
EN55020:2007+A11:2011
EN55024:2010
EN61000-3-2:2014
EN61000-3-3:2013

Supplementary Information:
The product herewith complies with the requirements of
the Low Voltage Directive 2006/95/EC
and the EMC Directive 2004/108/EC.

Clifford Chase
President / CEO
January 22, 2018

EMC/EMI

This equipment has been tested and found to comply with the limits for a Class B Digital device, pursuant to part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in residential installations. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. There is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment to an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Legal Notices

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Important Safety Instructions



WARNING: To reduce the risk of fire or electric shock, do not expose this appliance to rain or moisture.



CAUTION: To reduce the risk of fire or electric shock, do not remove screws. There are no user serviceable parts inside. Refer servicing to qualified service personnel.

1. Obey all warnings on the Axe-Fx III chassis and in this User Guide.
2. Keep away from sources of heat such as ducts, registers or appliances that produce heat.
3. Connect only to a standard grounded AC outlet of 100–240V, 47–63 Hz.
4. Keep the power cord in good condition. Do not kink, bend, or pinch.
5. If the cord becomes damaged, discard and replace it.
6. If not using your Axe-Fx III for extended periods of time, disconnect from AC power.
7. Protect the unit from rain and excessive moisture.
8. Refer servicing to qualified personnel only.
9. Stop operation of the unit and obtain service if:
   - Liquids or excessive moisture enter the unit.
   - The unit operates incorrectly or performance is inconsistent or erratic.
   - The unit has been dropped and/or the enclosure damaged.
10. Prolonged exposure to high volume levels can cause hearing damage and/or loss. The use of hearing protection in high volume situations is recommended.

A Manual for Online and Print Use

This manual is intended for use in desktop, tablet, and smart phone readers. It includes clickable links and bookmarks to make navigation and cross-reference easy. We recommend against printing, because firmware updates tend to make older manual versions fall out of date. That said, considerations have been taken for those who prefer paper. You are granted permission to print this PDF for personal use only. A copy center or online printer can print and bind a book for you from the PDF file. Hopefully those with screen readers can forgive the changes made herein to accommodate the print version: all links also include a page or section numbers, page spreads have extra margins towards the binding edge, and blank pages have been included to preserve page and chapter flow.

TABLE OF CONTENTS

1 INTRODUCTION 1
Welcome to the Axe-Fx III 1
Feature Summary 2
The Setup menu 3
Quick Connect Guide 4
Setting Levels 5
Humbuster™ Cables 6
Mono vs. Stereo 6
The Home Page: Presets 7
Intro to the Layout Grid 8
Intro to Scenes and Channels 9
Grid Editing: Quick start 10
Block Editing: Quick start 10
USB Overview 11
Fractal-Bot & Axe-Edit III 11
Connecting Pedals Directly 12
Global Expression Pedal Setup 13
Global Volume Pedal Setup 13

2 HARDWARE OVERVIEW 14
The Front Panel 14
The Rear Panel 16

3 USB 18
Computer Integration 18
USB Inputs and Outputs 19
Mac OS X System Audio 20
USB Re-Amping 21

4 SETTING UP 22
General Principles 22
FRFR/Direct 23
FRFR monitor + FRFR to Front-of-House 24
Power Amp & Guitar Speakers 25
Power Amp & Cab + FRFR/Direct 26
Four-Cable Method (“4CM”) 27
FX Processor only (“Post”) 28
FX Processor only (“Pre”) 29
Inserting Outboard Gear 30
Multi I/O Setups 31

5 PRESETS 32
Overview 32
The Layout Grid 33
Working with Blocks 33
Connector Cables 35
Block Inventory 37
Example Preset Grids 38
Editing Effect Blocks 39
Saving Changes 40
Preset CPU Limits 41

6 SCENES & CHANNELS 42
Overview 42
Changing Channels 43
Setting Up Channels 43
Selecting Scenes 44
The Default Scene 44
Selecting Scenes & Channels Remotely 45
Program Change Mapping 46
Sending MIDI with Scenes 46
Scene levels 46
Scenes, Channels & Modifiers 47
Scene Revert 47

7 LEVELING PRESETS 48
A Method for Leveling 49
Block Bypass Levels 50

8 BLOCKS GUIDE 51

9 MODIFIERS 143
Creating a Modifier 143
Modifier Tutorial: Wah Pedal 144
Modifier Tips And Tricks 144
Modifier Sources Overview 145
Modifier Parameters 146
Internal Controllers 149
Metronome 150
External Controllers 150
Modifiers List 150
Tutorial: Scene Controllers 151

10 TEMPO 152
Synchronizing Sound Parameters 153

11 THE TUNER 154
Advanced Tuner Functions 154
Mini Tuners 154

12 SETUP MENU 156
The Foot Controllers Menu 156
The Global Settings Menu 156
The I/O Menu 159
The IR Capture Utility 162
The MIDI/Remote Menu 163
The Utilities Menu 166

13 FC PROGRAMMING 167

14 IR CAPTURE 168
Method 1 Setup: Mic Only 169
Method 2 Setup: Mic + Di 170
Capturing Impulse Responses 171

15 ADDITIONAL TOPICS 172
Fractal-Bot 172
Backing Up & Restoring 172
Firmware Updates 173
Recovery 174
Getting Help 174
Loading User Cabs 175
Axe-Change 176
Frequently Asked Questions 177
Shortcuts 179
Spillover 180
LFO Waveforms & Phase 181
Tempo Cross Reference 182
MIDI Reference Tables 183

16 SPECIFICATIONS 186
MIDI Implementation 188

WARRANTY 189

EULA 190

1 INTRODUCTION

WELCOME TO THE AXE-FX III

The Axe-Fx III is the latest product in our long line of industry-leading guitar processors, and the world's most powerful unit of its kind. Axe-Fx III runs our latest "Ares" amp models, plus UltraRes™ speaker cab simulations and legendary effect algorithms, all with significantly more raw power than any previous Fractal Audio Systems product. Our vision for the new platform was to exceed anything done previously — to create a product that sounds better, does more, and is easier to use than ever before.

Fractal Audio Systems is known for accurate amp modeling with the sound and feel of real tube amps. The Axe-Fx III carries this torch high, with all of our current models and room for many new ones to come, all running on a new platform designed for the future. "Ares" picks up where "Quantum" left off, with improvements and upgrades coming all the time. Some may ask: "how can it still be getting better after all these years?" In the Axe-Fx III, we discover that there is always room for improvement and evolution when the commitment to excellency is real.

The hardware has been completely re-designed around a streamlined new workflow. The centerpiece is a dazzling new high-contrast full-color display with 30x the resolution of previous products. This display works alongside five new endless-rotary push encoders. These "Push-knobs" do different things on different pages, with labels to show dynamic functions. The end result is a user experience that's immediately more intuitive. The display also excels at animation for meters, modifiers, and more.

Power and resources are vastly expanded. Two 1-GHz "Keystone" DSPs benchmark at 2.8x the power of the Axe-Fx II, and there are two additional processors dedicated to graphics and USB. Peripherals include 4Gb of blazing fast PC1600 DDR3 memory, hundreds of Mb of FLASH memory, a proprietary FPGA and more. The number of inputs and outputs has more than doubled in comparison to previous products. Routing flexibility has also been upgraded with dedicated Input and Output blocks for highly flexible preset and rig design. All 1/4" outputs feature our Humbuster™ technology to help reduce the noise caused by ground loops.

Better sound quality has been achieved through an audiophile-grade signal path. The difference shines through in lower noise floor, pristine clarity, and better USB audio performance than most dedicated audio interfaces. Computer audio integration now includes a professional-quality 24-bit/48k 8×8 interface. Record multiple tracks of processed audio at the same time, plus a DI. The re-amping workflow is improved with independent monitoring of backing tracks and re-processed guitar tones while you dial in a sound.

Physically, the Axe-Fx III continues our tradition of combining extraordinary design with manufacturing quality rugged enough to withstand the rigors of life on the road. The unit is housed in a steel enclosure with a one-piece anodized aluminum faceplate/bezel featuring a glowing Axe-Fx III logo.

Axe-Edit™ III, a fully re-tooled version of our editor for Mac or PC, is packed with great new features including Quickbuild mode, a Scene Manager, and window scaling on monitors up to 5k and beyond.

Of course the unit features upgradeable firmware, meaning the best is very likely yet to come via our free updates.

We hope you will enjoy using the Axe-Fx III as much as we've enjoyed creating (and playing) it.

FEATURE SUMMARY

- Rugged 3U steel chassis with a sleek anodized aluminum front panel and illuminated “Axe-Fx III” logo.
- Two 1-GHz “Keystone” DSPs benchmark at 2.8x the power of the Axe-Fx II.
- 4Gb blazing fast PC1600 DDR3 memory, hundreds of Mb of FLASH memory, proprietary FPGA.
- Dedicated processors for graphics and USB.
- VALUE knob optical encoder lifespan rated for 1,000,000+ rotations.
- Audiophile-quality signal path yields the best sound quality, lowest THD, and lowest noise floor of any product we’ve ever created by far.
- Auto-switching front/rear instrument inputs, both with “Secret Sauce IV”.
- Three stereo inputs: 1 with XLR/1/4" combo jacks, two with balanced 1/4" inputs.
- Four stereo outputs: 1 with XLR outs, 1 with both XLR and 1/4", and 2 with 1/4".
- Humbuster technology on all 1/4" outputs helps reduce noisy ground loops.
- Front panel headphones output.
- Front panel LED meter bridge for instant visibility of levels.
- Easy-to-use interface allows intuitive and efficient editing.
- Stunning color display with 30x the resolution of our previous products.
- Animated on-screen metering for every block, input, output, and more.
- 5 soft-knobs with on-screen labels for turn and push functions.
- 512 Preset memories, each an entire rig with amp(s), cab(s) and effects.
- New 14×6 grid layout allows larger presets.
- New “Zoom” view shows the entire grid on one page.
- 8 nameable “Scenes” per preset store settings like “presets within a preset”.
- Revolutionary new “Channels” concept for fast, seamless switching of up to four channels per block.
- Built-in tools to copy Channels and Scenes within a preset.
- Hundreds of accurate amp models with real tube amp sound and feel.
- New Cabinet block with a “Cab-Lab” style multichannel mixer loads up to four IRs.
- 2,048 Factory cabs, 189 “Legacy” Cabs (from Axe-Fx II), 2,048 User Cabs and 16 Scratchpad memories.
- Factory IRs from Fractal Audio, AustinBuddy, Celestion, Dr. Bonkers, ML Sound Lab, OwnHammer, John Petrucci, Chris Traynor, James Santiago, Valhallir, York Audio, and more.
- Presets can now contain up to four instances of Compressor, Drive, and Delay blocks.
- New blocks: Plex Delay, Ten-Tap Delay, Multiplexer, Realtime Analyzer and more.
- Effect Types create classic sounds at the turn of a knob. Types include 37 drives, 44 reverbs, 32 delays, and many more.
- New fully modeled analog Tri-stereo Chorus type.
- Improved Pitch block with faster pitch detection.
- Improved Reverb is even richer and smoother than ever.
- New fully parametric “Input EQ” in Amp block.
- Fully revamped 5.8 minute stereo Looper with undo.
- Tone Matching to clone any tone, now in UltraRes.
- Built-in IR Capture Utility shows data in either Time or Frequency domain.
- New Input and Output blocks make routing more flexible than ever.
- Improved and faster Tuner featuring a new interface.
- 8×8 USB Audio with better audio performance than most dedicated interfaces.
- Dedicated USB Input block for reamping/processing tracks.
- MIDI-over-USB between Axe-Fx III and a computer is 10× faster than the same connection on Axe-Fx II.
- MIDI IN, OUT and THRU ports.
- Rich MIDI implementation allows extensive remote control.
- The new “Scene MIDI” block enables each scene to transmit up to 8 PC/CC messages when loaded.
- Optional MIDI PC/Bank transmission when a new preset is loaded.
- Two on-board PEDAL jacks for expression pedals like the Fractal Audio EV-1 and EV-2.
- New FASLINK 2 connector for our next-gen “FC” Series Foot Controllers.
- New “Client-Server” architecture allows multiple FC series foot controllers and Axe-Edit to control the Axe-Fx III and update each other seamlessly and instantly.
- Daisy-chain FC series foot controllers for a “Mega Pedalboard” or to allow player and tech to share switching responsibilities on “cloned” systems.
- Future proof: as ever, fully upgradeable firmware means free updates for years to come.
- Built-in backup firmware allows recovery in the event of complications during update.
- And much more.

THE SETUP MENU

Throughout this manual, you will be directed to the **Setup** menu of the Axe-Fx III for a range of options including I/O, MIDI, global settings, and more. It includes the following areas:

- The **Foot Controllers** menu includes all settings for a connected FC Series controller.
- The **Global Settings** menu includes options which govern the behavior of the Axe-Fx III.
- The **I/O** menu contains settings for all inputs and outputs, including levels, audio options, and switches or expression pedals connected directly to the Axe-Fx III.
- The **IR Capture** utility is used to create impulse responses for use in the Cab block.
- The **MIDI/Remote** menu contains MIDI settings, plus several pages of remote controller assignments for the many remote-controlled functions of the Axe-Fx III.
- The **Utilities** menu contains informative readouts plus a control to adjust screen brightness.

Access and use the **Setup** menu as follows:

- Press **HOME**.
- Push the "**E**" soft knob for the **SETUP** main menu.
- Use the **NAV** buttons to select a sub-menu and press **ENTER** to make a selection.
- Use **PAGE** and **NAV** buttons to get around, **VALUE** and knobs **A–E** to make changes.
- Setting changes under **SETUP** never need to be stored. All changes take effect immediately.
- Press **HOME** at any time to return home.

Use these links for instant access to the **Setup** menu topics in this manual:

SETUP

- **Foot Controllers** — Section 13
- **Global Settings** — Page 156
- **I/O** — Page 158
- **IR Capture** — Section 14
- **MIDI/Remote** — Page 163
- **Utilities** — Page 166

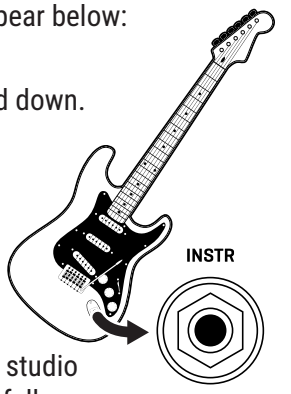
QUICK CONNECT GUIDE

Perhaps the best and most flexible way to enjoy your Axe-Fx III is through a full-range system such as studio monitors, a high-quality PA, or full-range cabinets designed specifically for guitar. All of the factory presets are designed for this type of setup.

The Axe-Fx III is incredibly flexible, however, and many other types of setups are supported, including those which integrate with tube amps, USB recording, 3rd-party outboard equipment, and much more.

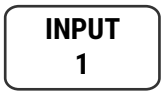
Find additional rig diagrams in **Section 4**.

The most basic setup instructions appear below:



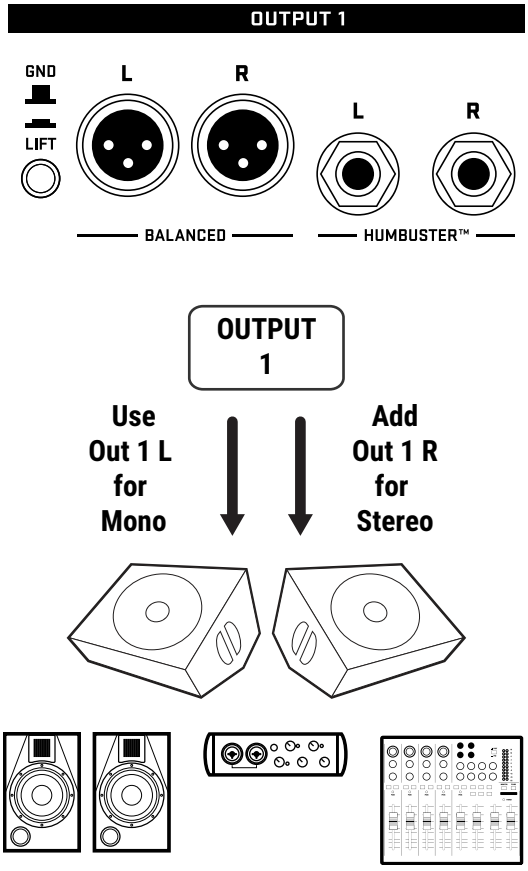
1 Begin with all level knobs turned down. Connect your guitar to the front or rear **Instrument** input (**Input 1**).

The Axe-Fx III is also perfect for bass and other instruments.



2 Connect **Output 1** to your mixer, studio monitors, audio interface, PA system, full-range speakers, power amp inputs, etc.

- For a mono rig, use **Out 1 Left**.
- For optional stereo, connect **Out 1 Right**.
- Use **XLR outputs** to connect to balanced inputs, or use the **1/4" outputs** when connecting to unbalanced inputs.
- Use **Humbuster cables** with **1/4" outputs** to reduce hum from ground loops.



FRFR speakers, Studio Monitors
Audio Interface or Mixer

3 Turn up the front panel **OUT 1** knob and adjust the level on your monitors as desired.

Explore factory presets by turning the **VALUE** wheel.

Select Scenes within a preset by using **NAV Up/Down**.

FAQ: WHY FULL RANGE?

A Full-Range Flat Response ("FRFR") system aims to reproduce the entire audio spectrum without compromise. In comparison, most guitar speakers are narrow range, with no ability to accurately reproduce extended lows and highs. A 1×12 open-back combo is never going to sound like a 4×12 stack. In comparison, full-range flat response studio monitors, high-quality PA speakers, and FRFR speakers designed specifically for guitar should be able to reproduce *anything* you play through them.

SETTING LEVELS

Setting proper levels is critical, but easy, with abundant meters to inform you about levels on the Axe-Fx III.

INPUT LEVELS

The Axe-Fx III comes ready-to-use for the typical guitar or bass, but you should check that your guitar's output is not clipping the inputs.

Connect a guitar to the **Instrument** input. Choose your loudest pickup setting and set all the guitar controls to "wide open." Play loudly to push the levels as you watch the front panel **INPUT 1** meter LEDs. A red LED on the input meter indicates a level of -6dB, which is safely below clipping. Adjust the input trim so your loudest playing "tickles the red" as follows:

- Page to the **Input** page of the **I/O** menu under **SETUP**.
- Reduce **Input 1/Instrument** input levels using the **A** knob.

In general, you should set input levels as high as possible without clipping.

Inputs 2, 3 and 4 can be adjusted using the same method. Each has its own **Input trim** parameter on the **Input** page of the **I/O** menu under **SETUP**.

You can also monitor all levels on the **Meters** page of the **Home** menu.

Except at very low settings (5% or less), input level adjustments do not affect gain levels. As you adjust the level to the A/D converter, its output compensates accordingly, so your signal remains exactly the same when it hits the grid and any virtual amps or effects.

OUTPUT LEVELS

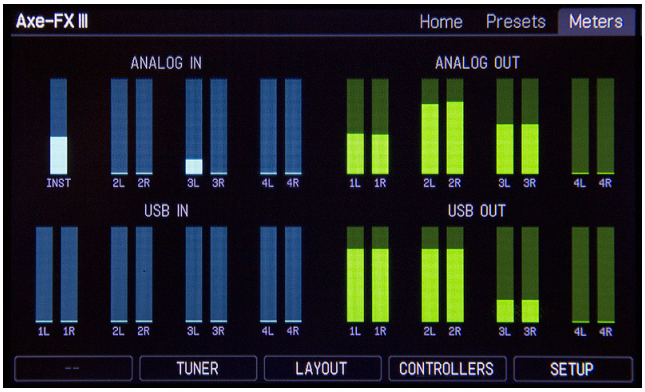
Output levels are very easy to monitor on the front panel meter bridge, and easy to control using the four front panel **LEVEL** knobs. You can also monitor output levels in higher resolution by paging to the **Meters** page from the **Home** page.

Outputs 1-2 are ready to be connected to consumer line inputs (-10 dbV). If you are connecting to professional-grade equipment operating at +4dBu, set the nominal output levels as follows:

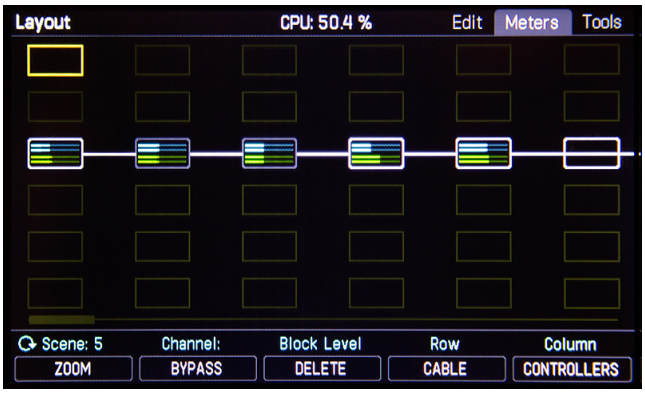
- Page to the **Audio** page of the **I/O** menu under **SETUP**.
- Adjust **Output 1: Output Level** or **Output 2: Output Level** to +4dBu.

Learn more about the **I/O Menu** on p. 158.

Section 7 on p. 48 is dedicated to the subject of adjusting preset levels.



The Meters page of the Home menu shows all I/O levels.



The Layout also has a Meters view showing block levels.

HUMBUSTER™ CABLES

Use XLR cables between the Axe-Fx III XLR outputs and the XLR inputs of mixers, monitors, and other devices.

When connecting to 1/4" balanced inputs, use an XLR-to-1/4" (TRS) cable or adapter to keep the connection balanced.

For unbalanced 1/4" inputs on other devices, standard 1/4" (guitar) cables may be used, but there is a better option. **Humbuster** technology on all 1/4" outputs of the Axe-Fx III can significantly reduce unwanted noise due to the common problem of a ground loop. For this to work, you must use a special Humbuster cable with one **TRS** end (like a balanced cable) and one **TS** end (like a guitar cable). The TRS end connects to the Axe-Fx III. The TS end connects to your amp.

Humbuster cables are available from http:/www.fractalaudio.com/cables. You can make your own by following the diagram below. Be sure to use high-quality connectors and shielded cable.



MONO VS. STEREO



The Axe-Fx III comes pre-configured for **stereo** but you can connect it in **mono** with no issues. If you are NOT running in stereo, you may find that certain presets or settings produce unexpected results. For example, a panner sounds like a tremolo when one channel is missing. A ping-pong delay may ping but never pong. Stereo enhancers or certain types of modulation may not be apparent at all. Tone may change completely if amps or cabs have been panned. Here is an overview of several scenarios, with recommended settings for each.

- **Stereo:** *No special settings required.*
- **Half-Stereo:** Leaving the Axe-Fx III in its default stereo configuration but connecting only one (Left) output results in a “half-stereo” setup. This works fine, aside from the exceptions mentioned above (ping-pong, panning, etc). *No special settings required.*
- **Dual Mono:** If you want to force your rig to mono, dual mono is one choice. Sonically, this is identical to half-stereo, with the same limitations, except that mono signal is produced at *both* the left and right jacks so you can connect to two monitors. To switch to dual mono, open **SETUP: I/O: Audio** and set **Mode** for the desired output to “COPY L->R”.
- **Summed Mono**: In this setup, left and right channels are added together resulting in an identical mono signal at both left and right outputs. This has the advantage of not discarding half of the sound, but summing has its own issues. For example, short delays or phase differences between channels can result in strange artifacts or even total cancellation. To switch to summed mono, open **SETUP: I/O: Audio** and set **Mode** for the desired output to “SUM L+R”.

On the Axe-Fx III, a flexible block-based I/O system makes it easy to use different outputs with different settings for different purposes. See “Input & Output Blocks” on p. 52.

All options of the **I/O** menu under **SETUP** are detailed in **Section 12**.

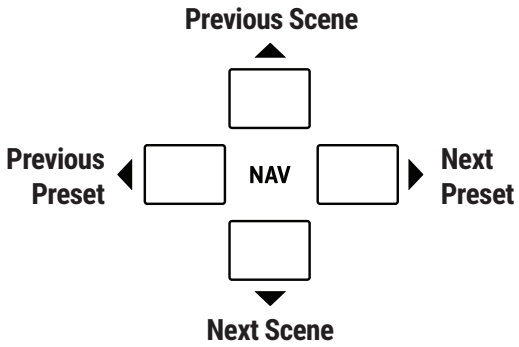
THE HOME PAGE: PRESETS

Once you have connected your Axe-Fx III you can begin to audition factory preset sounds. Remember that aside from a few templates, the factory presets are designed for use with full-range speakers / monitors / headphones. (See **Section 4** for details on many other types of setups.)

The Axe-Fx III contains 512 preset memories, each a fully independent rig with its own amps, cabs, effects, settings, controllers, and more. When you consider everything that is possible with Scenes and new Channels, a single preset can easily cover an entire song, or even a whole show.

Here's how to explore the factory presets:

- Press **HOME** to show the **Home** page, where you can switch presets.
- To change presets, turn the **VALUE** knob or use the **NAV** buttons.
- Many factory presets contain extra **Scenes**. You can switch between Scenes using the **NAV** buttons or Push-knob **A**.
  See p. 9 for an introduction to Scenes.



OTHER HOME MENU FUNCTIONS

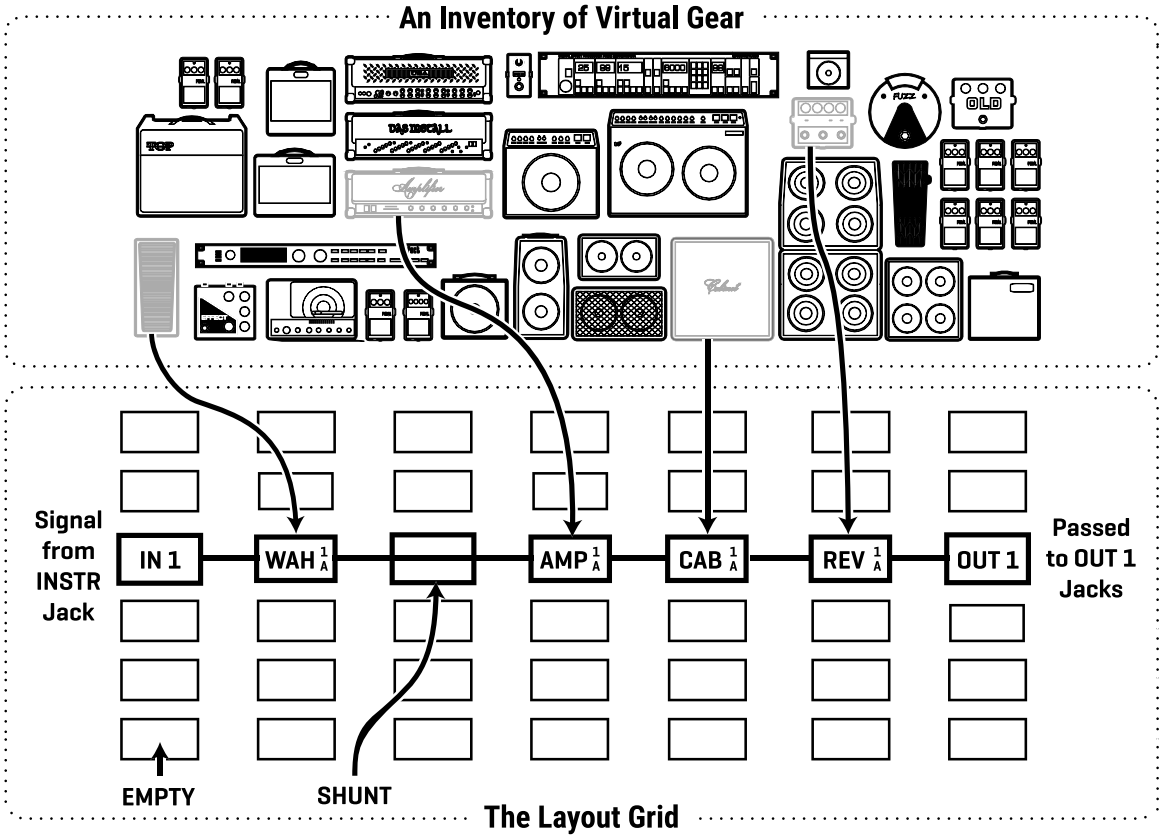
- The push functions of the five knobs provide access to other areas:
  - **Tuner** (Page 154)
  - **Layout grid** (Page 8 and Page 33)
  - **Controllers** (Page 149)
  - **Setup** (Page 3 and Page 156).
- A **mini-Tuner** appears at the top of the **Home** page itself!
- The **Home** menu also provides access to the **Presets directory** page, listing all presets in a scrolling list sorted by number.
  - **PAGE** to the **Presets** page to show the directory.
  - To sort alphabetically by name, press the "**Sort A–Z**" button.
  - The **ENTER** key loads the selected preset and returns to the **Home** page.
- The **Meters** page provides visual indication of all input and output levels, including USB.

INTRO TO THE LAYOUT GRID

Behind every Axe-Fx III preset is the **Grid**, a matrix of rows and columns that is the key to unlocking the true potential of this unit. In the world of traditional gear, our options are limited by budget, space, weight, and the limits of the equipment itself. Building a rig means hard choices. With the Axe-Fx III however, a limited selection of gear is instead replaced by a vast, ever-growing "**inventory**" of virtual amps, cabs, effects, and more. Every preset gives you the flexibility to design a unique rig composed of different components.

Let's lay out the terms of preset creation. To create a **preset**, virtual pieces of gear are selected from the **inventory** and placed as **blocks** into the slots of the **layout grid**. Each block represents a different component like a wah pedal, amp, or reverb unit. Blocks are then connected together using virtual **cables**. You can create splits, merges or multiple parallel paths as needed. Passive **shunts** carry signal through otherwise empty spaces on the grid. Special **input** and **output** blocks connect to the various jacks and USB signals of the Axe-Fx III.

The following illustration shows the inventory concept with several items placed on the grid as blocks.



Let's review what is happening above. Signal flow begins at the **Input** block on the left. It is then routed to a **Wah** block, which in turn feeds a **shunt**. The shunt has no effect on the sound and is shown only to introduce the idea of how it carries signal from one location to another. The shunt is connected to an **Amp** block (we might set its type to "Plexi 100W High"), which in turn feeds a **Cab** (one of the many "4×12" options, perhaps). This is connected to a **Reverb** and then to an **Output** block. In this limited example, many grid spaces are empty, and only half of the columns are actually shown! The size of a preset is limited only by the grid structure, block inventory, and total processing power ("CPU"). You will find that you can create very large virtual rigs with multiple amps, cabs, and effects.

QUICK TIP *A new "ZOOM" feature shows the whole grid at once. Look for the ZOOM button on Layout menu pages.*

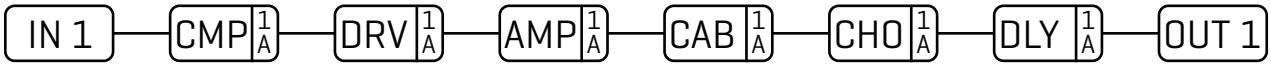
INTRO TO SCENES AND CHANNELS

Imagine a rack system like the ones used by guitar players over the past 35+ years. In this system, various effect pedals, amps, and rack units are connected with a central switching system, which places other components in or out of the signal path. Some of the effects may also have settings that the switcher can control remotely—like the channels on an amp, or preset selection on a delay. Using the switcher, you would set up different combinations and settings and then save each as a preset: Clean, Rhythm, Lead, etc.

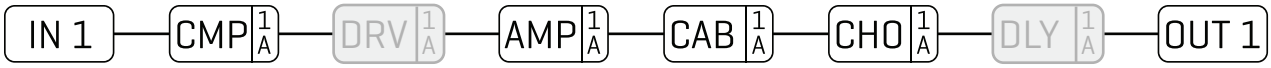
If (as discussed on the previous page) each Axe-Fx III preset is like an entire rig, then **Scenes** are similar to the presets of the system described above. A Scene stores whether each block is **engaged** or **bypassed**, and which of up to four "**channels**" it should be on. A Channel is like a preset within a block. Each block has up to four channels, and each channel has fully independent parameters. For example, **Channel A** of a **Drive** block might be a "clean boost", **B** could be a "screamer" type with low gain and medium level, **C** could be that same screamer "dimed", while **D** is a super-saturated fuzz. That's four different sounds from just one block, and almost every block has four channels! By combining different blocks and channels, Scenes offer an incredibly powerful way to create sounds.

Scenes also offer a big advantage over presets in that when you change scenes, the Axe-Fx III doesn't need to reload an entire rig. Instead, it just instantly sets the blocks and channels as needed for the new scene. Reverb and delay effects are easy to set up for "spillover" so tails can ring out. On the Axe-Fx III, scenes even have their own names, which is especially helpful.

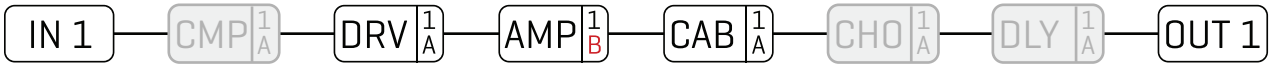
SCENES AND CHANNELS: A VISUAL EXAMPLE



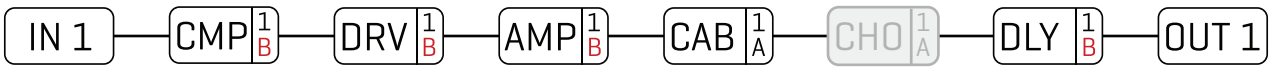
Here is our preset. The Input 1 block feeds a Compressor followed by a Drive, then into an Amp and Cab, followed by Chorus and a little Delay, ending up at the Output 1 block.



SCENE 1 – "Clean": For scene 1, the Drive and Delay blocks are bypassed. We dial in the Compressor, Amp, Cab, and Chorus for a classic clean tone. Notice that the amp says "1A". This means we are using **Amp 1**, set to channel A. Let's imagine it as the "ODS-100 Clean" model. We name the scene "Clean".



SCENE 2 – "Crunch": To create scene 2, we bypass the Compressor, Chorus and Delay and engage the Drive. The Channel of the Amp block is changed from "A" to "B", which we dial in as a "Euro Blue" model. Remember, each channel has a totally independent set of settings, so we can dial in every amp parameter exactly as we wish for this scene: Drive, Treble, Mid, Bass, Master, and many, many more. We dial in Channel "A" on the drive block with a good "screamer" sound by selecting the "TS808 OD" type. Let's name this scene "Crunch".



SCENE 3 – "Lead": Here's our soaring lead. The Chorus is bypassed. We've changed the compressor to Channel "B" and dialed it in for sustain. The amp is the same as the "Crunch" scene but Drive changes to "B", which we'll make a "Fat Rat". The Delay is on channel "B" dialed in with higher mix and feedback.

Learn more in "Scenes & Channels" on p. 42

GRID EDITING: QUICK START

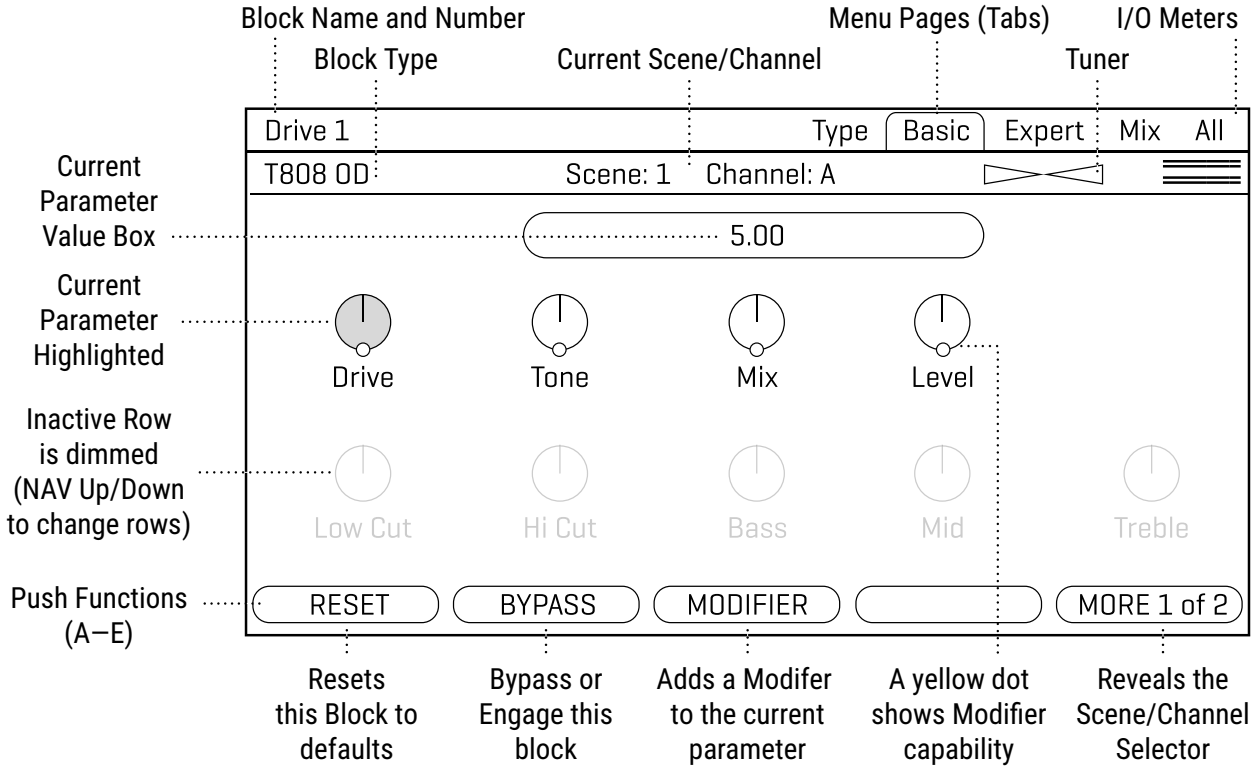
Learn more about the grid in Section 5. Meanwhile, here's a very quick primer:

- From the **Home** page, press **LAYOUT** (knob **C**) or **ENTER** to show the grid (the Edit page of the Layout menu). Note that the "push" function of the main VALUE knob also serves as a shortcut to Layout.
- Use the **NAV** buttons to move the cursor around the grid.
- Turn the **VALUE** knob to cycle through the inventory of blocks. When you find the block you want, press **ENTER** to confirm. Press **EXIT** to cancel changes.
- On the grid, "push" functions of the **B**, **C**, and **D** knobs allow you to toggle a block's **Bypass** state, **Delete** a block, or create/remove a connector **Cable** between any two blocks in adjacent columns.
- To save any changes, press **STORE, ENTER, ENTER.**

BLOCK EDITING: QUICK START

Learn more about editing blocks in Section 5. Meanwhile, here's a very quick primer:

- On the grid, select the desired block and press **EDIT** to open its menu.
- Use the **PAGE** buttons to navigate menu pages.
- Many blocks have a **TYPE** page, which allows you to dial in classic settings instantly. Examples include setting an amp to "USA Lead+", a drive to "Face Fuzz", or a reverb to "Large Spring".
- Use the **A**, **B**, **C**, **D**, and **E** knobs to edit on-screen parameters. From any block's Edit menu, press **EXIT** to return to the grid.
- To save any changes, press **STORE, ENTER, ENTER.**
- The stylized diagram below shows a typical edit menu page with annotations:



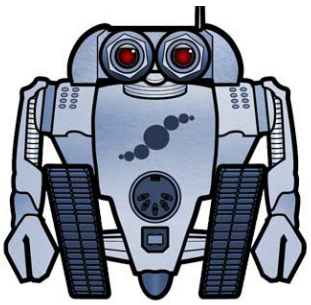
USB OVERVIEW

USB provides the Axe-Fx III with a host of great features. A 16-core, 500 MHz microcontroller provides 16 channels of flawless USB audio (8 in, 8 out) allowing the Axe-Fx III to be the center of your music workstation. The audiophile-grade signal path components and converters provide better audio performance than most dedicated USB audio interfaces. With 8×8 channels you can record multiple tracks of processed audio and DI signals, monitor backing tracks, audition stereo re-amps in real-time, and more. A USB audio sound source can even be placed on the grid for processing via the dedicated USB Input block.

USB is also required to use Axe-Edit and Fractal-Bot, the companion software products for the Axe-Fx III.

See Section 3: USB, on p. 18 for full details on the USB capabilities of the Axe-Fx III.

FRACTAL-BOT & AXE-EDIT III



In addition to providing audio and MIDI capabilities, connecting the Axe-Fx III to a computer via USB allows you to use companion software applications **Fractal-Bot** and **Axe-Edit III**.

Fractal-Bot is a small, simple program used to update the Axe-Fx III when new firmware is released. It includes tools to backup or restore presets and other custom settings, and can be used to install downloaded presets or cabs.

Axe-Edit III is a full-featured software editor/librarian for the Axe-Fx III. If you are comfortable with audio software or plug-ins, you will probably enjoy using this program to work with the Axe-Fx III. While every function of the Axe-Fx III can be operated from the unit's front panel, Axe-Edit III brings a new level of convenience, including some "power-user" routines that aren't possible on the unit itself (e.g. Block Library and Scene Swap). It also provides great tools for managing presets and banks, installing Cab Packs, and more.

You can download Axe-Edit III for Mac or Windows at http://www.fractalaudio.com.



CONNECTING PEDALS DIRECTLY

This section is for pedals connected to the local "Expression" jacks of the Axe-Fx III.
For pedals connected to a MIDI or FC Series controller, see the next page.

Each of the two local **Expression** jacks of the Axe-Fx III allows you to connect one expression pedal or one external footswitch. A connected pedal or switch can change effect parameters, or operate any of a long list of global functions. Many factory presets are ready for use with one or more pedals.

TO CONNECT & CALIBRATE AN EXPRESSION PEDAL...

Expression pedals for the Axe-Fx III should have a linear resistance taper and must have a maximum resistance in the range of 10–100kΩ. Expression pedals must be used with Tip-Ring-Sleeve (TRS) cables!

Connect your expression pedal to the desired jack with a TRS cable, then follow these instructions to calibrate:

- Page to the **Pedal** page of the **I/O** menu under **SETUP**.
- Ensure that **Pedal Type** is set to "CONTINUOUS".
- Navigate to the **Calibrate Pedal** function and press the **ENTER** button.
- Follow the on-screen instructions to perform calibration.

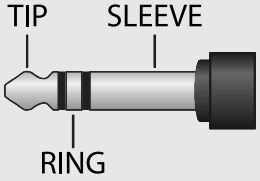
TO CONNECT A SWITCH...

Follow the instructions above and simply set **Type** to "MOMENTARY" or "LATCHING" based on what you're connecting. (If you're not sure, check with the switch manufacturer or try both settings to see which works better.) Calibration is not required. You can use a normal guitar cable (TS) for switches.

THE EV-SERIES EXPRESSION PEDAL

The Fractal Audio Systems EV series of expression pedals are perfect for use with all Fractal Audio Systems products. The EV-1 is a full-sized expression pedal. The EV-2 is a compact version. These pedals feature rugged cast metal casings, a high quality 100kΩ potentiometer, and built-in analog volume pedal capability. Learn more at https://www.fractalaudio.com.

FAQ:



WHAT IS TRS? "TRS" stands for TIP-RING-SLEEVE and describes the configuration of a 1/4" end plug or jack with three connectors. Normal guitar cables are "TS" (Tip-Sleeve) since they lack the ring required for a third contact. Expression pedals require TRS cables because full control voltage is transmitted to them on one contact (the tip), while less than full voltage is returned to on another (the ring) so the host device is able to sense and utilize the pedal position. The third contact (sleeve) is connected to ground.

GLOBAL EXPRESSION PEDAL SETUP

Once you have connected and calibrated a pedal according to the instructions on the previous page, or if you are using a MIDI or FC-connected pedal, you can globally assign it as “External 1”. Numerous factory presets are set up to respond to this global controller.

To set up “External 1” to respond to your connected pedal:

1. From the Home page, open **SETUP: MIDI/Remote**: **External**.
2. **NAV** to **External 1**.
3. Turn the **A** or **VALUE** knob to assign the external controller to your selected entry:
   - Select “**PEDAL 1”** or “**PEDAL 2”** for the onboard Expression jacks.
   - Select “**FC__ PEDAL __”** for a pedal connected via an FC series controller.
   - Select a **number** if your pedal is transmitting a MIDI CC#. For example, by default, the Fractal Audio Systems MFC-101 transmits CC#11 for XP1 and CC#16 for XP2.
4. **EXIT** when finished. Most of the first 100 Factory Presets contain a Wah block you can test with.

Learn more about assigning pedals and switches to sound parameters in Section 9: Modifiers.

GLOBAL VOLUME PEDAL SETUP

The Axe-Fx III allows remote control of the volume at all input and output blocks. If you have a second pedal, or if you want to use one pedal for global volume instead of expression, follow the instructions below to set it up.

Adjusting **Input volume** affects gain/distortion and the behavior of level-dependent blocks like the compressor or gate. Adjusting **Output volume** does *not* affect level-dependent blocks, but scales *everything* you hear including effect tails. For all other volume options, you can insert a **Volume** block anywhere in your preset and control it with a pedal as described in “Modifiers” on p. 143.

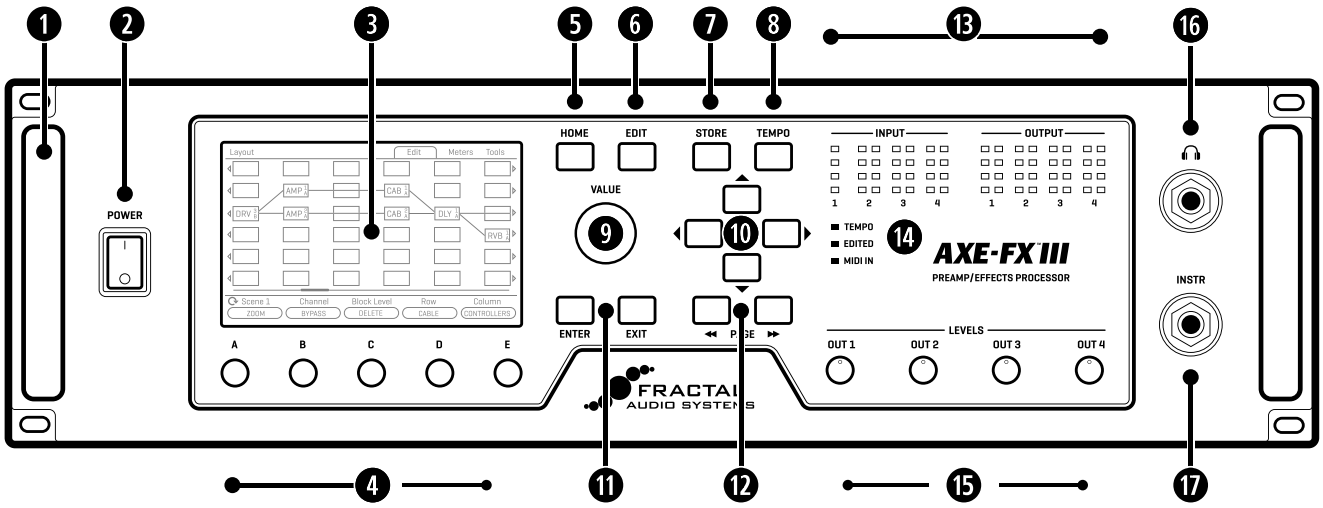
First ensure that you have an expression pedal connected and ready to use. Then:

1. From the **Home** page, open **SETUP: MIDI/Remote**: **Other**.
2. **NAV** to the entry for the volume you want to control: **Input 1, 2, 3, or 4**, or **Output 1, 2, 3, or 4**.
3. Turn the **A** or **VALUE** knob to assign a controller to your selected entry.
   - Select “**PEDAL 1** or **PEDAL 2”** for the onboard Expression jacks
   - Select “**FC__ PEDAL __”** for a pedal connected via an FC series controller.
   - Select a **number** if your pedal is transmitting a MIDI CC#. For example, by default, the Fractal Audio Systems MFC-101 transmits CC#11 for XP1 and CC#16 for XP2.
4. Test and **EXIT** when finished.

Learn about other options for global remote control in “The MIDI/Remote Menu” on p. 163.

2 HARDWARE OVERVIEW

THE FRONT PANEL



1 **Chassis** — The Axe-Fx III is housed in a rugged 3U steel chassis with a sleek anodized aluminum front panel. A pair of high-strength handles make the unit easy to manage.

2 **Power Switch** — The front panel power switch turns the Axe-Fx III on or off and features pop suppression.

3 **Color Display** — A large 800×480 color display features 30x the resolution of our previous models. This new display is optimized for readability in the most difficult conditions, with greater brightness and contrast than "off-the-shelf" displays.

4 **A, B, C, D, and E knobs** — Five endless rotary push-encoders perform different functions depending on which page is shown in the display. Most Edit pages show one or two rows of five knobs for easy 1:1 operation. On vertical menu pages, knob functions are indicated on-screen by labels ("A" through "E").
The Push functions of the five knobs are shown on-screen as buttons.

5 **HOME Button** — This button shows the **Home** menu—a convenient starting point for loading or editing presets, and accessing the tuner, controllers, metering, and global setup options.

6 **EDIT Button** — This button opens the **Edit** menu for a block selected on the Layout grid. You can also tap **EDIT** sequentially to step through all blocks in the current preset (top-to-bottom, left-to-right). See "Presets" on p. 32 for more on editing presets.

7 **STORE Button** — Enters the **Store** menu where you can save presets and enter names for presets or scenes. See "Saving Changes" on p. 40.

8 **TEMPO Button** — Tap this button once to show the **Tempo** page of the Controllers menu, or tap 2+ times to set a new tempo. The tempo can also be entered using MIDI or a remote switch. After setting the tempo, press EXIT to go back to back to wherever you were. See "Tempo" on p. 152.

9 **VALUE Knob** — The **VALUE** knob performs different functions on different menu pages. In the **Home** menu, it selects presets. In **Layout** (grid) pages, it is used to add or modify blocks on the grid. In **Edit** menus, it changes parameter values, selects from lists, and more. A hidden "push" function enters the Layout (grid) of the current preset from other pages or menus.

10 **NAV Buttons** — The four **NAV** buttons perform different functions on different menu pages. On the preset pages of the **Home** menu, they select and load Presets (Left/Right) and Scenes (Up/Down). In other menu pages, they select between on-screen parameters or options, moving the "focus" of the **VALUE** knob, as indicated by a blue highlight and brighter blue text.

11 **ENTER and EXIT Buttons** — The **ENTER** button executes commands, commits changes, accesses sub-menus, and more. **EXIT** works for cancel, escape, and various other functions.

12 **PAGE LEFT and PAGE RIGHT Buttons** — These buttons step through menu pages, shown as "tabs" at the top of the display.

13 **Meter Bridge** — Eight LED meters show the level of every input and output. For input meters, the red LED indicates -6dB. Set your Input Trims so your loudest playing "tickles the red" (see p. 5). For output meters, the red LED indicates -1 dBFS and is therefore a more immediate indication of clipping. You can also find on-screen meters on the **Meters** pages of the Home menu and the Layout.

14 **Status LEDs** — Three LEDs indicate important information. The **Tempo** LED flashes to show the current Tempo. The **Edited** LED lights whenever the current preset has been altered but not saved. The **MIDI In** LED lights while MIDI data is being received at the MIDI Input or via USB.

15 **Output Level Knobs 1–4** — These knobs independently control the volumes at the corresponding rear panel outputs. The Output 1 knob simultaneously controls Output 1 XLR and quarter-inch jacks, and the headphone output level. Outputs 3 and 4 are at unity gain when set fully clockwise.

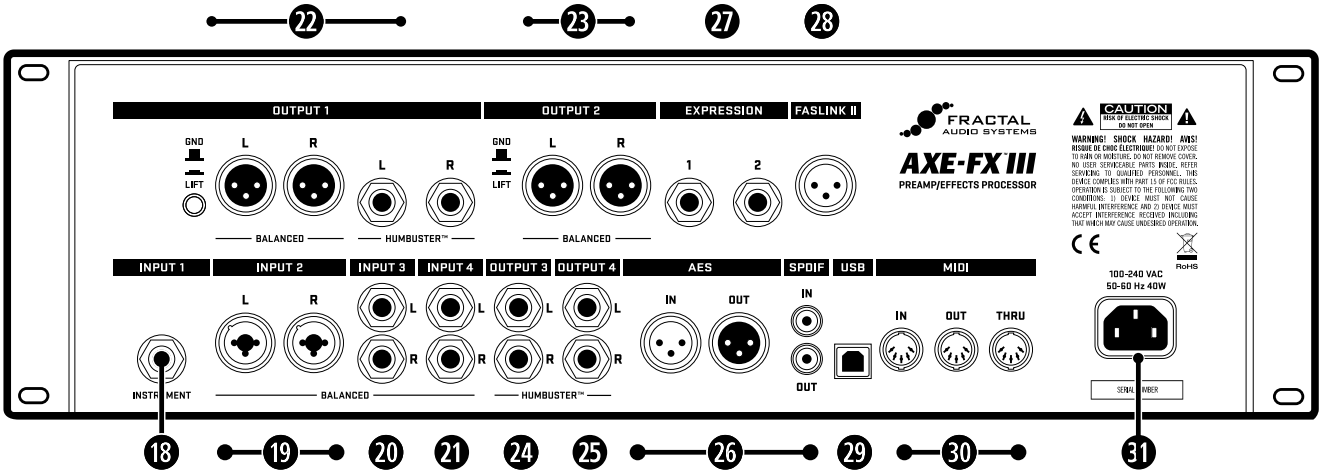
16 **Headphones Output** — **1/4" Stereo Jack** — Connect headphones here to monitor Output 1 (see above).

17 **Front Panel Instrument Input (mono)** — **(1) 1/4" Jack (unbalanced)** — Plug your instrument into this jack intended specifically for use with electric, acoustic, and bass guitars. See **Notes** below for details about both front and rear instrument inputs.

NOTES ABOUT FRONT & REAR INSTRUMENT INPUTS

- The front Instrument input *always* over-rides the rear. This allows you to build a rig using the rear input—with a wireless unit, for instance—and then change to the front input simply by plugging in a cable. There are no options for this. The input always switches automatically.
- Both front and rear Instrument inputs feature our **"Secret Sauce IV"** circuitry, which lowers the noise floor using a proprietary technique along with special analog input circuitry.
- The **Input 1** block sets the impedance for Input 1 automatically, or can be manually overridden for "loading" effects on your pickups. See "Input 1/Instrument Impedance" on p. 54.
- You can also connect guitars and other instruments to **Inputs 2, 3 or 4**. These are designated as LINE Level but accommodate high impedance sources so as not to load down guitar pickups.
- You can use traditional effect pedals between your guitar and the Axe-Fx III. You may need to trim the input when using a loud guitar or pedal. See "Setting Levels" on p. 5.

THE REAR PANEL



ANALOG INPUTS

Axe-Fx III inputs are extremely flexible. Any input can be placed as a block on the grid for use in any number of scenarios. Example rig diagrams are provided in **Section 4**.

18 **Input 1 [Instrument] (mono) — (1) 1/4" Jack (unbalanced)** — Plug your instrument into this jack intended specifically for use with electric, acoustic, and bass guitars. See **Notes** (previous page) for details.

19 **Input 2 L+R (stereo) — (2) XLR/1/4" Combo Jacks (balanced)** — These combo jacks combine XLR female and 1/4" jacks in one, accepting either balanced or unbalanced line level signals.

20 **Input 3 L+R (stereo) — (2) 1/4" Jacks (balanced)** — Input balanced or unbalanced line level signals.

21 **Input 4 L+R (stereo) — (2) 1/4" Jacks (balanced)** — Input balanced or unbalanced line level signals.

Balanced audio connections are resistant to noise and interference. It is best to use XLR or TRS (3-conductor) cables when connecting Axe-Fx III inputs 2, 3 and 4 to the balanced outputs of other devices. Regular TS-type “patch” cables (2-conductor) are sufficient when connecting to unbalanced outputs.

ANALOG OUTPUTS

As with inputs, outputs are placed as blocks on the grid for use in any number of scenarios; though in this case the various XLR and 1/4" Humbuster jacks suggest different uses for different outputs. Example rig diagrams are provided in “Setting Up” on p. 22.

Humbuster technology on all 1/4" outputs can significantly reduce hum from ground loops. Humbuster Cables are available at http://shop.fractalaudio.com. Learn more on p. 6.

22 **Output 1 L+R (stereo) — (2) XLR-Male (balanced) and (2) 1/4" Jacks (Humbuster)**
Use the XLR jacks to connect Output 1 to balanced inputs, employing the provided ground lift switch if necessary to reduce 60-cycle hum. Use the 1/4" unbalanced Humbuster outputs to connect to unbalanced inputs, such as those on guitar power amps and other devices.

Note that Output 1 is utilized as the main output in all factory presets.

㉓ **Output 2 L+R (Stereo) — (2) XLR-Male Jacks (balanced)** — Use the XLR jacks to connect to balanced inputs. Use the ground lift switch if necessary to reduce unwanted 60-cycle hum.

㉔ **Output 3 L+R – (Stereo) — (2) 1/4" Jacks (Humbuster)** — Use these 1/4" unbalanced outputs to connect to unbalanced inputs, such as those on some guitar power amps or other devices.

㉕ **Output 4 L+R – (Stereo) — (2) 1/4" Jacks (Humbuster)** — Use these 1/4" unbalanced outputs to connect to unbalanced inputs, such as those on some guitar power amps or other devices.

Outputs 3 and 4 are designed for unity gain applications such as the popular "Four-Cable Method" and other scenarios. Set the OUT 3 and OUT 4 Level knobs fully clockwise for unity gain. See the relevant rig diagrams in **Section 4** for more information.

㉖ **Digital Inputs and Outputs** — This includes both S/PDIF and AES/EBU format input and output jacks. Only one or the other pair can be active at any time depending on the setting of the **SPDIF/AES Select** parameter found in **SETUP: I/O: Audio**. These jacks transmit and receive at a fixed rate of 48k. The Axe-Fx III defaults to using its internal clock but can also use Word Clock signals received at the digital input. Find descriptions of these and other related parameters on "I/O: AUDIO Page" on p. 159.

㉗ **Expression Inputs — (2) 1/4" Jack** — These are used to connect an external expression pedal or switch to control various functions of the Axe-Fx III. See "Connecting Pedals Directly" on p. 12 for more information.

㉘ **FASLINK II Connector** — The FASLINK II connector allows you to connect the Axe-Fx III with our new FC-6 or FC-12 foot connectors. A standard XLR cable powers the FC without an external "wall wart" adapter, and carries 2-way communications.

NOTE: *The FASLINK II port on the Axe-Fx III is designed for our FC Series of foot controllers and is NOT compatible with the FASLINK port on our MFC-101 MIDI foot controller. Connecting an Axe-Fx III to an MFC-101 via FASLINK should not cause any damage, but it will not work.*

㉙ **USB** — This provides the Axe-Fx III with 8×8 USB audio and 2-way MIDI capabilities when connected to a compatible Mac or PC for use with Axe-Edit III, Fractal-Bot, a DAW or other audio/MIDI applications. Drivers are required for Windows. No driver is required for Mac OS X. Connect the Axe-Fx III via USB and it will appear under System Preferences and all Audio/MIDI apps.

㉚ **MIDI Ports** — The MIDI IN port of the Axe-Fx III allows you to control various MIDI functions including preset and scene selection, effect bypass, channel changes, parameter changes, and more. Messages received at the MIDI IN port are passed to the MIDI THRU port. The MIDI OUT port transmits MIDI data to connected devices.

See "The MIDI/Remote Menu" on p. 163 for details on how to configure the MIDI Channel and other options. MIDI messages can also be transmitted by presets or by the Scene MIDI block, see "The Scene MIDI Block" on p. 130.

㉛ **AC Power Receptacle** — Insert the supplied power cable and connect the other end to a grounded AC power receptacle. The Axe-Fx III has a universal power supply, which means it can be used around the world by simply changing the cable.

3 USB

COMPUTER INTEGRATION

With a USB connection to a computer, the Axe-Fx III functions as an 8-in/8-out 24-bits 48kHz ultra-low latency audio interface for Mac and Windows computers. A driver is required for use under Windows. This extremely high-quality audio interface is powerful enough to be the centerpiece of a Digital Audio Workstation (DAW).

In addition, a computer connection provides high-speed "MIDI-over-USB" for remote control of the Axe-Fx III, backups, updates, editing, and more.

WINDOWS MINIMUM REQUIREMENTS

OS: Windows 7 SP1 or newer (all versions compatible with x86 or x64).

CPU: Intel Core 2 @1.6 GHz or better, or AMD equivalent.

Memory: 1GB minimum.

USB: USB 2.0 support required.

Driver: The Axe-Fx III is fully class-compliant, but driver installation is still required for use under Windows.

The Windows driver can be downloaded at http://www.fractalaudio.com/support.

Step-by-step instructions are included with the installer.

MAC MINIMUM REQUIREMENTS

OS: OS X 10.6.8 for MIDI-over-USB (Fractal-Bot, Axe-Edit III, Cab-Lab, etc.).

OS X 10.9 or later required for USB audio. An issue in older OS X versions causes audio pops.

CPU: Intel Processor.

Memory: 512MB minimum.

USB: USB 2.0 support required.

Driver: No driver is required for Mac OS. The Axe-Fx III is class-compliant.

*Important: If you are using a USB-C to USB adapter on a newer Apple computer, always plug the USB-C end of the adapter into a USB-C or Thunderbolt 3 (USB-C) port on your Mac first, and **then** connect the USB cable of your Axe-Fx III to the adapter.*

MIDI-OVER-USB

The USB connection also carries high-speed "MIDI-over-USB". On the computer, the **Axe-Fx III MIDI In** port inputs MIDI data from the Axe-Fx III. The **Axe-Fx III MIDI Out** port transmits data from the computer to the Axe-Fx III.

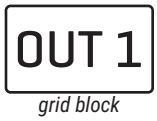
MIDI-over-USB performs the same functions as regular 5-pin MIDI, only faster. For example, the Axe-Fx III will process an incoming MIDI Program Change message whether that message is transmitted through the unit's own **MIDI IN** port, or via the **Axe-Fx III MIDI Out** port of a connected computer.

USB INPUTS AND OUTPUTS

Each of the 16 USB audio channels is “hard-wired” to a specific Axe-Fx III input or output.
As with all USB Audio devices, inputs and outputs are defined with respect to the computer.

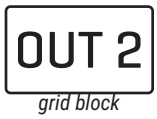
COMPUTER INPUTS 1+2

Source: The output of the **OUT 1** block on the grid.
Applications: Record the stereo processed output of the Axe-Fx III.



COMPUTER INPUTS 3+4

Source: The output of the **OUT 2** block on the grid.
Applications: Record a second stereo processed signal.



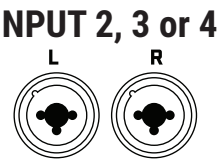
COMPUTER INPUTS 5+6

Source: The signal at the **Input 1/Instrument** jack (front or rear).
Note: Input 1 is mono, so the same signal is fed to both Left and Right channels.
Applications: Record a DI signal for re-amping.



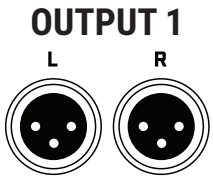
COMPUTER INPUTS 7+8

Source: The signal at your choice of **Input** jacks: 2, 3 or 4. (Left = 7, Right= 8)
Make this selection in **Setup : I/O : Audio**. (Default: Input 2)
Applications: Record a stereo source without processing.



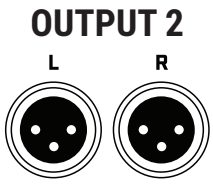
COMPUTER OUTPUTS 1+2

Destination: The **Output 1 D/A converter** which feeds all **Output 1** jacks, **Headphones**, and possibly **SPDIF/AES**. **USB audio** is added to the signal from the **OUT 1** grid block (if any) and this mix is passed to the output converters.
Applications: Play stereo audio from the computer through the Axe-Fx III.



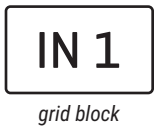
COMPUTER OUTPUTS 3+4

Destination: The **Output 2 D/A converter** which feeds the **Output 2** jacks, and possibly **SPDIF/AES**.
Applications: Play a second stereo signal through the Axe-Fx III. Ideal for backing tracks so they can be fully independent from processed guitar at **Output 1**.



COMPUTER OUTPUTS 5+6

Destination: The input of the **IN 1** block IF **Input 1 Source** is set to “USB”.
See “Input 1 Source” on p. 159 and “Input Blocks 1–4” on p. 52.
Applications: Route tracks to the Axe-Fx III for processing. Ideal for re-amping a DI.



COMPUTER OUTPUTS 7+8

Destination: The **IN U (USB)** grid block. Signal appearing here will be processed by the block’s noise gate, etc.
Applications: Route additional DAW track(s) to the Axe-Fx III for processing without requiring changes to **SETUP: I/O: Audio**.



MAC OS X SYSTEM AUDIO

By default, Mac OS X system audio plays through USB Channels 1+2 of a connected device.

Audio from apps such as iTunes will therefore be mixed in at output 1— which is also the destination for processed guitar signals in all factory presets. (See "Input & Output Blocks" on p. 52 for more on routing processed signals to different outputs.)

If you want backing tracks and guitar signals to appear at *different* output jacks, you might think at first that you need to change every factory preset to use a different output. Not so. A simple setting change on your computer allows you to change the Axe-Fx III channels for system audio including iTunes and other non-multitrack apps. To change this setting:

1. First ensure that the Axe-Fx III is selected as your output device.
   - Connect the Axe-Fx III to your Mac as described on p. 18.
   - Open the Output pane of Sound preferences (choose Apple menu > System Preferences, click Sound, then click Output) and select the Axe-Fx III.
2. Open **Audio MIDI Setup** (find it in Utilities, inside your computer's Applications folder).
3. Select the **Axe-Fx III** device from the list, then click **Configure Speakers**.
4. Make sure that the Configuration is set to "Stereo" and then select the desired left and right channels.
5. Click Done to finish.

EXAMPLE:

Let's say your Axe-Fx is connected to a mixer for live performance. You want to use the Axe-Fx III as your audio interface and have processed guitar on one set of mixer channels and iTunes backing tracks on a different set of mixer channels. Route Mac system audio to **Analog 3+4**. Then **Axe-Fx III output 1 L+R** will be processed guitar, and **Axe-Fx III output 2 L+R** will be the backing tracks. You also have the benefit of two Axe-Fx III front panel level knobs.

Of course, if you are using a multi-channel audio application such as Logic or Mainstage, you can simply select the desired outputs on a track-by-track basis. The above only applies to *system* audio.

USB RE-AMPING

The Axe-Fx III USB Audio capabilities are perfect for “re-amping”, a method in which the raw un-processed DI output of a guitar is recorded and then re-processed later through the amp, cab, and effects of your choice.

Re-amping has many benefits. First, it allows you to record when inspiration strikes, capturing a DI instead of obsessing over final tone. Later, you – or a mixing engineer – can redesign a sound as your track’s production advances. Also, punches and edits made on the DI track are essentially made inaudible by the re-amping process.

In comparison to other products, the Axe-Fx III allows you to use the unit in the best and most effective re-amping workflows, including being able to dial in new tones while simultaneously listening to the track for context.

NOTE: This method is for USB Re-Amping. For analog reamping, simply copy Input 1 to Output 3 or 4 (p. 161).

STEP 1: RECORDING

The following tutorial assumes that the Axe-Fx III has been selected as the audio interface in your Digital Audio Workstation (DAW), and that you have connected **Output 1 L/R** to studio monitors, or that you are monitoring through the Axe-Fx III headphones jack. The details for each step below may vary slightly from one DAW to another, but this guide should be easy to adapt to your own environment.

1. In your DAW, select **Axe-Fx III** as the main audio interface. Set the main outputs to **Axe-Fx III Outputs 1+2**.
2. Create a new project in the DAW, and set its sample rate to **48kHz**. Record or insert any backing tracks.
3. Connect your guitar to the **Instrument** input and select a preset well-suited to the material you’re recording.
4. For this tutorial, we will record both the processed output and the DI.
   - Create a **mono** track. Name this track something like “Guitar DI”. Set its input to **Input 5**. This is the DI output of the Axe-Fx III. This will record the signal at the **Input 1 Instrument** jack with no processing.
   - Create a **stereo** track. Name this track something like “Guitar Reference 1”. Set its inputs to **Axe-Fx III Inputs 1+2**. This will record the processed guitar signal sent on the grid to the **OUT 1** block.
5. **Arm** both of your guitar tracks for recording, making sure that **Software/Input Monitoring** is *NOT* enabled.
6. Hit **RECORD** and roll the track. You should see signal on both tracks. Be aware: while the processed track will show “hot” levels, the signal level on a DI track will appear to be *very low*. This is NORMAL! You are recording the instrument level signal exactly as it comes out of your guitar with no pre-amplfication.

STEP 2: RE-AMPING

For basic re-amping, we will temporarily change one setting on the Axe-Fx III. Press **HOME** and open **SETUP**. Navigate to the **I/O** menu, and change **Input 1 Source** to “USB (CHANNELS 5/6)**”**. Remember to change this back when you finish re-amping.

1. **MUTE** the “Guitar Reference 1” track.
2. Change the output of your “DI” track to **Axe-Fx III Output 5/6.**
3. To demonstrate that re-amping is working, choose a totally different preset on your Axe-Fx III. Solo your DI track, then rewind and hit PLAY. You should hear the DI being processed by your new Axe-Fx III preset.
4. Record the re-amp. Create a stereo track. Name this track something like “Guitar Re-Amp 1”. Set its inputs to **Axe-Fx III Inputs 1+2**.
5. Arm your re-amping track for **recording**, making sure that **Software/Input Monitoring** is *NOT* enabled.
6. Rewind, hit **RECORD**, and roll the track. The output of the preset is recorded.

Again, the method described here is basic, but it should serve as a guide to get you started. Review the previous page to inspire other workflow ideas.

4 SETTING UP

This section provides an overview of popular Axe-Fx III setups. This is just the beginning. The number of inputs and outputs, combined with the new modular Input/Output block approach, makes possible an incredible number of rig designs, whether combining applications or creating totally new ones.

GENERAL PRINCIPLES

POWER

- When creating a new setup, begin with all level knobs turned down and all devices powered off.
- The Axe-Fx III features pop suppression on startup, but speakers or monitors should still be powered on last whenever possible.
- Basic surge/spike protection is recommended.

AXE-FX III INPUTS

- When connecting a guitar to the Axe-Fx III, the front or rear **Input 1/Instrument** jacks are preferable. These feature "Secret Sauce IV" circuitry, which lowers the noise floor using a proprietary technique and special analog input circuitry. The front jack overrides the rear one, so you can connect without changing menu settings even when the rear jack is utilized in a rack system.
- You can also connect guitars and other instruments to **Inputs 2, 3 or 4** with no problems. These are designated as LINE level but accommodate high impedance sources so as not to load down guitar pickups.
- When connecting the outputs of other devices to inputs on the Axe-Fx III, use appropriate cables and connectors, i.e. for balanced signals use XLR cables or balanced 1/4" connectors. Regular guitar cables are fine for unbalanced connections to the inputs of the Axe-Fx III.
- Setting **Input Levels** correctly is important. See "Setting Levels" on p. 5.

AXE-FX III OUTPUTS

- Outputs 1 and 2 default to -10 dBV. You may wish to change these to +4 dBu when connecting to to professional audio equipment. Find these settings on the **Audio** page of the **I/O** menu under **SETUP**.
- If a connected device has **balanced XLR inputs**, it is best to use the XLR outputs of the Axe-Fx III with XLR cables.
- If a connected device has **balanced 1/4" TRS inputs**, it is best to use the balanced XLR outputs of the Axe-Fx III with XLR-to-TRS cables.
- If a connected device has **unbalanced inputs**, use the unbalanced 1/4" outputs of the Axe-Fx III. **Humbuster cables** are not required, but can help prevent the noise associated with ground loops. Purchase Humbuster cables at shop.fractalaudio.com or from your nearest Fractal Audio Systems dealer. (See p. 6.)

MONO/STEREO

- Any setup can easily be adapted for mono or stereo with some simple changes to input and output settings.
- See "Mono vs. Stereo" on p. 6 for an introduction, and "I/O: AUDIO Page" on p. 159 for more detail.

USING THE AMP BLOCK WITH TRADITIONAL GUITAR SPEAKERS

- When using the Axe-Fx III amp block into traditional (*not* FRFR) guitar speakers, certain advanced parameters warrant special consideration. On the "Speaker" page of the Amp block, set **Speaker Drive** to zero. You may also wish to zero **Speaker Compression**. Also consider matching **LF Resonance Frequency** to match the resonance of your connected cab — if you know it.

FRFR/DIRECT

Global Settings: Default
Presets: Factory or Custom

This type of setup takes full advantage of the ability of the Axe-Fx III to recreate all aspects of an "end-to-end" guitar chain—stompboxes, amps, cabs, post effects, and more. It is the most versatile and popular setup, representing the ultimate in tonal flexibility.

Output 1 is connected directly into full-range, flat response ("FRFR") speakers such as those offered especially for guitar players by a number of manufacturers. Such systems are typically self-amplified ("active" or "powered"), but may consist of separate "passive" full-range speakers and a full-range power amp. Other suitable FRFR systems include high-quality PA speakers (including "Personal Monitor" wedges) or studio monitors.

In some cases, signal from the Axe-Fx III may pass through a mixer first. That's fine, as long as you connect to LINE inputs. You do NOT need to use a mic preamp! Contact your mixer's manufacturer if you have any questions.

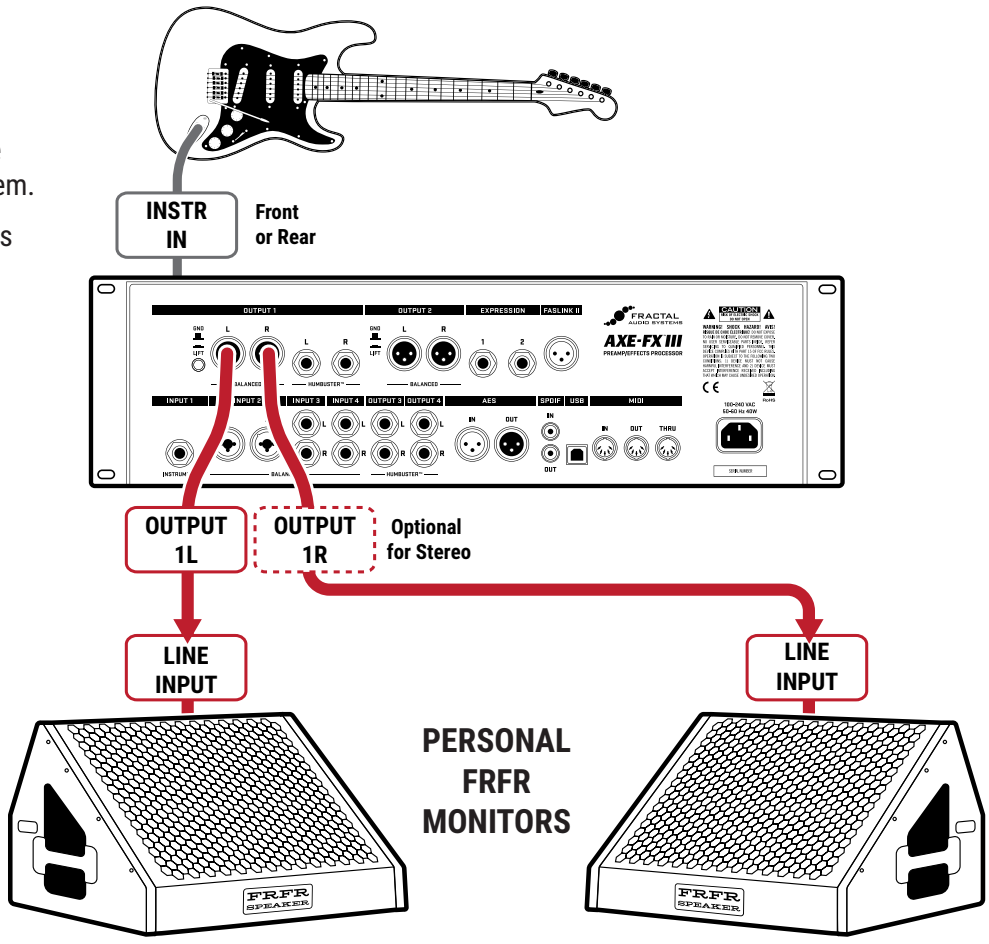
All Global and I/O settings are at default values. All Axe-Fx III factory presets are designed for this setup.

Adjust overall levels using the front panel **OUT 1** knob.

This is also the setup for a Digital Audio Workstation (DAW). Just add a computer connected via USB (don't forget drivers if you're running Windows) and the speakers connected to your Axe-Fx III become the main monitors for the DAW. Headphones can also be used.

CONNECTIONS

- Connect your guitar to **Input 1 (Instrument).**
- Connect **Output 1** to the input of your FRFR System.
- The presets used for this setup also work great with **headphones**.



FRFR MONITOR + FRFR TO FRONT-OF-HOUSE

Global Settings: "Out 2 Copy Out 1" Turned ON
Presets: Factory or Custom

This type of setup is similar to the previous FRFR/Direct setup except that it also sends signal to a mixer for front-of-house mixing (and possible band monitors). In this configuration, the Axe-Fx III creates a fully modeled end-to-end guitar signal path for the ultimate in tonal flexibility—stompboxes, amps, cabs, post effects, and more.

Output 1 is connected directly to your own monitors. Use full-range flat response ("FRFR") monitor speakers such as those offered especially for guitar players by a number of manufacturers. These might be self-powered or passive, with an external power amp. High-end PA monitor wedges can also be fantastic for this purpose.

Output 2 is used as an independent *copy* of Output 1. For this example, this is accomplished with a Global settings change rather than inserting an Output 2 block in every preset. (See "Connections" below for details.)

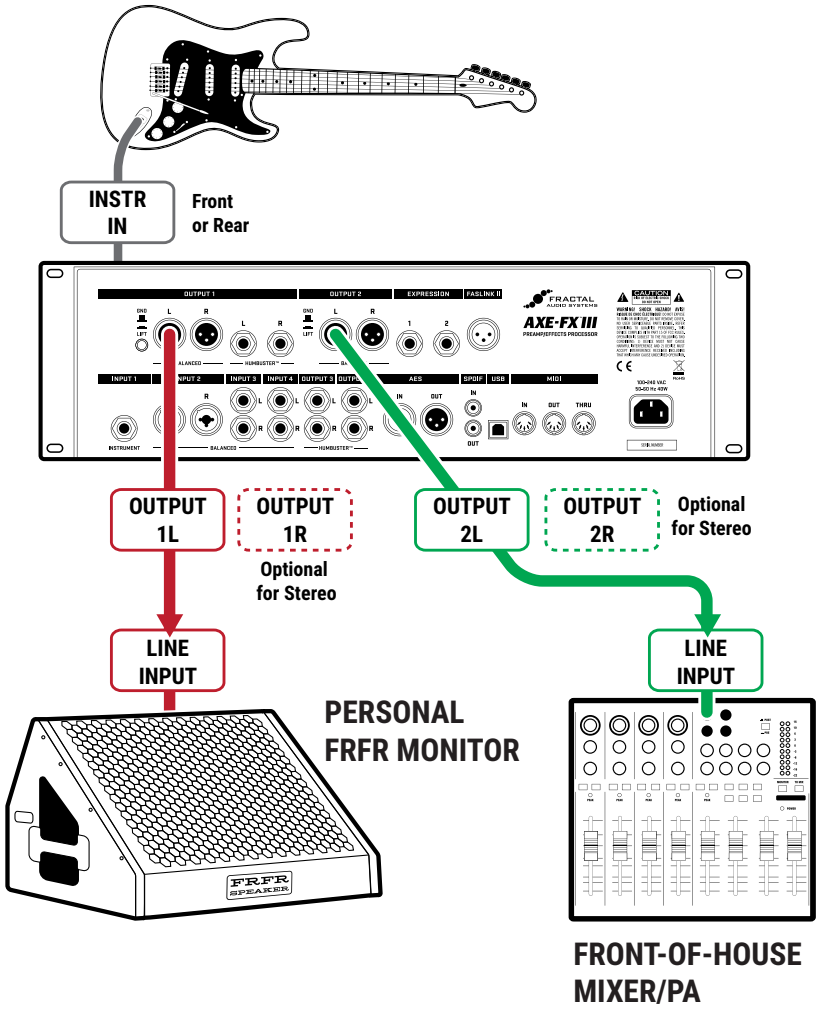
In the event that your monitors are stereo but front-of-house wants mono, consider whether you want to send them half-stereo or summed stereo and change **SETUP: I/O: OUTPUT 2 Mode** (see "Mono vs. Stereo" on p. 6).

Adjust you own monitor level using the front panel **OUT 1** knob.

Adjust your send level to front-of-house using the front panel **OUT 2** knob. One advantage of this setup is that you can adjust stage levels and house levels independently.

CONNECTIONS

- Connect your guitar to **Input 1 (Instrument).**
- Connect **Output 1** to the input of your FRFR System.
- In **SETUP: I/O: Audio**, set **Output 2 Copy Output 1** to "ON".
- Connect **Output 2** to the front-of house system.
  - This will presumably be a balanced connection, so use XLR cables if so.
  - If you are working with a sound technician, be sure to tell them that you are outputting a hot LINE level input — NOT a microphone level signal. They should use a balanced line input without a preamp.
  - Also tell them you are sending a fully processed "mix-ready" sound, and that — at least to start — they should initialize the channel with no EQ or processing.



POWER AMP & GUITAR SPEAKERS

Global Settings: Modified (see below)
Presets: Factory or Custom

This popular setup uses a power amp and traditional guitar speakers, which are *NOT* full-range or flat response, so speaker simulation is disabled globally. Blocks are used to recreate stompboxes, amps, and studio effects. Depending on your amp, power amp modeling may also need to be disabled globally.

DISABLE SPEAKER CAB MODELING

IMPORTANT: For this setup, bypass **Cabinet Modeling** under **Global Settings**.

- Navigate to **SETUP**: **Global Settings**: **Config**.
- Change **Cabinet Modeling** to "BYPASSED".

CONSIDER YOUR POWER AMP...

There are two categories of power amp that you can use with the Axe-Fx III. A sonically "**neutral**" power amp is generally the preferred variety. Such amps (mostly solid-state) allow you to use **Power Amp Modeling** in the Axe-Fx III, which accurately recreates the appropriate power amp tone and dynamics for your selected amp model.

The second category is "**non-neutral**" power amps, typically tube-powered, and made specifically for traditional guitar rigs. This includes not only rackmount amps, but the power amp in a tube head or combo—usable for a rig like this one when you plug directly in to the amp's FX RETURN jack. The non-neutral power amp can sound great, but you'll be "stuck" with the tone and dynamics of that particular amp, and you will almost certainly want to disable **Power Amp Modeling** as detailed below:

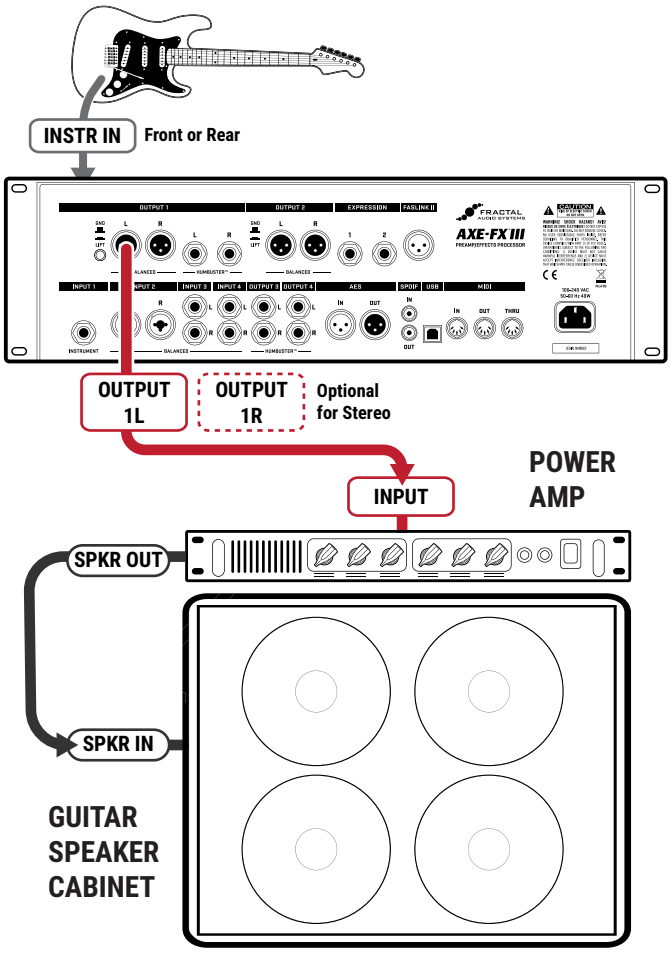
DISABLE POWER AMP MODELING

IMPORTANT: when using a non-neutral power amp, set **Power Amp Modeling** under **Global Settings** to **OFF**.

- Navigate to **SETUP**: **Global Settings**: **Config**.
- Change **Power Amp Modeling** to "OFF".

CONNECTIONS

- Connect your guitar to **Input 1 Instrument**.
- Connect **Output 1 L** to the input of your power amp, which is assumed to already be connected to a guitar speaker cabinet. You can also connect to the FX Return jack on many amp heads/combos.
- When using 1/4" outputs, **Humbuster** cables are recommended to reduce hum from ground loops.



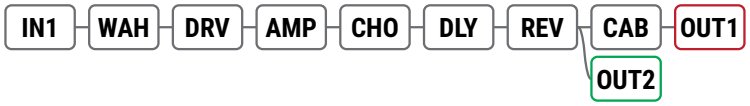
POWER AMP & CAB + FRFR/DIRECT

Global Settings: Default, check I/O Mono/Stereo settings
Presets: Custom

In this setup, custom presets simultaneously send two different signals—WITH speaker sims sent to a full-range front-of-house/monitors P.A. mix, and one WITHOUT speaker sims, sent to a "backline" rig consisting of a power amp and traditional guitar cab. For this example, we are assuming that the backline system uses a neutral power amp.

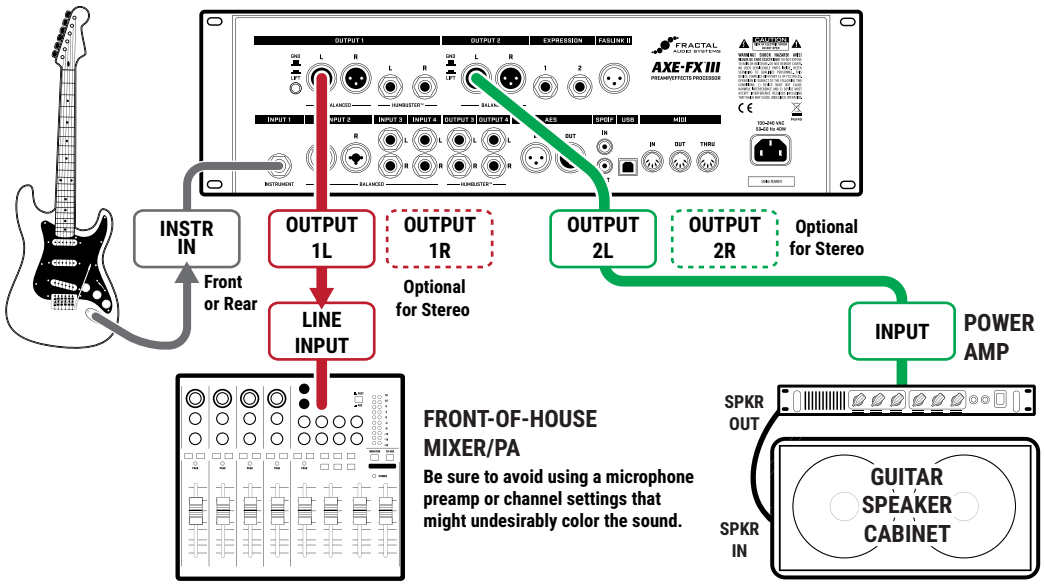
Output 1 connects to a full-range system—typically the house PA—sending a complete virtual rig tone, including simulated speakers. This provides all the benefits of going direct: versatile, consistent tone without "bleed" at very controllable volume levels for a better overall mix.

Output 2 sends a separate signal, almost identical to the first, but without speaker simulation—because it will be going to a power amp and traditional guitar speaker. This approach provides a very familiar "backline" experience—great natural sustain and feedback, "moving air," and familiar stage volume and presence.



This setup requires **custom presets** with both **Output 1** and **Output 2** blocks on the grid. Output 1 is at the end of the chain (where it appears in factory presets) and Output 2 taps the signal just before the cab block as shown below.

Note that the cab block has been moved to after the effects, which may collapse them to mono at Output 1. If a stereo signal is required at Output 1, the cab block must be modified for stereo use. Another solution would be to split the signal after the amp into separate chains of [Cab—Effects—Out 1] and [No-Cab—Duplicate Effects—Out 2].



CONNECTIONS

- Connect your guitar to **Input 1 (Instrument)**.
- Connect **Output 1** to the Front-of-House PA. Inform the sound technician that the Axe-Fx III outputs a direct LINE level signal which should initially not require any EQ or settings typically used on a mic'd guitar cabinet.
- Connect **Output 2** to the input of your power amp. Again, we are assuming a neutral power amp is in use (see note, p. 25).
- If using the 1/4" outputs, **Humbuster** cables are recommended to reduce hum from ground loops.

FOUR-CABLE METHOD (“4CM”)

Global Settings: Default, but see “Tip” Below **Presets: Custom**

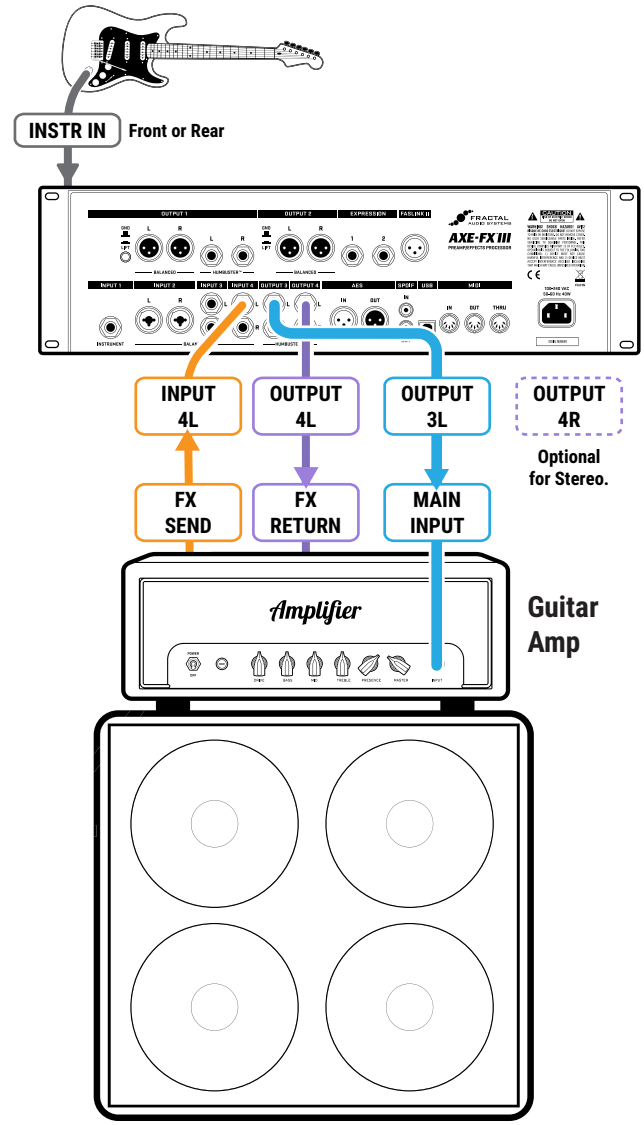
The Four-Cable Method (“4CM”) setup places the Axe-Fx III in two places at once with two separate processing chains. The first goes in front of your amp, where a chain of “pre” effects like wah and drive replace traditional stompboxes. The second placement is in the effects loop of the same amp, where a chain of “post” effects like delay and reverb appear. Although a head and separate cab are shown, many combo amps also offer an onboard effects loop and can be used as well.

4CM requires special presets with no Amp or Cab blocks. A stylized illustration appears below. Signal hits the Axe-Fx III first, where it is processed by **pre effects**. **Output 3** feeds the front of the amplifier. A chain of **post effects** is run in the amp’s FX loop using **Input 4** and **Output 4**. Note that the pre and post chains are not connected to each other at all on the grid. In fact, either chain can be as simple or as complex as desired.



CONNECTIONS

- 4CM requires custom presets with no Amp or Cab blocks (see above). A template is included (#382).
- Connect your guitar to **Input 1 (Instrument).**
- Connect **Output 3 L** to the input of your amp. Set the front panel **OUT 3** knob fully clockwise for unity gain operation. A **Humbuster** cable is recommended to reduce hum from ground loops.
- Connect your amp’s FX Send to **Input 4 L** on the Axe-Fx III. Input levels can be adjusted on the **Input** page of the **I/O** menu under **SETUP**. Set **Input 4 Mode** to “LEFT ONLY” on the **Audio** Page of the **I/O** menu under **SETUP**.
- Connect **Output 4 L** to the FX Return of your amp. Set the front panel **OUT 4** knob as desired for appropriate volume. A **Humbuster** cable is recommended. To extend this configuration for optional stereo, connect **Output 4 R** to the FX Return of a second amplifier, bypassing that unit’s preamp altogether.



TIP: OPTIMIZE FOR LOW NOISE

The optional **Boost/Pad** functions on Outputs 3 and 4 are designed to run the D/A converters at boosted levels, padding their outputs for even lower noise. To find the right setting, adjust either of these settings as high as possible without clipping the outputs, as indicated by a red LED on the front panel meters. You’ll actually hear the noise floor drop as you increase Boost/Pad. Find these options on the **Audio** page of the **I/O** menu under **SETUP**.

FX PROCESSOR ONLY ("POST")

Global Settings: check I/O Mono/Stereo settings, see "Tip" Below
Presets: Custom

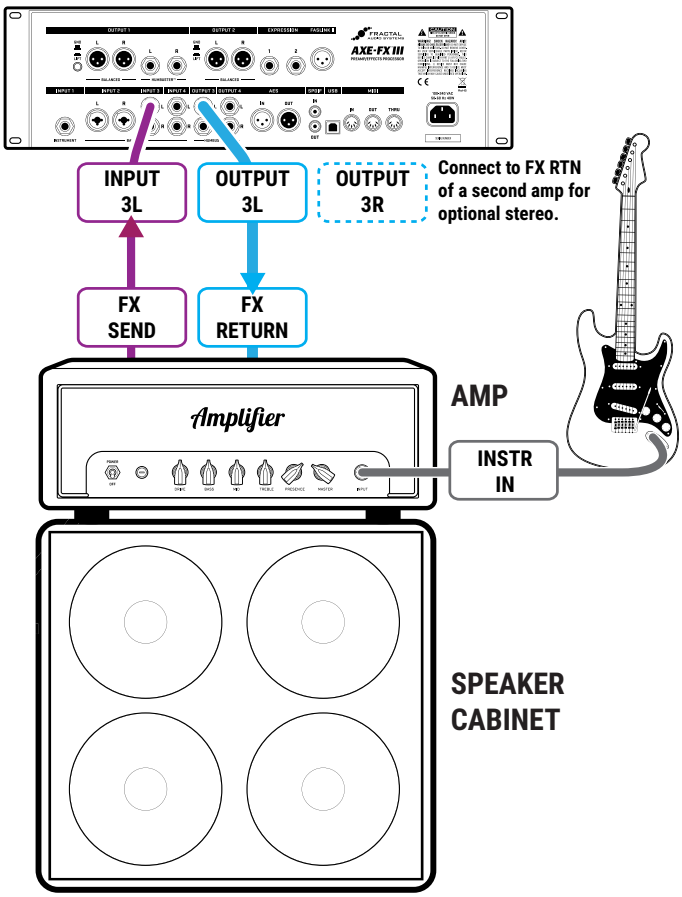
Although it was designed for complete rig replacement, the Axe-Fx III can also be put to superb use as a standalone FX processor, providing access to our industry leading effects for those who are not ready to make the switch to modeling.

This setup uses only three of the four cables used in the Four-Cable Method (p. 27) for operation in an amp's FX loop, providing post- but no pre-effects. Output 3 and 4 are designed for unity gain applications—which may or may not be important in this case depending on your amp. Here we'll use Output 3 and pair it with Input 3 for simplicity's sake.

You will need to create custom presets without Amp or Cab blocks, containing only those effects that sound good to you *after* preamp distortion: typically chorus, EQs, delays, reverbs, certain types of pitch shift, etc.

CONNECTIONS

- This setup requires custom presets with no Amp or Cab blocks.
- Connect your guitar to your amp's instrument input as usual.
- Connect the FX Send of your amp to **Input 3 L**. Input levels can be adjusted in **SETUP : I/O : Input**.
- In **SETUP: I/O: Audio**, set **Input 3 Mode** to "LEFT ONLY"**.**
- Connect **Output 3 L** to the FX Return of your amp. A **Humbuster** cable is recommended.
- Set the front panel **OUT 3** knob as desired for appropriate volume levels. If unity gain is required, set the knob fully clockwise.
  - To extend this configuration for optional **stereo**, you would typically connect **Output 3 R** to the FX Return of a second amplifier, bypassing that unit's preamp altogether.
  - Other types of stereo setup are of course possible, including using two totally different amps each connected on Left and Right sides of one I/O pair, or I/O 3 for one amp and I/O 4 for another. There are no limits except your imagination.



TIP: OPTIMIZE FOR LOW NOISE

The optional **Boost/Pad** functions on Outputs 3 and 4 are designed to run the D/A converters at boosted levels, padding their outputs for even lower noise. To find the right setting, adjust either of these settings as high as possible without clipping the outputs, as indicated by a red LED on the front panel meters. You'll actually hear the noise floor drop as you increase boost/pad. Find these options on the **Audio** page of the **I/O** menu under **SETUP**.

FX PROCESSOR ONLY (“PRE”)

Global Settings: check I/O Mono/Stereo settings, see also “Tip” below
Presets: Custom

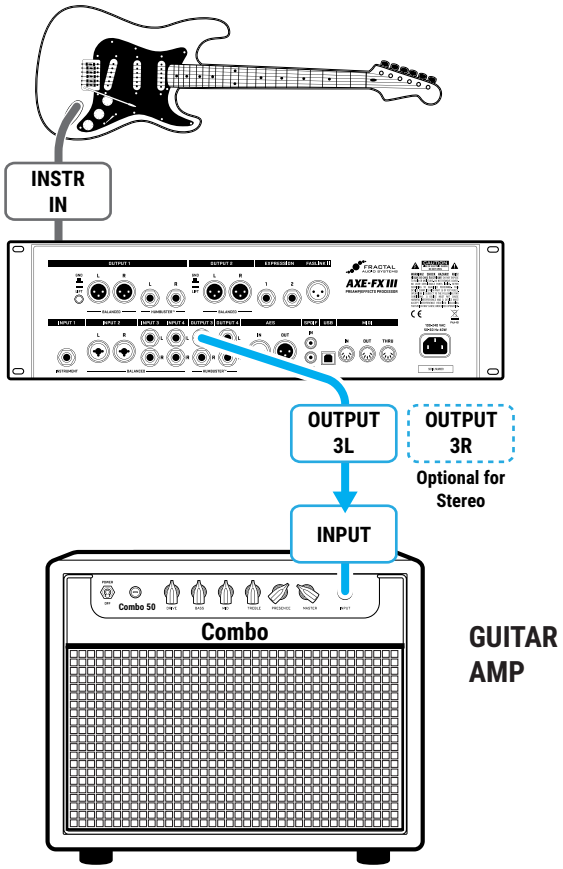
Here, the Axe-Fx III is used as the ultimate **virtual pedalboard**, providing access to our vast collection of industry-leading effects between guitar and amp, where traditional pedals might be placed. As with all Axe-Fx III setups, you are not limited by the stock offerings but can combine effects in any configuration imaginable, creating totally new sounds using the layout grid and its modular block architecture. Custom presets contain no Amp or Cab blocks.

This setup uses only two of the four cables used in the Four-Cable Method (p. 27). Outputs 3 and 4 are designed for unity gain applications—which is important in this setup. Here we’ll use **Output 3** and pair it with **Input 3** for simplicity’s sake.

You will need to create custom presets without Amp or Cab blocks, and those presets should be created with an understanding of how effects sound *in front of* your amp’s preamp and the distortion it generates.

CONNECTIONS

- This setup requires custom presets with no Amp or Cab blocks.
- Connect your guitar to **Input 1 Instrument** on the Axe-Fx III. Adjust input levels if required as detailed on p. 5.
- Connect **Output 3 L** to the input of your amp. A **Humbuster** cable is recommended.
- Set the front panel **OUT 3** knob fully clockwise for unity gain.
  - To extend this configuration for optional stereo, connect **Output 3 R** to the input of a second amplifier.
  - You might use the Axe-Fx III to select between two amps by changing Channels on the Output 3 block:
    - Channel A: Balance Center (Both Amps)
    - Channel B: Balance Left (Left Amp)
    - Channel C: Balance Right (Right Amp)
    - Channel D: (Not used)
    (See “Scenes & Channels” on p. 42 for more.)



TIP: OPTIMIZE FOR LOW NOISE

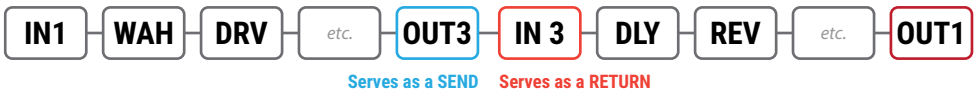
The optional **Boost/Pad** functions on Outputs 3 and 4 are designed to run the D/A converters at boosted levels, padding their outputs for even lower noise. To find the right setting, adjust either of these settings as high as possible without clipping the outputs, as indicated by a red LED on the front panel meters. You’ll actually hear the noise floor drop as you increase boost/pad. Find these options on the **Audio** page of the **I/O** menu under **SETUP**.

INSERTING OUTBOARD GEAR

Global Settings: As per your usual setup, check I/O Mono/Stereo Settings
Presets: Custom

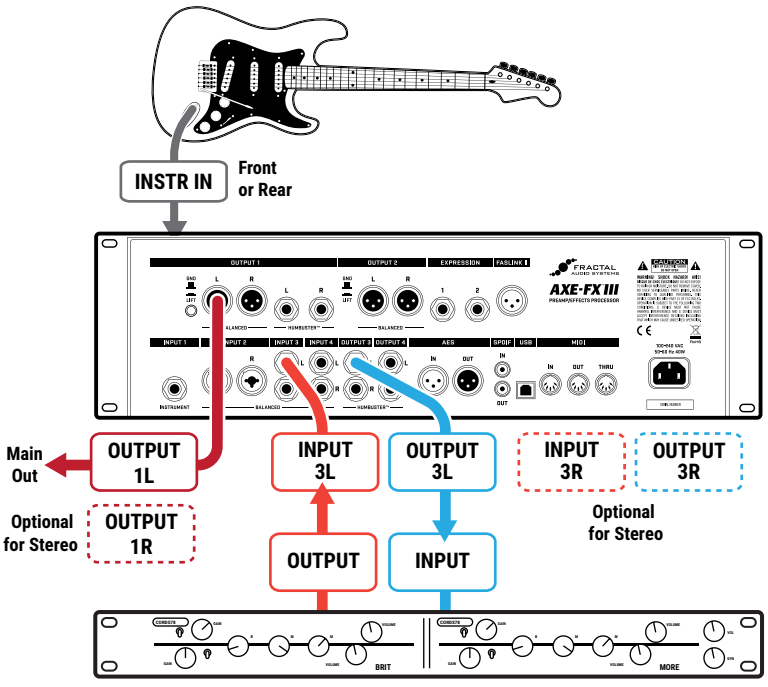
With the typical "direct" setup using only Input 1 and Output 1, the Axe-Fx III has enough I/O blocks left over for multiple "send-and-return" inserts of outboard gear. The simple example below imagines a 3rd-party device inserted using Output 3 (used as a "send") and Input 3 (used as a "return"). This setup can also be used with stereo inserts, or expanded with multiple I/O pairs for multiple outboard devices. Insert devices can even be added to other types of setups using these same techniques.

Custom presets are required for this type of setup, with output and input blocks added for the send and return points (see example below). The Axe-Fx III has no "FX LOOP" blocks, so inputs and outputs are placed individually on the grid. In a send/return pair, the Output block is connected to the Input block (as shown below), allowing the Input block (return) to serve as a master bypass control for the send/return loop.



CONNECTIONS

- Connect your guitar to **Input 1 (Instrument)**.
- This example assumes **Output 1** is connected to FRFR monitors.
- Connect **Output 3 L** as a "send" to the input of your outboard device. Adjust output levels using the front panel **OUT 3** knob with the following considerations:
  - Outputs 1 and 2 have options for –10 dbV and +4 dBu in **SETUP: I/O: Audio**.
  - Outputs 3 and 4 provide unity gain when the knob is fully clockwise.
  - You can also adjust the levels of an individual Output block.
- Connect the output of your outboard device to **Input 3 L** as a "return". Adjust levels on your device or use the trim controls under **SETUP: I/O: Input**.
- Optionally connect the right channel inputs and outputs for stereo insert devices.



OUTBOARD GEAR SUCH AS PREAMPS, PROCESSORS, PEDALS, ETC.

MULTI I/O SETUPS

The flexibility of the new Input and Output blocks allows some very cool and complex setups to be basically self-explanatory. The diagrams below show ideas for rigs which in the past might have required intricate explanations, but are now just presets.

The ideas in these layouts might inspire other setups, perhaps in combination with the rig setup diagrams found earlier in this section. For example, you might use extra Inputs and Outputs to insert outboard gear, or use different backline and front-of-house options. The only limit is your imagination.

For clarity only, all blocks are shown engaged instead of bypassed.

DUAL OUTPUT GUITAR

CPU: 82%

Here, Inputs 1 and 2 are used independently with the magnetic and piezo outputs of a dual-output guitar. The two signals eventually merge for some “mastering” effects before being passed to Output 1 in full layered stereo. Each chain has a huge complement of effects including amp and cab modeling. The amp for the piezo out might use a “Tube Preamp” model and an acoustic guitar simulation IR downloaded from Axe-Change.



FULLY INDEPENDENT GUITAR AND BASS

CPU: 83%

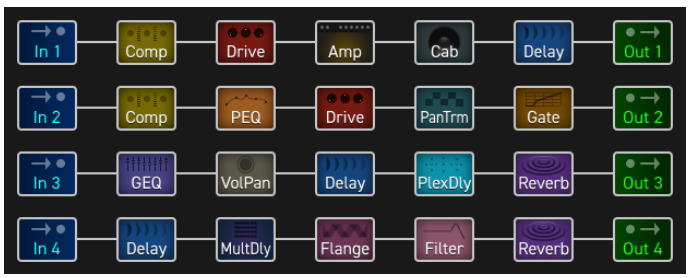
This rig processes full guitar and bass rigs at the same time! Guitar uses Input 1 and Output 1. Bass uses Input 2 and Output 2. This setup would have required two separate Axe-Fx II units to accomplish. You can easily imagine adapting a setup like this to process two electric guitars, or separate simultaneous electric and acoustic. By the way, these signal paths are copies of actual presets used by some artists who endorse Fractal Audio.



GUITAR CHAIN + MIXER EFFECTS

CPU: 84%

Let’s look at a more vertical example. Here, a mini FRFR guitar rig is combined with three rich sets of effects imagined to be integrated with a mixer for live performance. I/O pair 2 is used for a special vocal FX insert chain, complete with a raging Drive block. Chain 3 is a massive stereo shimmer for a keyboard rig, controlled by its own volume pedal. Chain 4 contains a bevy of effects, imagined to be used in different scenes on an Aux bus.



5 PRESETS

OVERVIEW

- The Axe-Fx III contains 512 preset slots, each a fully independent rig with its own amp, cab, effects, and more.
- The Axe-Fx III ships with various “Factory Presets” pre-loaded, but *every* preset is a “User” preset which can be changed or completely overwritten.
- If you want to restore the factory presets (or install the newer versions we occasionally release), you can download these from https://www.fractalaudio.com and re-install them using Fractal-Bot.
- Every preset has its own name that you can change as you store it.
- Presets are built on the “Grid” by inserting, connecting, and dialing in “Blocks”.
- Presets contain “Scenes” and Blocks contain “Channels”. Both of these subjects are covered separately in the section after this one.
- Compared to other products, the Axe-Fx III allows larger presets with greater capabilities. When you consider everything that is possible with Scenes and Channels, a single preset can easily cover an entire song, or even an entire gig.
- Please review “Intro to the Layout Grid” on p. 8 before reading this section.

LOADING PRESETS

Different areas of the Axe-Fx III provide different ways to load presets:

- **On the Home page** – Use the **NAV LEFT** and **NAV RIGHT** buttons or turn the **VALUE** knob.
- **On the “PRESETS” directory page of the HOME menu** – Select a preset and press **ENTER**. Presets in the directory are sorted in numerical order. To re-sort alphabetically, press the **SORT A–Z** button (Push-knob **A**).
- **FC CONTROLLER** – The FC Controllers offer a range of dedicated “preset select” options.
- **MIDI** – MIDI Bank and Program Changes select presets using the industry-standard “Controller 0 + PC” method. See “MIDI Reference Tables” on p. 183. MIDI Custom PC Mapping is also supported, see “Program Change Mapping” on p. 46.



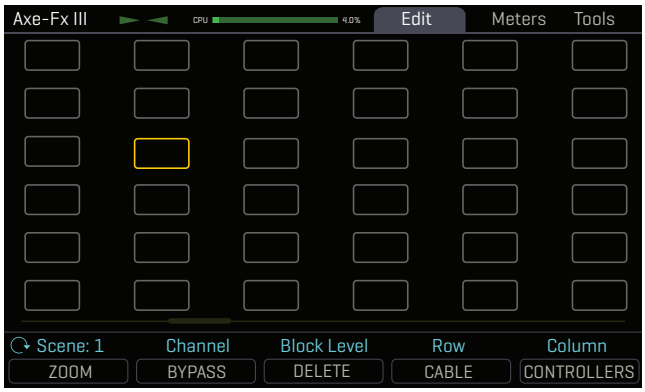
The Home Page

THE LAYOUT GRID

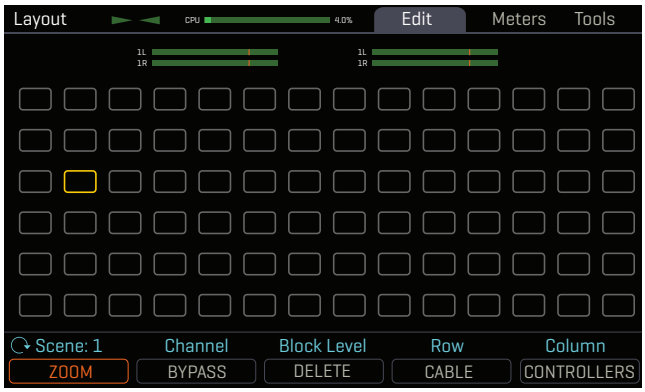
The grid is a 14×6 framework that presets are built upon. Think of grid rectangles as slots into which "blocks" are inserted and connected to determine signal path and processing. Each grid space and cable is stereo. You don't need separate left and right paths!

Access the grid from the **Home** page by pressing the **ENTER** button or the **LAYOUT** Push-knob.

In the default view, the Axe-Fx III display shows only a 6×6 section of the 14×6 grid. Navigate to off-screen areas using the **D** or **E** knobs, or the **NAV** buttons. A bottom scrollbar indicates where you are in the overall left-to-right layout. To show the entire grid at once, zoom out with the **ZOOM** button (Push-knob **A**).



The grid normally shows six rows and six columns. You can scroll left or right with navigation controls.



Zoom out to show the entire grid at once. This also reveals VU meters for leveling scenes and presets.

WORKING WITH BLOCKS

As explained in "Intro to the Layout Grid" on p. 8, the Axe-Fx III grid must be populated with **Blocks**—components pulled from a large inventory of amps, cabs, stompboxes, studio effects, mixers, and more. To interact with the grid you must move the **cursor** — a yellow rectangle controlled by the **D** or **E** knobs, or the **NAV** buttons. For simplicity, we'll refer simply to these controls as "**NAV**" going forward.

TO INSERT A BLOCK…

- **NAV** to the desired grid location.
- Turn the **VALUE** wheel to step through the list of blocks, including options for **SHUNT** (see below) and **NONE**.
- Press **ENTER** to confirm or **EXIT** to cancel changes.
- As you insert blocks, they are removed from the list, but every preset has its own complete inventory.

TO CHANGE OR REMOVE AN EXISTING BLOCK…

- Select the block and turn **VALUE** to change it to the desired type. To REMOVE a block, select "NONE".
- Press **ENTER** to confirm or **EXIT** to cancel changes.

DELETE

A shortcut makes it easy to remove an existing block or convert it to a shunt. Select the block and press the DELETE button (Push-knob C). A deleted block is replaced by a shunt. Delete any shunt to leave an empty space.

TO BYPASS OR ENGAGE A BLOCK ON THE GRID…

- Use **NAV** buttons to select the block.
- Press the **BYPASS** button (Push-knob **B**). Bypassed blocks are dimmed or "grayed out" on the grid.

RESETTING A BLOCK/CHANNEL

Blocks on the Axe-Fx III are "initialized" when you place them on the grid, but you may want a fresh start after making various changes. **RESET** takes only two button presses and can be performed at any time.

RESET

- If it isn't already open for editing, select the desired block on the grid and press **EDIT**.
- Press the **RESET** button (Push-knob **A**). You are prompted to reset the *current channel*.
- Press **ENTER** to confirm. You can reset other channels in a block as they are used.

SHUNTS

A shunt is a sonically transparent block—a mere tool which carries signal through otherwise empty grid locations. Your preset will be silent unless you make a complete connection from the Input block to the Output block, so place a shunt wherever you don't use another block like an Amp, Cab, or effect. Like grid spaces, a shunt is stereo; you don't need two for left/right.

TO INSERT A SHUNT…

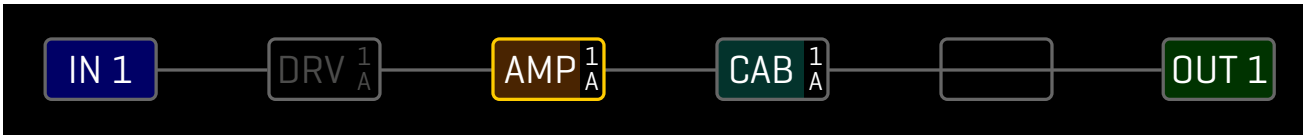
- **NAV** to the desired grid location and turn **VALUE** until "**SHUNT**" is displayed.
- Press **ENTER** to confirm or **EXIT** to cancel changes.

INPUTS AND OUTPUTS

Input and **Output** blocks are required to place input signals on the grid or pass signal to output jacks. All factory presets use Input 1 and Output 1. Other setups may require different Input and Output blocks.
See "Input & Output Blocks" on p. 52 for additional information.

BLOCK DISPLAY TYPES

The color display shows different grid block states as follows:



Input | Bypassed | Selected | Normal | Shunt | Output

GLOBAL BLOCKS

Global Blocks allow settings for a given block type to be automatically synchonized across multiple presets. For example, a global AMP block could be used in several presets. When any setting of that amp is changed on one, the others update automatically. Global blocks are detailed in Section 8 on p. 141

MOVING BLOCKS, ROWS, AND COLUMNS

The **Tools** page of the **Layout** menu contains various utilities to **MOVE** individual blocks or entire rows/columns UP, DOWN, LEFT, or RIGHT. When a block or a grid row/column is moved, it changes places with the item in the space it is moved to. This may result in certain connector cables being modified or removed, so be sure to observe how the elements of your preset are interconnected before proceeding with a move operation.

- Open the **Tools** page of the **Layout** menu.
- Select a function with the **FUNCTION** knob (**A**): Move Effect/Column/Row, Left/Right/Up/Down.
- Use the **NAV** buttons to select the target Block, Row, or Column you wish to move.
- Press **ENTER** to execute the move.

CONNECTOR CABLES

Just like physical gear, the blocks in the Axe-Fx III need to be connected together for signal to flow. This is done using virtual cables, which run from one grid block to another. With even one connector missing, your preset may be totally silent! Like shunts, connectors are *totally* transparent.

TO CREATE A CONNECTOR CABLE…

- On the grid, navigate to the block where you wish the cable to BEGIN. You can't start from an EMPTY grid space!
- Press the **CABLE** button (Push-knob **D**), or **ENTER**. The selected block and its neighbor to the right will alternatingly flash.
- **NAV** to the desired destination block, which must be in the next column to the right.
- It is possible to select an empty location. A shunt will be added there if you complete the cable.
  - Be sure to select a destination that is not *already* connected to the block you started from, or you will REMOVE that cable (see below).
- Press **ENTER**. To cancel without connecting, press **EXIT** instead.

TO REMOVE A CABLE CONNECTOR…

Cables are removed in much the same way as they are created.

- On the grid, **NAV** to the block where the cable begins.
- Press the **CABLE** button (Push-knob **D**), or **ENTER**. The selected block and its neighbor to the right will alternatingly flash.
- **NAV** to select the "other end" of the cable you wish to remove.
- Press **ENTER**. To cancel without removing, press **EXIT** instead.



SHORTCUT: To span multiple empty grid columns with a series of shunts and cables, select any block that is followed by a series of empty spaces, then **press-and-hold** the **ENTER** button. The intervening spaces will be automatically filled with shunts and connected with cables. Careful: any *existing* cables encountered along the way will be REMOVED!

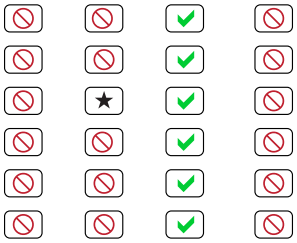


Remember, on the Axe-Fx III grid, each component is STEREO. Shunts, cables, and most blocks are stereo in/stereo out. The grid allows up to six full stereo paths, and you do NOT necessarily need to create parallel grid paths for stereo! Some blocks internally process audio in mono (such as Amp or Drive) but even these generally have **Input Select** *and* **Output Balance** *parameters.*

THE RULES OF AXE-FX III CABLES

- No cables = No sound. Even one missing link will break the chain.
- Signal flows from LEFT to RIGHT.
- A cable MUST originate from a BLOCK or a SHUNT. Empty locations are not viable origins.
- If you try to connect to an EMPTY location, a SHUNT will be created there.
- You can ONLY connect to blocks in the next column to the right.

The diagram below illustrates some of the ideas above:



The ★ represents the origin of a connector cable.

The ✔ shows valid possible destinations.

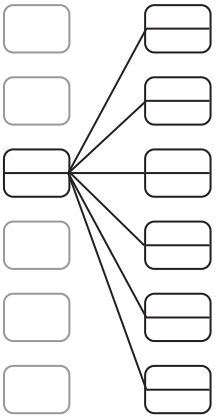
The ⃠ symbol shows destinations that are illegal/unavailable.

Any columns farther left or right would also be illegal/unavailable.

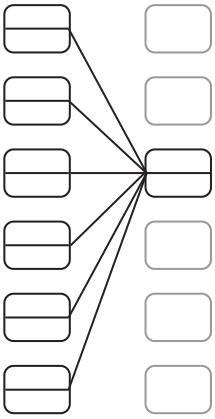
If the ★ were in a different ROW, every ✔ would still be in the same place.

- You may freely SPLIT or MERGE up to six ways at any point. This is sonically transparent and there is zero risk of signal degradation or phase problems. CROSSING is also possible. Here are some examples:

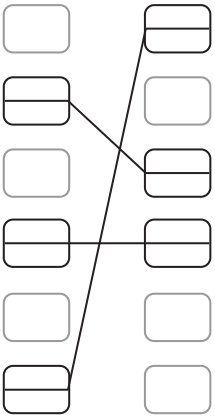
Split



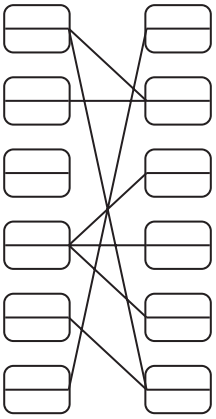
Merge



Cross



Complex



BLOCK INVENTORY

The following table contains an overview of every block. Each preset has the entire inventory to choose from.

BLOCK		DESCRIPTION	PER-PRESET	CHANNELS	TYPES
AMP	Amp	Here it is! 260+ amp models in one block!	2	4	260+
CAB	Cab	Speaker cab simulation offering our patented Ultra-Res Technology.	2	4	2237
CHO	Chorus	Create classic mono and stereo modulation effects including vibrato.	2	4	14
CMP	Compressor	Control dynamics and add sustain.	2	4	6
XVR	Crossover	Split a signal into high and low frequency components.	2	2	1
DLY	Delay	Up to 8000 ms of delay, with types for analog, digital, tape, and more.	4	4	18
DRV	Drive	25+ types including boost, overdrive, distortion, fuzz, and many more.	4	4	37
ENH	Enhancer	Classic and modern modes to create and control spatialization.	2	4	2
RTN	Feedback Return	Receives signal from the Feedback Send block.	2	1	1
SND	Feedback Send	Transmits signal to the Feedback Return block.	2	1	1
FLT	Filter	Includes low pass, high pass, band pass, and many other types.	4	4	12
FLG	Flanger	Various types cover everything from subtle modulation to extreme jet.	2	4	8
FOR	Formant	Create dynamic vowel sounds with this multi-mode formant filter.	2	4	1
GTE	Gate/Expander	Useful for everything from subtle control to extreme effects.	4	4	1
GEQ	Graphic EQ	A variety of modes allow easy, flexible tone sculpting.	4	4	12
IN	Input	Injects the signal from physical inputs onto the grid.	5	4	1
LPR	Looper	A powerful looper with great remote control options.	1	1	1
MGT	Megatap	This 40-tap delay creates fantastic sonic patterns.	2	2	1
MIX	Mixer	Allows you to mix up to six stereo signals.	4	4	1
MBC	Multiband Comp	Three-band compressor that is great for mastering or dynamic EQ.	2	4	1
MTD	Multitap Delay	A variety of special delays including diffuser, quad-tap, and more.	2	4	5
MUX	Multiplexer	This input selector routes one of many inputs to an output.	2	4	1
OUT	Output	Transmits signal to the corresponding physical output jacks.	4	4	1
PEQ	Parametric EQ	The 5-band parametric equalizer allows precise control of tone.	4	4	1
PHA	Phaser	A variety of vintage and cutting edge phaser effects, including ‘vibe.	2	4	8
PIT	Pitch Shift	Includes detune, harmonizer (intelligent/custom), whammy, and more.	2	4	12
PLX	Plex Delay	Up to eight delay lines interacting in a matrix. Gorgeous!	2	4	3
RES	Resonator	Resonant comb filters in parallel. Create chords and more.	2	2	1
REV	Reverb	World-class recreations of vintage springs, rooms, halls, and more.	2	4	44
RNG	Ring Mod	The extremely flexible ring modulator provides for a range of cool effects.	1	2	1
ROT	Rotary	Simulates a classic rotary speaker with multiple microphones.	2	4	1
RTA	Realtime Analyzer	Visually plots the frequency spectrum of an input signal.	1	1	1
SYN	Synth	A 3-voice monophonic synth that tracks what you play.	2	4	1
TTD	Ten-Tap Delay	Set the time, pan, and spacing of one to ten separate echoes.	2	4	2
TMA	Tone Match	Matches the Axe-Fx III to the sound of an amp, recording, or device.	1	1	1
TRM	Tremolo	Creates classic trem, plus auto-pan or extreme psycho acoustic effects.	2	4	2
VOC	Vocoder	Digital re-creation of the analog classic. “*Stop the human. Stop the intruder*.”	1	1	1
VOL	Volume/Pan	Simple volume block also offers channel input/output tools.	4	4	1
WAH	Wah	The classic wah, with multiple types based on classic originals.	2	4	8

Each preset also includes a set of **Controllers** for automation, and a **Scene MIDI** block.

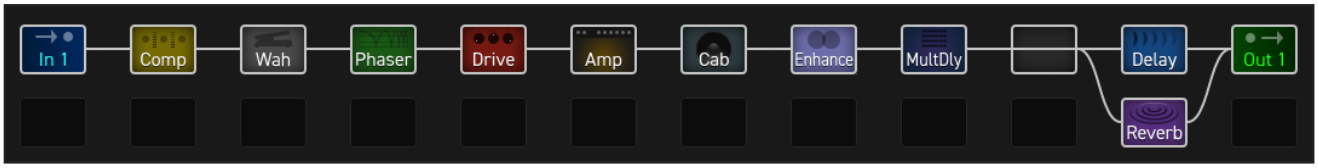
EXAMPLE PRESET GRIDS

Several preset illustrations appear below, to explain how blocks are combined to form presets.
These illustrations are screen-captures from Axe-Edit III, rather than representations of the small screen.

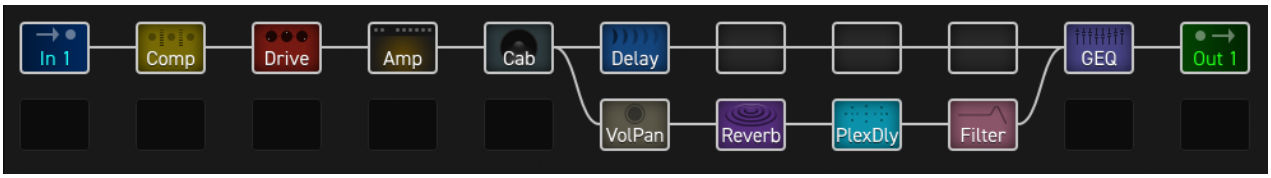
Ex 1: Sweet and Simple. Wah. Drive pedal. Amp and Cab. Delay and Reverb post.



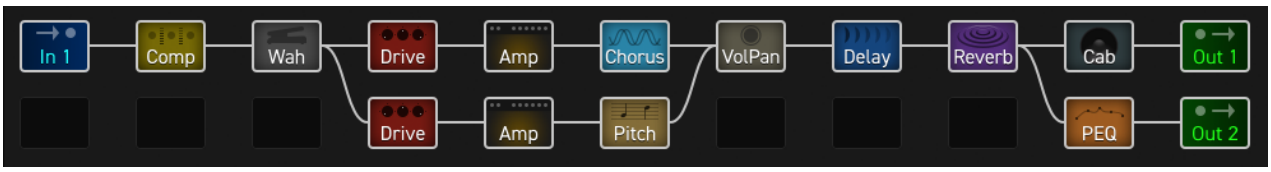
Ex 2: Here some more effects are added, and the reverb is run in parallel to the delay for a different sound.



Ex 3: An entire parallel chain for a shimmery, filtered effect with a VOLUME block for its own "Send" control.

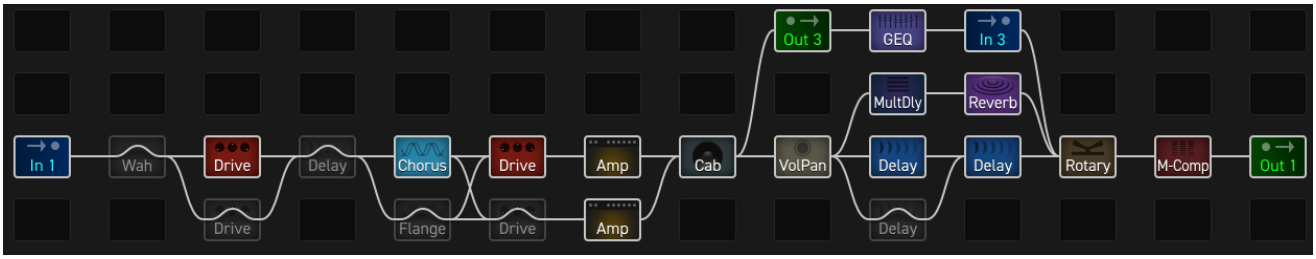


Ex 4: A complex preset with effects *before* and *after* two amps. A stereo cab feeds OUT 1 (Front of House), while Output 2 taps the signal with NO cab for use with a neutral power amp & real guitar speakers (on-stage backline).



Ex 5: A highly complex preset with two amps, parallel paths, outboard gear, and 4 each of drive and delay blocks.

(This preset is the first on this page to show some blocks which are bypassed.)



EDITING EFFECT BLOCKS

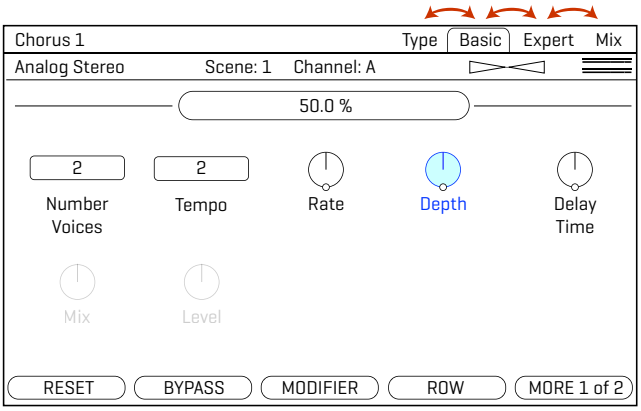
Blocks are fully programmable, allowing you to dial in every setting as desired. The **Edit** menu for every block contains one or more **pages,** each with multiple **parameters** which control various functions. Here's a quick guide covering how to access and work with the different types of Edit menu pages.

OPENING AN EDIT MENU

- On the **Layout** grid, select the desired block.
- OR... from any page, press **EDIT** to jump right to the **Edit** menu for the selected block (or the first block if none is selected).

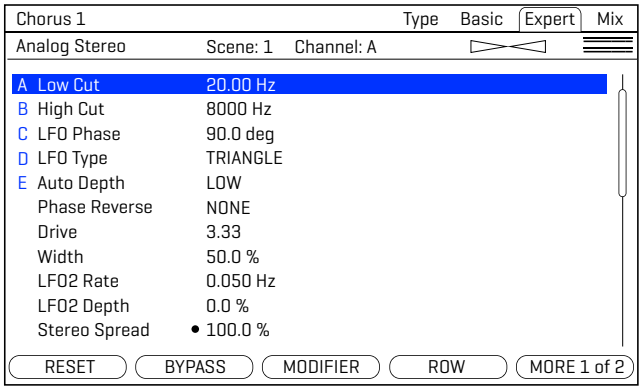
TURNING PAGES

- Most Edit menus have multiple pages, shown as "tabs" at the top of the menu (red arrows, right).
- Tap the **PAGE** buttons to page left or right.



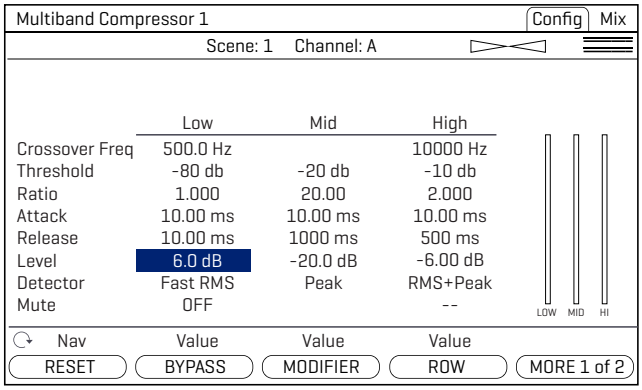
EDITING "KNOB" PAGES

- Most Edit menu pages show rows of up to five virtual controls like knobs, switches or selectors. To make changes, use the five knobs below the display.
- You will hear all changes in real time.
- Switch between rows using **NAV UP/DOWN**.



EDITING "MENU" PAGES

- Some pages have vertical lists of parameters.
- Use the **NAV UP/DOWN** buttons to step through the list or **NAV LEFT/RIGHT** buttons to jump.
- The **A** knob and the **VALUE** knob operate the selected parameter.
- **B**, **C**, **D**, and **E** knobs control other parameters in the list, as shown by labels on the list.



OTHER PAGE TYPES

- Some blocks have special Edit menus, with parameters arranged in rows and columns. Some of these include interactive meters or graphs. Use **NAV** and **VALUE** or **A**,**B**,**C**,**D**,**E** knobs to move around and make changes.
- For **TYPE** pages, such as Amp Type, just **NAV** through the list to make a selection and turn the page or press **EXIT** when finished. Selections take effect instantly.

SAVING CHANGES

After editing a preset, you will want to save the results.

Every preset in the Axe-Fx III can be modified. There are no permanent presets.

When you change the current preset in any way, the front panel "EDITED" LED lights until you STORE or load a new preset.

TO STORE A PRESET…

- Press **STORE** to show the STORE page.
- Press **ENTER** to display “Do you want to overwrite the Preset?”
- Press **ENTER** again to confirm, or **EXIT** to cancel.
- The message “SAVED!” is shown when saving is complete.

TO CHANGE THE PRESET NAME OR LOCATION…

The Axe-Fx III has 512 preset memory locations. You can edit the name of any preset while storing.

- Press **STORE** to show the STORE page.
- **NAV** down once to the PRESET line.
  - Turn the **B** knob to move the cursor.
  - The **C** knob selects **upper case** letters.
  - The **D** knob selects **lower case** letters.
  - The **E** knob selects **numbers**.
  - The **VALUE** knob selects ALL characters, including **symbols**.
  - You can use up to 31 characters in a preset name.
- You can also **NAV** to any Scene name to edit it in the same way.
- Press **ENTER** to Store, then press **ENTER** again to confirm.
- The message “SAVED!” is displayed and the new location (if any) is loaded.



Scene naming is a powerful new capability. Aside from stating what a specific scene is intended for, the names of un-used scenes can be used for short notes or reminders. Shorthand may help. For example, a “slash” could indicate expression capability on an “External” controller:
“FB /1, Time /2” could show that **Feedback** *is controlled by “External 1” and* **Mix** *by “External 2”.*

PRESET CPU LIMITS

Each block you place in a preset contributes to the total CPU load. You can check the current load at any time by viewing the mini CPU meter at the top of the grid.

The maximum allowable load is 90%. There are safeguards to prevent you from pushing the Axe-Fx III too far. First, you are stopped from inserting any block which might cause an overload by the message “INSUFFICIENT CPU”. The Axe-Fx III assumes that a block will be used to its limits when making this determination. If you manage to get the load above 90%, audio performance may be compromised.

If you are prevented from inserting an effect because of CPU limits, you can make changes to reduce the current CPU load and try again. You might remove an effect that is less important. Adjusting certain parameters can also help. But beware of trying to set parameters back to higher values after inserting the desired block. At extremely high CPU usage levels, the Axe-Fx III may become unstable.

Here are a few of the common block parameters whose settings have a measurable effect on CPU usage:

- **Amp: Output Comp**: Set to "0" for the least CPU usage.
- **Cab: Mute**: The more IRs you use per block, the higher the CPU usage.
- **Reverb: Quality** and **Echo Density**. Lower = less CPU. The “Spring” type uses the least CPU.
- **Compressor**: Set **Type** to one of the "PEDAL" options to use less CPU.
- **Phaser**: **Stages**. Lower = less CPU.
- **Filter**: **Order** and **Q**. Lower = less CPU.
- **Multitap Delay**: **Type**. Different types have different CPU requirements.
- **Modifiers** also affect CPU usage.

The Fractal Audio Wiki keeps a list of CPU saving ideas. Find it at:
https://wiki.fractalaudio.com/axefx2/index.php?title=Presets and_blocks.

CPU & USB

With its dedicated processor for USB, CPU usage is not increased when USB is connected.

6 SCENES & CHANNELS

OVERVIEW

Before reading this section, please review "Intro to Scenes and Channels" on p. 9.

Every Axe-Fx III preset consists of eight **Scenes**. Scenes don't need to be added or created—they're already there, ready to be used. Think of a Scene like a "preset within a preset," representing the preset in different states. Scenes can control which blocks are on or off, which Channel each block is set to, and more.

There are many benefits to using scenes. Selecting a Scene allows you to recall with one tap what would otherwise require "tap dancing" on foot switches. Scenes load instantly and changes can be gapless and seamless. Scenes offer the easiest way to ensure perfect "spillover" of time-based effects like delay and reverb. That's just the beginning… this incredibly powerful feature may become one of your favorite aspects of the Axe-Fx III.

The Channels feature is almost self explanatory. Most blocks have some number of channels (typically four), and for each channel you can set every parameter of that block to any setting. It's like having multiple blocks in one.

WHAT SCENES INCLUDE...



Each Scene stores all of the following:

1. The **Bypass** state of every block in the current preset.
2. The current **Channel** of every block in the current preset.
3. The setting for four virtual **Scene Controller** knobs, which adjust sound parameters across scenes.

Scenes can use the **Scene MIDI block** (p. 130) to send up to 8 MIDI PC or CC messages each time a new scene is selected.

Each scene also has its own Name.

WHAT SCENES *DO NOT* INCLUDE...

In short? *Everything else*. Scenes only affect the items listed above. Scenes cannot change which blocks are placed on the grid, or change the grid routing. You cannot tweak effect parameters per Scene (except through Scene Controllers), but remember, you can switch Channels.

WHAT EACH CHANNEL INCLUDES...

Each Channel stores its own settings for **every parameter in the block**. This gives you up to four independent sounds from a single block. Note that just like on a traditional pedal or processor, time-based blocks such as Delay and Reverb share one set of memory across different channels, so you may hear a "sweep" if you change time or size. To avoid this, just use different blocks from scene-to-scene.

FAQ: WHY NOT JUST CHANGE PRESETS? Preset changes give you total flexibility. You can change anything and everything in every preset, but there are also drawbacks. Synchronizing multiple presets is tedious, it takes care to get levels and spillover just right, and while preset changes are fast, Scene changes are even faster and can be seamless.

CHANGING CHANNELS

Different areas of the Axe-Fx III provide different ways to change Channels:

- **On the GRID** – Select the desired block and turn the **Channel** knob (**B**).
  The current Channel is shown above the knob.
- **While EDITING a BLOCK** – Press the **MORE** button (Push-knob **E**) to show the **Scene/Channel** strip.
  Use the **CHANNEL–/+** buttons (Push-knobs **C** & **D**) to select a channel.
  The current channel is shown near the top center of every page in the Edit menu.
- **FC CONTROLLER** – The FC Controllers offer a range of dedicated "Channel Select" options.
- **MIDI** – MIDI and other Global Controllers such as a connected external footswitch can be used to select Channels. See "Selecting Scenes & Channels Remotely" on p. 45.

SETTING UP CHANNELS

Whenever you edit a block, you are already programming at least one Channel (usually Channel A). Programming additional channels is easy: simply select the desired channel as described above and use your usual methods to set the various parameters to their new values. Repeat for each Channel in the block and then save the preset.

TO COPY A CHANNEL…

The **Tools** page of the **Layout** menu ("grid") provides a utility to copy one Channel to another. This works within a single block only; you can't copy a Channel from one block to another, or from one preset to another.

1. Navigate to the **Tools** page of the **Layout** menu (grid).
2. Turn the **VALUE** wheel until "COPY CHANNEL" is shown on the display.
3. Use the **D** knob to select the scene you want to copy *from*. Channel letters are shown above the knobs.
4. Use the **E** knob to select the location you want to copy *to*.
5. Press **ENTER** or the on-screen **EXECUTE** button (Push-knob **C**) to complete the copy.
6. Test your preset and remember to **STORE** to make the changes permanent.

SELECTING SCENES

Different areas of the Axe-Fx III provide different ways to select a Scene.

- **On the Home page** – Use the **NAV UP/DOWN** buttons or turn Push-knob **A**. The current scene is highlighted.
- **On the GRID** – Turn the **Scene** knob (Push-knob **A**). The current scene is shown above the knob.
- **While EDITING a BLOCK** – Press the **MORE** button (Push-knob **E**) to show the **Scene/Channel** strip. Use the **SCENE –/+** buttons (Push-knobs **A** & **B**).
  The current scene is shown near the top center of every page in the Edit menu.
- **FC CONTROLLER** – The FC Controllers offer a range of dedicated "Scene Select" options.
- **MIDI** – MIDI and other Global Controllers such as a connected external footswitch can be used to select scenes. See "Selecting Scenes & Channels Remotely" on p. 45.

THE DEFAULT SCENE

When a new preset is loaded, it automatically starts on whichever Scene that was selected when that preset was last saved. So, to set the **default scene** for any preset, simply select the desired Scene and then save the preset.

SETTING UP SCENES

Whenever you create a preset, you are configuring at least one Scene. Programming additional scenes is easy. Select the desired Scene and use your usual methods to turn effects ON or OFF and select channels. Repeat this for each Scene and then save your changes. Here are some step-by-step instructions for setting up scenes:

TO SET UP A NEW SCENE OR EDIT AN EXISTING ONE…

1. Load the desired preset and select the desired Scene.
2. Bypass or engage each block as desired.
   - On the Layout (grid) or while editing any block, use the **BYPASS** button (Push-knob **B**).
   - Or use any assigned footswitch or remote controller to bypass/engage the block.
3. Set the Channel for each block as described on the previous page.
4. Test and save your preset. On the **STORE** page, **NAV** to your new scene and give it a name (see p. 40).

Always test ALL scenes in your preset—even those you think you might not use.
Ensure that there are no unpleasant volume jumps or other surprises in case of accidental mis-steps later.

TO COPY ONE SCENE TO ANOTHER…

The **Tools** page of the **Layout** menu contains a tool to copy one scene to another. (not including the scene name). This works within a single preset; you can't copy a scene from one preset to another.

5. Navigate to the **Tools** page of the **Layout** menu (grid).
6. Turn the **VALUE** wheel until "COPY SCENE" is shown on the display.
7. Use the **D** knob to select the scene you want to copy *from*. Scene numbers are shown above the knob.
8. Use the **E** knob to select the location you want to copy *to*. You can also choose to copy one scene to "ALL".
9. Press **ENTER** or the on-screen **EXECUTE** button (Push-knob **C**) and the copy operation is completed.
10. Test your preset and remember to **STORE** to make the changes permanent.

SELECTING SCENES & CHANNELS REMOTELY

Scenes and Channels can be selected using MIDI or Global Controllers. Switches but not pedals are recommended in this case (though switches *can* be plugged in at "Pedal" jacks). Several options are provided.

SCENE INCREMENT AND DECREMENT

The **Scene Increment** and **Decrement** functions allow you to step up or down through scenes one at a time. You must first assign each function its own CC# or pedal/switch option in the **MIDI/Remote** menu. The function is triggered by any value of the controller.

SCENE SELECT

This option uses the value of a controller to select a specific Scene. You must first give the **Scene Select** function its own controller in the **MIDI/Remote** menu under **SETUP**.

The scene is set by the *value* of the controller (not the controller number itself... see FAQ below). Values begin at 0, while Scenes are numbered from 1, so "**Value plus one = Scene number**" (see table 1, right). The series continues, repeating scenes 1–8 across CC values up to 127[1].

EXAMPLE: Let's say you want your DAW to select scene 3. Open the **MIDI/Remote** menu and assign your desired CC# to "Scene Select". Let's say CC#34. Set your DAW to send CC#34 with a value of "2", which loads scene 3.

TABLE 1

CC Values, Scenes.

0 = Scene 1
1 = Scene 2
2 = Scene 3
3 = Scene 4
4 = Scene 5
5 = Scene 6
6 = Scene 7
7 = Scene 8

CHANNEL SELECT

This uses the value of a controller to select a specific Channel. Each block has its own dedicated setting for channel select, found across the listings on the **Channel** page of the **MIDI/Remote** menu under **SETUP**.

The Channel is set by the *value* of the controller (not the controller number itself... see FAQ below). Values begin at 0, which is Channel A, and it continues from there. (See table 2, right). As with Scene Select (above) the series continues, repeating Channels A–D across values to 127.

TABLE 2

CC Values/Channels

0 = Channel A
1 = Channel B
2 = Channel C
3 = Channel D

1 In mathematical terms, that's **Scene# = [(CC Value mod 8) +1]**

FAQ: MIDI CC Number and CC Value... What's the difference?

MIDI **Control Change** messages —aka "**CCs**"— have a **number** (0–127) and a **value** (0-127). The number is like an "ID" specifying *which* function you want to control. A simple example might be a MIDI expression pedal that sends CC#7 which is interpreted as Volume by a device. The Axe-Fx III lets you designate CCs for items in a long list found in the **MIDI/Remote** menu under **SETUP**.

Once a CC# is set to control a function, its **value** tells that function what to do. Some functions—like Volume—interpret data across a continuous range from 0-127. Other functions—like Bypass—simply toggle OFF for a low value and toggle ON for a high one. Other functions might be triggered by *any* value.

Different types of physical controllers transmit values in different ways. A **pedal** that rocks continuously from heel-to-toe sends a "stream" of values from 0–127. A **switch** sends a single value for OFF (typically 0) and another for ON (typically 127). Other MIDI controllers offer other options.

As described above, both the CC number and its value are the key to selecting Scenes and Channels via MIDI.

PROGRAM CHANGE MAPPING

Another way to select Scenes via MIDI is using **Custom PC Mapping**, which allows a single incoming MIDI **Program Change** (PC) message to select your choice of both **Preset *and* Scene**. This is a popular option in cases where a connected MIDI foot controller lacks the sophistication to transmit the MIDI messages required for scene selection.

The **PC Mapping** parameter on the **General** page of the **MIDI/Remote** menu under **SETUP** must be enabled for custom mapping to be in effect. With this option set to "ON", an internal table re-maps each incoming MIDI Program Change message so it can load your choice of preset and scene. The map is limited to 128 entries so MIDI BANK SELECT messages are ignored while PC Mapping is enabled.

With PC Mapping option set to "OFF", MIDI Program Changes load presets on a 1:1 basis and Bank Select commands are processed as usual as per "MIDI Reference Tables" on p. 183.

BUILDING THE MAP

The mapping table is on the **Mapping** page of the **MIDI/Remote** menu under **SETUP**.

To use it, follow this simple process:

1. **NAV** to the row for whichever Program Change message you want to re-map.
2. Use the **B** knob to set the desired value for **Map to Preset**.
   This is the preset that will load when the selected Program Change message is received.
3. Use the **C** knob to set the desired value for **Map to Scene**. You can choose a scene by its number, or select "AS SAVED" to load the default scene saved in your preset. (see p. 44 for more on Default Scene).
4. Repeat steps 2 through 4 for any remaining Program Change messages you wish to remap.
5. **EXIT** when finished. You do not need to store settings in the **SETUP** menu.

The custom map parameters remain intact even if you switch **PC Mapping** Off.

SENDING MIDI WITH SCENES

See "The Scene MIDI Block" on p. 130 for details on how each scene can transmit up to eight MIDI messages.

SCENE LEVELS

Each of the Output blocks includes eight parameters that allow you to cut or boost the level for any scene. Using these adjustments is a quick way to balance or boost scene levels when other level options are not available.

SCENES, CHANNELS & MODIFIERS

The Modifier system provides extensive automation and remote control of parameters on the Axe-Fx III.

Modifiers are detailed in Section 9 on p. 143 but here is a summary of important information about Scenes, Channels, and modifiers.

SCENE CONTROLLERS

Block parameter values are the same in every Scene of a preset. Of course, you can change the Channel, which is great but not the same thing. The **Scene Controller** system, however, provides a way for individual parameters or groups of parameters to have different values in every scene.

See "Tutorial: Scene Controllers" on p. 151.

CHANNELS & MODIFIERS

Modifiers are **shared** across Channels, meaning that any modifier you create in one channel will be present on all channels of that block. This is a great convenience in most cases, avoiding the need to apply the same modifier multiple times as you change channels.

In some cases, you may notice that if you change an effect **Type** from one channel to the next, certain parameters are either not present at all, or appear with different names. For example, in the Delay block, the **Mono Tape** delay has a parameter called "Head 2 Ratio", while this same parameter is called "L/R Time Ratio" in the **Dual Delay**. These differences are such that results should not be unpredictable.

SCENE REVERT

Scene changes you make from the Axe-Fx III front panel (or Axe-Edit III) remain in effect until you save them or discard them by loading a different preset without saving. Changes made via footswitches or MIDI, however, may work differently, depending on a Global setting called **Scene Revert**. With Scene Revert turned on, Scene changes are discarded as soon as you load a new Scene. Here are two examples so you can compare how Scene changes work in both modes:

Ex 1: SCENE REVERT OFF (default)

1. You load SCENE 1. DRIVE 1 is OFF.
2. You turn DRIVE 1 ON with a Footswitch.
3. You change to SCENE 2.
4. You change BACK to SCENE 1.
5. Drive will be ON as you last **left** it.

Ex 2: SCENE REVERT ON

1. You load SCENE 1. DRIVE 1 is OFF.
2. You turn DRIVE 1 ON with a Footswitch.
3. You change to SCENE 2.
4. You change BACK to SCENE 1.
5. Drive will be OFF as you last **saved** it.

TO ENABLE/DISABLE SCENE REVERT…

1. Page to the **General** page of the **MIDI/Remote** menu under **SETUP**.
2. **NAV** to SCENE REVERT and set it "ON" or "OFF" as desired.
3. **EXIT** to finish. (You do not need to STORE changes in SETUP.)

7 LEVELING PRESETS

This section is about balancing levels across Presets, Scenes and Channels, as compared to the topic of correctly setting *overall* input and output levels. Balancing levels requires both knowledge and tools. The challenge of getting levels "right" is not unique to the Axe-Fx III. Guitar players and audio technicians face the same obstacles when using traditional equipment: our hearing, speakers, and surroundings are all variable. On a platform as powerful and flexible as the Axe-Fx III, the number of options that affect volume can be daunting, but fortunately there are some good rules of thumb, all backed by a number of precision controls and meters. This section aims to provide a method you can rely on to normalize the levels of Presets, Scenes and Channels.

GENERAL ADVICE

- Our ears deceive us. A phenomenon known as the "equal-loudness contour" effect (aka "Fletcher Munsen" effect) results in different perceptions of tonal balance at different volume levels. At lower volumes, low and high frequencies seem to be relatively quieter. Learn more about this phenomenon and check your sounds at various levels, especially "gig" volume!
- Because of varying frequency response, different speaker systems also change our perceptions of relative volume. Set levels on the system you will perform through, or ideally, the best and most accurate speakers you can find. Be prepared to make adjustments on other systems. (This goes for your tone as well as your level.)
- Our surroundings, or the context of playback, also change perceptions of loudness. Two guitar sounds may appear to be relatively equal in level when you audition them alone, but sound totally different when you are "competing" in a mix against other instruments such as bass and drums, or when played in a problematic performance space. Adjust in context.

THE RULE OF THUMB

- There are a lot of different ways to adjust level on the Axe-Fx III. A good rule of thumb is to use the **Level** parameter of the **Amp** block. See "The Method" on the next page for an explanation of how to work through this process across Presets, Scenes and Channels.
- Above all, use common sense. Rely on meters, but don't set levels solely on how they look. **USE YOUR EARS!**

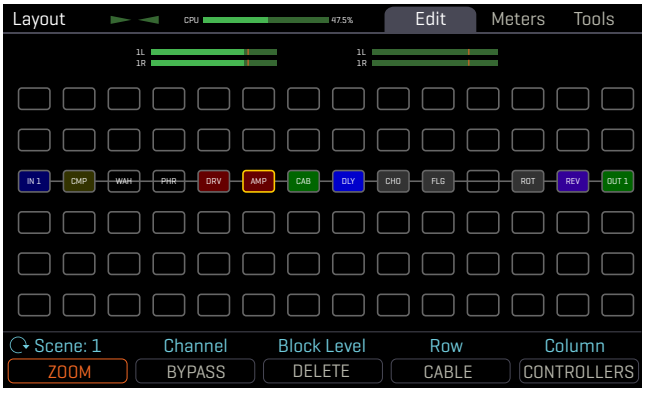
EXCEPTIONS TO THE RULE

- When following the Amp block(s) with any level-dependent dynamic effects, such as a **compressor** or **gate**, you will need to re-adjust those blocks after changing amp levels. A better option may be to "bend" the rule of thumb and adjust the output of the last dynamics block instead.
- This is also the case for any nonlinear effects such as **drive** which may in rare cases (such as Tape Saturation or Bitcrusher effects) be placed *after* the amp. Adjust **Level** for these blocks *instead* of the amp.
- If you have prominent **Drive**/**Saturation** in the **Preamp** simulation of the **Cab** block, adjust the **Level** of the Cab block instead.

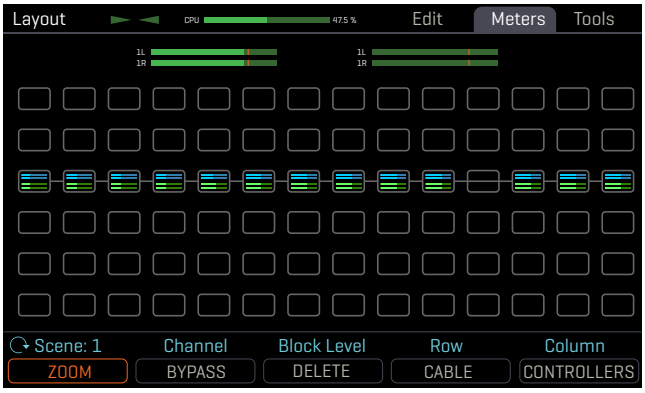
A METHOD FOR LEVELING

The new ZOOM view of the grid provides an excellent environment for leveling presets, scenes, and channels.

As you page between the **Edit** and **Meters** pages of the **Layout** (grid), you can see block names as well as block input and output level meters (blue and green, respectively). More importantly, while the grid is zoomed out, you can see larger **VU meters** for **Outputs 1 and 2** at the top of the display. The scale of these meters is -20 to +10 dB with a red line at 0. (This is the internal level, NOT a dBu reading.) These meters are great for setting levels.



The zoomed out grid shows VU meters atop the screen.



Page right to the Meters page for block meters.

To set levels, play your guitar and watch the VU meters. Remembering the rule of thumb — and assuming no exceptions — adjust the output level of the Amp block by selecting the block and turning the Block Level knob (**C**) until the level hovers around at the red line (see tip below about using the Looper). Different types of playing and different amp models and settings will activate the meter in different ways. Play consistently. Play chords and "chugs" — bass content may push the meters harder. If you want to compare the levels of two presets/scenes/channels, play the same type of material while watching the meters. Remember that whenever you adjust Drive, Master or other amp tone settings, you may need to revisit this process.

If you have two Amp blocks you may find it easier to adjust levels in a single block which sums their outputs, such as a Mixer, Volume block, or Cab.

The **Layout** view allows you to change Scenes (knob **A**) and Channels (knob **B**), so you can compare and adjust everything all within a single page.

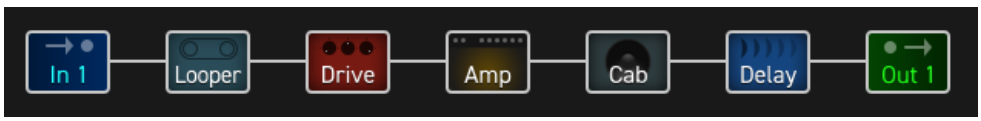
Remember this common sense principle: you must also use your ears. When you use the meters, a punchy clean amp may end up sounding louder overall than an overdriven amp with a ton of low end.

LEADS AND LOUDER SOUNDS

You now have a sense of how to normalize the level across Presets, Scenes, and Channels, but what if you specifically want some tones to be louder or quieter than others? A good approach is to begin with the loudest sound and make the other sounds *quieter*. This helps ensure that you will have plenty of headroom and avoid clipping. It's OK for certain tones to be very quiet: our products have an extremely low noise floor and do not suffer internally from the kinds of problems which plague analog gear at lower than "optimal" levels.

QUICK TIP

Make it easier to adjust levels by using the Looper block to "play" while you operate the level controls. Place the looper between the input and your first block.



BLOCK BYPASS LEVELS

Not just amps and cabs, but block effect settings also contribute to preset levels. Here are some tips to help dial in effects, so that levels can remain consistent as you bypass or engage blocks.

- Sometimes, effects are intended specifically to boost or cut levels. Other times, you want the volume to stay the same as an effect is dialed in. In either case, there's an easy method you can use to make correct settings. First, with the effect engaged, dial in the **Mix** so the blend of wet and dry sounds is as desired. Then, with one hand on the block's **Level** parameter and the other on its **Bypass** button (Push-knob **B**), switch the effect on and off, making level adjustments until the desired volume is achieved when the effect is engaged.
- **Time-based effects** like delay and reverb are usually best with their **Bypass Mode** (p. 55) set to "MUTE FX IN". This ensures that dry levels remain consistent when you engage or bypass the block. Because this setting masks possible changes to your "dry" signal, you should use the Meters page of the Layout menu to ensure that the level is roughly the same before and after these effects when they are bypassed.
- Pitch effects require special consideration. Depending on their particular type and settings, you may wish to treat them like a time-based effect with the "MUTE FX IN" setting, or you may wish to set **Bypass Mode** to "THRU" and use the first method.

8 BLOCKS GUIDE

The BLOCKS GUIDE covers all of the effect blocks, types, and parameters. It has its own table of contents:

BLOCKS GUIDE : CONTENTS

INPUT & OUTPUT BLOCKS 52
Noise Gate Parameters. 52
Output Mixer Parameters 53
COMMON MIX/LEVEL PARAMETERS 55
THE AMP BLOCK. 57
THE CAB BLOCK 66
THE CHORUS BLOCK. 69
THE COMPRESSOR BLOCK 71
Pedal/Optical and Common Parameters 71
Studio Parameters 72
Dynamics Processor Parameters 72
THE CROSSOVER BLOCK. 74
THE DELAY BLOCK 75
Mono Types and Common Parameters. 76
Stereo Delay Parameters 77
Dual Delay Parameters 77
Sweep Delay Parameters 77
Mono Tape Delay Parameters. 78
Reverse Delay Parameters 79
THE DRIVE BLOCK. 81
THE ENHANCER BLOCK. 84
Modern Enhancer Parameters 84
Classic Enhancer Parameters. 84
Stereoizer Parameters 84
THE FILTER BLOCK 85
THE FLANGER BLOCK 86
THE FORMANT BLOCK 88
THE GATE/EXPANDER BLOCK 89
THE GRAPHIC EQ BLOCK 90
THE LOOPER BLOCK 91
THE IR PLAYER BLOCK 93
THE MEGATAP BLOCK. 94
THE MIXER BLOCK 96
THE MULTIBAND COMPRESSOR BLOCK 97
THE MULTIPLEXER BLOCK 98
THE MULTITAP DELAY BLOCK 99
Quad-Tap and Common Parameters. 101
Quad Parallel Delay Parameters 103
Quad-Series Delay Parameters 103
Diffusor Parameters 104
Quad-Tape Delay Parameters 104
THE PARAMETRIC EQ BLOCK 106
THE PHASER BLOCK. 107
THE PITCH BLOCK 109
Pitch Block Common Parameters. 111
Dual Detune, Quad Detune 112
Dual Shift, Quad Shift, Quad Shift Delay 113
Harmony (Dual, Quad, Quad Delay). 114
Arpeggiator. 115
Octave Divider. 116
Whammy (Classic + Advanced) 117
Crystal Echoes 118
Custom Shifter 118
Virtual Capo 119
THE PLEX DELAY BLOCK. 120
Plex Delay and Common Parameters 120
Plex Detune Parameters 121
Plex Shift Parameters. 121
THE REALTIME ANALYZER BLOCK. 123
THE RESONATOR BLOCK. 124
THE REVERB BLOCK 125
THE RING MODULATOR BLOCK 128
THE ROTARY BLOCK 129
THE SCENE MIDI BLOCK 130
THE SEND BLOCK 131
THE RETURN BLOCK 131
THE SYNTH BLOCK. 133
THE TEN-TAP DELAY. 134
Ten-Tap Delay and Common Parameters 134
Rhythm Tap Delay Parameters 135
THE TONE MATCH BLOCK. 136
THE TREMOLO/PANNER BLOCK. 137
THE VOCODER BLOCK. 138
THE VOLUME/PAN BLOCK. 139
THE WAHWAH BLOCK. 140
GLOBAL BLOCKS. 141
Global Block Operations 142

INPUT & OUTPUT BLOCKS

5 Blocks, 4 Channels **IN**

4 Blocks, 4 Channels **OUT**

INPUT BLOCKS 1–4

Input blocks place signals on the grid. To process any input signal, a preset *must* have at least one Input block

Like all input meters, Input blocks are BLUE.

Each of the four numbered Input blocks inserts signal on the grid from the corresponding input:

The **Input 1** block uses the signal from Input 1.[1]
The **Input 2** block uses the signal from Input 2.
The **Input 3** block uses the signal from Input 3.
The **Input 4** block uses the signal from Input 4.

1 This may sound obvious at first, but unlike other inputs, Input 1 is not hard-wired to a physical jack, and may in fact be set to use your choice of three different signals: 1) **ANALOG** (the Input 1/Instrument jack), 2) **USB Output Channels 5/6** on a connected computer, or 3) The Digital **SPDIF or AES** input signal. To select the source for Input 1, use **Input 1 Select** on the **Audio** page of the **I/O** menu under **SETUP**.

INPUT USB

The **Input USB** block is fed signal from the **USB Outputs 7+8** ports of a connected computer. (See "USB" on p. 18.) There are many creative applications for the Input USB block such as creating "hardware plugin" chains via USB, or processing backing tracks and live guitar through "mastering" effects.

Noise Gate Parameters

Every Input block includes a noise gate. All signal that appears at the block is processed by the gate, whether analog, digital, or USB. The following gate parameters are offered:

Threshold – This determines how quiet the signal must be for the gate to close. The threshold can also be adjusted (or disabled) globally using the global **Noisegate Offset** parameter (see p. 157).

Ratio – This determines how much quieter the signal will be when the gate is closed. Think of Ratio as the lower number in a fraction expressing the output in dB. A ratio of "4" means you multiply the input in dB by 1/4 to find the output in dB. At lower ratio settings, noise is slightly reduced. At higher ratio settings, it is effectively silenced. A ratio of "1.00" would result in the gate having no effect.

Attack – determines how long it takes for the gate to open after signal levels exceed the threshold. You'll typically want this set low so the attack of your first notes can punch through naturally.

Release – determines how long it takes for the gate to close. Use a slow setting for a gradual decay to prevent your notes from being clipped suddenly or a fast setting for "djenty" clamping.

Output Level – Controls level at the output of the noise gate. This can be used to boost the overall preset input level which is useful, for example, when you really want to "smack" a Compressor or a Drive.

Gate Type – The gate has a "CLASSIC" type, which simply opens and closes like a classic noise gate pedal, and an "INTELLIGENT" type which uses a filter instead of only adjusting levels. The Intelligent noise also provides smart EMI filtering which reduces hum and buzz. For best results, set **AC Line Frequency** in the Global menu under **Settings** to match the actual line frequency of the AC outlet that the Axe-Fx III is plugged in to (i.e., "60 Hz" for USA/Canada/most of South America, or "50 Hz" for Europe/Australia/most of Asia, etc.)

BYPASSING INPUT BLOCKS

Input blocks have a **Bypass** switch. When bypassed, they behave exactly like a Shunt. That is to say, their **Bypass Mode** is permanently set to "THRU". While an Input block is NOT bypassed, it *discards* signal at its grid input.

OUTPUT BLOCKS 1–4

Output blocks carry signal from the grid to various outputs. To output any input signal, a preset *must* have at least one Output block.

Like all output meters, Output blocks are GREEN.

The **Output 1** block transmits signal to the physical Output 1 jacks[1] and to USB Inputs 1+2 on a connected computer.

The **Output 2** block transmits signal to the physical Output 2 jacks[2] and to USB Inputs 3+4 on a connected computer.

The **Output 3** block transmits signal to the physical Output 3 jacks[3].

The **Output 4** block transmits signal to the physical Output 4 jacks[3].

1 The Output 1 block signal may also be transmitted through the Digital outputs if **SPDIF/AES Out Source** is set to Output 1, and through Output 2 if **Output 2 Copy Output 1** is turned ON and the Output 2 block is not present on the grid.

2 The Output 2 block signal may also be transmitted through the Digital outputs if **SPDIF/AES Out Source** is set to Output 2.

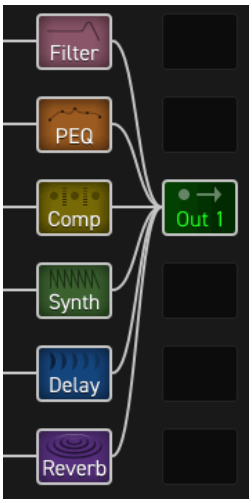
3 The Output 3 or 4 jacks can be set globally to echo signal from **Input 1**, though placing Output 3 or Output 4 blocks on the grid will override these settings (p. 161)

Output Mixer Parameters

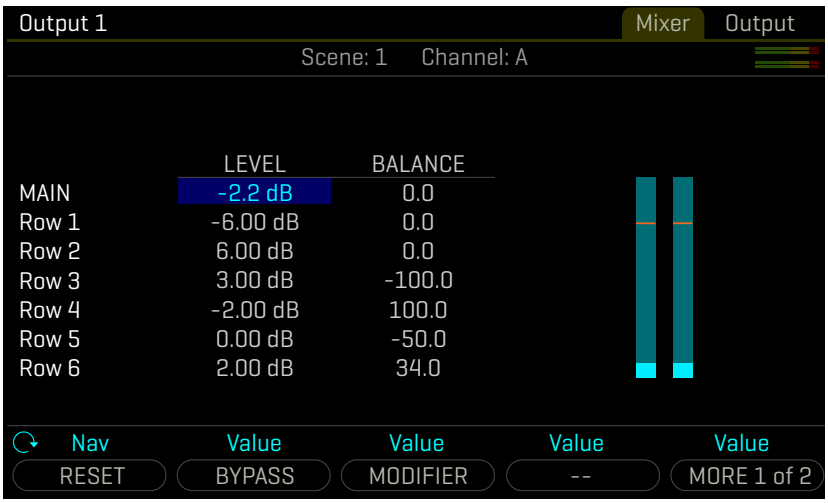
Every Output block includes a six-channel mixer. The six channels correspond to the six rows of the grid. A **Main** level controls the master mix. Each channel and Main also have **Balance** controls.

The following example demonstrates how this works. Six rows are fed into an Output 1 block, whose parameters determine how each of these six signals are outputted.

Grid



Output 1 Block Mixer Parameters



Above, six blocks are shown in six rows, with an Output block in the next column. It does not matter which *row* the Output block itself is in; the six channels of its mixer always correspond to the six rows of the column to its left. Those rows might contain blocks, shunts, or nothing.

- The Main output of the block is set to -2.2 dB. It's Balance is dead center (0.0).
- Row 1 (the Filter) will reach Output 1 at Level -6.00 dB, with Balance centered (0.0).
- Row 2 (the PEQ) will reach Output 1 at Level +6.00 dB with Balance centered (0.0).
- Row 3 (the Compressor) will reach Output 1 at Level +3.00 dB with Balance fully left (-100.0).
- Row 4 (the Synth) will reach Output 1 at Level –2.00 dB with Balance fully right (+100.0).
- Row 5 (the Delay) will reach Output 1 at Level ±0.00 dB with Balance 50.0% left (-50.0%).
- Row 6 (the Reverb) will reach Output 1 at Level +2.00 dB with Balance 34% right (+34.0).

BYPASSING OUTPUT BLOCKS

Output blocks have a **Bypass** switch. When an Output block is bypassed, it mutes the signal at the corresponding output AND on the grid. That is to say, its **Bypass Mode** is permanently set to “MUTE”.

INPUT & OUTPUT BLOCK CHANNELS

Input blocks and Output blocks each have four channels. This gives them multiple sets of parameters you can use to vary their operation from scene to scene.

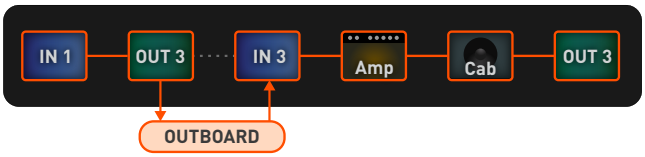
OUTPUT BLOCK SCENE LEVELS

Every output block is equipped with eight **Scene Level** parameters. Use these to cut or boost output for any scene.

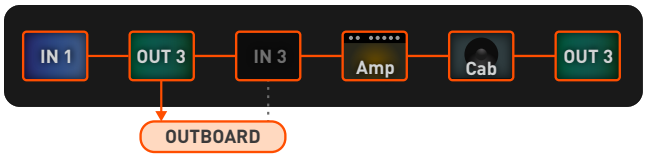
vINPUT & OUTPUT BLOCKS AS SEND & RETURN LOOP

The “FX Loop” block found on previous products no longer exists. Instead, individual Input and Output blocks must are placed separately on the grid. In the example below, signal flows (in red) from the Output block to outboard gear and back through the Input block. The Input block serves as a “master bypass” control for the send/return loop. The Output block on the grid is connected with a cable to the Input block as shown below, but the Input block ignores grid input signals when it is engaged (left image) and ignores external input signals when it is bypassed (right image).

Loop Active



Loop Inactive (Input Block Bypassed)



INPUT 1/INSTRUMENT IMPEDANCE

The **Input Instrument Impedance** parameter appears only on the **Input 1** block. This parameter changes the actual analog circuitry of the INSTR input jack to interact differently with your guitar pickups. This recreates the way that some classic effects (e.g. ‘Vibe) “load down” the pickups, causing a change in frequency response. In “AUTO” mode, the impedance is set automatically, based on the first active effect to follow the input. Normally the setting of “AUTO” is best, but you may also select any of the following values. This setting is saved with the channel.

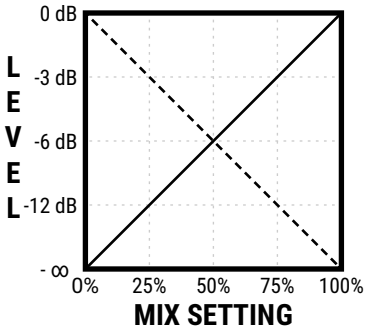
- 1MΩ
- 1MΩ+ Capacitor
- 230kΩ
- 230 kΩ + Capacitor
- 90 kΩ
- 90 kΩ + Capacitor
- 70 kΩ
- 70 kΩ + Capacitor
- 32 kΩ
- 32 kΩ + Capacitor
- 22 kΩ
- 22 kΩ + Capacitor

COMMON MIX/LEVEL PARAMETERS

Almost every block has a **Mix** page with parameters that determine how that block contributes to the preset. All **Mix** pages use one or more of the following standard parameters. Different blocks have different **Mix** page parameters because they require different approaches. You may need to change more than one parameter setting to achieve a desired result.

Mix – Determines the balance of wet and dry signals. In most cases, setting the mix by ear is best.

With the exception of a few blocks that use a constant power algorithm, Mix controls the dB levels of wet and dry signals in an inverse linear relationship. A mix setting of 50% results in both wet and dry being at -6 db in comparison to their maximum output levels. A diagram (right) illustrates this "mix law".
Note: See “The Delay Block” for details on how its Mix control works differently.



Level – As you would expect, this controls the overall output level of the block.

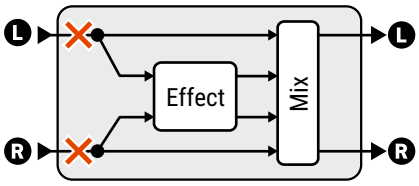
Balance – Every block outputs both left and right signals. As you adjust to the left or right, the opposite channel gets quieter. Both the wet and the dry are affected.

Bypass – Blocks have a dedicated Bypass parameter for the purpose of allowing Modifier control.

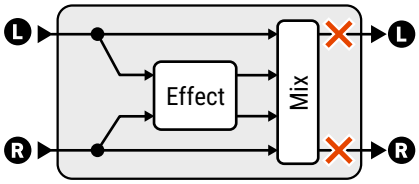
Bypass Mode – This setting determines what happens when the block is bypassed. The different options are detailed below. Not every option is available in every block.

MUTE – When the block is bypassed, both wet and dry are totally silenced.

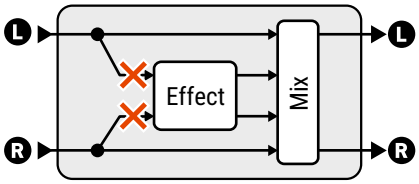
THRU – When bypassed, the block is completely disengaged. None of its parameters have any effect on the sound; it behaves exactly as if it were a shunt.



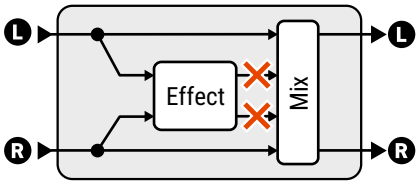
MUTE IN –When the block is bypassed, its inputs are muted. This silences both wet and dry but allows effect tails to ring out or spillover. Use this for time-based effects in parallel.



MUTE OUT – When the block is bypassed, all audio is totally silenced at the outputs. Effect tails are silenced immediately, but the input is still “listening” so tails may be heard when you engage the effect.



MUTE FX IN – When the block is bypassed, the inputs to the effect are muted, and the dry level is unaffected. This allows effect “tails” to ring. LEVEL and BALANCE (and even MIX) affect the dry even when the block is bypassed. Use this for time-based effects in series when you want spillover.



MUTE FX OUT – When the block is bypassed, the outputs of the effect are muted, and the dry level is unaffected. The effect is always “listening” so tails are heard already when you engage, but are silenced when you bypass. LEVEL and BALANCE affect the dry even when the block is bypassed.

Input Gain – Determines the amount of signal fed to the effect processor *within* the block. This simulates the way an AUX Send would normally feed an effect routed in parallel. It has no effect on the dry signal. It appears on the following blocks: Delay, Megatap Delay, Multitap Delay, Ten-Tap Delay, Pitch, Plex Delay, Resonator, Reverb.

Global Mix – This switch determines whether or not the **Mix** setting of the selected effect will be subject to an offset (±50%) applied using the global **Effects Mix** parameter found in the **SETUP: Global** menu.

This feature is provided so you can design presets with the built-in ability for fast mix compensation in playing environments that require more or less wet mix. It is offered on the following effect blocks: Chorus, Delay, Flanger, Megatap Delay, Multitap Delay, Phaser, Pitch, Plex Delay, Resonator, Reverb, Synth, Ten-Tap Delay, Vocoder.

THE AMP BLOCK

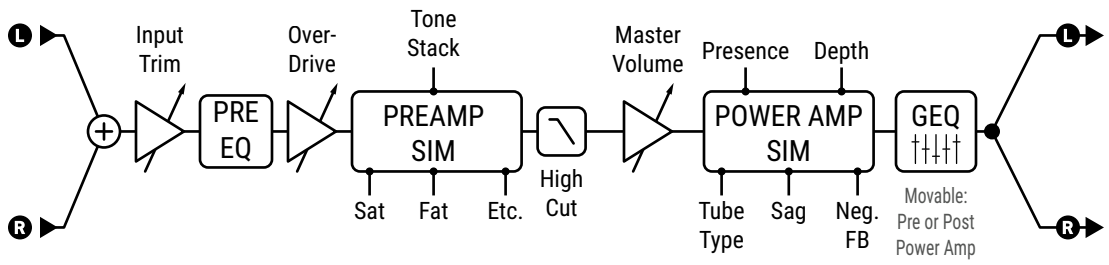
2 Blocks, 260+ Types, 4 Channels **AMP**

The Amp block reproduces the sounds of an impressive array of vintage and modern guitar and bass amplifiers, with 260+ different "types" based on stock, custom, and hybrid models. It uses our newest "Ares" Amp Modeling technology with component-level physical modeling which produces tone of a depth and quality that cannot be obtained using lesser modeling methods. Our amp models are the result of thousands of hours of incredibly detailed analysis of the actual amps that inspired them. Like "Quantum" Amp Modeling before it, Ares models the preamp tubes, tonestack, cathode follower, power supply, power amp, phase inverter, amp/speaker interaction, and more.

The Amp Block works hand-in-hand with the equally important Cab block. To learn about how amp and cab work together, try different amps through the same cab, or different cabs with the same amp. In other rig designs, the amp block is used with real guitar speakers and a power amp. In any configuration, the results are amazing: tight bass, powerful midrange, silky highs, and highly expressive touch sensitivity.

You can get great tone from these models using just the basic controls on the Type and Tone pages. Should you desire to dig deeper, you'll find many exciting parameters that allow you to tweak and adjust the most intricate aspects of amp tone. Don't become paralyzed by the options. The default settings are very accurate, and more than sufficient to achieve great results. An easy way to start branching out might be with Post EQ and Pre EQ. For dynamics, try Speaker Compression and Output Compression. To learn more, tap into the Fractal Audio community, and watch for "Tech Notes" on the Fractal Audio forum for tips. Above all, use your ears: rules are made to be broken; you have the tools here to create the future of tone.

THE AMP BLOCK



TYPE PAGE

Amp Type – Amp types are presented in an alphabetical list. Make a selection using the **VALUE** knob or **NAV** buttons. You do not need to press **ENTER**. There are over 260 different types mostly based on real-world amps. If you're unfamiliar with the models, the Fractal Audio wiki (a publicly managed document) contains many pages of useful information, especially "Yek's Guide to the Fractal Audio Amp Models".

As you change the amp type, you will notice that various other parameters change to different "starting values" for each type. This is perfectly normal and it helps ensure that a model sounds correct when selected. For example, if an original amp has no Master Volume control, meaning its power amp is "wide open", the model will set Master to "10.0" to ensure accuracy. (Of course, you can still adjust any parameter as desired from the recommended starting point.)

When you press the RESET button (Push-knob A) to initialize the current Channel, the amp type is not changed but all other parameters return to default values. You can also "soft reset" an amp by changing type to another model and then back. This leaves some custom settings intact, like tone knob positions.

TONE PAGE

Input Drive – (aka "Drive") sets the amount of preamp gain/distortion. Used in conjunction with the Master Volume, Input Drive determines whether the sound will be clean, slightly broken up, moderately overdriven, or fully distorted.

A treble peaker circuit found on the drive or volume control on many amps is also modeled. This can be heard as the low frequencies are attenuated more than the highs when the drive is turned down (and vice versa). For amps that have no Master Volume, Input Drive functions as the amp's volume control.

Bass, Mid, Treble – Adjust these as you would the controls on any amp.

While other modelers use simple filters to approximate amp tone controls, our products recreate the exact frequency and phase response characteristics of a classic passive tonestack. In most cases, knob positions can even be matched to settings on the original amps. (Though recognize that many types of amps were built inconsistently with different types of potentiometers from one run to another.)

Some models may provide tone controls NOT offered on the original amp. For example, many amps have no Mid control. To faithfully simulate such amps, set controls they are missing to "noon" (or "0" if you are using the "ACTIVE" Tonestack Type). Of course, you may still adjust these "bonus" controls to achieve tones the original amp does not have.

Please note that extreme tone and high gain settings can cause pickup squealing or excessive noise. This is especially true with **Tonestack Type** set to "ACTIVE."

Bright Switch – Many amplifiers contain a "treble peaker," included as a pull or toggle switch, or even hard-wired. Each amp type includes this control (even if the original mode does not). The effect may be subtle or quite pronounced depending on the amp type. This is also affected by the **Bright Cap** setting. If the original amp had no bright circuit, Bright is OFF by default but can still be turned on to apply circuit values suited to an amp of that general type. If the amp has a hard-wired treble peaker, the default state is ON.

Input Trim – Amps without an **Overdrive** control (below) will display **Input Trim** instead. This allows you to adjust for more or less preamp gain than the actual circuit being modeled. This is different from the **Input Drive** in that it does NOT interact with the surrounding circuitry to change frequency response as it is varied. In short, use Input Trim to adjust gain *without* also changing tone.

Overdrive – Amps with **Overdrive** will not display **Input Trim** (above). Adjusts overdrive. Note that Input Drive and Overdrive are applied to the appropriate points in the circuit for the amp being modeled, i.e. prior to the last triode stage or prior to the third triode.

Presence/Hi-Cut – Boosts (or cuts) the upper frequencies from the virtual power amp by varying the negative feedback frequency response. Increased presence can help a sound cut through. Note that settings for Presence don't necessarily correspond to knob positions on the amp being modeled.

Amps with no negative feedback circuits in their design cannot utilize a realistic presence control. Therefore, if **Negative Feedback** is set (manually or automatically) to "0", Presence becomes a high-shelf EQ at the output of the power amp, and its label changes to **Hi-Cut**. This allows you to control power amp highs even for amp models that wouldn't normally have this capability. When changing to a model with no negative feedback (i.e. Class-A, etc.), be sure to check your Hi-Cut settings as values higher than zero will darken the sound.

Depth – Boosts low frequencies from the virtual power amp by varying the negative feedback frequency response. It is set by default to an appropriate value when the amp type is selected, but this setting may be overridden.

Note: Presence and depth are disabled when Power Amp Modeling is turned off. For more on disabling Power Amp Modeling, see Supply Sag (p. 60) and "Global Settings: Config Page" on p. 156.

Master Volume – The almighty Master Volume is a very important control. It determines the distortion and dynamics characteristics of the power amp simulator, and its setting at any moment can dramatically change the amp's sound. As it is turned up, the tone controls will have less influence, and the sound will have more "bloom" and touch sensitivity. Settings for Master don't necessarily correspond to knob positions on the amp being modeled. With a little experimentation, you will learn to dial in different great sounding Input Drive and Master combinations.

When you select an amp type, the Master will change to an appropriate/typical setting for that amp. If a real amp doesn't have a Master, the "correct" setting will be applied—i.e. "10", or "wide open."

- At high settings, less Input Drive is usually required, especially for high-gain types.
- Amps designed for preamp distortion will typically sound great with the Master set low to prevent the tone becoming muddy or noisy. This includes the "USA Lead" types and others.
- Amps with negative feedback tend to have "crunchier" power amp distortion, which can get "raspy" when driven too hard. Experiment with the interactivity of Negative Feedback and Master on distortion tone.
- With Power Amp Modeling disabled, either globally or in one specific block, Master Volume becomes a simple level control with 40 dB of range. For more on disabling Power Amp Modeling, see Supply Sag (p. 60) and "Global Settings: Config Page" on p. 156.
- If more power amp gain is desired, **Master Volume Trim** in the Advanced menu can be used for increased range.

LEVEL – This controls volume with NO EFFECT on tone, making it ideal for setting the overall level of a preset.

PREAMP PAGE

Input Boost, Boost Type, Boost Level – The selectable input boost acts as a "clean boost" to replicating the common technique of setting a drive pedal's Drive knob to 0 and using its level control to hit an amp harder. Turn the switch on, choose the type you want based on real drive pedals, and set the level to drive your amp harder without needing a separate Drive block.

Fat Switch – Emphasizes midrange and adds "body" by shifting the tonestack center frequency.

Cut Switch – Reduces the amount of low frequencies coming into the amp simulation. This can be used to achieve a "tighter" tone or to reduce low-end "flub".

Saturation Switch – This engages a popular mod between the preamp and the tonestack for a thicker, more aggressive distortion character. The "ON (AUTH)" and "ON (IDEAL)" settings differ only in volume. "IDEAL" gives you the hotter output you wish a real amp had with saturation engaged. ;-)

Saturation Drive – Controls the amount of saturation (see above). The default value differs for each model.

Preamp Tube Type – Changes the characteristics of the virtual preamp tubes, based on real world examples.

Cathode Follower Compression – Sets the amount of compression in the virtual cathode follower. This interacts with other parameters listed in the Cathode Follower section of the Advanced page (Time, Ratio, etc.)

Preamp Bias – Sets the bias point of the last triode (not counting the cathode follower). Depending on the bias points of the previous stages, increasing or decreasing this value can alter both harmonic content and attack characteristics. Typically, if the previous stage has a negative bias then increasing this value will be more noticeable (and vice-versa).

Bright Cap – Sets the value of a virtual capacitor to determine the sonic effect of the **Bright** switch. Increasing this will make the preamp brighter and vice versa.

High Treble – Think of this as an extra tone control, useful to add "zing" or tame harsh highs.

POWER AMP PAGE

Power Tube Type – Changes the characteristics of the virtual amp power tubes. The virtual power amp includes modeling of the plate impedance of the power tubes. Plate characteristics are adjustable via **Dynamic Damping**, an advanced parameter. The Tube Type parameter allows you to select from common power tube types: 6AQ5, 6L6/5881, 6V6, 300B (triode), 6550, 6973, EL34/6CA7, EL84/6BQ5, KT66, KT88 are offered, as well as an "ideal tetrode" and an "ideal pentode".

Power Tube Grid Bias – Sets the bias point of the virtual power amp. Lower values approach pure Class-B operation. Higher values approach pure Class-A operation.

Negative Feedback – This controls the amount of negative feedback, or damping, in the power amp simulation. Higher values give a tighter and brighter sound but can sound harsh at very high master volume levels. Lower values give a loose and gritty sound and feel. Like many other power amp parameters, Negative Feedback is set to an appropriate value whenever you change the amp type, but it can be changed as desired. For example, you might dial in some negative feedback on a "Top Boost" amp type to give the power amp a more "American" sound while still retaining the preamp voicing.

Transformer Matching – Transformer Matching is an extremely powerful parameter. Lower values cause power tubes to clip later and therefore the phase inverter and grid clipping becomes more predominant. At higher settings, power tubes clip sooner and the **Resonance** settings on the Speaker page become more pronounced. For optimum results bring up the **Master** until the desired amount of power amp distortion is achieved, then adjust **Matching** until the character of the distortion is as desired. The various **LF and HF Resonance** parameters interact strongly with this parameter, so be sure to experiment with those as well when crafting a tone.

Bias Excursion – The higher the value, the more the bias shifts when the virtual power tubes are overdriven. Bias excursion pushes a power amp from Class-AB operation towards Class-B operation, which can result in crossover distortion. A little goes a long way, but too much can lead to what is referred to as "blocking distortion" which can make an amp sound unpleasant.

Power Tube Hardness – This controls the "shape" of the virtual power tube's transfer function. Higher values have a more abrupt transition into saturation.

Power Tube Mismatch – Use this to simulate a gain mismatch between the virtual push and pull power tubes. A value of zero represents perfectly matched tubes.

Bias Trem Frequency, Bias Trem Depth – These create true bias tremolo by varying the bias of the virtual power tubes. Bias Tremolo is very organic and varies based on a multitude of variables including power amp settings, damping, bias parameters, and more. It is also "self-ducking" and decreases as you play harder. On some amp types, extreme bias depth can result in excessive crossover distortion. On other amps the amount of tremolo can vary greatly between loud and soft playing. All this, however, is part of the allure of bias tremolo as it results in a particularly "organic" sound.

POWER SUPPLY PAGE

Supply Sag – This controls virtual power amp dynamics. Higher settings simulate higher power supply impedance, and thus greater tube plate voltage "droop," for a more compressed feel. This control interacts with the Master and will have little effect if the power amp is not being "pushed hard". As Master is increased, the power amp draws more current from its power supply and the Supply Sag control will have more effect.



IMPORTANT: Turning Supply Sag fully counterclockwise defeats power amp simulation for a single amp block! This allows it to be used into a real tube power amp without globally disabling **Power Amp Modeling** in **SETUP** : **Global** : **Config**. With Supply Sag at 0, **Master** works as a simple volume, **Depth** is deactivated, and **Presence** turns into a simple shelving filter.

B+ Time Constant – This interacts with the **Supply Sag** control because it makes the virtual power supply response slower or faster. When the supply is fast the amp will sag rapidly accentuating the pick attack and compressing after. Most guitar players like this, but setting it too fast will cause excessive AC ripple and ghost notes. For convenience the virtual power supply voltage (B+) is shown as a meter on this page when the Supply Sag control is selected. (The meter shows dB, relative to the idle voltage.)

Supply Type, AC Line Frequency – These select between AC and DC virtual power supply types. AC rectification and resulting supply ripple are modeled, and the line frequency is also selectable. Note that as with a real tube amp, the AC Supply can cause "ghost notes" when **Supply Sag** is low and **B+ Time Constant** is high. Lower B+ Time Constant values will make the amp feel "faster," but too low can also cause ghost notes.

Variac – This sets the relative AC line voltage into the amp simulation. Volume varies greatly when a Variac is used with a real amp, but the virtual variac compensates for this.

Screen Frequency, Screen Q – These set the resonant frequency of the virtual power tubes' screen filter, and the Q of that filter.

Level – Duplicated control from the Tone page.

SPEAKER PAGE

These parameters shape the virtual speaker impedance curve and the resulting resonances in the virtual power amp. Amp and speaker interaction affects tone by causing an increase in power amp response at certain frequencies. Note that setting **Negative Feedback** greater than "0" flattens the effect of the response curve.

LF Resonance Frequency, LF Resonance Q, LF Resonance – Guitar loudspeakers have strong low-frequency resonance. This shifts up slightly when the speaker is mounted in an enclosure. This resonance causes an increase in the power amplifier response due to the finite output impedance of the power amp.

HF Resonance Frequency, HF Resonance – A loudspeaker voice-coil presents an inductive load to the power amp at high frequencies. This inductive load, in conjunction with the output transformer capacitance, creates a high-frequency resonance at the specified frequency.

HF Slope – This parameter allows fine adjustment of the high-frequency impedance of the virtual voice coil (which affects the slope of the impedance curve). A speaker voice coil is "semi-inductive" due to eddy current losses in the motor. This presents an impedance to the power amp that is neither fully inductive nor fully resistive. The amount of resistive loss varies by brand and type. Reducing Slope simulates a speaker that is less inductive, increasing Slope simulates a speaker that is more inductive. Typical speakers range from 3.0 to 4.5 with the median being about 3.7. Lower values yield greater midrange while higher values are more scooped and sizzly.

Speaker Drive – This simulates distortion and gentle compression caused by pushing a speaker too far. It interacts with the Master which determines how hard the actual power amp is pushing. Don't overlook this when striving for "vintage" tones as it helps make edge-of-breakup tones sound like an old, well-played amp.

Speaker Compression – Even if you never touch most advanced parameters in the Amp block, don't overlook Speaker Compression, aka the "chunka chunka" parameter. It models the interaction of the power amp with the power compression of a virtual speaker. Typical guitar speakers compress between 3 and 6 dB depending upon construction, age, volume, etc. The default value is conservative and yields about 3 dB of compression. **Master, Presence**, and **Depth** will interact considerably with Speaker Compression, with higher causing more compression. A gain reduction meter shows the amount of Speaker Compression when this parameter row is selected. Note that this parameter does not reset to its default value when changing the Amp Type. Tip: Speaker Compression is a simulation of what happens in a real guitar speaker, so set it to zero when using a tube power amp and conventional guitar cab.

Speaker Compliance – Thios control changes the nonlinear behavior of the virtual speaker. Selecting a new amp model or resetting the block will set the value to 50% which is a typical value for guitar speakers.

Transformer LF, Transformer HF – These set the output transformer bandwidth.

Transformer Drive – This models core saturation in the virtual output transformer. Higher values simulate a smaller, more easily saturated transformer. The distortion produced by an overdriven output transformer isn't particularly pretty but it does play a role in the tone and without it the distortion would not be authentic.

INPUT EQ PAGE

The Input EQ parameters adjust a powerful set of filters at the input of the amp block, so its effect is heard before preamp distortion or a front-end tone stack.

Type, Frequency, Q, Gain – These parameters work together to define a powerful multi-mode filter that can be used for anything from a subtle bump to an extreme spike, or from gentle to extreme shelving.

Low Cut – Adjusts the cutoff frequency of a high-pass filter. This control removes bass frequencies and can be useful to tighten an amp's low end, removing "flub".

High Cut – Adjusts the cutoff frequency of a low-pass filter. This filter is actually placed between the preamp and the power amp, so it has a more dramatic effect than an EQ that might be placed before preamp distortion.

Definition – This control is a basic "tilt EQ" which adds highs/cuts lows, or vice versa.

OUTPUT EQ PAGE

The amp block includes a built-in graphic EQ, eliminating the need to use a separate block for tone-shaping. You can achieve different results by changing **EQ Location** in the Advanced menu. To reset the EQ to flat, tap **ENTER**.

To operate the EQ bands, use the **NAV** keys, and **VALUE** or **A,B,C**, **D** and **E**.
To change the number of bands, use the **NAV UP/DOWN** buttons.

EQ Types (Advanced page) include: 3-Band Console, 3-Band Passive, 4-Band Passive, 5-Band Passive, 5-Band Constant Q, 7-Band Constant Q, and 8-Band Constant Q, 5-Band "Mark", 7-Band Variable Q, and 8-Band Variable Q.

DYNAMICS PAGE

Input Dynamics – Sets the strength of an input dynamics processor. When set below zero, the amp type compresses, resulting in a smoother, less dynamic sound. When set greater than zero, the amp expands, resulting in a punchier, crunchier and more dynamic sound. Note that extreme values can have undesirable side-effects such as pumping or clipping.

Output Compression – This controls the ratio of a compressor specifically tailored to reduce the output dynamic range of the amp block. A gain reduction meter shows the amount of compression when this parameter row is selected.

Compressor Type – Sets the type of **Output Compression**. The "OUTPUT" type simply compresses the output. The "FEEDBACK" type likewise compresses the block output, but also applies dynamics to the input of the block, so you will also get more distortion as you play harder and less when you play softer or roll back the volume.

Compressor Threshold – Sets the threshold of **Output Compression**. A lower value causes compression to occur for quieter signals.

Compressor Clarity – Used in conjunction with the other **Output Compression** parameters, this adjusts the bass response of the compressor and can be used to add clarity to the low end.

Speaker Compression – Duplicated control from the Speaker page.

Speaker Time Constant – This adjusts the thermal time constant of the virtual voice coil, affecting the attack and release of virtual **Speaker Compression**. Lower values cause the voice coil to heat and cool faster and vice-versa.

Dynamic Presence – This models output transformer leakage inductance, resulting in a brightening of the tone when the virtual power amp is pushed. When playing softly or at lower gains, the influence of this control is lessened. Note that this only affects the power amp modeling and is dependent on the degree of power amp overdrive. This control can also be set negative to cause the tone to darken when playing harder. This can help dial in the sweet spot of an amp model. As the Master is increased, an amp becomes more liquid, more compressed and easier to play. However, the highs may get overly compressed, causing the amp to sound too dark. Dynamic Presence allows you to get the desired power amp drive and feel without high frequency loss.

Dynamic Depth – Analogous to Dynamic Presence, above. This increases low frequencies when the virtual amp is being pushed. While real amps don't display this behavior, it is a cool tone-shaping tool.

Level – Duplicated control from the Tone page.

ADVANCED PAGE

Input Select – The Amp block processes audio in mono. This control determines how incoming stereo signals will be processed. You can input only "LEFT" or "RIGHT" channels, or "SUM L+R" (the default setting).

Input Trim – Duplicated control from the Tone page.

EQ Type – This determines the number of bands for the amp block's built-in graphic equalizer, and whether it will be variable Q, constant Q, passive or console type. You can change this from the **Output EQ** page using the **NAV UP/DOWN** buttons.

EQ Location – Sets the position of the equalizer controlled by the **Post EQ** page. The default value of "OUTPUT" places the EQ at the output of the virtual power amp. "PRE PA" places the EQ between the preamp and power amp. The "INPUT" setting actually moves the Post EQ to the location of the Pre EQ: in front of the preamp.

Output Mode – The default value, "FRFR", is designed for use while using "Full Range/Flat Response" monitors, or while recording. The "Solid State Power Amp + Cab" ("SS PWR AMP +CAB") mode is intended for use while using a solid-state power amp and conventional guitar cab. In this mode speaker compression modeling behaves differently, relying on the speaker for compression while still simulating the interaction with the power amp.

NOTE: The SS PA +Cab Mode is not intended for use with "current drive" power amps, i.e. tube power amps, Class-D current feedback amps, etc. This mode CAN be used, however, with FRFR monitors in high volume applications where the monitor's speakers are compressing, thereby achieving a more dynamic response.

ADVANCED PAGE: PREAMP SECTION

Definition – This control is a basic "tilt EQ" which adds highs/cuts lows, or vice versa. It is located at the amp type's input, so its effect is heard before preamp distortion or a front-end tone stack.

Low Cut Freq – This control filters the lows at the input to the amp.

Hi Cut Freq – This filters the highs at the very end of the preamp simulation. Experiment with this to fine-tune your tone. For example, some of the higher gain amp types are characterized by fairly heavy filtering after the preamp stage. Increase for a brighter tone or decrease for a darker tone.

Tonestack Type – By default, the **Bass**, **Mid** and **Treble** controls operate as a "passive" tonestack: they simulate exactly the frequency and phase response of the classic passive tonestacks found in the original amplifiers many amp types are based on. This lets you change this behavior from "PASSIVE" to "ACTIVE", or to substitute the passive tonestack of another amp.

Selecting a substitute tonestack allows you to mix and match amps and tone stacks to create your own hybrids. This allows you to use, for example, a Plexi-type tonestack on a Blackface amp model.

Selecting the "ACTIVE" type gives each tone control +/– 12 dB boost/cut operation for up to twice the range of a typical amplifier. Since the active tone controls are more sensitive, small adjustments have bigger effects. For example, full PASSIVE treble for a high-gain British amp would be equivalent to only +5.0 dB ACTIVE, leaving 7 dB of additional headroom! Active tone controls do not interact like those of a typical amplifier, so when you adjust the treble, the mid and bass are not affected. This can make dialing in a certain tone easier and quicker than it might be with a PASSIVE tonestack.

Tonestack Freq – Sets the center frequency of the tone controls to determine their effect on the sound. This control works whether you are using Active, Passive, or substitute tone stacks. This parameter defaults to an appropriate value whenever you change the amp TYPE, but it can then be changed as desired. But if you subsequently change the **Tonestack Type**, the Tonestack Frequency will not necessarily be correct anymore.

Tonestack Location – Lets you change the location of the tone stack. "PRE" places the tone stack at the input to the virtual preamp, "POST" places the stack between the preamp and power amp. "MID" places it between the last two triode stages, and "END" places it after the power amp (which is physically impossible with a real amp).

Preamp Sag – Turning this ON causes the amp block to behave like an integrated tube head or combo amp. Turning this OFF simulates a separate preamp and power amp.

Triode 1 Plate Freq, Triode 2 Plate Freq – These parameters set the cutoff frequency of the last two triodes in the chain. Many amps have a capacitor across this triode's plate resistor. This capacitor is used to smooth the response and reduce noise. You can adjust the amount of capacitance, and the resulting frequency, using these parameters.

Bias Excursion – Not to be confused with Bias Excursion on the Power Amp page, this is a separate parameter for the preamp. The higher the value, the more the bias shifts when the virtual tubes are overdriven.

ADVANCED PAGE: CATHODE FOLLOWER SECTION

Cathode Follower Compression – Determines the amount of compression in the virtual cathode follower. This parameter interacts with the other parameters listed below.

Cathode Follower Harmonics – Simulates harmonics that occur naturally inside an amp as tubes interact. Higher values increase the interaction between virtual tubes, yielding "softer" distortion.

Cathode Follower Type – Selects between "AUTHENTIC" and "IDEAL" types.

Cathode Follower Time, **Cathode Follower Ratio** – These parameters determine the attack time and ratio used for **Cathode Follower Compression**.

Cathode Follower Hardness – For models that use cathode followers, this adjust the shape, warmth, smoothness and decay of cathode follower distortion.

ADVANCED PAGE: POWER AMP SECTION

Master Vol Trim – Allows you to adjust the range of **Master Volume**. Increasing the value above 1.0 will cause more gain in the virtual power amp and vice versa.

Master Vol Cap – Sets the value of the bright cap across the **Master Volume** control.

Master Vol Location – Sets the location of the **Master Volume** control. Most amps have the Master Volume before the phase inverter ("Pre PI"). On some amps (like the "Class-A" types) the Master Volume comes after the phase inverter ("PI"). A third option, "pre-triode," is the default for "Hipower" amp types.

Presence Freq – This multiplier alters the center frequency of the **Presence** and **Dynamic Presence** controls.

Depth Freq – This multiplier alters the center frequency of the **Depth** and **Dynamic Depth** controls.

P.A. Cathode Resonance – There are two types of power tube bias: fixed bias and cathode bias. In a cathode biased amp a resistor is placed between the power tube cathode and ground thereby self-biasing the tube. This parameter sets the value of the virtual cathode resistor. Higher values result in a more negative bias and push operation towards Class-B, resulting in more crossover distortion.

P.A. Cathode Time Const – This sets the time constant of the virtual RC cathode network for amp types that are cathode biased.

Dynamic Damping – Controls the reduction in plate impedance as a function of grid voltage. Higher values result in a more focused and forward midrange.

PI Bias Excursion – Controls how much the virtual phase inverter bias shifts when overdriven.

Pwr Tube Bias Excursion – Controls how much the grid bias shifts when the virtual power tube grids are overdriven.

Excursion Time, **Recovery Time** – These parameters are related to **Pwr Tube Bias Excursion**. They control how rapidly the coupling capacitor charges or bleeds off as the virtual power tube grids are conducting or not.

ADVANCED PAGE: OUTPUT SECTION

The Amp block has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See "Common Mix/Level Parameters" on p. 55.

THE CAB BLOCK

2 Blocks, 4 Channels **CAB**

The Speaker Cabinet simulator, or Cab block, recreates the tonal characteristics of any number of speaker cabs. The block also recreates the sound of various mic preamps, room reverb, “air”, "proximity", and simulates millimeter-resolution “mic distance” for accurate inter-microphone phase effects.

Speaker simulation is performed using our exclusive UltraRes™ impulse responses (IRs) with 2,048 factory cabs to choose from, plus a “Legacy” bank with all 189 factory cabs from the Axe-Fx II. UltraRes enhances the spectral resolution of an IR without extreme CPU burden or intolerable latency.

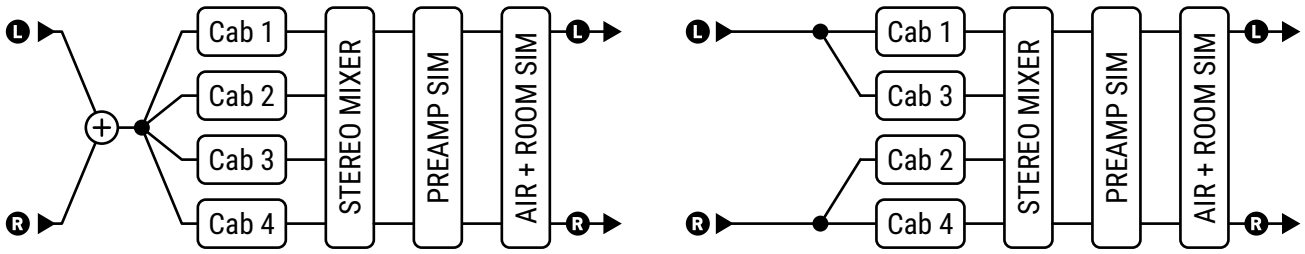
Factory content includes selections from the best of today’s IR producers and artists including Fractal Audio, AustinBuddy, Celestion, ML Sound Lab, Ownhammer, Chris Broderick, John Petrucci, Chris Traynor & James Santiago, Valhallir, York Audio, Dr. Bonkers, and more.

An additional 2,048 “User Cab” memories allow you to load our Cab Packs (including any of those compatible with our previous products) or 3rd party IRs, and a built-in utility allows you to capture and save your own speaker IRs —now with 16 “Scratchpad” locations!

The Cab block in the Axe-Fx III features a four-channel mixer based on our popular Cab-Lab software. It provides the capability to mix and remix IRs on-the-fly as you would do with real mics on a real speaker cabinet.

If you are new to the Cab block, you may enjoy reading “The Sound of IMPULSE RESPONSES” on p. 68 before jumping in.

THE CAB BLOCK



Mono Input Modes *Stereo Input Mode*

CABS PAGE

The **Cabs** page allows you to select up to four different IRs, with a full column of parameters for each one. The names of currently selected IRs are shown at the top of the page. Each of the following parameters is available for every IR.

Bank – Selects from the various banks containing cabs. The available banks are:

- **Factory 1, Factory 2:** Two banks of 1,024 built-in IRs contain a wealth of selections from the best or today’s IR producers and artists, including many taken from Fractal Audio Cab packs produced prior to March 2018.
- **User 1, User 2**: Two banks of 1,024 IRs are blank until you load them yourself. Learn more in “Loading User Cabs” on p. 175
- **Legacy**: This bank contains all 189 factory cabs from the Axe-Fx II XL+.
- **Scratchpad**: The 16 Scratchpad locations are designed to allow you to “audition” cabs before committing them to a User Cab slot. This capability is especially useful when capturing IRs or using Cab-Lab (https://www.fractalaudio.com/cab-lab-3). Please note that the contents of the Scratchpads are cleared every time you power off.

Cab Number – This selects a specific IR by its number within the current bank.



Cab Picker – With the cab number field selected, press **ENTER** to show the **Cab Picker**. This lists all IRs in the current bank and makes it easy to scroll through the list to find what you are looking for. UltraRes IRs are shown in violet. Non-UltraRes IRs are shown in green. Cabs are initially shown in numbered order, but the **Sort A–Z button** (Push-knob **A**) re-sorts the list alphabetically. The **Filter** button filters the list, selecting a different cabinet size with each press (or showing FILTER:OFF when the list is not filtered. Filters include: 1x4, 1x6, 1x8, 1x10, 1x12, 1x15, 2x10, 2x12, 4x10, 4x12. Press **EXIT** to leave the **Cab Picker** and return to the **Cabs** page.

Cab 1, 2, 3, 4 Level – This parameter may seem straightforward enough, but it can actually be a bit tricky until you understand how it works. Rather than simply controlling the overall level of each Cab IR as you might expect, this sets *relative* levels between the IRs when more than one is loaded. The final blend is then normalized[1] and controlled via the **Level** parameter on the **Mix** page—a master control for all four Cab IRs. So, for example:

- When only **ONE** Cab IR is in use and the other three are **muted**, this control will have no effect since *relative* level doesn't apply without two or more Cab IRs to compare.
- When **TWO (or more)** Cab IRs are loaded, you only need to adjust the one(s) you want quieter. Think about it: reducing Cab 1 to –2dB and Cab 2 to –4 dB is the same as leaving Cab 1 alone and setting Cab 2 to –2 dB. Only the *difference* counts!
- With more than one Cab IR in use, you will hear the *tonal* blend change, but the overall volume will remain relatively the same because again, the blend is normalized as you adjust. Do keep in mind however that because of their frequency contents, some cabs may be *perceived* as louder or quieter than others.

1 Normalization in this case makes the output signal as loud as it can be without distorting.

Mute/Solo – **Mute** turns any of the four IRs ON or OFF. When an IR channel is muted, it uses no CPU, making this one of the parameters you can turn to in order to reduce CPU burden. For reasons discussed above under **Level**, the overall level will not change when you mute one or more IRs. **Solo** mutes all IRs except the selected one.

Pan – Sets where a given IR will be heard in the stereo field.

Mic Distance – Sets the distance of the virtual microphone in millimeters, causing sub-millisecond delays that can be used to create inter-mic phase effects.

Proximity – Simulates the classic "proximity" effect of microphones on a cab, causing an increase in bass or low frequency response as proximity is increased.

Smoothing – Reduces the prevalence of peaks and valleys in the IR, reducing the notches that are characteristic of a mic'd speaker. Some may find this effect brings the sound closer to that of a cab "in the room."

PREAMP PAGE

A microphone on a guitar speaker is subject to the pleasing musical distortion generated by a mic preamp. It might range from subtle "warmth" to full-on "nasty". Real mic preamps also offer tone controls which change the sound of the mic'd speaker. The Cab block includes controls to simulate these effects.

Drive – Sets the overall gain of the simulated preamp. Increase for more drive. A VU meter below the knob shows the level into the virtual preamp. As you turn it up and the VU meter approaches or exceeds the 0 dB marker, you will begin to overdrive the preamp.

Saturation – This controls the ratio of even/odd harmonics in mic preamp distortion.

Bass, Mid, Treble – These adjust the tone of the virtual mic preamp.

Preamp Mode – Choose between "HIGH QUALITY" and "ECONOMY" modes which use more or less CPU.

Preamp Type – Selects from various preamp types, each with their own drive characteristics. Options include: Tube, Bipolar, FET 1, FET 2, Transformer, Tape 70us, Tape 50us, Tape 35us, Vintage, Modern, Exciter. Set to "NONE" to bypass the virtual preamp completely, saving some CPU.

Low-Cut/Hi-Cut/Filter Slope – These adjusts the cutoff points of high-pass and low-pass filters after the preamp. Increase Low Cut to reduce boom. Decrease High Cut to darken the tone. Slope allows you to select between first-order (6 dB/octave) or second-order (12 dB/octave) filters, with options for both at 6, both at 12, and one or the other at 6/12 . Note that these controls still work when preamp is set to "NONE"!

ADVANCED PAGE

Air, Air Frequency – Allow add high-frequency "air", with a cutoff frequency to control brightness.

Proximity Freq – Tunes the frequency range over which the "proximity" effect occurs. The amount is set per-IR on the **Cabs** page.

Room Level, Room Size – These controls determine the level and virtual size of a room reverb simulation, built into the Cab block. Increase to add room ambience.

Mic Spacing – Increases delay times inside the room reverb simulation by simulating the distance of the room microphone from the sound source.

TIP: To convert Axe-Fx II Mic Distance values for use on the Axe-Fx III, multiply the old value in milliseconds by 343.

MIX PAGE

Input Mode – Selects the input mode of the Cab block. Three mono models allow selecting LEFT, RIGHT or SUM L+R, while the STEREO mode processes L+R input channels independently. (Compare the diagrams on p. 66.)

THE SOUND OF IMPULSE RESPONSES

Fractal Audio Systems amp models and speaker simulation technology are incredibly accurate, yet some listeners find the sound of IRs unfamiliar at first. This is because impulse responses typically recreate the sound of "close miking." When you mic a guitar speaker, the mic "hears" something very different to what you might hear. Our ears are by definition "neutral" whereas a mic has distinct "color." We typically listen at a distance (and speaker tone is very different as we move around) while a mic is inches away and stationary, focusing on the desirable sound at a specific spot.

As guitarists, we are accustomed to the sound of a speaker "in the room," but this is not what our audiences hear. For recording and performing, the **close mic'd** sound is essentially a universal standard. THIS is the sound that the Cab block is designed to reproduce, and this explains why not only guitarists, but recording and front-of-house engineers have embraced its use. (Of course our Amp models can also be used with a traditional guitar speaker as demonstrated in many of the rig designs in Section 4: Setting Up).

If you are new to using mics on a guitar amp, you will find the Cab block is a fantastic way to learn more. To get started, listen to single IRs, or explore the factory presets which combine several at once. For almost a century, artists, producers and engineers have refined the craft of placing or blending mics to achieve a desired tone. Many classic techniques are easy to recreate. Try a tried-and-true "recipe" like blending one "bottomy" and one "edgy" mic, or try something totally original.

The factory content includes dozens of speakers with different mics in different positions. You may also enjoy "Mix" IRs by Fractal Audio or 3rd-parties, which bring a producer's experience to you in a single IR. In any case, recognize that the sound of IRs is the very sound of the speakers and mics they capture.

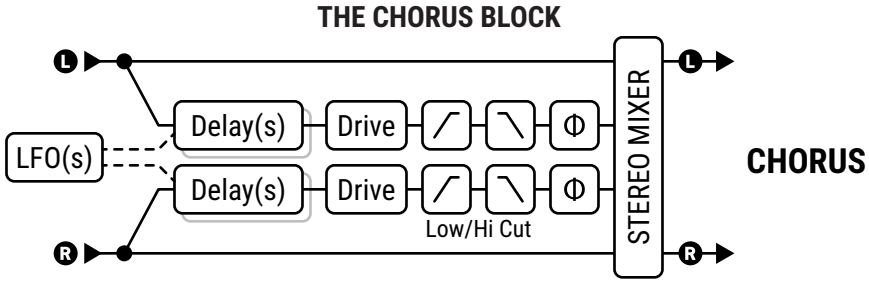
User Cabs add a whole new dimension to the Cab block. See "Loading User Cabs" on p. 175 for more on this.

THE CHORUS BLOCK

2 Blocks, 16 Types, 4 Channels **CHO**

A chorus unit creates one or more delayed copies of the input signal and modulates each of these to create the layered effect of different voices. Used subtly, the effect can be dimensional and liquid, while more extreme settings can produce a vibrato or "Leslie" effect. The Chorus block offers a high-quality, multi-voice stereo chorus capable of producing anything from exceptionally smooth ensemble effects to a wildly detuned warble, with special modes for some rare vintage effects such as tape chorus, dimension chorus, and tri-chorus.

Chorus is heard on many "80s" tones, especially the "LA Session" sounds of Steve Lukather and Mike Landau. Tom Scholz, Alex Lifeson, Mike Stern, Eric Johnson and many others have famously used chorus.



Basic Chorus shown. Some types use variant algorithms.

TYPE PAGE

Type – Make a selection using the **VALUE** or **NAV** buttons with no need to press **ENTER**. This control instantly sets other parameters for different useful sound settings. Types include: **Digital Mono**, **Digital Stereo**, **Analog Mono**, **Analog Stereo**, **Japan CE-2**, **Warm Stereo**, **80'S Style**, **Triangle Chorus**, **8-Voice Stereo**, **Vintage Tape**, **Dimension (1/2/3)**, **4-Voice Analog**, **8-Voice Analog**, **Stereo Tri-Chorus**, and **Dual Chorus**.

All **Analog** types use an algorithm that models classic bucket brigade (BBD) units. **Vintage Tape** uses a unique algorithm that models the behavior of a tape chorus. **Tri-Stereo Chorus** is based on the coveted original.

BASIC PAGE

Number Of Voices – Sets the number of voices or delay lines. Increasing voices increases the fullness of the effect. Use two voices for a vintage chorus, or use up to eight for a lush, multi-layered ensemble.

Rate – Controls the speed at which the chorus oscillates. Use low settings with higher depths for slow-moving sounds. Increase the rate and depth for vibrato effects. Set fully counterclockwise to sync the rate to the **LFO1 controller** (p. 149). When shown in parentheses, it is being set automatically by the **Tempo** parameter.

When the Dual Delay type is selected, individual left and right Rate controls are shown.

Tempo – Sets the **Rate** in rhythmic relation to the tempo. For example, if **Tempo** is set to "1/4" and the Tempo is 120 BPM, the rate will automatically be set to 2 Hz (BPM/60 = Hz). To ignore tempo, set to "NONE".

Depth – Sets the amount of delay modulation, which determines the amount of detune heard from each voice.

Tip: Rate and Depth are usually adjusted inversely (high rate/low depth or low rate/high depth), but other settings can also produce interesting effects. For greater depth, turn Auto Depth on the Expert page to "OFF".

When the Dual Delay type is selected, individual left and right Depth controls are shown.

Delay Time – Adjusts the minimum delay time from 0.01–50.0 ms. Lower values create a more singular sound.

Mix – Sets the ratio of wet and dry. A setting of 50% produces a classic effect. Set Mix to 100% for pure wet or vibrato effects, while higher values go beyond "double tracking" towards "slap back" effects.

EXPERT PAGE

Low Cut – Adjusts the cutoff frequency of a high-pass filter at the output of the processed signal. This control removes bass frequencies and can be useful to create chorus effects designed for bass guitar.

High Cut – Adjusts the cutoff frequency of a low-pass filter at the output of the processed signal. Decreasing this value creates a darker chorus effect reminiscent of an age when typical effects were unable to reproduce the full frequency spectrum. Lower this to achieve those sounds some might call "warm."

LFO Phase – Adjusts the phase differential between left and right voices, shifting the detune effects for less or more apparent difference between opposing voices.

LFO Type – Sets the "shape" of the modulation. Sine and Triangle are the most commonly used waveforms.

Note: Whenever the number of voices is set to more than two, the LFO type will change automatically to "SINE." If the number of voices is greater than two and the LFO type is changed to something other than "SINE," the number of voices will be reset to two.

See also "LFO Waveforms & Phase" on p. 181.

Auto Depth – Scales **Depth** to create a consistent sound at any **Rate**. This control simplifies dialing in "musical" results. For precise control or wild sounds, you may wish to turn it off.

Phase Reverse – Allows left, right or both channels of the effect to be phase-inverted.

Drive – This control allows you to simulate the gentle distortion produced by overdriving an "analog bucket brigade" delay chip of the type used in many vintage chorus effects. Set to zero for "pristine clean."

Width – Widens the sound, creating a difference between left and right delay times by scaling the right time downwards from the value set (see **Delay Time**, above) toward 1 ms as Width goes from 0–100%.

LFO2 Rate – Adjusts the rate of the secondary LFO. This LFO modulates the primary LFO and can be used to create more interesting effects.

LFO2 Depth – Adjusts the depth of the secondary LFO.

Stereo Spread – Controls stereo width by setting the pan position of the two delays from hard-panned (100%) to dead center (0%).

Dimension Mode – Allows simulating the famous "Dimension"-style rackmount and pedal chorus units:

- OFF: Dimension mode is not active.
- LOW: A neutral version with no tonal coloration.
- MED: Classic Dimension processing buttons 1-3. Set Rate to 0.25 - 0.50 Hz and Depth to taste.
- HIGH: Classic Dimension processing button 4. Rate and Depth as above.

Right Time Ratio – When the Dual Delay type is selected, this scales the time for the right chorus.

MIX PAGE

The Chorus block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode** and **Global Mix** parameters.
See "Common Mix/Level Parameters" on p. 55.

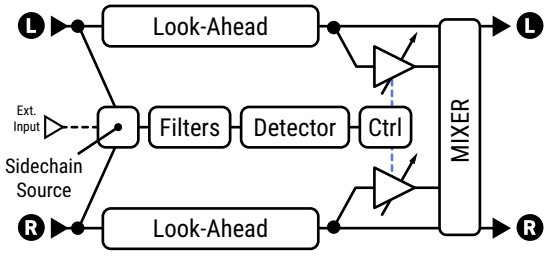
THE COMPRESSOR BLOCK

2 Blocks, 5 Types, 4 Channels **CMP**

A compressor reduces the difference between loud and soft sounds by reducing the louder portions of a signal. The designated reduction occurs whenever the measured signal exceeds a set threshold. While a compressor reduces the volume of loud sections, it can simultaneously boost overall level for greater perceived sustain.

In guitar pedalboards, a compressor is often placed at the start of an effects chain, though using the effect can increase noise or even squealing. In the recording studio, a compressor is typically placed towards the end of a signal chain to smooth irregular levels or provide special effects. This block includes both pedal and studio-type compressors (detailed below) and allows side-chaining and sophisticated detector signal filtering.

THE COMPRESSOR BLOCK



Studio Type shown

TYPE PAGE

Type – Compressor types include **Studio Compressor**, **Pedal Compressor 1**, **Pedal Compressor 2**, **Dynamics Processor**, and **Optical Compressor**.

The **Studio** type simulates popular high-end "Feed Forward" stereo studio compressors. In comparison, the **Pedal** and **Optical** types simulate classic "Feedback" compressors where the detector listens to the output signal instead of the input. **Pedal 2** uses a smoother detector which pumps less, while the **Optical** compressor offers different light types with different control characteristics. The **Dynamics** type allows compression or expansion with a single control.

Make a selection using **VALUE** or **NAV** buttons with no need to press **ENTER**.

CONFIG PAGE

Pedal/Optical and Common Parameters

Compression – Determines how compressed the signal will be, decreasing dynamics and increasing sustain. Set low for light compression or high for total squash. An on-screen meter shows gain reduction.

Attack Time – This sets how fast the compressor reduces the volume once the threshold is exceeded. Higher settings will produce a popping attack since more of the uncompressed signal has time to be heard.

Release Time – Determines how quickly the output volume returns to normal once the input level falls below the compressor's threshold. Fast release rates allow the compressor to bounce back and produce snappy attack, but a setting that is too fast can cause distortion if used in conjunction with fast attack times and high compression or ratio. Slow release times can keep the entire signal quiet, reducing the gain of passages even though they are below the set threshold.

In general the release rate should be set slightly faster than the natural release rate of the program material. An easy way to set the release rate is to strum a chord, watch the gain reduction meter (on PG2 of the Edit menu) and set Release Time so the decay observed is slightly faster than the natural decay of the instrument.

Look Ahead Time – Despite fast attack times, a compressor can fail to "catch" very fast transients. This setting introduces a short audio delay so the gain control stage can have "advanced notice" from the detector, which is side-chained with no delay. Higher look-ahead values help reduce "popping," especially when heavy compression is used on very percussive sources.

Level – Sets the output level of the compressor.

Auto Attack/Release – Turns dynamic control on or off. Turning this switch ON automatically varies the Attack and Release rates according to the program material; the compressor will respond to faster transients with a faster attack.

Light Type – (Optical Compressor-type only) Three light types each cause the compressor to respond differently. Choose the type found in your favorite hardware compressor, or whichever one sounds best to you.

Studio Parameters

The **Studio** compressor has the same parameters as the Pedal/Optical Types, with these exceptions:

Threshold – Sets the level above which automatic volume reduction will occur. When the input exceeds the threshold, the compressor reduces the output. **Ratio** (below) determines *how much.*

Ratio – Sets the input-to-output ratio for signals above the **Threshold**. A ratio of "2.00" (2:1 compression) means that the compressed portion of the signal will be 1/2 as loud as it was at the input. A ratio of "10.00" (10:1 compression) means that the compressed portion of the signal will be 1/10th as loud as it was at the input.

Setting Ratio to "INFINITE" turns the compressor into a "limiter," which has an absolute ceiling or "brick wall" which nothing can exceed.

Auto Makeup – When turned ON, this increases the output level to maintain perceived loudness at the current threshold and ratio. The **Level** knob can still be used.

Detector Type – Selects whether the compressor will use RMS ("Root Mean Square" ...formerly known in our products as "Fast RMS"), PEAK detection, or RMS+PEAK detection. RMS is "smooth" and generally used to even out the level of the program material over a long period of time. Peak detection, commonly used with guitar, is useful for fast limiting. RMS+ Peak combines attributes of both: the speed of a peak and the smoothness of RMS.

Knee Type – The knee control "softens" the operation of the threshold and the ratio, introducing gain reduction gradually as signals approach the threshold. With high compression ratios, a hard knee may produce abrupt gain changes. A soft knee produces a more "transparent" effect since it causes the compressor to engage gradually.

Mix – Sets the ratio of wet (compressed) and dry sounds. This would normally be set to 100% but lower settings allow parallel compression, also known as "New York compression."

Dynamics Processor Parameters

When Type is set to **Dynamics Processor**, the following parameter appears:

Dynamics – When set *below* zero, compression occurs and dynamics are de-emphasized. When set *above* zero, expansion occurs and dynamics are exaggerated. Expansion is basically the inverse of compression. Rather than making loud signals quieter, it makes *quiet* signals quieter, as any signal below the threshold is reduced.

Other controls for this type are the same as their counterparts from the pedal compressors.

SIDECHAIN PAGE

This page is used not only for side-chaining, but also to filter signals before they reach the detector of the compressor. A graph shows the effect of the filters.

Sidechain Source – determines which signal is used to feed the compressor's detector. "BLOCK L+R" is the normal setting which selects the compressor's input (sum of the rows feeding the block). You can also use the input from a designated row, and any of the input jacks. The "BLOCK L" and "BLOCK R" options are useful when the compressor follows an effect with one side out of phase (delay, chorus, enhancer). Remember that the Pedal/Optical types use a feedback architecture, in which the output signal is also fed to the detector.

Low Cut, High Cut – These set the cutoff frequencies of high- and lowpass filters shaping the detector signal.

Emphasis – By emphasizing the high frequencies, this high-shelf EQ can prevent thumpy lows from causing the compressor to pump.

Filter Type, **Frequency**, **Q**, and **Gain** – Use this variable filter to selectively boost or cut the detector signal.

MIX PAGE

This block has **Mix**, **Level**, **Bypass**, and **Bypass Mode** parameters.
See "Common Mix/Level Parameters" on p. 55.

FAQ: WHAT IS "PUMPING"? Pumping is an effect that occurs when a sudden strong peak causes a compressor to reduce levels such that the entire signal audibly dips and then returns. The event that triggers pumping might be brief, and can be in any part of the frequency spectrum. To reduce pumping caused by low or high end spikes, you can use sidechain filtering to make the compressor less sensitive to such peaks. Very fast and very slow release times tend to make a compressor seem more resistant to pumping, but can bring issues of their own. Another possibility is to switch to a multiband compressor, which handles lows, mids and highs individually. Now, it cannot go without saying that pumping can also be used as an effect in itself, as when a kick drum is set up as a sidechain source to give electric guitars a bouncing sound common in dance and industrial music.

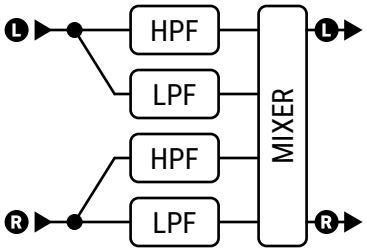
THE CROSSOVER BLOCK

2 Blocks, 2 Channels **XVR**

A crossover splits an audio signal into high and low frequency components so they can then be handled separately. This two-way stereo crossover block contains 4th-order Linkwitz-Reilly filters for excellent separation. You can use two crossover blocks to create a three-way crossover by feeding the output of one into the other.

Applications include splitting a signal into two different amps, or creating "multi-band" effects such as polyfuzz or hi-lo chorus.

THE CROSSOVER BLOCK



CONFIG PAGE

Frequency – Sets the crossover frequency of the filters.

Frequency Multiplier – When set to "×10," the crossover frequency is multiplied by ten.

Lo Level Left, **Lo Level Right** – These set the low-frequency level on left and right channels.

Hi Level Left, **Hi Level Right** – These set the high-frequency level on left and right channels.

Low Pan Left, Hi Pan Left – These controls allow panning the two Left input bands anywhere in the output.

Low Pan Right, **Hi Pan Right** – These controls allow panning the two Right input bands anywhere in the output.

MIX PAGE

This block has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See "Common Mix/Level Parameters" on p. 55.

THE DELAY BLOCK

4 Blocks, 18 Types, 4 Channels **DLY**

The Delay block creates classic, modern, and innovative echo effects. Any delay records an input and then plays it back later in time, creating the effect of an echo…echo…echo. Modified tape recorders were once used for this purpose, but these had sound quality, noise, and reliability concerns. Solid-state ("analog") delays provided an alternative to tape but had shortcomings of their own. The advent of digital technology paved the way for delays with pristine sound, longer times, and superior flexibility, plus the ability to use additional processing to simulate the favorable "nostalgic" qualities of tape, analog, and even lo-fi digital predecessors.

In addition to this block type, additional delay blocks include Multitap, Plex, Megatap, and Ten Tap delay.

TYPE PAGE

Type – This control sets various parameters of the Delay block to achieve popular delay effects instantly. See the table below for a listing of available types. Make a selection using **VALUE** or **NAV** buttons with no need to press **ENTER**.

TYPE	DESCRIPTION
Digital Mono	Full-range, pristine modern digital delay in mono.
Digital Stereo	As above, but stereo in/stereo out.
Analog Mono	Features the frequency response and character of an analog delay.
Analog Stereo	As above, but stereo in/stereo out.
Mono Tape	Special type with dedicated algorithm featuring Motor Speed and other tape controls.
Stereo Tape	A stereo delay with the sonic properties of a tape echo.
Ping-Pong Delay	Echoes bounce from left to right automatically.
Dual Delay	Two delays in one, each with a full set of controls, plus cross-feedback options!
Reverse Delay	This type samples continuously and plays backwards snippets of audio.
Sweep Delay	An LFO controls stereo bandpass filters at the output of a stereo delay.
Ducking Delay	This stereo delay has dynamic ducking to lower echo levels when you play harder.
Vintage Digital	Uses bit-depth reduction for a lo-fi vibe.
2290 w/ Modulation	Based on the legendary original from Denmark.
Ambient Stereo	Ultra-wide echoes with reverb-like diffusion.
Deluxe Mind Guy	Recreates the sound of a classic delay pedal.
Mono BBD	Mono delay with vintage "bucket brigade delay" tone.
Stereo BBD	As above, but stereo in/stereo out.
Lo-Fi Tape	As mono tape, but very low fidelity to create the sound of a worn-out tape.

A WORD ON "SPILLOVER"

Like the reverb, the Delay block is capable of "spillover," which means that effect tails ring out when the effect is bypassed or when you change scenes/presets. For more on this subject, please see "Spillover" on p. 180.

CONFIG PAGE

CONFIG PAGE: CONFIGURE SECTION

Mono Types and Common Parameters

The **Mono Delay** algorithm has a great set of features including drive, reverb-like diffusion, dual-LFO modulation, EQ in the feedback path, bit reduction, ducking, and more.

Time – Sets the time of the delay in milliseconds. When shown in parentheses, its value is being set automatically by the **Tempo** parameter (see below). Set **Tempo** to "NONE" to regain manual control.

Tempo – Syncs the **Time** parameter (see above) in rhythmic relation to the tempo. For example, if the tempo is 120 BPM, and Tempo is "1/4" (one echo per beat), time will be 500 ms. To ignore the tempo, set to "NONE."

Master Feedback – Scales any and all feedback parameters. Note that the range of this control is 0–200%, making it possible (easy, in fact) to overload the feedback loop.

Feedback – Sets the amount of delay feedback (a.k.a. regeneration) to determine the number of repeats. Negative values phase-invert the signal in the feedback loop.

Echo Pan – Positions the echo in the stereo output of the Delay block. Note that this is different than the **Balance** control, which acts on both wet and dry signals.

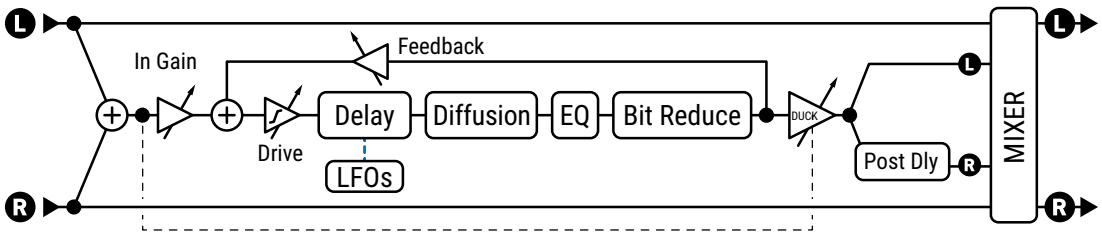
Repeat Hold – This switch defeats the inputs of the delay and "captures" the current feedback loop, which plays infinitely, as long as this switch remains ON.

Drive – Determines the amount of distortion in the delay path. Use this to simulate the way cascading feedback overloads a tape or analog delay.

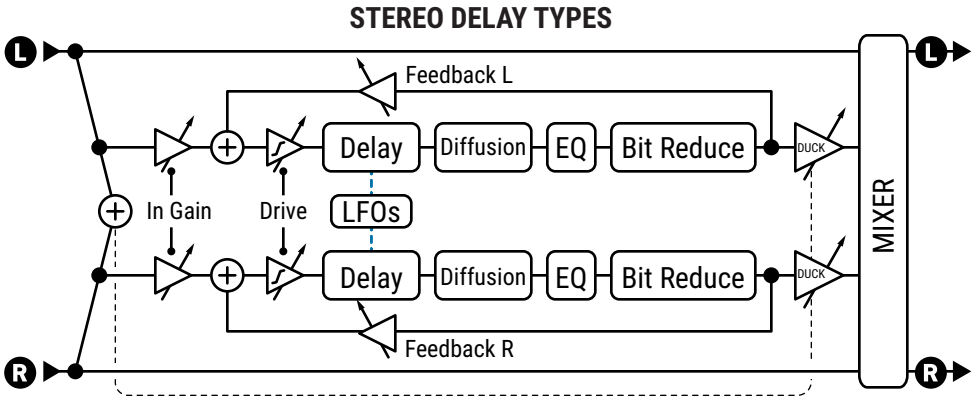
Bit Reduction – This control makes it possible to create the lo-fi sounds of vintage digital delays. The number shown is the number of bits to be subtracted from 24-bit full scale. To create a 16-bit delay, for example, set Bit Reduction to "8" (24 – 8 = 16). Bit Reduction is often used with high-frequency roll-off.

Right Post Delay – Widens the sound of a mono echo by adding 0–100ms of delay at the right (wet) output.

MONO DELAY TYPES



Stereo Delay Parameters



Most parameters for the **Stereo Delay** types work the same as those for the mono types with the exception that the wet signal is stereo-in/stereo out and several additional parameters are available.

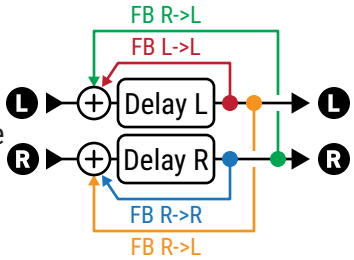
L/R Time Ratio – Sets the right channel time as a percentage of the left. 100% results in both channels having equal delay time. Settings just below 100% will widen the echo sound, while ratios corresponding to whole number relationships like 7:8 (87.5%), 3:4 (75%) or 1:2 (50%) will create syncopated rhythm.

Spread (Width) – Controls the overall stereo image width by setting the pan position of the two delays from hard panned (100%) to dead center (0%) to swapped hard pan (-100%).

Feedback L, **Feedback R** – Stereo delays have independent left and right feedback controls. To preserve "tail" balance, **Feedback R** will be adjusted automatically as **L/R Time Ratio** is adjusted. You may override automatic settings by setting a new value manually. Negative feedback values invert the signal phase in the feedback loop.

Dual Delay Parameters

Dual Delay is identical to the Stereo Delays with the exception that left and right delays are fully independent, with their own **Time**, **Tempo**, **Level**, **Pan** and **Feedback** controls. Dual Delay features a feedback network with four separate parameters. **Feedback L->L** and **Feedback R->R** are the normal feedback controls for each of the two delay lines. **Feedback L->R** and **Feedback R->L** are cross-feedback controls. This structure is illustrated in the diagram at right.



Sweep Delay Parameters

Sweep Delay uses the same algorithm as the Stereo Delay, but adds an LFO-driven stereo bandpass filter after the outputs of the delay.

Start Freq, Stop Freq – These controls set the range of the filter sweeps.

Resonance – Sets the resonance of the filter. Some might describe this as the "intensity" control.

Sweep Type – Sets the waveform of the LFO that controls the sweeps. See "LFO Waveforms & Phase" on p. 181.

Sweep Rate – Sets the speed of the sweeps.

Sweep Tempo – Locks the **Sweep Rate** parameter in rhythmic relation to the **Tempo**.

Sweep Phase – Adjusts the phase of the right outputs of the LFO. Set above zero for stereo effects.

Mono Tape Delay Parameters

The **Mono Tape** algorithm is quite different from other delays. It features two heads and a unique motor speed control with some very cool "analog" behaviors.

Motor Speed – Sets the speed of the tape motor from 50% to 200%. This parameter can be modified in real time, making it possible to “time warp” delay playback.

Head 1 Time – Sets the distance between the virtual record and play heads, in milliseconds. Note that time heard will be shorter if **Motor Speed** is increased above 1.0, or longer if below 1.0.

Head 1 Tempo – Locks the **Head 1 Time** parameter in rhythmic relation to the **Tempo**. See the **Tempo** section under the **Mono Delay** config for more information on tempo and time.

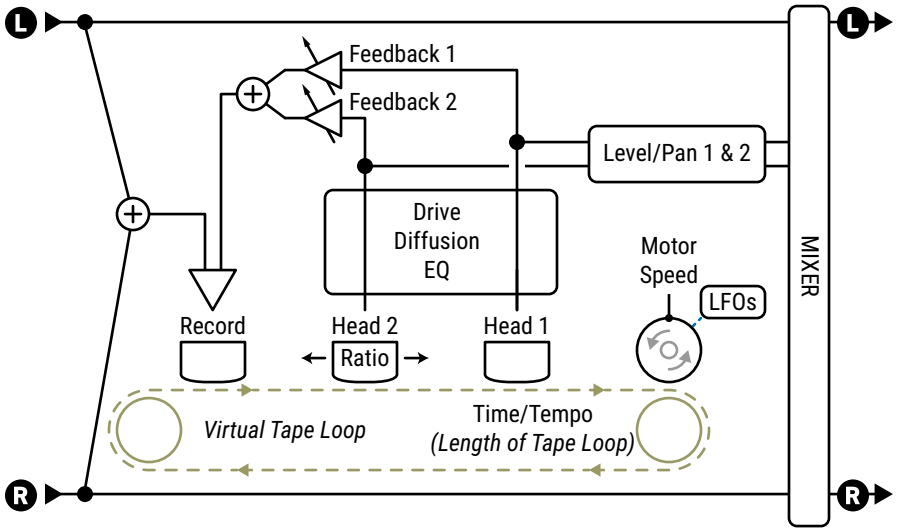
Head 1 Ratio – The Mono Tape Delay has two heads, or “taps” on the loop. This control sets the relative position of the second playback head from zero to a maximum of 100%—the **Head 1 Time** value. Settings close to 100% (e.g. 98%) will widen the echo sound, while values expressing whole number relationships like 7:8 (87.5%), 3:4 (75%), 2:3 (66%) or 1:2 (50%) will create familiar rhythmic patterns.

Level 1, Level 2 – These set playback output level for each of the two heads.

Feedback 1, Feedback 2 – Set the amount from each playback head to be routed back to the recording head to create feedback or “regeneration”. Higher values create a greater number of echoes over time. Because each head replays its own feedback signals plus those of the other head, the sound can very quickly become dense or even out of control—and quite loud. Increase feedback settings slowly, and watch the front panel clip LED as a warning indicator. Lowering **Master Feedback** a bit can help. Negative values phase invert the signal in the feedback loop.

Pan 1, Pan 2 – These position the output of each head in the stereo listening field.

THE MONO TAPE DELAY



Reverse Delay Parameters

The **Reverse Delay** is based on the Mono Delay type, with differences noted below. The Reverse Delay simulates the impossibility of a performance from the future being heard backwards in the present. It does so by using a delay line to first record for a set time period and then to play that recording backwards. While the first recording plays, the next snippet is being recorded so that reverse playback appears to continue seamlessly. If you can think of your performance as a train, this is like individually reversing each car in place instead of flipping the whole thing from front-to-back.

To hear only the reverse audio, make sure **Mix** is set to "100%."

Time – Sets the length of time that the delay line will "record" before reverse playback begins. This can also be synced using the **Tempo** parameter.

Run – When this is turned ON, reverse playback runs and backward audio is produced. Turning this OFF mutes playback, but recording continues in the background. This switch can be remotely operated with a modifier (such as a footswitch or the Envelope controller) to stop and start playback automatically.

Trigger Restart – The Reverse Delay records audio for a duration determined by the **Time** parameter. While one clip is being recorded, the previous one is played back in reverse. This process runs continuously in the background, meaning the start and end of reversed phrases might not align with the moment you begin playing, even if the recording time is synced to tempo. This parameter helps you gain control over this process. When set to ON, it forces the recording process to re-start at the same moment the **Run** switch turns on. Now, your footswitch (or envelope or MIDI or other Modifier source) not only starts reverse playback, but synchronizes the start of recording too.

Crossfade Time – This extends recording outside of the time window, so backward snippets can overlap briefly and crossfade to create a more smoothly continuous performance.

EQ PAGE

Parameters on this page color the tone of the delay sound (not the dry sound). Use it to produce echoes with the tonal color of vintage technology, or EQ creatively. An on-screen display shows the EQ that is dialed in.

Low Cut, High Cut, Filter Slope, Q – These define high- and lowpass filters for broad EQ adjustments. **Low Cut** and **High Cut** set the frequencies. **Filter Slope** results in more gradual or steeper cutoff. **Q** sets the resonance of both filters, as shown in the display.

Freq 1, Q1, Gain 1, and **Freq 2, Q2, Gain 2** – A pair of 2-band peaking parametric EQs allow boosting or cutting at selected frequencies. EQ 1 is lower range (20 Hz–2KHz) and EQ 2 is higher range (100 Hz–10KHz).

MODULATION PAGE

This page includes parameters for delay time modulation, plus the ducker, diffuser and output phase controls.

LFO Depth Range – Sets the overall delay time modulation depth to "LOW" or "HIGH" range.

Phase Reverse – Allows the "LEFT", "RIGHT", or "BOTH" delay line outputs to be phase inverted.

MODULATION PAGE: LFO1/LFO2 SECTIONS

LFO1 Type, LFO2 Type – Sets the "shape" of the modulation. See "LFO Waveforms & Phase" on p. 181 for more on LFO types. Remember that the shift in pitch is determined by the slope of the LFO, so a "TRIANGLE" waveform actually creates a sound that you might expect from a "SQUARE" waveform.

LFO1 Target, LFO2 Target – Sets whether the "LEFT", "RIGHT" or "BOTH" delay line(s) will be modulated. Mono types (including Reverse, Tape, etc.) use only the left delay line.

LFO1 Rate, LFO2 Rate – Sets the delay time modulation speed. When shown in parentheses, it is being set automatically by a **Tempo** parameter (see below). Set the Tempo to "NONE" for manual control.

LFO1 Tempo, LFO2 Tempo – Sets the LFO rate in rhythmic relation to the Tempo. For example, if the tempo is set to "1/4" and the Tempo is 120 BPM, the LFO rate will automatically be set to 2 Hz (BPM/60 = Hz). To ignore the Tempo, set these controls to "NONE".

LFO1 Depth, LFO2 Depth – Sets the depth of delay time modulation. This setting is scaled by the setting for **LFO Depth Range** (above).

LFO1 Phase, LFO2 Phase – Sets the LFO phase offset for the right delay line. This has no effect on Mono delays (including Reverse, Tape, etc.).

MODULATION PAGE: DUCKING SECTIONS

The **Modulation** page also contains the ducking controls. Ducking causes the "wet" level to be lowered automatically when the level of your playing goes above a set threshold. Then, when you play more quietly or pause, the effect volume increases so that the echoes fill the spaces.

Ducker Atten – Attenuation sets the amount by which the effect volume will duck (decrease). A setting of 20 dB, for example, will decrease the echoes by 20 dB when the input level is above the threshold. Set to 0 to defeat the ducker.

Ducker Threshold – Sets the trigger level of the ducker. If the input signal exceeds this value, the delayed signal will be reduced by the amount set with the **Attenuation** control.

Ducker Release – Sets how long it takes for the delay signal to return to normal when the input goes back above the threshold. A short value here causes the ducked echoes to return to full volume very quickly. Longer times cause the levels to swell back more gradually.

MODULATION PAGE: DIFFUSER SECTION

Diffusion – Sets the amount of diffusion, which creates a reverb-like effect to the echoes.

Diffusion Time – Sets the delay time for the diffuser.

MIX PAGE

This block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode, Input Gain,** and **Global Mix** parameters. For convenience, Mix parameters also appear on the **Config** page. See "Common Mix/Level Parameters" on p. 55.

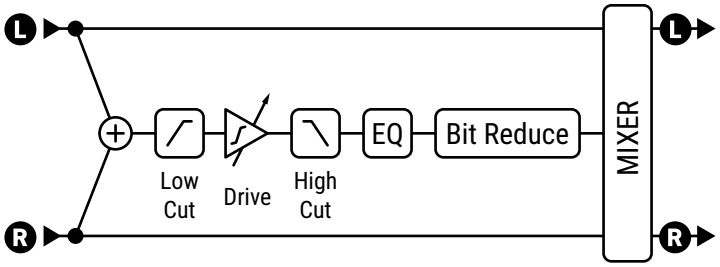
THE DRIVE BLOCK

4 Blocks, 37 Types, 4 Channels **DRV**

The Drive block contains over 36 different, classic stompbox models, ranging from clean boost to insane gain. The overdrive types ("OD" or "Drive") are based on a cold-cathode tube model and give a warm, mellow drive. The Boost types don't distort much unless the drive is set quite high, but they are great to push an amp or reshape its tone. Distortion types are based on a variety of tube and solid-state models. Fuzz types are based on a hard-clipping distortion and give that beautiful nasty tone.

Drive effects include the basic controls you will find on their real world equivalents: tone, drive, and level, plus EQ controls that allow you to go beyond the original models. Expert parameters allow you to perform advanced modifications and create totally new drive types.

THE DRIVE BLOCK



TYPE PAGE

Type – This selects from different models. Select using **VALUE** or **NAV** buttons with no need to press **ENTER**.

TYPE	DESCRIPTION
BB Pre	Based on the Xotic® Pedals BB preamp®.
Bender Fuzz	Based on the classic Tone Bender circuit.
Bit Crusher	Based on a black box we found in the trash outside *Studio Harshclip*.
Blackglass 7K	Based on Darkglass™ Microtubes B7K bass preamp and drive.
Blues OD	Based on the Marshall™ Bluesbreaker®.
Esoteric ACB	Based on the Xotic® AC Booster®.
Esoteric RCB	Based on the Xotic® RC Booster®.
Eternal Love	Based on a Lovepedal® Eternity.
Face Fuzz	Based on Dallas Arbiter Fuzz Face®.
FAS LED-Drive	Designed by Fractal Audio Systems based on LED clipping.
FAS Boost	Our own take on the boost pedal.
Fat Rat	A modified version of the Rat Dist. A bit fuller and smoother.
FET Boost	A gentle, smooth clipping booster with tone controls.
FET Preamp	Based on… you guessed it: a FET preamp.
Full OD	Based on the Fulltone™ Fulldrive OD Pedal.
Hard Fuzz	A hard-clipping, 60s-style fuzz.
M-Zone Dist	Simulates the Boss™ Metal Zone™, popular for extreme gain settings.
Master Fuzz	Based on the Maestro Fuzztone, aka "Satisfaction" fuzz.
Micro Boost	Based on MXR™ Micro Amp, an opamp-driven clean boost pedal.

Mid Boost	A custom mid-boost overdrive.
Octave Dist	An octave distortion based on the Tycobrahe® Octavia®.
Pi Fuzz	Based on the Big Muff® Pi fuzz.
Plus Dist	Based on the MXR™ Distortion Plus.
Rat Dist	Based on the Pro Co™ Rat Distortion.
Ruckus	Based on the Suhr™ Riot, a “versatile high-gain distortion pedal”.
SDD Preamp	Based on the preamp section of the famed SDD digital delay.
Shimmer Drive	An original creation primarily intended as a boost for amps without master volume.
Shred Dist	Based on the Marshall™ Shred Master®.
Super OD	Based on the Boss™ Super Overdrive.
T808 MOD	Captures the most popular TS overdrive mods.
T808 OD	Based on the Ibanez™ TS-9® Tube Screamer overdrive.
Tape Dist	Simulates the clipping of an overdriven reel-to-reel tape deck.
Timothy	Based on Paul Cochrane's “Timmy” pedal.
Treble Boost	Based on a classic treble booster.
Tube Drv 3-knob	Based on the Tube Driver pedal, that actually contained a 12AX7 (3-knob version).
Tube Drv 4-knob	Based on the Tube Driver pedal, 4-knob version.
Zen Master	Based on Hermida® Zen Drive.

BASIC PAGE

This page contains basic parameters you might find as knobs or switches on actual drive pedals. You can generally use only **Drive**, **Tone**, and **Level** controls and achieve excellent results, but experimenting with the cut and EQ controls unlocks some powerful tone-shaping opportunities.

Drive – Sets the amount of gain/overdrive/distortion/fuzz/boost.

Note: Just as with analog gear, a high-gain drive before a high-gain amp may cause noise and squealing.

Tone – Determines the high/low character of the drive simulation, just as the tone knob on a pedal would.

Mix – Controls the ratio of dry to wet. This should normally be set to “100%.”

Level – Sets the output level. Even a clean-sounding drive can be used to “push” an amp to more distortion.

Low Cut – Controls the frequency of the input highpass filter. Increase to prevent "flubby" distortion.

High Cut – Controls the frequency of the output lowpass filter. Lower this for a darker sound.

Bass/Treble – These adjust the low end and high end from ±12 dB.

Mid – Sets mid boost or cut (±12 dB). Set **Mid Frequency** on the Expert page.

EXPERT PAGE

Clip Type – Controls the type of clipping circuit used, based on accurate models of analog components: 4558/ Diode, FET, Full Wave Rectifier, Germanium, Hard, HV Tube, LED, LV Tube, Op-Amp, Silicon, Soft Variable, Null.

Clip Hardness – The “VARIABLE” clip type (above) allows you to dial in a custom clip shape. Low values give a smooth, focused tone while high values give a harder, brasher sound.

Bias – Sets the bias point for the clipping circuit. Varying this setting controls the relative amount of even and odd harmonics. Set very high or very low for a unique “sputtering” effect. Use caution, as setting this too high or too low with certain clipper types can render the block inaudible.

Slew Limit – Limits the large-signal frequency response. Turning up this control simulates the limited high-frequency response inherent in drive pedals using early op-amps. This parameter defaults to an appropriate value when a type is selected.

Mid Frequency – Sets the frequency for the middle band of the EQ on the Basic page.

Bit Reduce – Creates digital distortion by reducing the resolution of the audio signal. The number shown is the number of bits that will be subtracted from 24-bit full scale. To create 4-bit audio, for example, set Bit Reduce to “20.” Yes, it is supposed to sound really gritty and nasty!

Sample Rate – Another nasty lo-fi distortion control, Sample Rate reduction creates intentional aliasing effects.

EQ PAGE

This block has a built-in 10-band, 2/3 octave Graphic EQ well-suited to fine sculpting of guitar tones. The EQ can be enabled/disabled via the **Graphic EQ** ON/OFF parameter on the Expert page.

MIX PAGE

This block has **Mix**, **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See “Common Mix/Level Parameters” on p. 55.

Input Select – The Drive block processes audio in mono. This control determines how incoming stereo signals will be processed. Options include inputting only "LEFT" or "RIGHT" channels or "SUM L+R" (default).

THE ENHANCER BLOCK

2 Blocks, 2 Types, 4 Channels **ENH**

The Enhancer block is used to widen stereo separation, and even converts mono sounds to stereo.

TYPE PAGE

Type – Select using the **VALUE** or **NAV** buttons with no need to press **ENTER**.

- **Modern Enhancer** relies on frequency-based separation of left and right channels to create a widening effect. In comparison to the Classic Enhancer, it does not introduce the risk of phase issues when the signal is later summed to mono, and therefore poses less of a risk for use in presets..
- **Classic Enhancer** delays the right channel by a very small amount to increase the apparent stereo separation between the left and right channels. It also provides individual left and right phase and pan controls. You can use these with or without Width settings as a "channel converter" to reduce width, merge to mono, switch L/R channels, or perform other types of manipulation.
- **Stereoizer** uses multiple high-order filters to create an enhanced stereo image.

CONFIG PAGE

Modern Enhancer Parameters

Width – Determines the character of the effect, shaping the degree of separation.

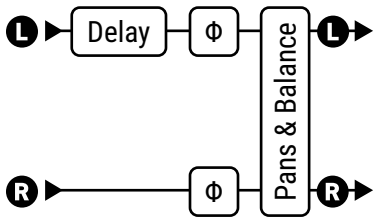
Depth – Determines how far to the left and right opposing frequency bands will be spread.

Low Cut, Hi Cut – These set crossover frequencies to determine which parts of the signal will be enhanced, and which parts will pass through unmodified. Increasing Low Cut causes lower frequencies to pass through unaffected. Decreasing Hi Cut allows high frequencies to pass through unaffected.

Classic Enhancer Parameters

Width – Sets a delay for the right channel from 0–20 ms. Adjust until the desired effect is achieved. Certain frequencies may cancel each other out at various settings, and effects can vary based on the position of the listener. To set a desired number of milliseconds, divide by 20. For example, 7/20 = 35% so to delay the right channel by 7 ms, set Width to 35%.

Invert – Allows phase inversion of the left or right channel (or both). Use in conjunction with the width control to adjust apparent widening.



The Classic Enhancer

Pan L, Pan R – These parameters to control the pan positions of the left and right output signals.

Balance – This changes the relative volumes of the left or right outputs. Sometimes the Classic type causes the Haas effect, which creates a perceived shift in pan away from the delayed side. The Balance control may be used to compensate for this.

Stereoizer Parameters

The stereoizer has **Width** and **Depth** controls which determine how separated the output signal will be.

THE FILTER BLOCK

4 Blocks, 12 Types, 4 Channels **FIL**

The stereo Filter block can be used for simple or spectacular sound shaping. It allows a variety of different effects with real-time control of useful parameters. It can also be used as a “boost” by setting the Type to “NULL” or as a treble (or mid... or bass) booster with great flexibility. Resonant (high "Q") filters bring a distinctive "synth" quality when used after distortion.

The Filter block is equipped with individual left and right pan controls. These allow you to adjust the stereo positioning of the left and right output signals. You can use these controls to turn a stereo signal into two mono outputs (set both to “0”), to reduce the stereo separation, or for general stereo manipulation.

EDIT PAGE PARAMETERS

Type – This selects from different filter types using **VALUE** or **NAV** buttons with no need to press **ENTER**:

- **Null**
- **Lowpass**
- **Bandpass**
- **Highpass**
- **Low Shelf**
- **High Shelf**
- **Peaking**
- **Notch**
- **Tilt EQ**
- **Low Shelf 2 (passive type)**
- **High Shelf 2 (passive type)**
- **Peaking 2 (Variable Q type)**

Notes:

- The **Null** type has no effect on frequency response but allows gain, phase, pan and other adjustments to be made.
- **Low Shelf 2** and **High Shelf 2** filters recreate the quirky “overshoot” behavior of analog shelving filters found on classic mixing consoles. Set the **Q** between 0.5 and 0.707 to recreate classic sounds or experiment with other settings for different effects.
- The on-screen display makes it easy to see the effects of different filter types.

Frequency – Sets the center frequency of the filter.

Order – Selects different filter slopes. 2nd = 12 dB per octave, 4th = 24 dB per octave. A 4th order filter is steeper and more extreme or "squelchy".

Q – Sets the “Q” of the filter. Higher values give sharper responses.

Gain – Sets the gain at the center frequency for the shelving and peaking filter types.

Lowcut, Hicut – These first-order filters provide additional tone-shaping capabilities.

Pan L, Pan R – These controls allow you to adjust the placement of the left and right output signals for stereo width adjustment or stereo-to-mono conversion.

Phase Invert – Controls the phase of the output signal with options for “NONE”, “LEFT”, “RIGHT”, ”BOTH”.

The filter block also has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See “Common Mix/Level Parameters” on p. 55.

THE FLANGER BLOCK

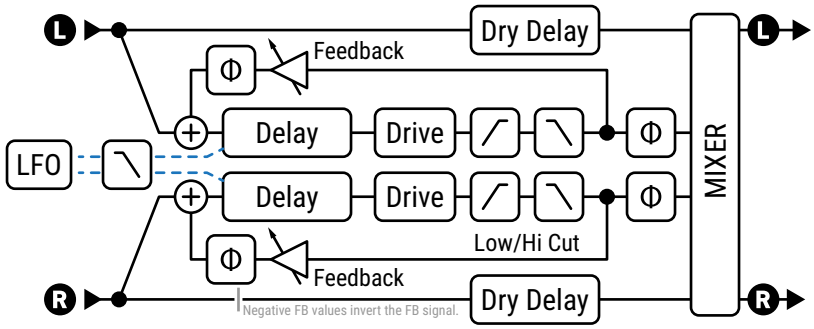
2 Blocks, 8 Types, 4 Channels **FLG**

The sound of a flanger effect can range from subtle chorusing, to jet plane, to robotic drainpipe. The effect was intended to duplicate the sweeping comb-filter sound created when one of two tape decks playing synchronized material is shifted out of time by pressing a finger on the "flange" of the tape reel — hence the term (unless you ask Beatles producer George Martin...). Classic examples of tape flanging can be heard in *Itchycoo Park* by the Small Faces, or *Listen to the Music* by The Doobie Brothers.

As technology advanced, pedal and rackmount flangers appeared, simulating the flange effect, and these have an amazing legacy in their own right. *Unchained* by Van Halen, *The Spirit of Radio* by Rush, and *Barracuda* by Heart all feature prominent flanger effects.

The flanger has many variants, but almost all of them have a "feedback" control (sometimes called "regeneration" or "intensity") which returns some of the output signal to the input and intensifies the characteristic flanger sweep, creating a real attention-getter. With feedback reduced, a Flanger crosses over into Chorus territory, as exemplified by Andy Summers' signature use of the "Mistress" type flanger.

THE FLANGER BLOCK



TYPE PAGE

Type – This control instantly sets other Flanger parameters for useful sound settings. Make a selection using **VALUE** or **NAV** buttons with no need to press **ENTER**. Types include: **Digital Mono**, **Digital Stereo**, **Analog Mono**, **Analog Stereo**, **Thru-Zero**, **Stereo Jet**, **Zero Flanger** and **Pop Flanger**.

BASIC PAGE

Rate – Controls the frequency of the Low Frequency Oscillator, which varies delay time to create the sweep. Use low settings rate with high depths for slow-moving sounds. Increase the rate for vibrato effects. Set fully counterclockwise to sync the rate to the **LFO1 controller** (p. 149). When shown in parentheses, it is being set automatically by the **Tempo** parameter (see below). Set Tempo to "NONE" for manual control.

Tempo – Locks the flanger rate in rhythmic relation to the tempo. For example, if the tempo is set to "1/4" and the tempo is 120 BPM, the rate will automatically be set to 2 Hz (BPM/60 = Hz). To ignore the tempo set **Tempo** to "NONE".

Delay – Resonant flanging is created using a very short delay with feedback. This parameter adjusts the minimum delay time from 0.01 to 10.00ms.

Depth – Sets the depth of delay time modulation. Higher depths increase the amount of flanging as the resonant frequency falls and rises. Depth *adds* to the Delay time, so you can precisely set the range of the effect by first dialing in the sound with Depth at zero and then increasing depth until you hear the low range you want. It helps to turn up Feedback while you do this, so you can focus on the pitch of the effect.

Feedback – This is the famous "regeneration" control discussed above. Feedback sets the amount of wet signal fed back to the input of the flanger. Extreme values give the flanger a more intense quality as it produces sharp singing resonances or clanging metallic overtones.

Note that this control ranges from -100 to +100. Negative values invert the signal phase in the feedback loop, creating sounds with a different character that is more hollow-sounding and vocal (almost like the sound of oscillator sync on a synthesizer (e.g. *Why Can't this be Love* by VH or *Let's Go* by The Cars). In either direction, extreme feedback settings may cause a siren-like ringing oscillation.

Low Cut, **High Cut** – These filter the wet portion of the effect signal, gently rolling off lows or highs at the set frequency with a slope of 6db.

Drive – This control allows you to simulate the soft distortion produced by overdriving an analog "bucket brigade" delay chip of the type used in many vintage flanger effects. Set to zero for "pristine clean."

EXPERT PAGE

LFO Type – Sets the shape of the modulation waveform.

LFO Phase – Adjusts the phase difference between the left and right LFO waveforms. For maximum stereo spread, set this to 180 degrees. For mono flanging, set this to zero.

See also "LFO Waveforms & Phase" on p. 181.

LFO Bypass Reset – This allows you to force the flanger to start from a fixed point in its sweep cycle when you engage the effect. With the default value of "OFF" the LFO will cycle freely, even when the effect is off.

LFO Hicut – Lowering this control filters the LFO waveform, rounding the edges of sharp turns in its shape. The highest setting of 100Hz is basically unfiltered. Certain waveform types (saw, square, random) can otherwise cause clicks or pops as their values jump from one extreme to another. When you adjust the **Rate** of the flanger, you may need to re-visit this setting. A slower LFO needs a lower Hicut setting to preserve the same effect.

Auto Depth – Scales **Depth** to create a consistent sound at any **Rate**. This control simplifies dialing in the sweet spot, but for precise control or wild sounds you may wish to turn it off.

Through Zero – Setting this to "ON" adds a delay to the dry path equal to half the sweep depth. This can emulate true tape deck flanging, where one of the two tapes is first ahead of, and then behind, the other.

Dry Delay – When **Through Zero** (above) is engaged, this allows shifting the cancellation point from the center of the waveform to the edge or anywhere in between.

Phase Reverse – Controls the phase of the wet output signal. Either or both channels can be inverted. Use to increase the effect of through zero flanging.

Stereo Spread – Controls stereo width by setting the pan position of the two delays from hard panned (100%) to dead mono (0%).

MIX PAGE

The Flanger block also has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode** and **Global Mix** parameters.
See "Common Mix/Level Parameters" on p. 55.

THE FORMANT BLOCK

2 Blocks, 4 Channels **FRM**

Although the wah effect can be said to mimic the sound of a human voice, it obviously falls a little short in this regard. The talk-box, a system which plays guitar sounds through a tube into a real human mouth, comes far closer to the sound of actual speech, but is a bit unwieldy, requiring a special speaker, mouth tube, and microphone. The Formant filter makes “talk-box” and other vocal effects possible without such a complex apparatus.

A formant is a prominent resonant peak. The human vocal tract is capable of producing different formants which we recognize as different vowel sounds. Filters can be used to reproduce these. For example, the vowel sound “eee” can be reproduced with a bank of narrow bandpass filters with various frequencies and amplitudes.

The Formant block can even blend dynamically between up to three different vowel sounds, called the Start, Mid, and End vowels. It usually sounds best when placed after distortion, although there are no hard and fast rules. The block is stereo in/stereo out and processes left and right channels independently.

CONFIG PAGE

Start Vowel – Sets the start vowel sound from choices shown in the table. *(Regional variations in English pronunciation may result in differences of opinion over which words best represent each sound. Use your ears.*)

Middle Vowel – Sets the mid vowel sound.

End Vowel – Sets the end vowel sound.

Resonance – Sets the resonance of the filters. Higher resonance yields a more dramatic effect.

Control – This controls morphing between the three vowel sounds. The start vowel is generated with the knob counterclockwise, the mid vowel at 12 o’clock, and the end vowel at fully clockwise. You can assign this to an LFO, an expression pedal, or other modifier sources. See **Section 9**.

The Formant block also has **Mix**, **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters. See “Common Mix/Level Parameters” on p. 55.

Vowels Table

AAA as in Bat
EEE as in Bait
III as in Beet
OHH as in Boat
OOO as in Boot
EHH as in Bet
AHH as in Bot
AWW as in Bought
UHH as in But
ERR as in Bert

THE GATE/EXPANDER BLOCK

4 Blocks, 4 Channels **GTE**

The Gate/Expander is sort of a “reverse compressor” that increases the difference between louder sounds and softer sounds by lowering the volume of soft sounds even further. When set up to completely silence incoming signals below a certain threshold, the expander is called a gate.

CONFIG PAGE

Threshold – This determines how quiet the signal must be before expansion or gating occurs. When input level is lower than the threshold, the expander reduces output volume based on the **Ratio**.

Ratio – This determines how much quieter the signal will be when signal levels are below the threshold. Think of ratio as the lower number in a fraction expressing the output in dB. A ratio of “4” means you multiply the input (in dB) by 1/4 to find the output level (in dB). At lower ratio settings, noise is slightly reduced. At higher ratio settings, it is effectively silenced. A ratio of “1.00” would result in the gate/expander having no effect.

Attack – determines how long it takes for the gate to open when signal levels exceed the threshold. You’ll typically want this set low so the attack of your first notes can punch through naturally. Set very high for a “slow gear” effect.

Release – determines how long it takes for the gate to close after signal falls below the threshold. Use a slow setting for a gradual decay to prevent your sustained notes from being gated abruptly, or use a fast setting to have the gate close quickly, stopping noise even between individual notes or phrases.

Hold – Sets how long the Gate/Expander holds the gate open once the threshold has been exceeded.

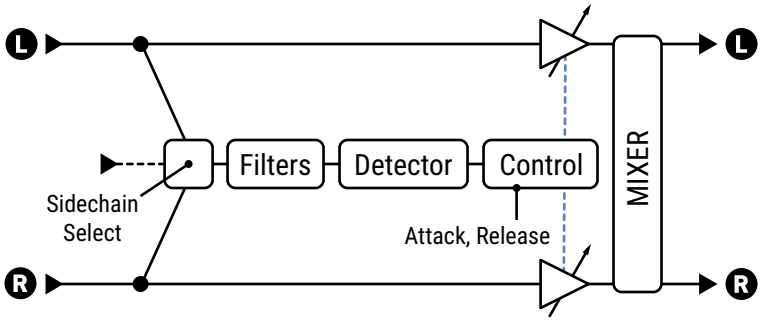
Sidechain Select – Selects the sidechain input source. “BLOCK L+R” is the normal setting and selects the block input (the sum of all rows feeding the block). The other settings allow isolating a single row or input as the sidechain input. By using a row or input as the sidechain input, you can use the Gate/Expander as a ducker or de-esser.

Low Cut/High Cut – These sets the frequency of low- and hi-pass filters on the sidechain input. The filters only shape the signal feeding the detector. They do not affect the tone of the signal at the outputs.

MIX PAGE

The Gate/Expander block has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See “Common Mix/Level Parameters” on p. 55.

THE GATE/EXPANDER BLOCK



THE GRAPHIC EQ BLOCK

4 Blocks, 12 Types, 4 Channels **GEQ**

The Graphic EQ is a stereo multi-band equalizer consisting of a number of filters spread across the frequency spectrum. This powerful and precise sound sculpting tool is extremely easy to use and understand. Each filter has just one setting, indicating how much to boost or cut at a given frequency.

Type determines the number of bands, and their behavior. Each band can boost or cut up to 12 dB. The outermost bands are shelving filters. The 10-, 8-, 7-, and 5-band types can be run as **Constant Q** or **Variable Q**. In a Variable Q equalizer, the bandwidth varies as a function of boost/cut level, such that lower boosts/cut levels result in a lower Q for a more broad equalization.

	Band 1	Band 2	Band 3	Band 4	Band 5	Band 6	Band 7	Band 8	Band 9	Band 10
10-Band	31	63	125	250	500	1000	2000	4000	8000	16k
8-Band	80	160	320	640	1250	2.5K	5K	10K		
7-Band	100	200	400	800	1.6K	3.2K	6.4K			
5-Band	80	240	750	2.2K	6.6K					

The 10 Band “2/3 Octave” types center the filter frequencies on a narrower range well-suited to fine tonal sculpting.

Several **Passive** EQ types are also offered: **5 band**: (Low, Low Mid, Mid, High Mid, and High), **4-band** (Low, Low Mid, High Mid, and High), and **3-band** (Low, Mid, High), plus a **3-band Console** type. These models capture the quirky “overshoot” behavior of analog filters found on classic consoles.

Master Q (on the Mix page) adjusts the Q or bandwidth of all bands. A value of 1.0 sets the Q to the default value (typically one octave). Lower values increase the bandwidth, making neighboring bands overlap more, while higher values make each band more narrow and focused.

The Graphic EQ block also has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See “Common Mix/Level Parameters” on p. 55.

QUICK TIP While you’re working on the EQ page, the **A–E** knobs beneath the display allow you to control five bands at once. To switch from the one range of bands to the next, tap **NAV UP** or **NAV DOWN**.

FAQ: HOW TO EQ? This question is discussed in “The Parametric EQ Block” on p. 106 Most of what is stated there also applies to a Graphic EQ. The Graphic EQ has the advantage of being easier to use, but it doesn’t offer the ability to “sweep” frequencies like a parametric equalizer, or see precisely what the EQ curve will be. Not to worry: move from one band to the next and you’ll be able to learn what you **like**, what you **lack** and what you want **less** of.

THE LOOPER BLOCK

1 Block, 1 Channel **LPR**

The Looper block allows you to create multi-layered looped audio performances in real time. It has a maximum recording time of approximately 300 seconds (5 minutes) in stereo and features “undo” capability to remove the last recording or overdub. It can be operated from the front panel, or remotely via MIDI/remote or a connected FC Controller.

CONTROL PAGE PARAMETERS

Some Looper functions are fired using soft knobs A–E as virtual “buttons”, through a foot controller, or via MIDI.

Record – If the looper is stopped and you press Record, recording begins. If the looper is playing, the Record button normally toggles OVERDUB on or off, but you can change this to PLAY or STOP using the **Record 2nd Press** parameter on the CONFIG page.

Play – This starts and stops playback. If the looper is recording or overdubbing when you press Play, recording or overdubbing stops and playback continues.

Erase – This clears the looper memory. (This function is available only while the looper is stopped.)

Undo – This undoes the last pass of recording or overdubbing. Pressing Undo when the loop is stopped erases all loop data. Pressing Undo during playback undoes the last recorded layer. Pressing Undo again during playback restores the undone layer.

Once – Pressing Once during playback causes the Looper stop automatically when it reaches the end of the loop. Pressing **Once** again while in the middle of a one-shot playback instantly starts the loop, re-triggering playback like a *ph- ph- ph- ph- phrase sampler*. Pressing Once while recording stops the loop. If playback is stopped, Once starts playback from the beginning, plays through to the end, and then stops.

Reverse – This toggles the direction of playback.

Additional Looper Functions on the Config page are accessed by turning knobs A–E:

Trim Start, **Trim End** – These parameters can be used to adjust the start and end points of the loop. The graph on the Control page provides visualization of the loop and the start/end points. Use the A knob to adjust trim start and the E knob to adjust Trim End.

Threshold – Threshold recording causes the looper to start recording automatically when you begin playing. When set to a value greater than -80 dB recording will not start until the input signal exceeds the set value. The Record icon will blink when the Looper is armed for recording and will turn solid when recording starts. The meter at the bottom of the page can be used to help set the Threshold. Use the soft knob or main Value knob to adjust the threshold value (or use the Record Threshold parameter on the Config page).

CONFIG PAGE PARAMETERS

Playback Level – This adjusts the level of the looper.

Dry Level – This adjusts the level of the level of the dry signal passing through the Looper.

Overdub Level – This works like a "Loop Decay" control, attenuating the level of previously recorded material in comparison to overdubs. (Think of it as similar to the "feedback" control on a delay). If you never want old layers to decay, set this to 0 dB, but be aware that layers can add up and potentially cause clipping.

Dry Level – This allows you to set the dry level that passes through the looper block.

Loop Quantize – This forces the loop length to a whole number of beats, as defined by the tempo.

Record Threshold – This is a repeat of the same parameter on the **Control** page.

Record 2nd Press – There are three options for what happens when you press RECORD again while Recording:

Overdub – Recording ends, Playback begins, and Overdub recording is turned on.

Play – Recording ends, Playback begins

Stop – Recording ends, Playback is stopped.

Speed – This is a repeat of the same parameter on the **Control** page.

Low Cut, High Cut – These allow gentle filtering of the looper at its output. These controls affect playback only and have no effect on the sound of material that is being recorded.

The Looper block also has **Bypass** and **Bypass Mode** parameters.
See "Common Mix/Level Parameters" on p. 55.

THE IR PLAYER BLOCK

2 Blocks, 4 Channels **IRP**

IR is short for “Impulse Response” — a type of “sonic capture” used by Fractal Audio products for speaker cab simulation and other creative purposes. The IR Player block is in fact a highly simplified version of the Cab block (p. 66), and like the Cab Block, you can load Factory or User Cabs into it. The difference is that you get only one IR “slot” and none of the bells and whistles, like virtual mic pre, “air” simulation, etc. The IR Player block can also be used to load the output of a Tone Match, leaving the Tone Match block istelf available for other purposes.

Bank, **Number** – Use these parameters to select the desired cab IR.

Mode – Selects between Ultra-Res and Standard processing.

Low Cut, **High Cut**, **Filter Slope** – These can be used to apply some basic EQ to the output of the IR player.

Input Mode – Selects the input mode of the Cab block. Three mono models allow selecting LEFT, RIGHT or SUM L+R, while the STEREO mode processes L+R input channels independently.

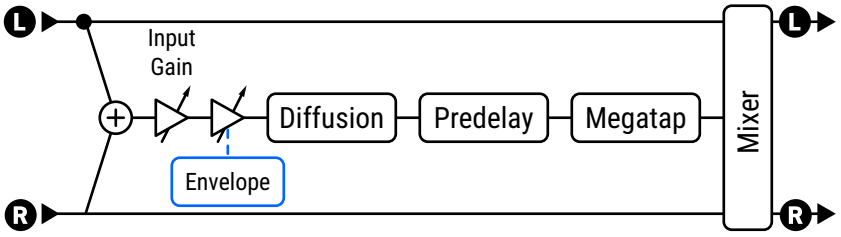
The IR Player block also has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See “Common Mix/Level Parameters” on p. 55.

THE MEGATAP BLOCK

2 Blocks, 2 Channels **MGT**

The Megatap Delay is a 4 second, 64-tap delay line with parametric control of time, amplitude, and panning. This effect can be used to create interesting sonic patterns/textures or to increase density before other effects.

THE MEGATAP BLOCK



CONFIG PAGE

Input Gain – Sets the input level to the effect, allowing for interesting control opportunities when a modifier is attached. For "swell" effects, the Megatap block also includes a built-in envelope follower which scales the input gain independently of this setting.

Master Level – Controls the overall level of the delay.

Delay Time – Sets the delay time of the last tap.

Number Of Taps – Sets the number of taps (repeats) from 1 to 64. Taps are distributed between the dry signal and the last tap, whose position is set by the **Delay Time** parameter (above). The three examples below shows a delay time of 1000ms with 2 taps, 6 taps and 24 taps.



The **Time Alpha** parameter (see below) changes the timing between taps.

Predelay – This delays the entire Megatap effect by up to 1 second. Because the megatap can create echo density that borders on reverb, this affects perceptions of space caused by the effect.

Time Alpha – Sets the rate of time change across the taps.
An Alpha of 1.00 spaces all taps evenly across the delay time, as shown in the examples above.
Alpha values lower than 1 *decrease* the time between each tap and the next:



Alpha values greater than 1 *increase* the time between each tap and the next:



Time Randomize – This scatters the positions of the individual taps along the time line.

Amplitude Shape, **Amplitude Alpha** – These control changes to the level of individual taps over time. Shape sets a pattern of changes (increasing, decreasing, etc.) and Alpha determines how the change varies over time.

Pan Shape, **Pan Alpha** – These control changes to the panning of individual taps over time. Shape sets a pattern of changes (increasing, decreasing, etc.) and Alpha determines how great the change will be over time. With a shape setting of “CONSTANT”, Pan Alpha becomes a simple pan control (0% = left, 100% = right).

Diffusion Mix, Diffusion Time – These set the level and time of an input diffuser, which adds a reverb-like effect to signals prior to the Megatap, thickening the tail and smearing transients to reduce the prevalence of individual echoes.

CONFIG PAGE: ENVELOPE FOLLOWER SECTION

Delays are frequently used after a volume pedal or volume envelope for “swells” of ambient sound. The Megatap block has a built-in input envelope follower which controls the input gain to the effect.

Threshold, Attack, Release – These set the threshold level and times for the input envelope. Tip: For a basic volume “roll up” setting, try Threshold: -70.0 dB; Attack: About 400 ms; Release: about 15 ms.

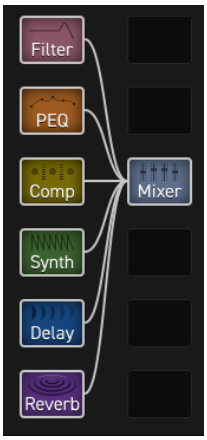
MIX PAGE

The Megatap block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode**, **Input Gain** and **Global Mix** parameters. See “Common Mix/Level Parameters” on p. 55 .

THE MIXER BLOCK

4 Blocks, 4 Channels **MIX**

The Mixer block can mix up to six stereo signals into one stereo or mono downmix. It has six inputs and six pairs of gain and balance controls.



The six mixer channels are “hard-wired” to the six grid rows of the column directly to the left of the mixer block. In the example shown on the right, the **Filter** is on Row 1, so the **Row 1 Gain** and **Row 1 Balance** parameters of the mixer will determine how the signal from the filter is passed to the mixer’s output. The **PEQ** is on Row 2, so the **Row 2 Gain** and **Row 3 Balance** parameters of the mixer will determine how the signal from the PEQ is passed to the mixer’s output, and so on.

Empty rows are still counted. For example, if only the Compressor block were connected to the mixer, the Row 3 controls would still be used because the compressor is on row 3.

All six mixer inputs are stereo. The **Balance** for a given row controls the left/right input balance for that channel. When the balance for an input channel is set fully left, only the left channel of that channel will be heard at the mixer’s output. (Remember that even blocks which process audio in mono still have both Left and Right outs, as well as their own output Balance controls.

Gain Rows 1–6 – These controls scale the level of signals at the six mixer inputs from 0–100%.

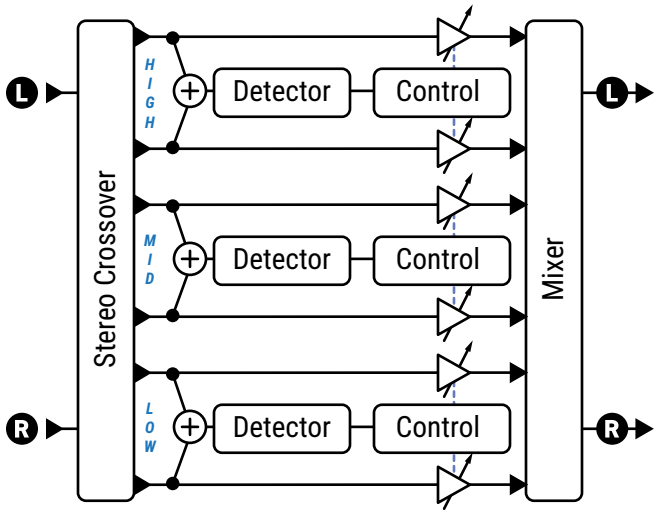
Balance Rows 1–6 – These controls set the left-right balance of signals at the six mixer inputs.

Mono/Stereo – This determines whether the output of the mixer is stereo or summed mono.

The mixer also has a master **Output Level** and **Output Balance** controls. The mixer block cannot be bypassed.

THE MULTIBAND COMPRESSOR BLOCK 2 Blocks, 4 Channels MBC

The Multiband Compressor can be used to shape dynamics and tone, providing independent compressor settings for low, mid, and high frequency bands. The basic principle of a Multiband Compressor is that the input is divided into different frequency components using a crossover. Compression is applied to the bands individually before they are recombined. This allows you to apply different types or amounts of compression to each. Multiband compression is a standard mastering tool and can greatly improve a final mix or a complex guitar sound. It makes the continuity of low frequencies less susceptible to pumping caused by the snap of pick attack.



CONFIG PAGE PARAMETERS

Crossover Frequency, **Frequency Range** – These two pairs of parameters set the ranges and crossover frequencies that divide the Low and Mid bands (from 50–500 Hz or 500–5000 Hz), and the Mid and High bands (from 100–1000 Hz or 1000–10,000 Hz).

Threshold – Sets the threshold above which compression will occur.

Ratio – Sets the input-to-output ratio for signals above the threshold. A ratio of 2.00 (2:1) means an increase of 2 dB is needed at the input to produce an increase of 1 dB at the output.

Attack – This sets how fast the compressor acts once the threshold is exceeded. Higher settings may produce a popping attack since more of the uncompressed signal has time to come through before compression is applied.

Release Time – Determines how quickly the output returns to normal once the input level falls below the compressor's threshold. Fast release rates allow the compressor to bounce back quickly, but a setting that is too fast can cause distortion if used in conjunction with fast attack times and heavy compression. Slow release times keep the signal quiet longer, for example when a "spike" reduces the level which then requires time to return.

Ratio, **Threshold**, **Attack** and **Release** parameters work together to determine the amount and responsiveness of compression for each band. The effects of these settings can be observed in on-screen meters.

Level – Sets the output level of the selected band.

Detector – Selects whether the compressors use RMS ("Root Mean Square") or Peak detection. RMS is "smooth" and works well to even out program material over longer time periods. Peak detection, commonly used with guitar, is useful for fast limiting.

Mute – Mutes the output of the band. By muting two bands, you can solo the third. By muting one band, you can focus on its contribution to the overall mix.

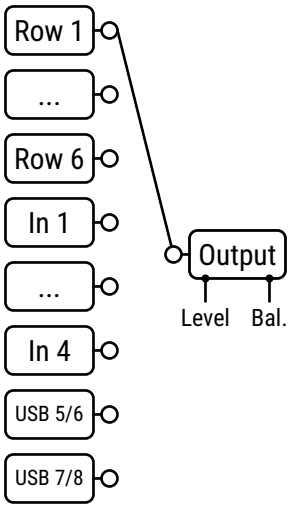
Crossover Slope – The filters dividing the bands can be set to 12 or 24 dB per Octave.

MIX PAGE

The Multiband Compressor block has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See "Common Mix/Level Parameters" on p. 55 .

THE MULTIPLEXER BLOCK

2 Blocks, 6 Channels **MUX**



The Multiplexer block or “MUX” is an input switching powerhouse. It combines various input sources into one block—including two USB options—and can also be used for on-grid switching between rows. No gap is heard when switching sources.

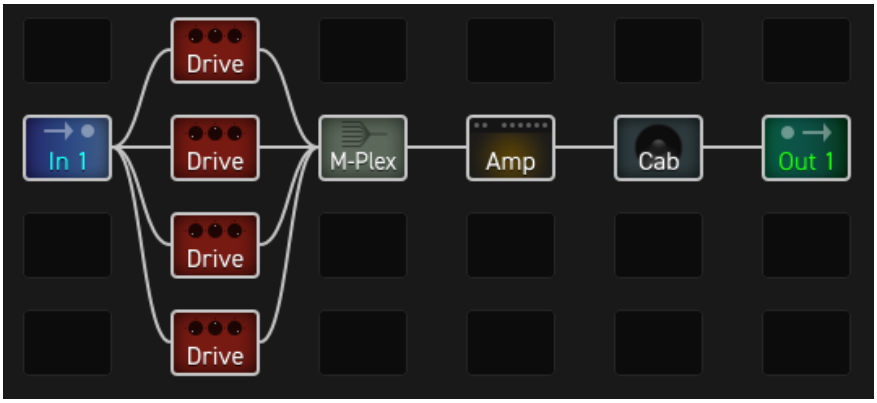
CONFIG PAGE

Input Select – This lists all of the possible input sources and selects which one is active. Only one input can be active at a time. (For multiple inputs, use the **Mixer** block.) Signals from **Inputs 1–4** and **USB 5/6** and **7/8** are present in the Multiplexer without the corresponding blocks needing to be placed on the grid at all. A modifier can be attached to the **Input Select** switch.

This block also has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters. See “Common Mix/Level Parameters” on p. 55.

EXAMPLE

The example below shows one possible use of the Multiplexer, used here to select any one of four different drive pedals.



THE MULTITAP DELAY BLOCK

2 Blocks, 6 Types, 4 Channels **MTD**

The Multitap delay uses up to four delay "taps" to create echoes in intricate patterns and rhythms. To understand the concept of taps, imagine one tape with multiple playback heads, or a turntable with multiple stylus arms. The source material remains the same, but each tap picks it up at a different place/time and can be panned, filtered, and modulated independently. The different types provide a fantastic range of creative possibilities.

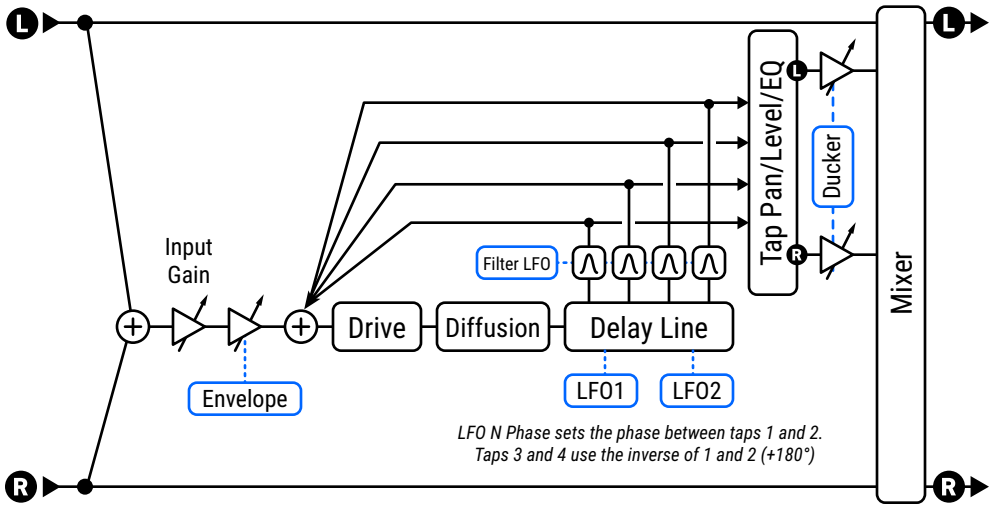
TYPE PAGE

Type – Make a selection using the **VALUE** or **NAV** buttons with no need to press **ENTER**.

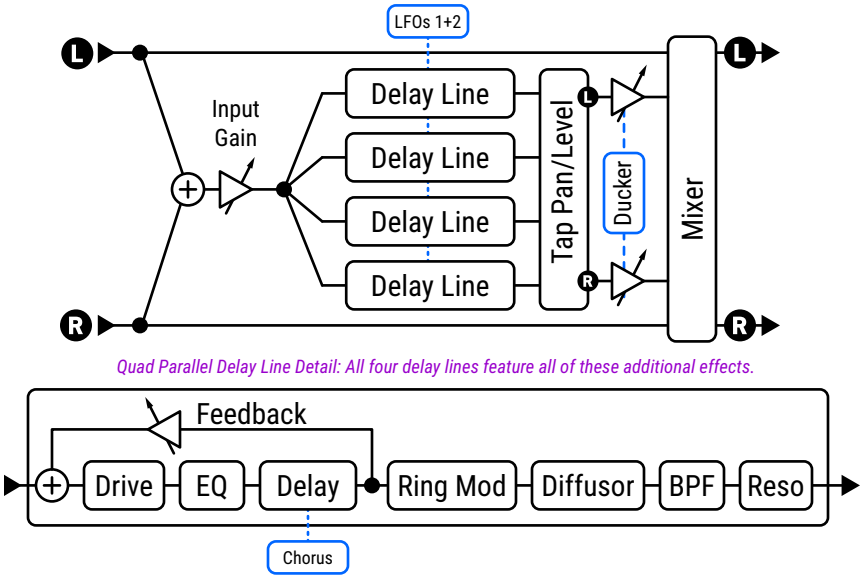
- **Quad-Tap Delay** – offers four independent delay taps with many extras. A truly creative delay experience.
- **Quad Parallel Delay** – extensive delay options with built in ring mod, diffuser, bandpass filter and a resonator.
- **Quad-Series Delay** – uses a very cool and unique feedback structure plus resonator and ring modulation.
- **Diffusor** – creates a plex of compound feedback, offering the ability to add density or even create reverbs.
- **Quad Tape Delay** – uses our proprietary tape delay model with four taps for "space echo" effects.
- **Quad Tap Band Delay** – Identical to the Quad Tap delay, except the Bandpass filters are outside the FB loop.

Several of the above types are illustrated below and on the next page.

THE QUAD-TAP DELAY

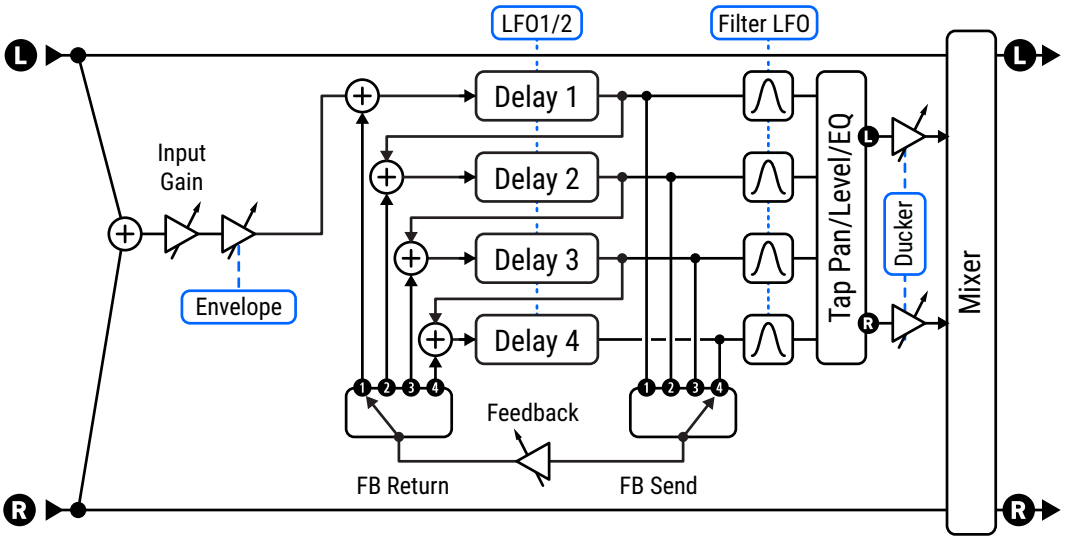


THE QUAD-PARALLEL DELAY



Quad Parallel Delay Line Detail: All four delay lines feature all of these additional effects.

THE QUAD-SERIES DELAY



CONFIG PAGE

Quad-Tap and Common Parameters

CONFIG PAGE: MASTERS SECTION

Master parameters are detailed below. These scale the effects of other parameters and can be controlled with a modifier for interesting real-time changes. If a given type doesn't have certain controls, it will also lack the corresponding master for those controls (ex: the Diffusor type has no Pan controls, so it lacks Master Pan).

Master Time – Scales the times of all taps.

Master Feedback – Scales the feedback settings for all taps.

Master Level – Scales the output levels of all taps.

Master Pan – Scales all tap pan amounts, essentially acting as a "width" or "spread" control. Negative values reverse pan positions in left and right channels.

Master Frequency , **Master Q** – These scales the filter frequency and slope for all taps. You can create dynamic filter effects by using a modifier to change this parameter in real-time, but don't set Q values too high or too low, or the filter may be difficult to notice.

Master Res Time, **Master Res Feedback** – For types with resonators, these scale time and feedback settings.

Master Ring Mod Frequency, **Master Ring Mod Mix** – For types with ring mod, these scale the effects.

Master Rate , **Master Depth** – These scale the Rate and Depth of all LFOs

Low Cut Frequency, **High Cut Frequency** – These simple filters affect the entire wet mix.

Drive – Use drive to add distortion to the echoes and can be used to add a saturation effect.

CONFIG PAGE: DELAY LINES SECTION

Time 1,2,3,4 – Sets the time when the selected tap will be heard, from 0-2000 ms. When any Time is shown in parentheses, it is being set automatically by the corresponding **Tempo** parameter (see below). Set Tempo to "NONE" for manual control.

Tempo 1,2,3,4 – Sets the corresponding **Time** parameter in rhythmic relation to the Tempo. For example, if Tempo 1 is set to "1/4" (one echo per beat), the time for Tap 1 will match the tempo.

Feedback 1,2,3,4 – These set the overall feedback blend. See the various diagrams to better understand how feedback is created. Negative values cause the signal to be phase inverted.

CONFIG PAGE: LEVEL/PAN SECTION

Level 1,2,3,4 – These set the level of the selected tap in the stereo mix.

Pan 1,2,3,4 – These set the pan for the selected tap in the stereo mix.

CONFIG PAGE: FILTERS SECTION

Four **bandpass** filters inside the feedback loop allow gentle or extreme tone sculpting of the individual tap delays.

Frequency 1,2,3,4 and **Q 1,2,3,4** control the center frequency and bandwidth of the filters. These filters use a constant power algorithm so higher Q values increase the gain of the filter and can cause extreme effects (guard against runaway feedback!) To get a sense of the sound and "shape" of these filters, you can study the Filter block when it is set to a 2nd order Bandpass type. TIP: To disable these filters, set **Master Q** to its minimum settting.

CONFIG PAGE: LFOS SECTION

LFO1 and LFO2 modulate Delay Times. Each LFO has the following parameters:

Type – Sets the shape of the modulation LFO. See also "LFO Waveforms & Phase" on p. 181.

Phase – Offsets the phase of the modulation LFO across the four delay lines. Specifically, the value shown is the phase for **Tap 2**. Tap 3 is 180° out of phase with Tap 1, and Tap 4 is 180° out of phase with Tap 2. So, for example, the default setting of 90° results in phase/offsets of: 0°, 90°, 180°, 270°.

Rate – These set the speed of the modulation LFO. When any **Rate** is shown in parentheses, it is being overridden by a **Tempo** parameter (see below). Set Tempo to "NONE" for manual control.

Tempo – Sets the LFO rate in rhythmic relation to the Tempo. To ignore the Tempo, set these controls to "NONE".

Depth – Sets the depth for the modulation LFO.

CHORUS SUBSECTION

For types with Chorus, four individual LFOs modulate the four delay voices independently.

Chorus LFO Rate, Chorus LFO Depth (1,2,3,4) – For types with chorus, each voice also has an independent Chorus LFO. By setting the delay times short (2–20 ms), you can create chorus effects even more lush than those of the legendary Tri Chorus or the Quad Chorus from previous generation products by Fractal Audio.

FILTER LFO SUBSECTION

The filter LFO has the same basic controls as LFO1 or 2 (detailed above). The difference is that it modulates the frequencies of the four bandpass filters (Sthe FILTERS section is detailed on p. 101)

CONFIG PAGE: DIFFUSER SECTION

Diffusion Mix, **Diffusion Time** – Sets the level and time of the diffuser, which introduces a reverb-like effect.

CONFIG PAGE: DUCKER SECTIONS

Ducking causes the wet level to be lowered automatically when the level of your playing goes above a set threshold. When you play more quietly or pause, the effect volume increases so that the echoes fill the space.

Threshold – Sets the trigger level of the ducker. If the input signal exceeds this value, the delayed signal will be reduced by the amount set with the **Attenuation** control.

Attenuation – Sets the amount by which the effect volume will duck (decrease). A setting of 20 dB, for example, will decrease the echoes by 20 dB when the input level is above the threshold. Set to "0" to defeat the ducker.

Release Time – Sets how long it takes for the delay signal to return to normal when the input goes back above the threshold. A short value here causes the ducked echoes to return to full volume very quickly. Longer times cause the levels to swell back more gradually.

CONFIG PAGE: ENVELOPE FOLLOWER SECTION

Delays are frequently used after a volume pedal or volume envelope for "swells" of ambient sound. The Multitap block has a built-in **input envelope follower** which controls the input gain to the effect.

Threshold, Attack Time, Release Time – These set the threshold level and times for the input envelope. Tip: For a basic volume "roll up" setting, try Threshold: -70.0 dB; Attack: About 400 ms; Release: about 15 ms. Increase **Attack Time** for a slower fade in.

Quad Parallel Delay Parameters

The **Quad Parallel Delay** type (illustrated on p. p. 100) features four delay lines, each with a great collection of effects. (Four of each: Ring Modulator, Diffuser, Bandpass filter, Resonator, and Chorus.) Most parameters are similar to those covered under Common Parameters (p. 101). Exceptions are detailed below.

CONFIG PAGE: RESONATORS SECTION

Resonator Time (1,2,3,4) and **Resonator Feedback (1,2,3,4)** – These control the settings for four resonant comb filters, one per delay line. These can be used to create resonant, metallic, flanger-like tones tones. Be sure to adjust Master Resonator feedback above the default value of 0% if you want to hear these resonators.

CONFIG PAGE: RING MODULATORS SECTION

Ring Mod Frequency (1,2,3,4) and **Ring Mod Mix (1,2,3,4)** These control the settings for four ring modulators, one per delay line. Be sure to adjust Master Ring Mode Mix above the default value of 0% if you want to hear these effects.

Quad-Series Delay Parameters

The **Quad-Series Delay** places four delays in series with a very cool and unique feedback structure. The delay lines are connected end-to-end so that their times are compounded as the signal travels from one to the next. Each line has its own output tap, however, so you can hear each tap as it enters the next in line. Feedback is controlled by four parameters:

Feedback – Sets the total feedback.

Feedback Send – Specifies which delay output (1–4) should be fed back to the input.

Feedback Return – Specifies which delay input (1–4) the feedback is returned to.

Diffusor Parameters

A Diffusor uses feedback delays to increase density and layers. It “smears” transients to create interesting reverb-like effects. At some Time and Feedback settings, taps can be heard individually, but diffusion is typically used to create more of a “wall of sound” effect. The Diffusor type in the Multitap Delay block is comprised of four two-second diffusors in series using the following parameters:

Master Feedback – Sets the feedback amount to determine density. Together with the individual delay time settings, this determines the character of the effect and the amount of “smear.”

LFO 1 Rate – Sets the rate of modulation to add a chorus-like sound to the tail of the effect.

LFO 1 Tempo – Synchronizes modulation to a rhythmic value in relation to the Tempo.

LFO 1 Depth – Sets the modulation depth to determine the intensity of time variations/chorusing.

Time 1,2,3,4 – Sets the time of each diffuser from 0-2000 ms. When shown in parentheses, it is being set automatically by the corresponding **Tempo** parameter (see below). Set Tempo to “NONE” for manual control.

Tempo 1,2,3,4 – Sets the corresponding **Time** parameter in rhythmic relation to the Tempo. For example, if the Tempo is 120 BPM, and Tempo is set to “1/4” (one echo per beat), Time will be 500 ms. To ignore the Tempo, set to “NONE.”

DIFFUSOR EXAMPLE

Since the Diffusor borrows many of its default settings from the other Multitap delay types, you may not immediately get classic sounds when you select it. Try the following settings for a lush reverb-like blanket effect.

Begin by resetting the channel and then setting its type to “DIFFUSOR”. Now make the following settings:

Time 1: 400 ms **Time 2**: 647 ms **Time 3**: 1047 ms **Time 4**: 1694 ms
Change all four **Tempo** parameters to “NONE”.

Now we’re ready to listen to these very long times. Play a short sharp note or noise and listen to how the echoes build in density as they feed back into each other. Now lower the Master Time to 50% and repeat the test. Now lower Master Time further to about 20% and you’ll start to understand how what sounds like a clutter of echoes at long times becomes a rich reverb-like diffusion effect at shorter times. Experiment with other Time and Feedback settings, modulation, and even Tempo to vary the texture of the effect.

Quad-Tape Delay Parameters

The Quad-Tape Delay type is almost identical to the Quad-Tap type with a key exception being that it uses our proprietary Tape Delay model featuring a **Motor Speed** control to vary the speed of the motor/tape. To hear the classic “time warp” effect, set feedback controls high enough and then vary Motor Speed. This type does not have an input envelope follower built-in.

MIX PAGE

The Multitap Delay block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode**, **Input Gain** and **Global Mix** parameters. See “Common Mix/Level Parameters” on p. 55.

THE PARAMETRIC EQ BLOCK

4 Blocks, 4 Channels **PEQ**

The five-band parametric equalizer is a precise and flexible tone-shaping tool, one that is definitely worth getting to know and understand. A parametric EQ lets you select specific frequencies to focus on, adjust boost or cut amount, and specify how the change should affect neighboring frequencies. While you work, a graphical display depicts the response, with all EQ band parameters shown below on one screen for total convenience.

EQ PAGE

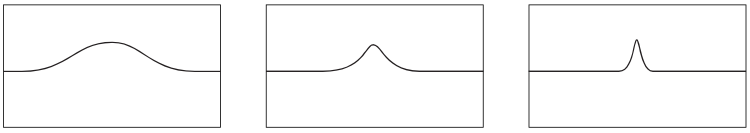
Type – Selectable band types provide even greater flexibility.

- **Shelving** – This type equally boosts or cuts all frequencies above or below the specified frequency, forming a "shelf." The typical bass or treble controls of most devices are shelving EQs.
- **Shelving 2** – A "Passive EQ" type, with the "overshoot" you'd expect from a vintage EQ.
- **Peaking** – A peak filter cuts or boosts around a center frequency.
- **Peaking 2** – In comparison, this variant responds differently to low Q settings.
- **Low-Blocking** – The blocking filter only allows frequencies above the cutoff frequency to pass.
- **High-Blocking** – The blocking filter only allows frequencies below the cutoff frequency to pass.

Frequency – Sets the center or cutoff frequency for the selected band.

Q – This sets the filter bandwidth around the center or cutoff frequency. Set Q low for a broad, gently-sloping boost/cut, or set high for a narrow, sharp boost/cut.

Three examples below demonstrate different Q values for a "Peaking" band. No values are changed except Q.



Q: 0.3 **Q: 1.0** **Q: 3.0**

Gain – Sets the strength of the effect for a band from –12 to +12 dB.

MIX PAGE

The Parametric Equalizer block has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See "Common Mix/Level Parameters" on p. 55.

FAQ: **HOW SHOULD I EQ?** Equalization can be as intricate an art as musicianship itself. Engineers, producers and artists use EQs to compliment, correct, and even create. The following simple tips will help you use EQ blocks to their fullest.

1. EQ works completely differently *before* distortion vs. *after* distortion. Try it each way. Try both at once!
2. If you hear something you don't like (such as "boom" "honk" or "fizz") use EQ to cut it. If you need help finding the frequency to cut, try boosting first, then sweep around until you find the objectionable area. Turn your boost into a cut and fine tune. Experiment with shelving, blocking or peaking types for this application.
3. If a sound seems to be missing something, try adding it in with an EQ boost. You can use the same "sweeping" technique described above, but remember, "you can't put back what isn't there," meaning that a change to some other aspect of the tone (amp? cab?) may be needed to achieve what you're after...
4. EQ in context and at appropriate volume levels. See **Section 7** for more on what this means and why...

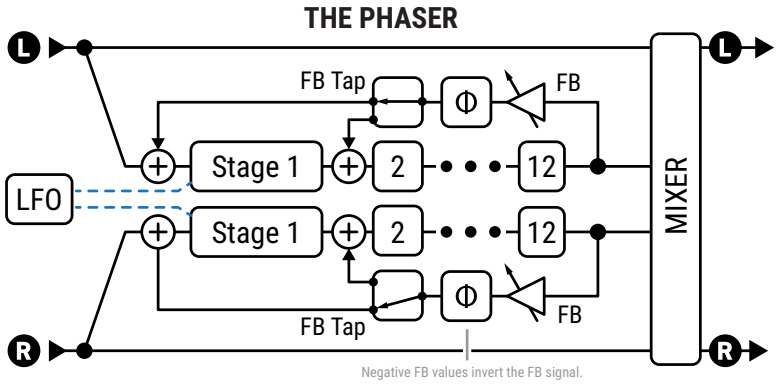
THE PHASER BLOCK

2 Blocks, 8 Types, 4 Channels **PHA**

The Phaser or Phase Shifter uses a series of all-pass filters and a feedback circuit to create a distinctive effect. The phase cancellation and reinforcement caused by the filters creates notches and peaks, and when these are shifted using an LFO, a distinctive swooshing sweep is created. Phaser effects can sound watery or airy. Unlike a chorus or flanger, there is no time/pitch modulation involved.

Examples of phasing include *Eruption* by Van Halen, *Hole in my Life* by The Police, *Have a Cigar* by Pink Floyd, *Machine Gun* by Jimi Hendrix, *Pink Napkins* by Frank Zappa, and various funk/disco/fusion records of the 70s (where phasers were used on guitar, but even more frequently on keyboards, especially electric piano and clavinet).

The Phaser block allows from 2 to 12 stages to be cascaded for subtle or extreme phasing, with positive or negative feedback for distinctly different sounds. A flexible, stereo LFO can imitate the classics or provide new modulation possibilities not possible on any vintage phaser. The phaser even includes a special mode based on the Shin-ei™ Uni-vibe™ effect. The phaser is stereo in, stereo out.



PHASER PARAMETERS

Type – Sets other parameters for different useful sound settings. Types include: **Digital Mono**, **Digital Stereo**, **Script 45**, **Script 90**, **Block 90**, **Classic Vibe**, **Stereo 8-Stage** and **Barberpole**.

Rate – Sets the frequency of the LFO that controls the "sweep." Set fully counterclockwise to sync to the **LFO1 controller** (p. 149). When shown in parentheses, it is being set automatically by the **Tempo** parameter (see below). Set Tempo to "NONE" for manual control.

Tempo – Synchronizes the rate of the Phaser LFO in rhythmic relation to the Tempo. For example, if the Tempo is 120 BPM, and Tempo is set to a "1/4," the LFO rate will be 2 Hz (120 BPM / 60 seconds = 2). Tempo is not used in "Barberpole" type.

Depth – Sets the depth of the LFO that controls the "sweep." Set higher for more dramatic phasing effects.

Feedback – Feedback, known on some phasers as "regeneration" or "resonance," controls how pronounced the peaks and notches are. This control is largely responsible for the iconic sound we associate with a Phaser.

Frequency – (Start Frequency) sets the starting frequency of the first stage filter. This is the lowest point of the cycle. The **Depth** control determines the highest.

Frequency Span – Sets the span of the all-pass filters. Higher values increase separation of the phase notches.

Tone – A simple tone control (wet only).

Phaser Mode – This parameter is set by the Type control but you can also change it manually. “Vibe” mode is based on a Uni-vibe™ effect, while the “Barberpole” type creates a sound evocative of a “Shepard” tone.

Order – Sets the number of phase shifting circuits—or “stages”—in increments of two. Different settings sound distinctly different. For a more pronounced effect, increase the order.

Lfo Type – Selects the “shape” of the LFO, determining how the sweep changes over time. See “LFO Waveforms & Phase” on p. 181.

Lfo Phase – Sets the phase difference of the right waveform of the Phaser LFO. Values greater than 0° produce stereo phasing. A setting of 180° reproduces the legendary “reverse sync” setting of the classic 70s “Bi–Phase” (which incidentally was a 6th order phaser in this mode).

Bulb Bias – In Vibe mode, this controls the quiescent current of the virtual light bulb. Varying this parameter controls how “lumpy” the frequency sweep is. Unlike on the vintage pedal, center frequency doesn’t change with bias, making it easier to control the sweep range.

Feedback Tap – Selects which phaser stage the feedback is returned to. Typically the feedback goes from output to input, but some types require it to be fed back to the second stage. (Stages are numbered from 0, so to return at the second stage, select “1”.)

LFO Reset on Bypass – Makes the phaser start from a fixed point in its sweep cycle when you engage the effect. With the default value of “OFF” the LFO will cycle freely, even when the effect is off.

Sweep Direction – Unique to the Barberpole type, this determines whether the sweep is up or down.

Filter Q – Unique to the Barberpole type. Together with **Feedback**, it controls the intensity of the effect.



BARBERPOLE WARNING: Extreme settings for Filter Q and Feedback can cause the Barberpole phaser to become loud and even unstable. Also be warned that the output level of the block can increase considerably as you lower the value for Order with this type selected.

MIX PARAMETERS

The Phaser block also has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode** and **Global Mix** parameters.
See “Common Mix/Level Parameters” on p. 55.

THE PITCH BLOCK

2 Blocks, 14 Types, 4 Channels **PIT**

The Pitch block uses our latest high-speed pitch detection and incredibly smooth shifting to produce a great range of different effects.

Octave Divider effects add one or two voices down. They have been around since before the dawn of digital.

Detune effects use micro-shifts to create chorus or double-tracked sounds.

Fixed pitch shifters allow you to add parallel harmonies or transpose without using a capo ("polyphonic shifting").

Intelligent shifters create scale-based harmonies, where each shifted note is drawn from a given scale/key (with options for both traditional and custom scales).

Whammy effects are designed to be controlled by an expression pedal. Tom Morello with Rage Against the Machine helped put this type of effect on the map, and now it has become a staple, having been adopted by players from David Gilmour to Steve Vai to Jack White.

Pitch Crystals use forward or reverse granular time stretching to create their distinctive sound.

The **Virtual Capo** can be used to tune up or drop tune the entire guitar.

In the Pitch block, the types above are bolstered by various add-ons including multi-voice capability, built-in LFO modulation, stereo panning, EQ, delays, feedback, modifier control and more. In addition to the many new parameters and features of this block, the pitch detector is faster and more stable than ever before and can be even assigned to any input.

Because the Pitch block is larger with more types and more variations than most blocks, this section is divided as follows:

"Pitch Block Types" on p. 110 provides an overview and covers similarities and differences between types.

"Pitch Block Common Parameters" on p. 111 covers parameters that apply to multiple types.

Pages p. 112 through p. 118 cover the different types in detail.

PITCH BLOCK TYPES

Type – Make a selection using the **VALUE** or **NAV** buttons with no need to press **ENTER**.

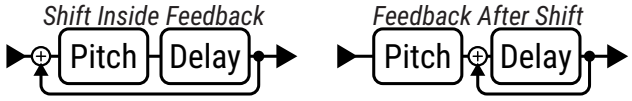
Types are summarized below.

- **Dual Detune** and **Quad Detune** are used for chorus-type effects based on precision detuning of each voice. Both types include an LFO for modulation, and the Quad type also allows feedback.
- **Dual Shift**, **Quad Shift** and **Quad Shift Delay** provide a range of options including mono- or polyphonic shifts.
- **Dual Harmony** and **Quad Harmony** and **Quad Harmony Delay** provide diatonic or "intelligent" pitch shifting.
- **Classic Whammy** and **Advanced Whammy** are single-voice shifters intended for foot control.
- **Octave Divider** provides two voices for easy one and two octave shifts down.
- **Crystal Echoes** uses splice-based shifting to achieve an effect worthy of its name.
- **Arpeggiator** uses diatonic or "intelligent" pitch shifting with a 16-step sequencer to shift in melodic patterns.
- **Custom Shifter** works with global scales to specify a custom shift amount for each of the 12 chromatic tones.

PITCH TYPES COMPARISON TABLE

Type	Voices	Detune per Voice	Shift per Voice	Delay per Voice	Feedback
Dual Detune	2	±50 cents +LFO	No	0–250 ms	None
Quad Detune	4	±50 cents +LFO	No	0–250 ms	Shift Inside
Dual Shift	2	±50 cents +LFO	±24 semitones	0–2000 ms	Shift Inside
Quad Shift	4	±50 cents +LFO	±24 semitones	0–2000 ms	None
Quad Shift Delay	4	±50 cents +LFO	±24 semitones	0–2000 ms	After Shift
Dual Harmony	2	±50 cents +LFO	±25 Scale Degrees (Intelligent)	0–2000 ms	None
Quad Harmony	4	±50 cents +LFO	±25 Scale Degrees (Intelligent)	0–2000 ms	None
Quad Harmony Delay	4	±50 cents +LFO	±25 Scale Degrees (Intelligent)	0–2000 ms	After Shift
Custom Shift	2	±50 cents, NO LFO	±24 semitones (Intelligent)	0–2000 ms	None
Arpeggiator	1	None	±24 Scale Degrees (Intelligent)	None	--
Classic Whammy	1	None	±1–2 octaves	None	--
Advanced Whammy	1	None	±24 semitones	None	--
Octave Divider	2	None	–1 and –2 octaves	None	--
Crystal Echoes	2	±50 cents, NO LFO	±24 semitones	0–2000 ms	Shift Inside
Virtual Capo	1	±50 cents, NO LFO	±24 semitones	None	None

Note: The table refers to two different feedback types used by the Pitch block: **Shift Inside Feedback** and **Feedback After Shift**. With the pitch shift *inside* the feedback loop of the delay, each repeat is shifted again, so the pitch gets higher and higher or lower and lower over time. With the feedback return point after the pitch shift, each echo sounds the same: an identical copy of the original shifted signal. This difference is illustrated below:



Pitch Block Common Parameters

The following are shared by multiple types within the Pitch block. Any key exceptions are noted below.

CONFIG PAGE

Input Mode – The Pitch block normally sums input signals to mono before processing. All Detune and Shift types allow stereo pitch shifting:

- **Dual Detune, Dual Shift**: In Dual types, when **Input Mode** is set to "STEREO", the left input signal is routed to voice 1 and the right is routed to voice 2.
- **Quad Detune, Quad Shift, Quad Shift Delay**: In Quad types, when **Input Mode** is set to "STEREO", the left input signal is routed to voices 1+3 and the right is routed to voice 2+4.

LFO Rate, **LFO Tempo**, **LFO Depth** – Most Pitch types with detune also have an LFO to modulate the detune. Use these controls to add a chorus or vibrato effect. LFO Rate sets the speed of modulation and can be set in rhythmic relation to the tempo using LFO Tempo (set to "NONE" for manual control). LFO Depth sets the range of pitch modulation. Modulation phase is varied by 90° from voice to voice (0°, 90°, 180°, 270°).

MASTER PARAMETERS

Master Level – Scales the output levels of all voices.

Master Pan – Scales all voice pan amounts, essentially acting as a "width" or "spread" control. Negative values reverse pan positions in left and right channels.

Master Feedback – Scales the feedback settings for all voice delay lines.

Master Pitch – Scales shift amounts for all voices. Does *not* scale detune values.

Master Delay – Scales **Delay Time** amounts for all voices.

DETECTOR PARAMETERS

Detector Source – This allows you to select a source for the shifter to use in pitch calculations. Detune and Crystal types do not use the detector.

- **Block In** – Pitch information is detected and analyzed at the input of the Pitch block. This mode allows the detector to track the pitch of a delay or effect tail even after you've stopped playing. This mode also considers the internal feedback of the Pitch block when calculating pitch.
- **Input 1/2/3/4** – In this mode, the pitch information comes from the signal at the designated input. This is unaffected by other effects in the current preset and can be an excellent choice for single note tracking.

Tracking Mode – In Shift or Whammy types, set this to optimize the pitch shifter for tracking single notes ("MONO") or chords ("POLY"). You can also turn tracking "OFF" for vintage-style "fixed" shifting; this type can waver or flutter depending upon the note(s) played, but this effect can be desirable when simulating certain gear.

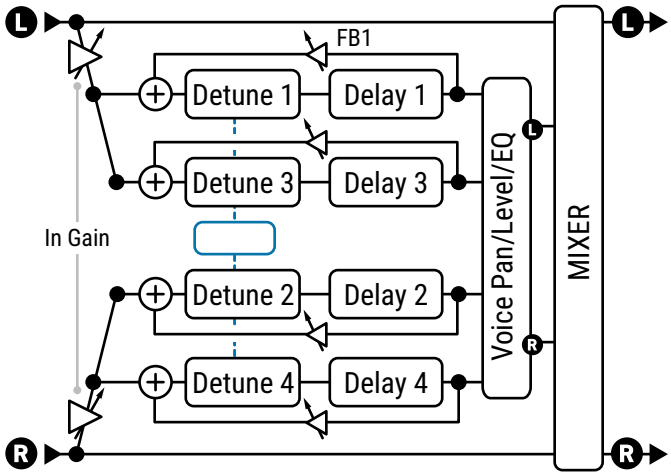
Tracking – This control allows fine-tuning of the pitch shifter "splice length." For large shifts, adjustments can help improve the quality of the shifted note.

Temperament – Switches between "Equal" and "Just" temperaments. Just temperament, while technically slightly out of tune compared to the fretted notes, uses "pure" ratios defined by the harmonic overtone series and can be used for "sweeter" harmonies, especially when followed by distortion.

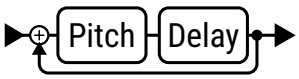
Dual Detune, Quad Detune

Two different detune algorithms effects provide rich doubling and chorus-like effects, and more.

THE QUAD DETUNE



The four-voice **Quad Detune** is shown here. In comparison, the **Dual Detune** has only two voices and no feedback. Both types have stereo input modes (as shown here) or can sum left and right input signals to mono before processing. Quad Detune feedback wraps around the detuners.



Detune 1,2,3,4 – These set the detune amount for each voice ±50 cents.

Delay 1, 2, 3, 4 – These set the delay time of the selected voice from 0–250 ms.

Feedback Mode – (Quad Detune only) Sets whether feedback is "INDIVIDUAL" for each voice (as shown above) or "SUMMED", which sends the summed output of all four voices to the input of every voice.

WARNING! Summed feedback can quickly get out of control. Reduce Feedback before changing the mode.

Feedback 1, 2, 3, 4 – (Quad Detune only) These set the feedback amounts for each voice. Feedback can be used for beautiful flanger effects or create cascading spirals bordering on the deranged.

Level 1, 2, 3, 4 – These set the volume level of each voice.

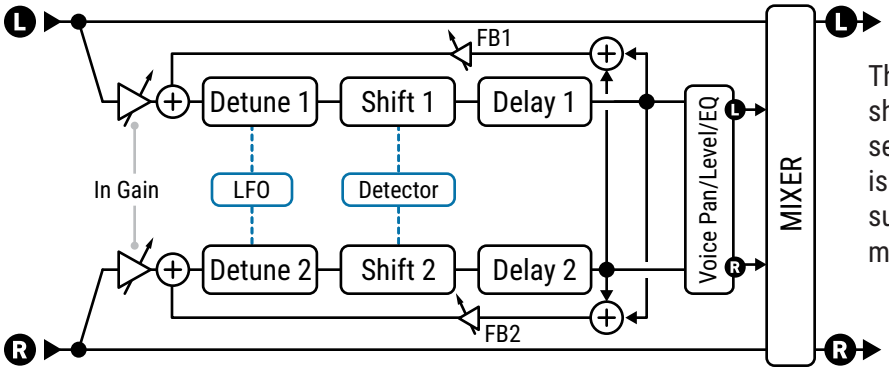
Pan 1, 2, 3, 4 – These set the pan position for each voice.

Master, **LFO**, and **Mix** parameters for the Detune types are covered under "Pitch Block Common Parameters" on p. 111.

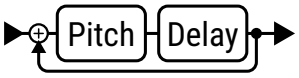
Dual Shift, Quad Shift, Quad Shift Delay

Three different algorithms effects provide fixed shifting effects.

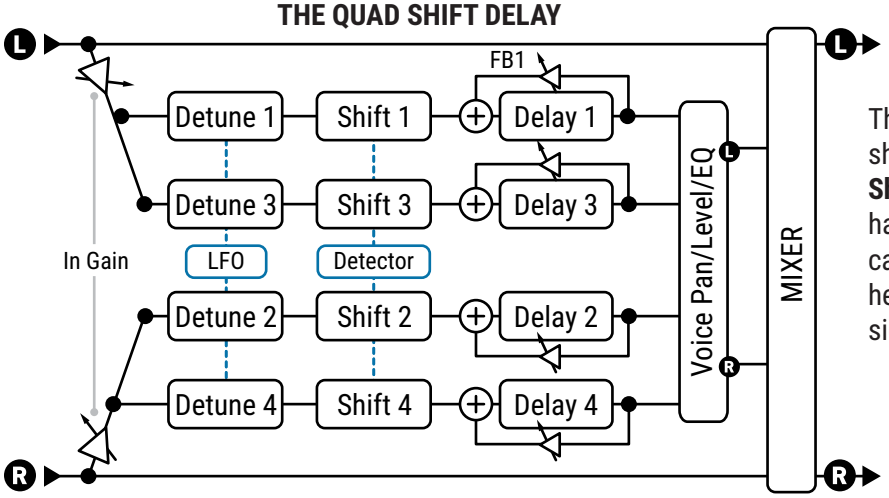
THE DUAL SHIFT



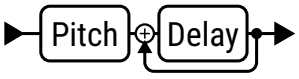
The two-voice **Dual Shift** is shown here. The Detector input is selectable. The stereo input mode is shown here. Mono input mode sums left and right input signals to mono before processing.



THE QUAD SHIFT DELAY



The four-voice **Quad Shift Delay** is shown here. In comparison, the **Quad Shift** has no feedback. Both types have selectable Detector inputs and can use stereo input modes (as shown here) or can sum left and right input signals to mono before processing.



Detune 1,2,3,4 – These set the detune amount for each voice from ±50 cents.

Shift 1,2,3,4 – These set the shift amount for each voice ±24 semitones.

Delay 1,2,3,4 – These set the time for the corresponding voice from 0-2000 ms. When shown in parentheses, it is being set automatically by the corresponding **Tempo** parameter (see below). Set Tempo to "NONE" for manual control.

Delay Tempo 1,2,3,4 – These set the corresponding **Delay** time parameter in rhythmic relation to the Tempo.

Feedback 1, 2, 3, 4 – These set the feedback amounts for each voice. In the **Dual Shift**, the feedback return is *before* the shifters, so successive repeats continue to get higher or lower in pitch. Feedback is also summed, so each voice crosses over to the feedback of the other. The **Quad Shift** has no feedback. In the **Quad Shift Delay**, the feedback return is after the shifters, so successive repeats do not change in pitch. Feedback is per-voice so voices do not cross over in the feedback tail.

Level 1, 2, 3, 4 – These set the volume level of each voice.

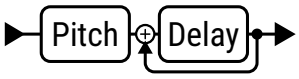
Pan 1, 2, 3, 4 – These set the pan position for each voice.

Master, **LFO**, and **Mix** parameters for the Shift types are covered under "Pitch Block Common Parameters" on p. 111.

Harmony (Dual, Quad, Quad Delay)

In comparison to **Shift** types, the Harmony types use a **Key** and **Scale** to determine the amount of shift. Three different algorithms provide three different variations of this effect. Structurally, they are all very similar to the **Quad Shift Delay** (pictured on p. 113) in that the shifted voices are mixed in parallel to the dry signal, with detune and delay on every voice. Only the **Quad Shift Harmony** type has feedback parameters, with the return located after the shifter so successive repeats are NOT shifted again and again.

The following unique parameters are used for the Harmony shifters:



Key – Sets the key for the harmony.

Scale – Sets the scale or mode into which notes will be shifted. A table of **Scale Types** is provided on p. 116.

Learn – While learn is "ON", the **Key** parameter will automatically change to whatever single note you play. Assign to a footswitch for key changes right in the middle of a phrase! Turn this "OFF" again to return to normal harmonizer function.

Harmony 1,2,3,4 – These set the scale degree that each voice will sound.

It is important to recognize that this is NOT a "semitones" control as used in the **Shift** types, but a specification of which note in the scale should be used. To hear how it works, or to audition any scale, set the **Key** to "G," play the open G string and change the **Harmony** value.

If the current scale contains more or less than seven notes, not including the tonic, (diminished, whole tone, custom, etc.), you may need to use your ears or do a bit of math to identify how its degrees play out over multiple octaves.

Glide Time – sets the rate at which the harmony shifts from the one pitch to another as different notes are played.

CUSTOM SCALE PARAMETERS

Each of the **Harmony** types provide a set of parameters for creating a basic custom scale of up to eight notes. In comparison, the **Custom Shift** type (page p. 118) provides a more extensive way to dial in custom harmonies with up to 12 defined notes.

To use the custom scale, first set the **Scale** parameter (above) to "CUSTOM".

Number of Notes – Sets the number of notes in your custom scale from four to eight.

Tonic – This parameter has no effect on how the scale will sound in actual use but instead allows you to see an example of your custom scale transposed to any key. Change this to whatever makes it easiest for you to dial in the custom notes. Remember: what matters when dialing in a scale is the intervals between the notes.

Note 1,2,3…8 – These are the notes of your custom scale relative to the TONIC. Set to define the scale degrees. Each scale degree must be at least ½ step higher than the previous.

Arpeggiator

The Arpeggiator uses a 16-step sequencer to control the shift amount of a harmonizer so that complex melodic patterns can be created from a single note. This is an “intelligent” shifter which works within a designated key/scale as you play different notes. So, in the key of C (Ionian) Major, the note “C natural” will arpeggiate as C-E-G (C major), but the note D natural will arpeggiate as D-F-A (d minor)

The Arpeggiator parameters are identical to those of the Harmony types (previous page) except as noted below. The Arpeggio type has no feedback parameters.

Key – Sets the key for the arpeggios.

Scale – Sets the scale or mode into which notes will be shifted. A table of **Scale Types** is provided on p. 116.

Arpeggiator Run – When set to “ON,” the sequence starts. When set to “OFF,” the sequence stops and resets to the beginning. Attach the Envelope Follower (p xx) to re-trigger the sequence on each new note.

Arpeggiator Steps – Sets the number of steps in the pitch sequencer.

Arpeggiator Repeats – Sets the number of times the sequence will repeat once triggered. Set to “INFINITE” to loop.

Arpeggiator Tempo – Sets the duration of each sequencer step as a rhythmic value in relation to the Tempo.

Amplitude Shape, Pan Shape – Specifies how the volume or pan changes as the arpeggiator cycles.

Amplitude Alpha, Pan Alpha – Sets the acceleration of the rate of volume or pan change. A setting of 0% results in no effect, while 100% results in an extreme effect.

Stage 1, 2, 3… 16 Shift – These set the number of scale degrees that each note of the arpeggiator will be shifted above or below the note played.

Let’s look at the example of a four-stage arpeggio with values of 0, 2, 4, and 7.

We’ll set Key to C and Scale to Major (Ionian). When we play a C, the arpeggiator will play C-E-G-C’, because:

- C + 0 scale degrees = **C**
- C + 2 scale degrees = **E** (C…D, E)
- C + 4 scale degrees = **G** (C…D, E, F, G)
- C + 7 scale degrees = **C’** (C…D, E, F, G, A, B, C’)

If we play a D, the arpeggiator will play D-F-A-D’, because:

- D + 0 scale degrees = **D**
- D + 2 scale degrees = **F** (D…E, F)
- D + 4 scale degrees = **A** (D…E, F, G, A)
- D + 7 scale degrees = **D’** (D…E, F, G, A, B, C’, D’)

Remember that the notes of the arpeggios and the steps required to arrive at them are drawn only from the designated **Key** and **Scale**. So in the example above, **C** produced a **C major** arpeggio and **D** produced a **D minor** arpeggio: both naturally occurring in the key of C major.

Scales with more or fewer than seven notes in one octave (diminished, whole tone, custom, etc.) can require some mental math, and it is sometimes easiest to just use your ears.

Tip: By choosing the **CHROMATIC** scale, you can create a pattern that ignores the notes you play and simply shifts pitch by a specified number of semitones.

SCALE TYPES AND FORMULAS

Here is a set of scale formulas for all scales used in the **Harmony** and **Arpeggiator** pitch types. If scale names or formulas look slightly different from what you've learned, remember this joke:

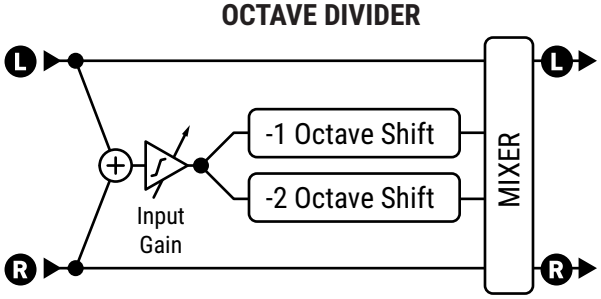
Q: How many musicians does it take to describe a scale?
A: Well, let's see there's Bird, Yardbird, Zoizeau, Charlie, Satchmo, Pops, Satchel Mouth, Dipper Mouth, Louis… so that's TWO right there…"

Scale Name	1	2	3	4	5	6	7	8
IONIAN (MAJOR)	1	2	3	4	5	6	7	
DORIAN	1	2	♭3	4	5	6	♭7	
PHRYGIAN	1	♭2	♭3	4	5	♭6	♭7	
LYDIAN	1	2	3	♯4	5	6	7	
MIXOLYDIAN	1	2	3	4	5	6	♭7	
AEOLIAN (MINOR)	1	2	♭3	4	5	♭6	♭7	
LOCRIAN	1	♭2	♭3	4	♭5	♭6	♭7	
MELODIC MINOR	1	2	♭3	4	5	6	7	
HARMONIC MINOR	1	2	♭3	4	5	♭6	7	
DIMINISHED (whole-half)	1	2	♭3	4	♭5	♭6	♮6	7
WHOLE TONE	1	2	3	♯4	♯5	♭7		
DOMINANT 7 (aka dim. half-whole)	1	♭2	♯2	3	♯4	5	6	♭7
DIMINISHED/WHOLETONE	1	♭2	♯2	3	♯4	♯5	♭7	
PENTATONIC MAJOR	1	2	3	5	6			
PENTATONIC MINOR	1	♭3	4	5	♭7			
BLUES	1	♭3	4	♭5	5	♭7		
CHROMATIC	All 12 tones							

Octave Divider

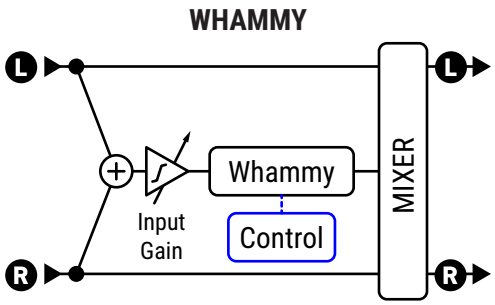
The Octave Divider simulates the classic analog effect which turns the input into a square wave and then lowers the frequency using a "flip-flop" circuit. Like the classic effect, this effect really only works as intended with single notes and would usually be placed before distortion.

Voice 1 and Voice 2 are fixed at one and two octaves down. Each has its own Pan and Level controls.



Whammy (Classic + Advanced)

The whammy effect, first introduced in 1991, is a relative newcomer to the field of guitar effects. The Classic Whammy brings all of the classic sounds of this pedal. Its **Whammy Control** parameter is designed to be operated remotely using a modifier (Section 9), typically assigned to some source controlled by a pedal (though don't let this convention limit your creativity. Sequencer and LFO pitch effects can be great too, as demonstrated in *Map of the Problematique* by Muse.



Whammy Mode – The **Classic Whammy** uses a single parameter to select between the following shift options:

- Up 1 Octave
- Down 1 Octave
- Up 2 Octaves
- Down 2 Octaves
- Up 1 Octave, Down 1 Octave
- Up 2 Octaves, Down 2 Octaves

Start Shift, **Stop Shift** – The **Advanced Whammy** type uses two parameters to specify the shift in semitones. This allows you to customize the range for effects such as whole-step "bender", 4th down/5th up, or anything else you can imagine within a range of ±24 semitones.

Whammy Control – This controls the whammy and would normally be assigned to a pedal, switch, or some other controller using a modifier (Section 9).

Tracking Mode – Use this to optimize the pitch shifter for tracking single notes ("MONO") or chords ("POLY"). You can also turn tracking "OFF" for vintage-style "fixed" shifting.

FAQ:

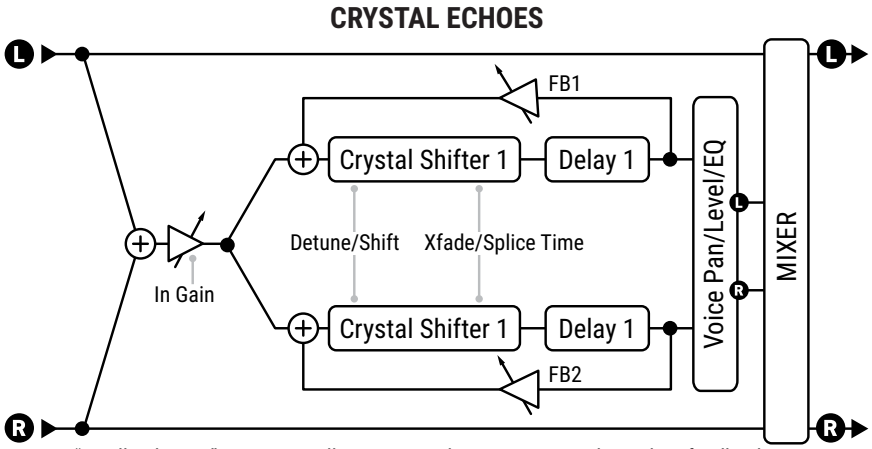
Key	Shift
E♭/D♯	-1
D	-2
D♭/C♯	-3
C	-4
B	-5
B♭/A♯	-6
A	-7
A♭/G♯	-8
G	-9
G♭/F♯	-10
F	-11
E	Srsly?

WHAT ABOUT DROP TUNING? Drop tuning pedals use polyphonic pitch shifting to simulate the sound of a lowered guitar tuning. The **Advanced Whammy** is perfect for this type of effect. Place it at the beginning of your signal path, prior to any Drive or Amp blocks. To make setup easy, just RESET the desired pitch block channel and then re-select the **Advanced Whammy** type. Dial in the drop tune with just a few simple changes. The most important setting is found on the **Mix** page. Set the **Mix** parameter to 100% so that only the "wet" portion of the effect is heard. Also check that **Tracking Mode** is set to "POLY" and **Whammy Control** is at 100%. For the drop, set **Stop Shift** to the desired value. (Ideally, you should have If you have a modifier on **Whammy Control**, perhaps for use in a different channel, just set **Start Shift** to match the value for **Stop Shift**.)

Assuming you are in standard "E" tuning, a table of drop values you can use for any tuning is shown here. Of course, you can also use this effect for "up tuning" effects. As with any tuning effect, the farther you shift in either direction, the less natural it sounds.

Crystal Echoes

The **Crystal Echoes** type or "Crystals Shifter" is an interesting variation on traditional pitch effects. The input signal is sliced into samples, which can be overlapped (crossfaded) reversed, shifted, detuned, and processed by a delay with feedback wrapped around the crystal shifter.



"Feedback Type" parameter allows summed, crossover or independent feedback.

Feedback Type – Selects the type of feedback. "DUAL" sends the individual voices back to their respective delay lines. "BOTH" mixes the voices and sends them back to both delay lines. "PING-PONG" sends each voice to the opposite delay line.

Splice 1 Time, Splice 2 Time – Pitch shifting breaks a signal into pieces called "granules." These are manipulated individually and then "spliced" back together. This parameter sets the length of the granules in milliseconds.

Splice 1 Tempo, Splice 2 Tempo – Sets splice time in rhythmic relation to the Tempo.

Direction – Determines whether "granules" of audio are played back forward or reversed. To understand how reverse works, imagine a word whose individual letters have been mirror-imaged but are still in the correct left-to-right order ("Fractal" vs. "ꟻɹɐʇɔɐl"). In this case, the letters are short snippets of audio. These are reversed (and possibly pitch-shifted) but are played back in the order in which they were recorded. The length of the snippets for a given delay line depends on its **Splice Time** setting.

Crossfade – Sets the amount of overlap in the audio granules. A low setting will make the edges more discrete, while higher settings blend splices together.

Custom Shifter

The Custom Shifter is identical to the **Dual Harmony** (p. 114) except that instead it uses global **custom scales**.

Scale 1, Scale 2 – These select the custom scale to use for each voice. Custom scales are created on the **Scales** page of the **Global Settings** menu under **SETUP** (p. 158). Every custom scale consists of 12 values, one for each note in the chromatic scale, so the shifter knows precisely how to shift any note you play. For example, if you entered **'C' SHIFT : +7 SEMITONES** and played the note C, the shifter would produce a G since G is 7 semitones above C. The range is two octaves up or down.

Key – This transposes both custom scales to the desired key. All Custom Scales are assumed to have a root of A, so this shift is relative to that note. For example, if you wanted to transpose a custom scale down one whole step, you would change the Key to G since G is one whole step below A.

Virtual Capo

The Virtual Capo type is is a one voice shifter that can be used to tune up or drop tune the entire guitar. Its controls are very simple. When you select this type, the mix is set automatically to 100% wet.

Shift controls the amount of shift, up to 2 octaves up or down in semitones (aka *half steps* i.e. *number of frets*)

Detune adds an additional slight shift in cents (each cent is 1/100th of a semitone.

MIX PAGE

The Pitch block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode**, **Input Gain** and **Global Mix** parameters.
See "Common Mix/Level Parameters" on p. 55.

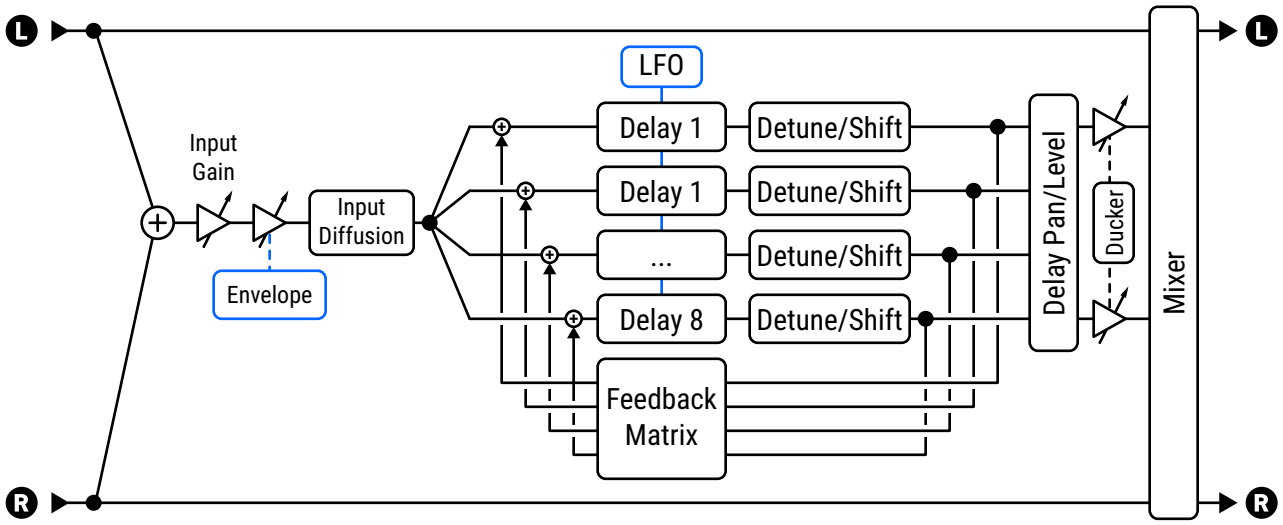
Low Cut, **High Cut** – These filter the wet signal only, gently rolling off lows or highs with a slope of 6db.

THE PLEX DELAY BLOCK

2 Blocks, 3 Types, 4 Channels **PLX**

The Plex Delay block uses a feedback matrix so that each of multiple delay lines is fed back to itself and all the others. The effect can produce very smooth, reverb-like atmospheric effects. When combined with modulation or detune, the result is a huge and lush-sounding space with qualities of echo, reverb, and chorus all at once.

In comparison to other Plex effects, the Plex Delay block allows the use of 4, 6 or 8 delay lines at once.



This diagram shows Plex Shifter. Plex Detune lacks "Shift" parameters. Plex Delay has no pitch effects.

TYPE PAGE

Type – Make a selection using the **VALUE** or **NAV** buttons with no need to press **ENTER**.

- **Plex Delay** is the most basic type of this algorithm, with up to eight lines, delay modulation, input diffusion, built-in ducking, input envelope controls, and more.
- **Plex Detune** is based on Plex Delay, but adds eight high-quality pitch shifters with a range of ±50 cents to the output of the delay taps. These create a chorusing effect in the plex tail.
- **Plex Shifter** adds eight high-quality pitch shifters with a range of ±24 semitones for everything from angelic shimmers above, to infrasonic depths below.

CONFIG PAGE

Plex Delay and Common Parameters

CONFIG PAGE: CONFIGURATION SECTION

Number of Delays – Sets the number of delay times in the matrix. Higher values increase the density of the effect but also use slightly more CPU. Note that other parameters will always be shown in sets of eight, even when Number of Delays is set lower.

Decay Time – Adjusts the feedback matrix to control the time required for echoes to fade. Use caution, as high decay times can result in instability.

Diffusion – This is "the" plex effect, determining how much cross-feedback there is between delay lines.

Low Cut, High Cut – These set the cutoff frequencies for gentle high- and low-pass filters in the feedback matrix.

CONFIG PAGE: MASTERS SECTION

Master Time – Scales all of the delay times in the block.

Master Level – Scales the output levels all of the delay times in the block.

Master Pan – Scales all pan values, essentially acting as a width or spread control. Negative values reverse left and right channels.

CONFIG PAGE: TIME AND TEMPO SECTIONS

Delay Time 1–8 – These set the times for the eight delay lines. If a tempo value appears in (parentheses) this indicates that the corresponding delay has a **Tempo** value set and cannot be changed manually.

Tempo 1–8 – These set the tempo for the eight delay lines. See "Synchronizing Sound Parameters" on p. 153.

CONFIG PAGE: LEVEL AND PAN SECTIONS

Level 1–8 – These set the levels for the eight delay lines.

Pan 1–8 – These set the pans for the eight delay lines.

CONFIG PAGE: ENVELOPE FOLLOWER SECTION

Plex delays are frequently used after a volume pedal or volume envelope for "swells" of ambient sound. The Plex Delay block has a built-in **input envelope follower** which controls the effect send level *inside* the block.

Threshold, Attack, Release – These set the threshold level and times for the input envelope. Tip: For a basic volume "roll up" setting, try Threshold: -70.0 dB; Attack: About 400 ms; Release: about 15 ms.

Plex Detune Parameters

Plex Detune offers these additional parameters:

Crossfade – Sets the amount of overlap used in the granules of the pitch shifters. Lower settings give a "grainy" sound, while higher values have a smooth blended sound.

Detune 1–8 – Sets the amount of detune within a range of ±50 cents. Small values create a subtle chorus, higher settings create descending or ascending cascades.

Master Detune – This scales the settings for the eight detune parameters.

Plex Shift Parameters

Plex Shift offers these additional parameters:

Direction – Determines whether "granules" of audio in the pitch shifter are played back forward or reversed. To understand how this works, imagine a word whose individual letters have been mirror-imaged but are still in the correct left-to-right order ("Fractal" vs. "ꟻɹɐɔʇɐl"). In this case, the letters are very short snippets of audio. These are reversed (and possibly pitch-shifted) but are played back in the order in which they were recorded. The length of the snippets for a given delay line depends on its **Time** setting.

Shift 1–8 – Sets the amount of detune within a range of ±24 semitones. This is where "Shimmer" is born.

MODULATION PAGE

Ducker Attenuation – Sets how much the ducker reduces delay levels. A setting of 20 dB will decrease the echoes by 20 dB when the input level is above the threshold. Set to "0" to defeat the ducker.

Ducker Threshold – Sets the trigger level of the ducker. If the input signal exceeds this value, the delayed signal will be reduced by the amount set with the **Attenuation** control.

Ducker Release Time – Sets how long it takes for the delay signal to return to normal when the input drops below the threshold. A short value here will cause the ducked echoes to return to full volume the moment you stop playing. Longer times will cause the level to swell back gradually.

Input Diffusion, Diffusion Time – These set the level and time of an input diffuser, which adds a reverb-like effect to signals prior to the Plex, thickening effect tails and smearing transients to reduce the prevalence of individual echoes.

LFO Depth, LFO Rate – These set the depth and rate of modulation, which adds a chorusing effect.

LFO Tempo – Synchronizes the rate of LFO modulation to the tempo.

MIX PAGE

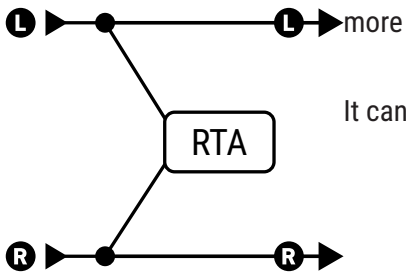
This block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode**, **Input Gain** and **Global Mix** parameters.
See “Common Mix/Level Parameters” on p. 55 .

THE REALTIME ANALYZER BLOCK

1 Block, 0 Channels **RTA**

The Realtime Analyzer block is an animated high-resolution multiband spectrum analyzer with parametric control. You can “set-it-and-forget-it” or adjust parameters to match your specific needs. You can tap grid signal anywhere to feed the RTA, allowing you to observe the frequency content.

CONFIG PAGE



Decay Time – Sets the decay or release time. The higher this value, the more slowly each measurement will “fall off” in the display.

Input Select – Selects how the RTA handles incoming stereo signals. It can monitor left or right channels individually, or sum them together.

Bands – Sets the resolution of the display to 32, 64 or 128 bands.

Window Type – Selects between different window functions to determine how data is displayed. If you are not familiar with “analyzer windowing”, you easily find more information online.

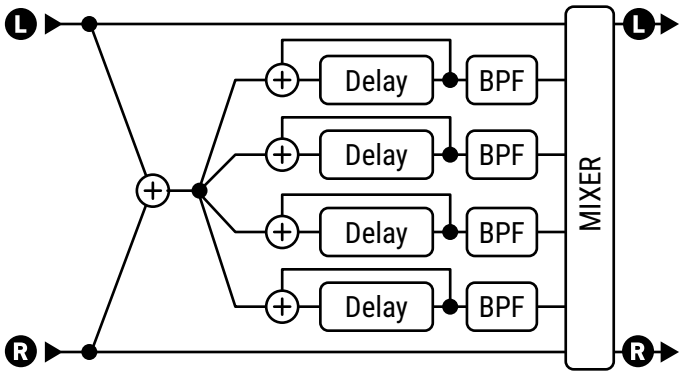
The RTA block has no channels and cannot be bypassed. Any signal at the block input is passed unmodified to the block output. In other words, sonically, it behaves exactly as a shunt.

THE RESONATOR BLOCK

2 Blocks, 2 Channels **RES**

The Resonator uses four parallel delays with feedback to create resonant comb filters. By tuning the resonant frequencies, chords and other interesting sounds can be created. It is useful for metallic droning effects, or for creating polyphonic harmony from non-melodic sources like percussion or speech. Because the delays can be extremely short, the resonator can also be used with a modifier on Master Frequency to create a "high flanger" effect. Four bandpass filters use the same frequencies as the delays. They are shown here in the "post" position but can also be used "pre".

THE RESONATOR



CONFIG PAGE PARAMETERS

Mode – The Resonator can operate in one of two modes.

- **MANUAL**: Resonator/filter frequencies are set individually as desired from 100–10,000 Hz.
- **CHORD**: Resonator/filter frequencies are set automatically based on a setting for the **Chord** parameter, which selects from various chord types (Major, Minor, etc.).

Master Freq/Frequency – In "MANUAL" Mode, this scales the frequencies set manually for the four resonators/filters. In "CHORD" Mode, this is replaced by **Frequency**, which sets the frequency for the chord root.

Master Level – Scales all the output levels.

Master Pan – Scales all the output pans. Use negative values to reverse the stereo image.

Master Feedback – Scales the feedback of all four resonators.

Master Q – Scales the Q of all four bandpass filters.

Input Mode – Selects between "MONO", where left and right input signals are summed to all four resonators (as illustrated above) and "STEREO", in which the left input channel feeds resonators 1+2 and the right feeds 3+4.

Frequency 1–4 – Sets the resonant frequency of the corresponding filter.

Feedback 1–4 – Sets the resonance of the corresponding filter by varying the feedback.

Filter Location 1–4 – Selects the position of the corresponding bandpass filter in relation to its resonator.

Filter Q 1–4 – Sets the Q of the corresponding bandpass filter.

Level 1–4 – Sets the output level of the corresponding resonator.

Pan 1–4 – Sets the pan position of the corresponding resonator.

MIX PAGE

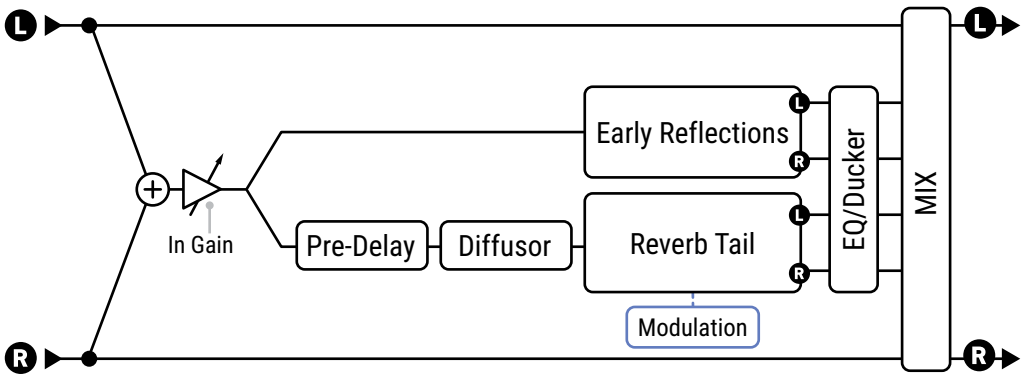
The resonator block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode**, **Input Gain** and **Global Mix** parameters.
See "Common Mix/Level Parameters" on p. 55

THE REVERB BLOCK

2 Blocks, 44 Types, 4 Channels **REV**

As an effect, reverb was one of the first. Since the dawn of music, musicians in small dead sounding places have wished they could hear the rich sounds their instruments make in a large reverberant spaces. Early simulators used metal tanks, springs or plates. Tube and solid state variants emerged, but reverb, like delay, was truly revolutionized by digital technology. Our latest reverb algorithm is at the leading edge of digital reverbs, and likely to be one of the finest you have ever experienced, with the ability to emulate real spaces, vintage plates and springs, classic digital effects, and more. The many "types" allow dialing in mix-ready sounds at the turn of a knob, while an extensive parameter list makes it easy to customize.

THE REVERB



TYPE PAGE

Type – Make a selection using the **VALUE** or **NAV** buttons with no need to press **ENTER**. There are dozens of reverb types including Halls, Chambers, Rooms, Cathedrals, Studios, Plates, Springs, and many more.

BASIC PAGE PARAMETERS

Time – Sets the decay time. This is the amount of time it takes for the reverb to vanish beyond the point of perception. This is also known as the "t60" time, referring to the amount of time required for the reverb to decay to 0.001 of its initial value (-60 dB).

Predelay – Adds extra delay before the reverb starts. The **Size** control (see below) automatically imparts a certain amount of delay before the reverb starts. Use this control to add more delay if desired. For example, if the Size is low, the reverb will start almost immediately. You can use this control to add some delay before the reverb starts but keep the small-sounding size.

Size – Sets the size of the space or spring. This controls the length of time it takes for an echo to bounce between virtual surfaces. Higher settings increase the echo time and the delay before the reverb starts.

Large values can make the reverb more "grainy" as the time between the individual repeats increases. Lower settings smooth out the reverb, but very small values will create a metallic sound.

As the size increases, the reverb will become somewhat darker as more high frequencies are absorbed.

Mix and **Level** are duplicated from the MIX page.

Crossover Frequency, **Low Frequency Time**, **High Frequency Time** – The reverb algorithm is actually multi-band, reflecting the way that sound energy dissipates differently in real spaces. This allows for very natural effects. These parameters control decay times for the two bands, plus the crossover frequency.

Mod Depth, **Mod Rate** – These parameters controls modulation in the reverb tail for a dynamic effect similar to chorusing. Modulation helps fill out the soundstage and makes the reverb sound fuller. For non-pitched instruments like drums, modulation may be undesirable. (Set Depth to zero to defeat.) To dial in modulation, turn the Mix to max, adjust Depth and Rate, and then set the Mix as desired again.

EQ PAGE

Parameters on this page color the tone of the reverb but not the dry sound. Use these settings to change how the reverb sits in the mix of your overall sound, or to produce creative effects. An on-screen display shows the EQ curve as you adjust other parameters.

Low Cut, High Cut – These define high- and lowpass filters for broad EQ adjustments.

Low Mid Frequency, Low Mid Q, Low Mid Gain and **High Mid Frequency, High Mid Q, High Mid Gain** – A pair of 2-band peaking parametric EQs allow boosting or cutting at selected frequencies. Low Mid is lower range (20 Hz–2KHz) and High Mid is higher range (100 Hz–10KHz).

MIX PAGE

The Reverb block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode, Input Gain,** and **Global Mix** parameters.
See "Common Mix/Level Parameters" on p. 55.

The **Mix** page also contains the ducking controls. Ducking causes the "wet" level to be lowered automatically when the level of your playing goes above a set threshold. Then, when you play more quietly or pause, the effect volume increases so that the echoes fill the spaces.

Ducker Atten – Attenuation sets the amount by which the effect volume will duck (decrease). A setting of 20 dB, for example, will decrease the reverb by 20 dB when the input level is above the threshold. Set to 0 to defeat the ducker.

Ducker Threshold – Sets the trigger level of the ducker. If the input signal exceeds this value, the reverb signal will be reduced by the amount set with the **Attenuation** control.

Ducker Release – Sets how long it takes for the reverb signal to return to normal when the input goes back above the threshold. A short value here causes the ducked echoes to return to full volume very quickly. Longer times cause the levels to swell back more gradually.

"ALL" PARAMETERS PAGE

In addition to the parameters listed above, the **All** page includes the following parameters:

Early Level – Adjusts the relative level of the early reflections (has no effect for the "Spring "types.)

Late Level – Adjusts the relative level of the reverb tail.

Echo Density – Controls the density of the reverb tail. Higher values give a smoother sound. Lower values allow the individual repeats to be more easily discerned. This determines the overall smoothness of the tail. Large **Size** values will also make the individual echoes more apparent, as will lower settings for the various Diffusion parameters. For legato signals, lower values may be used. For short, percussive signals, a higher value may be more desirable as the reverb tail will more smoothly fill the "space". This is a common parameter to lower to reduce CPU usage.

Early Diffusion – This sets the amount of diffusion in the early reflections. Higher values result in less distinct echoes. Lower values result in clear, distinct reflections.

Early Diffusion Time – This scales the delay time of the early reflection diffusers. Adjust this control to control the size and character of the simulated environment.

Early Decay – This parameter controls the decay rate of the early reflections. (Higher value = faster decay.)

Mic Spacing – Sets the stereo width of early reflections by simulating mic separation in the virtual space.

Late Input Mix – this routes the early reflections to the input of the late reverb "tail" generator. It uses a proprietary decorrelation technique which eliminates the metallic qualities common in other products.

Quality – There are two "Quality" options in the Reverb block: "NORMAL" and "HIGH". High quality provides the utmost in sound quality and performance. Normal quality provides excellent sound quality suitable for most guitar reverb needs, but conserves CPU power when you need it.

Hold – When Hold is activated the wet input to the block is muted and the **Time** is set to "infinity". This can be used to achieve drone or "pad" sounds. You can assign this easily to a modifier for remote control!

Wall Diffusion – Controls how quickly the reverb tail's density builds. Lower values makes discrete echoes last longer. Higher values cause the echo density to build rapidly for a more "washy" effect.

Input Diffusion, **Diffusion Time**– The reverb includes an additional diffuser at the input, which "smears" transients for a smoother overall effect. These parameters control the mix and time of this diffuser.

Stereo Spread – Varies the overall stereo image from full width (100%) to dead mono (0%).

SPRING PARAMETERS

The following three parameters only have an effect when one of the Spring reverbs is selected on the Type page.

Number Of Springs – This sets the number of springs. Using more springs produces a more dense reverb.

Spring Tone – This determines the tonal color of the virtual springs. Lower values create a darker tone.

Spring Drive – This allows overdriving the spring reverb circuit.

ALL PAGE: MODULATION SECTION

Mod Depth, **Mod Rate** – These parameters controls modulation in the reverb tail, creating a rich chorus effect.

LFO Phase – Determines how modulation is applied to the right side of the reverb with respect to the left.

ALL PAGE: DUCKER SECTION

These parameters are covered under the **Mix Page** section (previous page.)

A WORD ON "SPILLOVER"

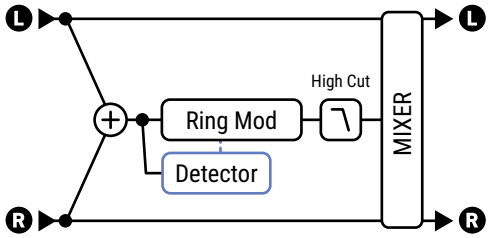
Like the delay, the Reverb block is capable of "spillover," which means that effect tails ring out when the effect is bypassed or when you change scenes or presets. For more on this subject, please see "Spillover" on p. 180.

THE RING MODULATOR BLOCK

1 Block, 2 Channels **RNG**

Ring Modulators are probably best-known for sci-fi synth and fusion electric piano sounds (or for Dalek/X-wing pilot voices), but they have been used to great effect on guitar as well. Check out the solo on *Paranoid* by Black Sabbath, *Satisfaction* by DEVO, or the *Blow by Blow* and *Wired* albums by Jeff Beck. A Ring Modulator uses fast changes in amplitude to create tones that are harmonically unrelated to the input. The Ring Modulator block goes beyond the classic effect in that the modulation frequency can track the pitch of the input signal for musically predictable results across the fingerboard.

THE RING MODULATOR



CONFIG PAGE PARAMETERS

Type – Selects between Classic and two different Single Sideband types.

Frequency – Sets the frequency of the oscillator used for amplitude modulation.

Frequency Multiplier – Scales the oscillator frequency from .25–4x.

Pitch Tracking – When this is set to "ON," the frequency of the oscillator tracks the pitch of notes played at the input of the block, allowing for a completely different effect. The **Multiplier** is still applied, allowing for tones that range from very pure-sounding to totally exterminated.

Hi Cut Frequency – Reduce this value to filter highs from the Ring Modulator sound.

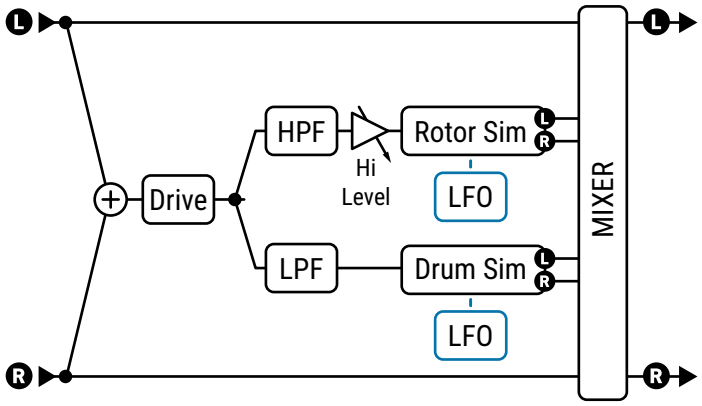
The Ring Modulator block also has the following Mix parameters: **Mix**, **Level**, **Balance**, **Bypass**, and **Bypass Mode**. See "Common Mix/Level Parameters" on p. 55.

THE ROTARY BLOCK

2 Blocks, 4 Channels **ROT**

A Hammond B3 Organ without a Leslie cabinet is like a sandwich without bread. Guitar players also revel in the wonderful, spinning, 3D sound of the Leslie and its brethren. The classic unit contains a spinning slotted drum and a rotating horn called a rotor. A low-frequency speaker is aimed into the drum while high frequencies are sent to the spinning horn. The result is unmistakable: from schmaltzy hockey-game to Steppenwolf, the rotary is ubiquitous. Drum-only rotary speakers have also been produced, with Stevie Ray Vaughan's *Cold Shot* offering an example of this type of sound. The Rotary block reproduces all these classic sounds and offers more control. Also, it doesn't weigh 300 pounds and doesn't require four guys to move up a flight of stairs.

THE ROTARY



ROTARY PARAMETERS

Rate – Controls the spin rate for both the drum and rotor. Assign a modifier for real-time control. You can even set the rate to 0.0 to "park" the rotary. To make the drum spin more slowly, as it might in a real rotary, adjust **Low Rate Multiplier** (below). When shown in parenthesis, it is being controlled by the tempo (below).

Tempo – Locks the rate in rhythmic relation to the Tempo. Set to "NONE" for manual rate control.

Drive – Give your virtual rotary speaker the grit and grind of the classic with this overdrive control.

Hi Depth – Sets the modulation depth of the rotor. To simulate a drum-only cabinet, reduce this value fully.

Hi Level – Sets the output level of the rotor. Use this to balance the level between the drum and rotor.

Low Depth – Sets the modulation depth of the drum. Higher settings provide a more pronounced throb.

Rotor Length – Adjusts the length of the virtual high-frequency horn. Larger values increase the amount of Doppler shift and result in a more intense effect.

Low Rate Multiplier – Adjusts the speed of drum compared to the rotor.

Low Time Constant, Hi Time Constant – Sets acceleration/deceleration rates for the drum (low) and rotor (hi) so they realistically "spin up" or down when you change the Rate or Tempo.

LF Mic Spacing, HF Mic Spacing – These set the placement of the (neutral-sounding) virtual mics, determining the stereo width of the effect. Setting zero (default) simulates a mono mic on the drum.

Mic Distance – These set the distance of the virtual mics.

Stereo Spread – Controls stereo width from full stereo (100%) to true mono (0%). (You'd be surprised at how many rotary recordings are actually mono!)

The Rotary block also has **Mix**, **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See "Common Mix/Level Parameters" on p. 55.

THE SCENE MIDI BLOCK

1 Block, 0 Channels **MIDI**

The Scene MIDI block is unlike other blocks in that instead of processing audio, it transmits MIDI messages through the **MIDI OUT** port of the Axe-Fx III. It is set up through a single page with no types or channels. It must be manually placed on the grid for per-Scene MIDI messages to be transmitted at all. It cannot be bypassed, but any scene can be set up to transmit no messages.

Once the Scene MIDI block has been placed on the grid, press **EDIT** to show its menu page.

- The **current Scene number** is shown at the top of the display.
- The eight table rows represent **eight different messages** that can be sent for the current scene.
- The four table columns are used to program MIDI messages. For each message, you can select:
  - **Type**: Determines whether the message will be a "PC" (Program Change) or a "CC" (Control Change).
  - **Channel**: Sets the MIDI Channel for the selected message.
  - **Number**: Sets the CC number, if the message is a Control Change.
  - **Value**: Sets the actual PC or CC message data.
- Beneath the table, buttons for **Scene Down** (-) and **Scene Up** (+) change the scene.
- The **Clear Row** button clears all parameter values for the current row in the current scene.
- The **Test Row** block allows you to transmit a single row for testing purposes.

Several examples of Scene MIDI block programming appear below. Unused commands are not shown.

EXAMPLE 1:

In this example, Scenes 1 and 2 each transmit two different PCs to devices on channels 3 and 4.

Scene: 1

	TYPE	CHANNEL	NUMBER	VALUE
Command 1	PC	3	--	11
Command 2	PC	4	--	64

Scene: 2

	TYPE	CHANNEL	NUMBER	VALUE
Command 1	PC	3	--	14
Command 2	PC	4	--	59

EXAMPLE 2:

In this example, Scenes 1 and 2 each transmit two different CCs to devices on channels 3 and 4.
Imagining two toggle switches, the values swap from one scene to the next.

Scene: 1

	TYPE	CHANNEL	NUMBER	VALUE
Command 1	CC	3	0	80
Command 2	CC	4	127	81

Scene: 2

	TYPE	CHANNEL	NUMBER	VALUE
Command 1	CC	3	127	80
Command 2	CC	4	0	81

EXAMPLE 3:

In this example, Scenes 1 transmits a CC and a PC to a device on Channel 9.
Scene 2 transmits nothing to the device on Channel 9, but sends different messages to three other devices.

Scene: 1

	TYPE	CHANNEL	NUMBER	VALUE
Command 1	CC	9	0	1
Command 2	PC	9	--	21
Command 3	--	--	--	--

Scene: 2

	TYPE	CHANNEL	NUMBER	VALUE
Command 1	PC	5	--	100
Command 2	PC	6	--	19
Command 3	CC	7	16	63

THE SEND BLOCK

2 Blocks, 0 Channels **SND**

The Send block passes audio to the Return block (see below). On the grid, the Send block behaves exactly like a Shunt. It does not process audio and cannot be bypassed. It has no parameters.

THE RETURN BLOCK

2 Blocks, 0 Channels **RTN**

The Return block receives audio from the Send block. Send and Return blocks must be used in pairs. There are two pairs of Send and Return blocks available. Receive 1 receives audio only from Send 1, while Receive 2 receives audio only from Send 2.

Send and Return blocks allow you to route signal from any point to any other point, bending the rule that signal must only flow from left to right. No connection will be visible between the two blocks, but signal will still flow from the output of the Send to the input of the Return. Both blocks must be used for either to function.

Send and Return blocks have two main uses: (1) creating feedback loops and (2) extending the length of effects chains beyond the size of the grid.

PARAMETERS

Return Level – Scales the level of the signal from the Send block that is passed to the grid by the Return block.

WARNING: Use these blocks with caution, as you can easily program an unstable loop and cause internal clipping and/or dangerously high sound levels. Begin with the Return Level at 0% and bring it up slowly. If you start to hear squealing or runaway feedback, immediately bring the value to minimum and analyze your routing for possible causes of instability.

Output Level – This adds an additional boost or cut to the signal received from the Send block.

This block has **Balance**, **Bypass**, and **Bypass Mode** parameters. See “Common Mix/Level Parameters” on p. 55.

FEEDBACK LOOPS

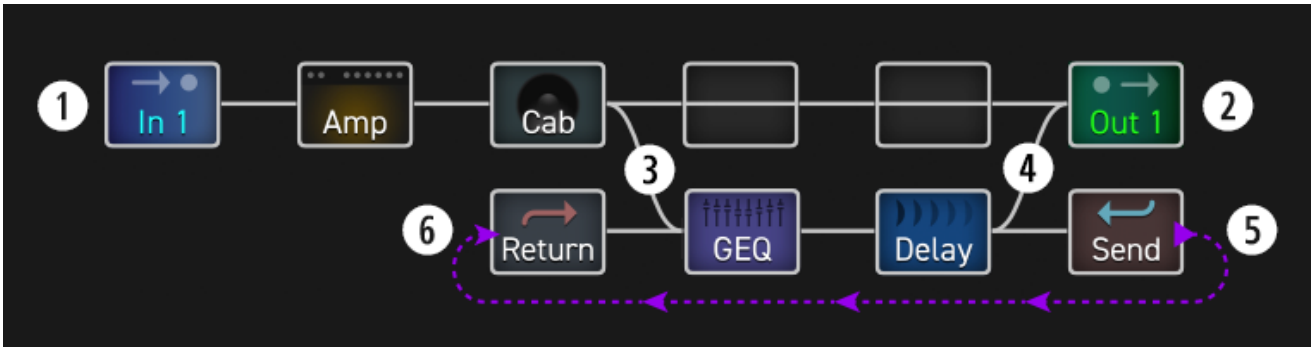
Feedback loops allow you to combine effects in interesting ways and then route the output back to the input. The figure below demonstrates this by using an overlay of a preset illustration rendered in Axe-Edit.

Signal enters the grid at (1), passes through the Amp and Cab blocks, and reaches the output (2), where it is heard in the speakers as you play. Meanwhile, a tap of this main line at (3) carries signal to a Graphic EQ, and a Delay set to 500 ms, with no feedback and mix at 100%. Because of the mix setting, nothing comes out of the delay for 500 ms. Then, the first echo is routed by a cable (4) to the output (2). This echo *simultaneously* enters the Send block (5) and is routed to the Return (6). From here it again passes through the GEQ—and hits the delay again. Each previous echo that cycles through has one extra pass of any other effects in the loop. The purple dotted line shows the imagined path that signal takes between Send and Return.

This feedback loop of **GEQ ▸ Delay ▸ Send ▸ Return** would continue forever if the **Return Level** parameter didn't make it a bit quieter each time so that it eventually fades away.

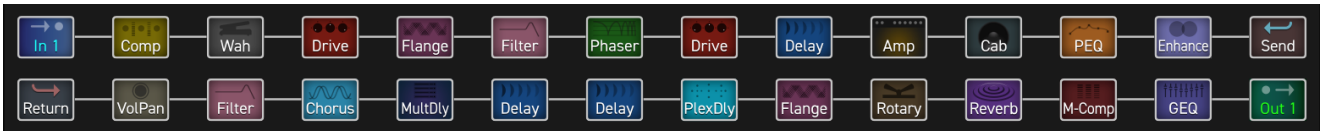
So to review, the **Send** "transports" signal to the **Return**, where **Return Level** controls the amount of feedback.

Countless variations of Send/Return loop presets are possible when you use different effects, vary their order, or enter and tap the loop in different places.



EXTENDED LAYOUTS

The 14x6 grid will suffice for the vast majority of long, complex routings. However, sometimes the "need to exceed" means building an effects chain that is longer than the number of columns available in the grid. The Send and Return blocks can be used for exactly this purpose. Place a Send block at the end of your first chain and place a Return block at the beginning of another, setting Return Level to "100%". Continue through other effects to an output as illustrated below. By the way, this real preset with 28 blocks hits just over 80% CPU usage!



THE SYNTH BLOCK

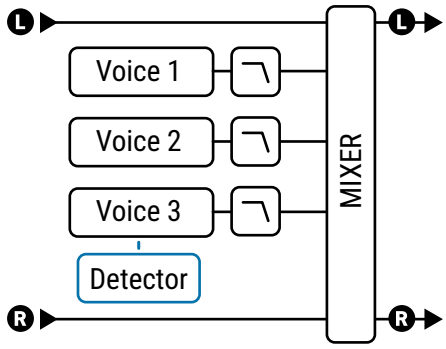
2 Blocks, 4 Channels **SYN**

The Synth block is a 3-oscillator monophonic synthesizer that can be used for a fantastic range of monophonic guitar synth sounds, special effects, and as a tone generator for testing purposes.

A monophonic synthesizer tracks single note playing.

Modifiers and controllers may be used to affect a variety of synth functions for real time effects.

THE SYNTH BLOCK



SYNTH PARAMETERS

Each of the three synth voices has its own oscillator and filter with the following parameters:

Type – Sets the oscillator waveform to Sine, Triangle, Square, Sawtooth, Random, White Noise, Pink Noise, or OFF. (Setting a synth voice to "OFF" helps conserve CPU).

Track – Selects the type of input tracking.

- **OFF** – Allows the frequency and level to be set manually via the FREQ and LEVEL controls.
- **ENV ONLY** – Selects the level to be controlled by the envelope while frequency is set manually.
- **PITCH+ENV** – Selects the frequency and level to be controlled by the pitch and envelope of the input.

Frequency – If input tracking is set to "OFF" or "ENV ONLY," this parameter sets the oscillator frequency.

Shift – Shifts the frequency of the oscillator up or down in semitone steps.

Tune – Detunes the oscillator slightly. The oscillator can be detuned +/- 50 cents.

Duty Cycle – When using the TRIANGLE or SQUARE waveforms, this parameter controls the symmetry or pulse width of the waveform.

Pan – Controls the panning of the oscillator.

Filter Frequency – Sets the cutoff frequency of a low-pass filter after the oscillator.

Filter Q – Sets the Q or resonance of the post-oscillator filter.

Attack – Sets the attack time of the envelope follower on the input.

Level – Each voice also has its own level control, located on the Mix page.

The Synth block also has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode** and **Global Mix** parameters.
See "Common Mix/Level Parameters" on p. 55.

THE TEN-TAP DELAY

2 Blocks, 2 Types, 4 Channels **TTD**

The Ten-Tap Delay is fairly self-explanatory, but has a number of features which allow it to be used for some very creative effects. Echoes sweep across the stereo field or play in unique rhythms with this exciting block.

TYPE PAGE

Type – Select a type using the **VALUE** or **NAV** buttons with no need to press **ENTER**.

- **Ten-Tap Delay** – The Ten-Tap Delay provides a unique way to control the time, pan, and spacing of one to ten separate echoes. Instead of feedback, it uses an innovative **Decay** control to determine how the level of the ten taps changes over time. The levels of individual delay taps can also be adjusted from -80 to +20 dB. Pan is set as a **Shape** that can change automatically as the taps progress.
- **Rhythm Tap Delay** – The Rhythm Tap Delay uses the same algorithm as the Ten-Tap Delay but allows you to create a custom rhythm of repeats. You can enter the rhythm in three ways:
  1. By specifying the number of milliseconds between each tap and the previous one.
  2. By specifying some number of quantized time units ("divs") between each tap and the previous one.
  3. By using the **LEARN** function to tap a rhythm with the **ENTER** button.

CONFIG PAGE

Ten-Tap Delay and Common Parameters

Mono/Stereo – Sets the mode of the Ten-Tap Delay. In mono mode, twice as much delay per tap is possible.

Delay Time – Sets the time between delay taps. When shown in parentheses, it is being set automatically by **Delay Tempo** (below). Set **Delay Tempo** (below) to "NONE" for manual control.

Delay Tempo – Sets the Delay Time in rhythmic relation to the Tempo.

Number Of Taps – Sets the exact number of repeats.

Decay – Sets how rapidly the volumes of the repeats decay over time. Remember that each tap also has an individual **Level** control which can be instead of **Decay** — or alongside it — to create custom patterns of level. The following diagram illustrates the effect of decay. The first bold line is the original signal.



Number of Taps: 10
Decay: 0.0%

Number of Taps: 10
Decay: 95.0%

Taps: 3
Decay: 0.0%

Taps: 3
Decay: 95.0%

Shuffle – Sets the amount of time-offset for the odd-numbered taps to give a shuffle feel to the repeats.

Spread – In Stereo mode, this sets the spread of the repeats. At maximum, the left channel is panned fully left and the right channel fully right.

Ratio – In Stereo mode, adjusts the left time as a percentage of the right.

Pan Shape – Controls the shape of the panning as a function of the tap number. The repeats can move slowly from one side to the other ("increasing" or "decreasing"), stay "constant," or move back and forth ("sine").

Pan Alpha – Controls how quickly the repeats move as a function of tap number and pan shape. Higher values produce a more pronounced effect. To alternate left-to-right, set Shape to "SINE" and Pan Alpha to maximum.

Low Cut, High Cut – These set the cutoff frequencies of high- and lowpass filters to adjust effect tone.

Tap Level 1–10 – Adjusts the relative level of the selected tap from -80 to +20 dB.

Rhythm Tap Delay Parameters

The parameters of the Rhythm Tap delay include those of the Ten-Tap delay except Delay Time/Tempo and Shuffle. It has the following additional parameters for creating unique rhythmic echo patterns:

Feedback – Sets the feedback level from the final repeat to the input of the delay line. This is just like feedback in a regular Delay block, except instead of determining how many times a single echo will repeat, this is how much the entire pattern will repeat.

Quantize – Changes **Tap Time** units (see below) from milliseconds to rhythmic subdivisions, or "Divs". The value of Quantize is the length of each div.

Learn – Use this function to enter a rhythm by tapping the **ENTER** button.

1. **NAV** to the **Learn** parameter.
2. Change the value of **Learn** to "TAP ENTER".
3. Tap the rhythm you want using the **ENTER** button. Your first tap represents the original (dry) signal.
4. If **Quantize** is "NONE", tap times will be set in milliseconds. Otherwise, each tap will be rounded to the nearest rhythmic subdivision. With Quantize turned ON, it is helpful if you use a metronome to guide you.
5. When finished tapping, change the value of **Learn** to "DONE".

Tap Time 1–10 – These ten parameters sets the time of each tap relative to the one before it. Times are set in milliseconds or divisions ("divs"). The length of each div is whatever value you set for **Quantize** (and re-calculated based on the Tempo). For example, if Quantize is set to "1/16", each div is 1/16th note, and all tap times will sound some whole number of 16ths after the preceding tap. If Quantize is "OFF", you can enter millisecond values directly or use the Learn feature (above). Times that have been learned may be further adjusted manually.

MIX PAGE

This block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode**, **Input Gain** and **Global Mix** parameters.
See "Common Mix/Level Parameters" on p. 55.

THE TONE MATCH BLOCK

1 Blocks, 0 Channels **TMA**

The Tone Match block changes the sound of a preset to match a reference signal, such as a recording or the feed from a mic on a live amp. The result can be saved into a preset, or exported as a User Cab for use in other presets. This block is detailed in a separate “Tone Match Mini Manual” available at:

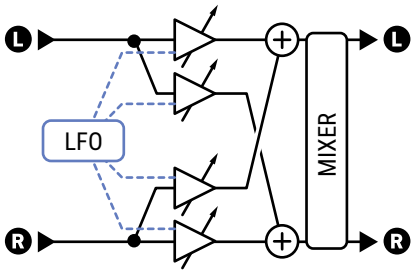
https://www.fractalaudio.com/downloads/manuals/axe-fx-2/Axe-Fx-II-Tone-Match-Manual.pdf

THE TREMOLO/PANNER BLOCK

2 Blocks, 2 Types, 4 Channels **TRM**

As its name suggests, the Tremolo/Panner block has two uses. A **tremolo** effect turns the overall volume up and down in a pulsing rhythmic way. You can hear tremolo sounds on *Gimme Shelter* by the Rolling Stones, *Boulevard of Broken Dreams* by Green Day and at the end of *Rumble* by Link Wray. Like a tremolo, a **panner** also varies volume, but when the left and right channels are changed in opposite directions, the "auto-pan" effect creates the illusion of motion in the stereo field. Panning covers anything from slow subtle swings to psychotic gyrations. Countless recordings use pan controls at the console to create dimensional stereo effects. Listen (in headphones, even) to *EXP* on *Axis: Bold as Love* by Jimi Hendrix, the middle section of *Whole Lotta Love* by Led Zeppelin, or the guitar solo in *Owner of a Lonely Heart* by Yes.

THE TREMOLO/PANNER



TREMOLO/PANNER PARAMETERS

Type – Chooses between **Tremolo** and **Panner** effects.

Rate – Sets the frequency of the LFO that controls the effect. Set fully counterclockwise to sync to the **LFO1 controller** (p. 149). When shown in parentheses, rate is being set automatically by the **Tempo** parameter (see below). Set Tempo to "NONE" for manual control.

Tempo – Synchronizes the rate of the LFO in rhythmic relation to the Tempo. For example, if the Tempo is 120 BPM, and Tempo is set to a "1/4," the LFO rate will be 2 Hz (120 BPM / 60 seconds = 2).

Depth/Width – Sets the depth of the LFO that controls the effect. Set higher for more dramatic effects. In the panner type, when **Width** is set to values above 100%, psychoacoustic effects create the illusion of panning beyond the boundaries of the normal stereo image.

LFO Type – Sets the shape of the modulation waveform.

LFO Duty Cycle – Sets the symmetry of the modulation waveform.

LFO Phase – Adjusts the phase difference between the left and right LFO waveforms. Use 0° for a true Tremolo, or 180° for perfect panning.

See also "LFO Waveforms & Phase" on p. 181.

Start Phase – (Tremolo only) This allows you to force the effect to start from a fixed point in its cycle when you engage the effect.

Pan Center – (Panner only) this shifts the apparent center of the stereo image.

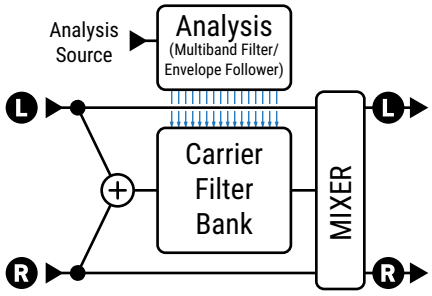
MIX PAGE

The Tremolo/Panner block has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See "Common Mix/Level Parameters" on p. 55.

THE VOCODER BLOCK

1 Blocks, 1 Channel **VOC**

The Vocoder block is a digital recreation of the classic analog effect. The vocoder, created by Homer Dudley, was originally designed as means of compressing human speech for transmission over narrow-band carrier channels. In the 1970s, Robert Moog and Wendy Carlos pioneered the use of the vocoder for musical applications. This block pays faithful homage to those early analog vocoders with up to 24 bands and a "constant-Q" approach. It can make your guitar "talk" but can also be used with the Synth block to create robot voices and other effects.



When using the vocoder with your guitar or another instrument, resist the urge to sing along. Speaking in a clear monotone voice yields the best results.

VOCODER PARAMETERS

Analysis Source – This selects the input to provide the analysis or modulator signal. This is typically used for vocal input, while the "carrier" or "synthesis" signal is derived from the block's input on the grid.

Analysis Channel – This selects whether the left or right channel is used for analysis.

Number Of Bands – Selects the number of bands to use in the analysis and synthesis filter banks.

Min Frequency – Sets the frequency of the lowest filter band.

Max Frequency – Sets the frequency of the highest filter band.

Filter Resonance – Sets the Q, or bandwidth, of the filters. Higher values yield narrower filters.

Frequency Shift – Shifts the frequency of the carrier/synthesis bands relative to the analysis bands. This allows you to change the character of the vocoding to produce "anonymous mob informant" or "chipmunk" effects.

Freeze – This locks the output of the envelope followers to hold the vocal formant.

Highpass Mix – This sets the amount of high-pass filtered signal to mix in with the output. This can be used to increase the intelligibility of the vocoding by allowing certain consonants and air sounds to pass right through.

Attack – Sets attack filter time for the envelope followers.

Decay – Sets release filter time for the envelope followers.

Master Level – Sets the master level for all the synthesis filter outputs.

Master Pan – Sets master panning for all the synthesis filter outputs. Individual control of the filter output levels and panning is provided on dedicated menu pages. You can use these controls to fine-tune the filter bank response and control the panning of the filter outputs.

Level 1–24 – These set the out level for each of the 16 bands.

Pan 1–24 – These set the out pan for each of the 16 bands.

MIX PAGE

This block has **Mix**, **Level**, **Balance**, **Bypass**, **Bypass Mode**, **Input Gain** and **Global Mix** parameters.
See "Common Mix/Level Parameters" on p. 55.

THE VOLUME/PAN BLOCK

4 Blocks, 4 Channels **VOL**

The Volume/Pan block can be used as a dynamic control with an external expression pedal, but is also very useful as a static boost or cut, to mute the signal, pan, or isolate left/right channels.

Type – Selects between a simple volume control and “Auto-Swell” based on the rare vintage “slow gear” effect, which replicates the sound of a guitarist rolling their volume knob up before the attack of each note.

Volume – Scales the output level out of the block. Assign a modifier to create a volume pedal.

Threshold and **Attack** – Use these to set the threshold and volume “roll up” speed when the type is “Auto-Swell”

Balance – Sets the balance (L/R) out of the block.

Taper – Sets the “taper” of the volume control. “LINEAR” selects a linear taper. Log 30A, 20A, 15A, 10A and 5A select various types of potentiometer typically used for volume control.

Input Select –This control determines how incoming stereo signals will be passed. Options include STEREO, LEFT ONLY, or RIGHT ONLY.

Pan Left, Pan Right – Controls the panning of the left and right output signals.

Bypass Mode – Determines whether input signals should pass through or be muted when this block is bypassed.

Level – Sets the output level of the block independent of the setting for the **Volume** control.

THE WAHWAH BLOCK

2 Blocks, 8 Types, 4 Channels **WAH**

Country guitar virtuoso Chet Atkins created his own Wah-wah pedal in the late 1950s but the effect really took off after Clapton used it on *Tales of Brave Ulysses* with Cream in 1967. From Jimi Hendrix's *Voodoo Chile* to, well, Stevie Ray Vaughan's *Voodoo Chile*—and right up to the present— the wah pedal, with its onomatopoetic name, remains one of the most cherished and recognizable guitar effect sounds of all time.

The Wah block is the embodiment of this great legacy but with modern reliability and control and a smooth feel without the problems that can plague the electromechanical version. Place a wah before distortion for a classic sound or after distortion for more of a synth-like sound. The effect is stereo-in/stereo-out.

PARAMETERS

Type – Selects between different wah types based on classic vintage and cutting edge modern designs.

Type	Based On
FAS Wah	A custom Bandpass wah by Fractal Audio Systems.
Clyde	Based on an original Vox Clyde McCoy wah.
Cry Babe	Based on a Dunlop Cry Baby.
VX846	Based on a Vox V846-HW hand-wired wah.
Color-Tone	Based on a Colorsound wah.
Funk Wah	Modeled after the "Shaft" sound.
Mortal	Based on a Morley wah/volume pedal.
VX845	Based on a Vox V845.

Wah Control – Sets the position of the wah. Normally you would assign this parameter to a pedal for real-time control, but you can also set it to an LFO or "park" it manually (so you can play the guitar on the MTV).

Control Taper – Defines the sweep or feel of the wah with selections representing various potentiometer tapers.

Resonance Q – Sets the resonance ("Q") of the filter. Higher values give a more pronounced response.

Q Tracking – Some classic wah pedals become less resonant as you push the pedal down. This control mimics that effect, so that as frequency is increased, the resonance decreases, becoming less "peaky" and harsh.

Low Cut Frequency – Applies a highpass filter, as created by the internal coupling capacitor of a real wah pedal.

Minimum Frequency – Sets the frequency of the filter when the frequency control is at its lowest value. This can be adjusted to match the range of your instrument or preference.

Maximum Frequency – Sets the frequency of the filter when the frequency control is at its highest value.

Drive – This simulates overdriving the circuit of the wah pedal.

Inductor Bias – This adjusts the DC offset of the virtual inductor which interacts with the **Drive** parameter to replicate the subtle (and awesome) sound of some of the most coveted wah pedals.

Fat – This is actually a mix control, adding dry signal to the wah for up to 50/50 wet/dry mix.

EQ PAGE

A built-in 8-band, 2/3 octave Graphic EQ can be enabled/disabled via the **EQ ON/OFF** parameter on the Mix page.

MIX PAGE

The Wah block has **Level**, **Balance**, **Bypass**, and **Bypass Mode** parameters.
See "Common Mix/Level Parameters" on p. 55.

GLOBAL BLOCKS

Global Blocks provide a convenient way to synchronize parameter changes across multiple presets.

If you use multiple presets and want to keep certain sound settings the same in all of them, you might imagine needing to repeat those changes into every preset, one by one. Another idea might be to use Axe-Edit to copy/paste the desired blocks from one preset to another. Global Blocks eliminate these the need for such tedious, repetitive tasks. Any change to any Global Block in one preset is made automatically across all other presets where that Global Block is used.

Each block type that supports Global Blocks is backed with eight Global Blocks of its own. For example, there are eight DIFFERENT Global “Amp 1” blocks and an additional eight DIFFERENT Global “Amp 2” blocks. Global blocks are stored outside the preset area in “System Memory.” All parameters in all channels can be governed by Global Blocks. All modifiers, however are stored in the preset itself and are NOT part of the Global Block.

GLOBAL BLOCKS SUPPORT

Most of the 40+ block types support the use of global blocks. (The following are NOT included: Crossover, FB Send/Return, IR Player, Looper, Megatap, MIDI, Mixer, Multiband Comp, Multiplexer, RTA, Resonator, Synth, 10-tap delay, Vocoder.)

GLOBAL BLOCKS EXAMPLE

Imagine a set of five different presets used for a live rig. Each of them has one amp block (AMP 1) with the same settings, but this amp is surrounded by totally different effect chains. Now, depending on the venue, your guitar, the backing band, the audience, the weather, you name it... You may want to make changes to your amp block, just as you would any physical amp. Turn up the treble. Turn down the gain. Adjust the Graphic EQ. Using the simple operations described on the next page, it becomes easy to link the amp blocks in these presets together, so modifying any one of them changes all of them.

First, you would open one of the presets where the amp is dialed in nicely. Now, save that Amp’s settings as a Global Block by selecting Amp 1 and choosing LINK TO GLOBAL BLOCK. Let’s say we choose “Global Block 1”. Don’t forget to save the preset!

Now, open each of the remaining presets, select AMP 1, and choose LINK TO AND LOAD FROM GLOBAL BLOCK, selecting “Global Block 1”. The settings from Amp 1: Global Block 1 appear automatically, and the link is created. Don’t forget to save the preset! Repeat as needed for other presets.

Other applications for Global Blocks include using them to change gate threshold settings as you change from an active guitar or bass to a passive one, globally adjusting the level of a volume block used as a “boost”, dialing in individual effects that are used with different amps in different presets, and much more.

Global Block Operations

Four different operations enable presets to interact with Global Blocks. You can find these operations on the **Tools** page of the **Layout** menu (from the Grid, page right to "Tools"). To execute a function, select the desired grid block using the **NAV** buttons, use the **VALUE** wheel to choose the desired Global Block function, turning the **E** knob if indicated to select the desired Global Block number, and press **ENTER** or the "Execute" button (Push-knob **C**).

Using Axe-Edit makes working with Global Blocks even easier, and also provides a high level view of where they are used in your preset.

LINK TO GLOBAL BLOCK

The **Link to Global Block** function "creates" a Global Block from a block in your preset, and then creates a link between the two. This is always the first step when you want to begin using Global Blocks. Any existing settings in the selected Global Block is over-written with the settings from the block in your preset. After a link is created, any time you save changes, the Global Block is updated. Any time another linked preset is changed and saved, this preset will be updated.

Always save your preset immediately after creating a link to a Global Block.

LOAD FROM AND LINK TO GLOBAL BLOCK

The **Load from and Link to Global Block** function loads existing Global Block data into the selected block in your preset, and *then* creates a link between the two. You would normally use this function to link a set of presets to a Global Block that you first created using the **Link to Global Block** function (above). After a link is created, any time you save changes, the Global Block is updated. Any time another linked preset is changed and saved, this preset will be updated.

Again, always save your preset immediately after creating a link to a Global Block!

LOAD FROM GLOBAL BLOCK

The **Load from Global Block** function loads existing Global Block data into the selected block in your preset, but does NOT create a link between the two. This allows you to use Global Block data as a "template" for populating local block settings without the usual link.

UNLINK FROM GLOBAL BLOCK.

This operation totally disconnects the selected block from any Global Blocks.

9 MODIFIERS

Modifiers allow sound parameters to be remote-controlled or automated in real time. For example, you might use an expression pedal to operate a wah or whammy, or use an LFO controller (Low Frequency Oscillator) to sweep a filter. MIDI can also be used to adjust effects, control volume, and more. Modifiers are easy to use, but provide incredible depth of power for those who like to push the limits.

CREATING A MODIFIER

The process of creating a modifier begins at the parameter you want to control. Parameters that can be controlled are marked with a circular symbol, shown below on a knob and a switch. When a modifier is already present, the circle will be solid yellow, like an LED that has been turned ON. You will see the same symbol whether the target parameter is on a knob page, on a text page, or any other type of menu.



The open circle means you can add a modifier here.

The filled circle means a modifier is already present.

Here, a switch instead of a knob allows a modifier.

This parameter has no option for a modifier.

TO CREATE A MODIFIER...

- Select any parameter that supports a modifier. ◯
- Press the **ENTER** button or press the **MODIFIER** Push-knob to show the **Modifier** menu.
- On the **Source** page, select a **SOURCE** to control the parameter. Learn more about Sources on p. 145.
- Press **PAGE RIGHT** to switch to the **Modify** page.
- Learn more about Modifier parameters on p. 146. At a minimum, check that **MIN** and **MAX** are set to the lowest and highest values you want to hear as the source changes.
- Press **EXIT** to leave the modifier menu.
- Modifiers animate on-screen knobs, faders switches and graphs. For text-only menu parameters, a bar graph shows the value of the source.
- Modifier changes are saved when you **STORE** the preset.

TO EDIT OR REMOVE A MODIFIER...

- To edit an existing modifier, use the same process as creating a modifier.
- To remove a modifier, change its **SOURCE** to “NONE”.

MODIFIER TUTORIAL: WAH PEDAL

Below you will find a basic step-by-step tutorial on setting up the modifier for a Wah pedal.

For this example, let's assume:

- A **Wah block** is in your preset.
- You have an **Expression pedal** ready (see p. 12 for how to connect pedals directly to the Axe-Fx III).

Here are the steps:



1. On the Grid, **NAV** to your Wah block and press **EDIT** to show its menu.
2. Turn to the **Config** page of the Wah block's Edit menu.
3. Notice the yellow Modifier dot symbol beneath the on-screen **Wah Control** parameter.
4. **NAV** to select that parameter (or just give its knob a twiddle). It will be highlighted when selected.
5. Press **ENTER** or the **MODIFIER** Push-knob to show the **Modifier** menu.
6. On the **Source** page, change the **SOURCE** to your pedal. Depending on how your pedal is connected this might be "Pedal 1" (the on-board expression jack of the Axe-Fx III), or it might be one of the "External Controllers" (p. 150), or it might be one of the "FC" pedal options.
7. Test! The Wah should work when you move the pedal. You'll also see the dot move on the graph. If it doesn't work, check your source or make sure the pedal is properly connected and calibrated.
8. Save the preset by pressing **STORE**, **ENTER**, **ENTER**.

The Wah in this example is controlled by an expression pedal. An expression pedal can be used to control other parameters like whammy control, rotary rate, delay feedback, and many, many more.

There are also many different sources we could use instead of a pedal. Try an LFO for a modulating auto-wah, or an Envelope Follower for some funky Mu-Tron action. And MIDI opens up a world of controller options.

MODIFIER TIPS AND TRICKS

- The same source can be assigned to multiple modifiers at the same time. For example, one pedal might adjust Chorus Rate and Chorus Depth at the same time — even in different ways.
- The modifiers for a given block will normally be shared across all of its channels, but you can also limit a modifier so it applies only to any one channel—your choice. Change the "Channel" setting inside the modifier menu to do so.
- Modifiers use a very small amount of CPU power while you use them. Test any presets which seem close to the edge.
- All of the Internal Controllers (Tempo, LFOs, ADSRs, etc.) are part of a **Controllers** block. This block has four channels, so you can have up to four different sets of Controller values in one preset.
- The Axe-Fx III has a dedicated **Bypass** parameter in every block that can be bypassed, and this parameter allows you to assign a modifier. *Bypass Mode is not used as it was on previous Fractal Audio products.*

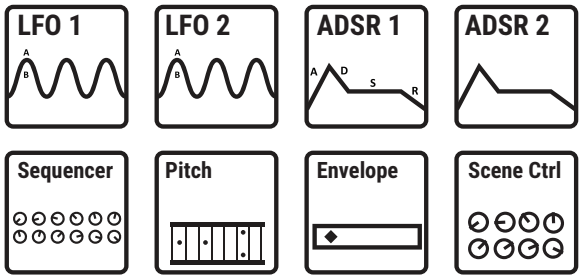
MODIFIER SOURCES OVERVIEW

Our previous example used a pedal, but there are in fact over 60 different choices to use for modifier sources.

INTERNAL CONTROLLERS

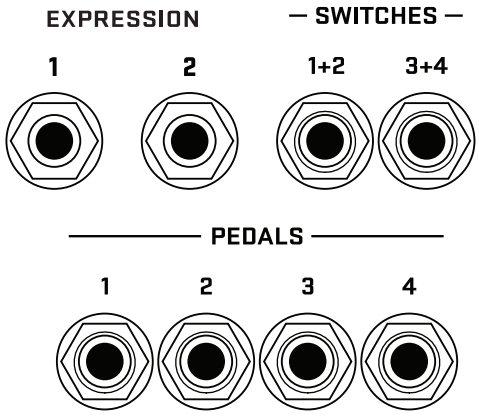
Internal controllers are built into the Axe-Fx III and can be programmed in every preset. You will find one **Tempo**, two **LFOs** (Low Frequency Oscillators), two **ADSRs**, a **Sequencer**, a **Pitch Detector**, an **Envelope Follower**, and four **Scene Controllers**. See p. 149 for more.

To show the **Controllers** menu, press the **CONTROLLERS** button on the **Home** page (Push-knob **D**) or **Layout** menu (**E**), or press the **TEMPO** button once.



ONBOARD & FC PEDALS & SWITCHES

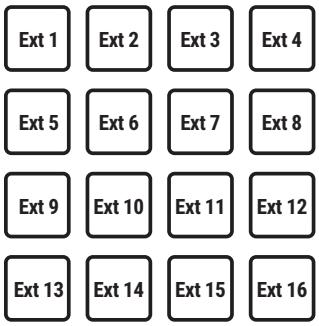
The onboard **Expression** jacks and the **Pedal** or **Switch** jacks of a connected **FC** series controller can be assigned directly as modifier sources. In comparison to previous products, it is no longer necessary to assign these to an **External Controller** first (though this is still possible).



EXTERNAL CONTROLLERS

External controllers are modifier sources which in turn have their own custom global assignments to determine *what controls them*. External Controllers are a great choice for presets you might share, because different people will use different controllers in different setups.

For example: you send your friend a preset containing a Wah operated by **External 1**. On your system, External 1 is globally assigned to “PEDAL 1”— one of the onboard expression jacks. On your friend’s system, External 1 is assigned to “CC#16” because an older MIDI controller is being used. The Wah works perfectly on both rigs with no changes required!



The Axe-Fx III contains 16 external controllers.
Each one can be assigned to your choice of:

- Any **MIDI Control Change** (CC#) message.
- One of the onboard **Expression** (Pedal) jacks.
- One of the **Pedal** or **Switch** jacks of a connected FC series controller.

Learn more about External controllers on p. 150.

MODIFIER PARAMETERS

The default modifier settings are linear. You change the source and the parameter follows in direct relationship. Modifier parameters allow you to change this relationship. You might tune the taper of a pedal, or make it work in reverse. Maybe you want the parameter attached to a footswitch to glide instead of snap. Here is an overview of the additional parameters you can use to create these effects and many more. Find these options on the **Modify** page when you set up any modifier.

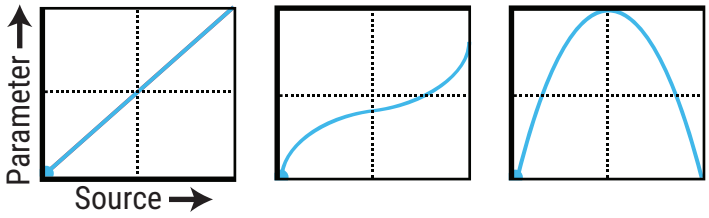
Min and **Max** determine the range of parameter change. These are extremely important parameters.

> *EXAMPLE: The modifier for a pedal controlling* **Delay Feedback** *has Min at "10%" and Max at "50%". The feedback sweeps only from 10% to 50 as the pedal is moved, even though this parameter's actual range is from -100% to +100%.*

Start, **Mid**, **End**, **Slope**, **Scale**, and **Offset** are used to create custom curves which re-map the relationship between the modifier source and the value of the target parameter. On the Modify page, a **graph** shows the relationship between the source (x-axis) and the parameter (y-axis). A dot on the graph tracks the source.

The default settings (first graph below) create a 1:1 linear relationship (the blue line) between source and parameter. As the source changes, the parameter tracks it directly.

The second two graphs below show examples of the kinds of non-linear curves you can create using Slope, Scale, and Offset. The pages which follow contain more examples.



Attack and Release: These values apply "damping" to slow the rate at which the target parameter value chases the source. **Attack** determines the rate of change as the source is increased, and **Release** controls the rate as it decreases. At low settings, these add just a little smoothing. Try settings of about 5 ms to "relax" a pedal or to ease the edges of a square LFO to eliminate clicks and pops. Higher settings can cause sound changes to be extremely slow and lazy.

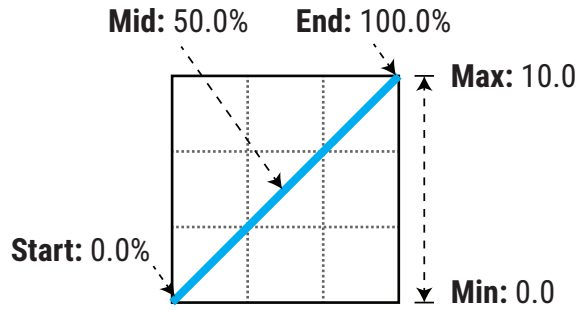
Auto-Engage works with **Off Value** to create effects which turn ON or OFF automatically whenever the source controller is moved. This is typically used with a Wah pedal so you don't need a toe switch. (Follow the example on p. 144 and set AUTO-ENGAGE to "SLOW POS" to try it out!) Find additional information on **Auto-Engage** on p. 148.

PC Reset sets the value of an external source when a preset first loads. This allows you to override the actual position of an external controller until it is moved or updated. Here's how it works: after applying the modifier, exit to the parameter again on its Edit page. Notice that the value of that parameter can be edited as usual. If **PC Reset** is ON, the value you set and then save will be used from preset load until the source is changed (i.e. the pedal has been moved).

Update Rate controls how often the modifier is refreshed. The setting of slow is actually very fast, and fine for the vast majority of applications. The faster settings require additional CPU but provide even smoother sound performance when ultra fast changes are required (while using a fast LFO for instance). Check this setting if you think you hear "zipper noise" while a modifier is in use.

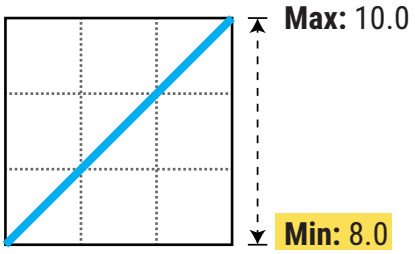
UNDERSTANDING MODIFIER MIN AND MAX

MIN and **MAX** set the range of a modifier. This lets you fine-tune modifiers in very cool ways. In the example below, imagine a volume pedal with different settings. Using the principles on this page, you should be able to achieve a wide range of modifier control scenarios.

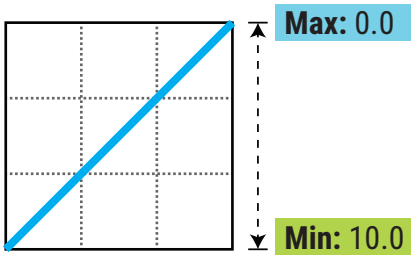


Here is your basic "vanilla" volume pedal.

The pedal "start" (heel position) sets the volume to Min: 0.00; silent. The pedal "end" (toe down position) has the volume at Max: 10 or all the way up.
As you rock the pedal forward, the dot follows the line. It moves up as it goes and the volume does too.



Now, something different: a "boost" pedal with Min set to 8 and Max still at 10. In the heel position, the volume will go only as low as 8, instead of 0 as in the previous example.

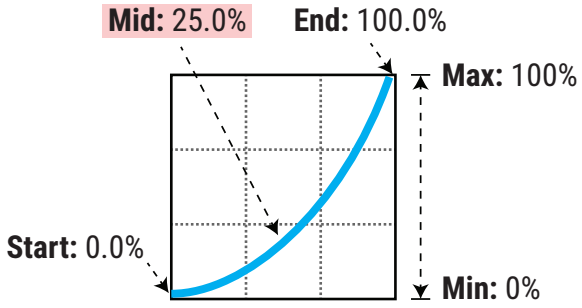


Here is a different twist: Min is at 10.0 — the loudest setting, while Max is at 0.0 — silent.

It looks the same, but this is a reverse volume pedal!

USING "MID" TO CREATE BASIC CURVES

Let's imagine a modifier on the feedback of a delay to demonstrate changes to **response curves**. By now you should be familiar with the default settings, so let's skip right to the curved version:

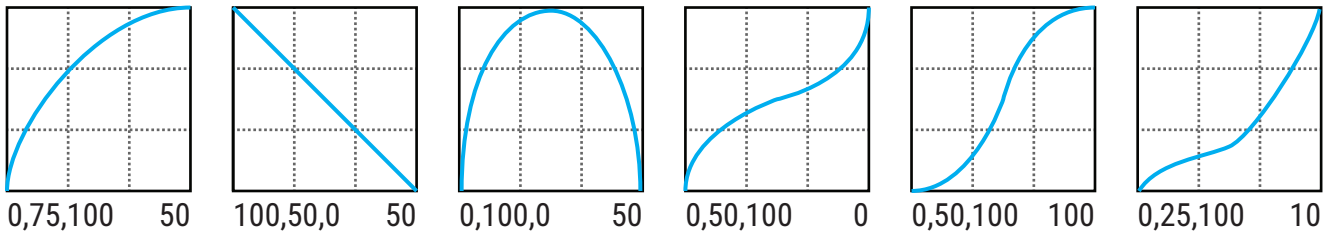


As we adjust Mid, the graph shows the curve. At 25%, the response is "tapered" so that with the pedal half way forward, Feedback is only 25% between Min and Max.

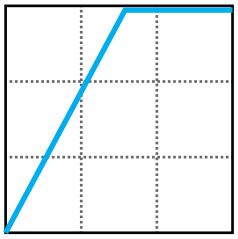
This type of curve provides gradual control of feedback across the lower end of the range, and then a more rapid change towards the top.

ADVANCED MODIFIER CURVES AND SHAPES

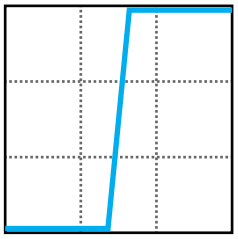
You can also adjust **Start**, **End**, and **Slope** (which introduces an "S" or "backwards S" shape). The numbers beneath each example below show the settings for **Start**, **Mid**, **End** and **Slope**. With a bit of experimentation, you will learn to achieve desired modifier effects quickly. Try dialing in the shapes below for practice:



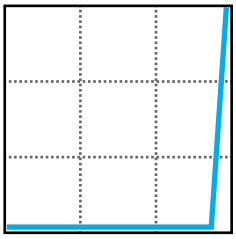
Scale and **Offset** also re-map modifier response. **Scale** adds vertical exaggeration or compression, while **Offset** moves the entire curve up or down on the Y-axis. Segments outside the graph boundaries will be clipped and replaced by line segments. The examples below show some interesting possible applications.



START:	0.0%
MID:	50%
END:	100%
SCALE:	2.0
OFFSET:	100%



START:	0.0%
MID:	50%
END:	100%
SCALE:	10.0
OFFSET:	0%



START:	47.2%
MID:	0%
END:	60.6%
SCALE:	10.0
OFFSET:	-100%

AUTO-ENGAGE TURNS EFFECTS ON/OFF AUTOMATICALLY

If you have used a Wah pedal with no toe-switch, you will instantly understand the principle of "auto engage". This type of Wah turns on automatically when you rock it forward it and then bypasses when you pull it back. That's the idea behind **Auto-Engage**.

Auto-Engage engages or bypasses a block when the **Source** of a modifier changes. Once you try it, we believe you will quickly find that this capability comfortably eliminates the need for expression pedal "toe" switches.

Set it up using two parameters on the Modify page:

Auto-Engage – Determines whether or not the block will automatically engage or bypass. FAST, MEDIUM and SLOW settings determine how quickly the effect turns ON/OFF once Auto-Engage is triggered. Use SLOW settings to "relax" auto-engage, so your effect doesn't switch off too suddenly while you're still using it.

- The three POSITION ("POS") options trigger the effect based on **OFF VALUE** (see below).
- The three SPEED ("SPD") options engage the effect when the controller is moved quickly.
- Set to "OFF" to disable Auto-Engage.

Off Value – Sets the *position* threshold that the source must cross for auto-engage to occur. When **Off Value** is set below 50%, the effect is bypassed when the controller goes *below* that value. If **Off Value** is set to 50% or higher, the effect is bypassed when the controller goes *above* that value.

For "heel down = bypassed", set to 5%. For "toe down = bypassed", try 95%.

MODIFIERS AND CHANNELS

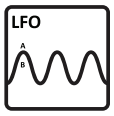
A modifier is normally shared across all of the channels of the block where it exists. However, you can also limit it to operating on just a *single* channel—your choice—using the **Channel** parameter in the modifer menu.

INTERNAL CONTROLLERS

Internal Controllers can be programmed per-preset for use as modifier sources. To access these parameters, push the **CONTROLLERS** button on the **Home** page (Push-knob **D**) or press **TEMPO**.

TEMPO Tempo appears in the Controllers menu. For more information, see Section 10: Tempo.

LFO1 + LFO2



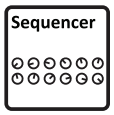
A Low Frequency Oscillator ("LFO") generates control signals in the form of a variety of periodic wave shapes. Examples of LFOs in use include the pulse of a tremolo, the back and forth sweep of a phaser, or the random filter in *Ship Ahoy* by Frank Zappa.

Type sets the waveform or shape. See p. 181 for a table of all waveform shapes. You can set the **Rate** or lock the LFO to the current **Tempo**. You can vary the overall **Depth** and **Duty**, or symmetry. The **Run** parameter starts and stops the LFO. This can be used to keep it from drifting out of time.

Each LFO outputs "A" and "B" signals, each of which is an independent modifier source on the list. You can change the **Phase** of output B with respect to A.

Quantize changes smooth waveforms into stepped ones. Try it on the "TRI" waveform for example.

SEQUENCER



The Step-Sequencer generates control patterns by looping through 2–32 "stages." You set the **Value** for each stage, the **Number** of stages, and the **Rate** or **Tempo**. The **Run** switch allows you to start or stop the sequence. Damping slurs the change between steps.

ENVELOPE FOLLOWER



This converts input level to a control signal, responding variably to your playing dynamics. The classic example is a touch-wah, where the frequency of the wah varies based on how hard you play. You can set **Attack** and **Release** times independently, set the trigger **Threshold**, and adjust the **Gain** at the trigger signal input, which can be set to Input 1 or Input 2.

ADSR1 + ADSR2



The two ADSR sources are envelope generators with **Attack, Decay, Sustain** and **Release** parameters (thus "ADSR"). The ADSR is triggered whenever the signal level exceeds its **Threshold** measured at the selected input (1 or 2). The envelope has three self-explanatory **modes (Once, Loop,** and **Sustain)** and can optionally be set to **Retrigger** every time the threshold is exceeded.

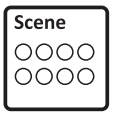
In comparison to the Envelope Follower, the ADSR envelope *generator* always creates the same control signal, but only triggers when you play hard enough.

PITCH DETECTOR



The Pitch Detector analyzes the pitch of your playing and outputs a low value for low notes and a high value for high notes.

SCENE CONTROLLERS



The four Scene Controllers provide a way to change sound settings from one Scene to another. Think of each Scene Controller as a "virtual knob" whose setting can programmed with a different value in every Scene. By assigning this knob as a modifier source, you can create "per-scene" parameters.

For example, you might assign Scene Controller 1 to **Input Drive** in the **Amp** block with a value of 10% in Scene 1, 50% in Scene 2, 75% in Scene 3, and so on.

As with any source, a scene controller can be assigned to multiple parameters at the same time, each with its own modifier settings.

A Scene Controllers tutorial appears on p. 151. For more on Scenes, see Section 6 on p. 42.

QUICK TIP

The Internal Controllers menu offers four channels for four sets of independent settings for ALL of the items listed above: four tempos per preset, four sequencer settings, etc. When you change the channel, ALL of the controllers switch to the new channel and the new settings. Axe-Edit III provides a way to copy/paste Controller channels.

METRONOME

A metronome is included for convenience. The Metronome is not technically a controller, but its settings are accessed through the Tempo page of the Controllers menu. To enable the metronome press the Tempo button and adjust the level for the desired output(s). Note: the metronome levels persist across presets and are reset to OFF at power on.

EXTERNAL CONTROLLERS

External Controllers are modifier sources that are in turn controlled themselves via external MIDI or a connected expression pedal or switch. For example, if the source of a modifier on a Wah effect is set to “External 1” and External 1 is set to “MIDI CC#16”; a connected MIDI expression pedal transmitting CC#16 then operates the Wah.

The global assignments for External Controllers are set on the **External Control** page of the **MIDI/Remote** menu under **SETUP**. You can assign a MIDI CC#, an on-board Expression pedal or switch, or a Pedal or Switch on a connected FC series controller. Choosing an External Controller as the source of your modifier is the same as assigning any other source (see “To Create a Modifier...” on p. 143). To change which CC# or pedal/switch operates an External Controller:

1. Open the **MIDI/Remote** menu in **SETUP** and page to **External Ctrl**.
2. Use **NAV** to select the External Controller you want to change (ex: “External Control 1”).
3. Use **VALUE** to change the assignment. You can also select “NONE” to disable the selected controller.
4. **EXIT** when finished.

EXTERNAL CONTROLLER INITIAL VALUES

External Controllers assigned to MIDI are given a value of “0” until some MIDI data is received to change them. You can imagine how a missing or broken MIDI controller might therefore present a big problem...

The **External Controller Initial Value** parameters can change the startup value for a controller to 100%. Then, when the Axe-Fx III is powered on, any modifiers assigned to the missing controller will be all the way *up* instead of all the way down. To change the initial value for an External Controller:

- Page to the **General** page of the **MIDI/Remote** menu under **SETUP**.
- Use **NAV** to select to whichever controller you want to change (ex: “External Control 1”).
- Use **VALUE** to change the setting.
- **EXIT** when finished.

MODIFIERS LIST

The last page of the **Controllers** menu lists all of the modifiers in the current preset. You can jump to edit each modifier from this list by pressing **ENTER**.

As you’ll gather from looking at the list, the maximum number of modifiers per preset is 24.

TUTORIAL: SCENE CONTROLLERS

In this tutorial, we will set up a Scene Controller to adjust the Input Drive of an Amp block. While it is easy enough to achieve different amp sounds within a preset simply by switching the channel, this example provides an excellent and easy way to understand how Scene Controllers work.

We will begin with the Wrecker Rocket preset (#030), though any preset with an amp block could be used. For this example, we chose an amp model where the Drive parameter setting will be more pronounced, and no pre-existing scenes are set up.

1. Load the factory preset "#030 Wrecker Rocket."
2. The first step in using a Scene Controller is to assign it as a **Modifier Source** to the parameter of your choice. Enter the **Layout** grid of the preset, navigate to the **Amp** block, and press **EDIT**.
3. Use the **PAGE** buttons to locate the **Tone** page of the **Amp** block.
4. **NAV** to the **Input Drive** parameter and notice the open yellow modifier "ring" beneath the knob.
5. With the **Input Drive** parameter highlighted, press **ENTER** to display the **Modifier** menu.
6. On the **Source** page, select **Scene Controller 1.** We don't need any custom settings on the **Modify** page.
7. Press **EXIT** to return to the **Amp** menu. Notice that the open yellow modifier symbol is now a **solid yellow dot**.
8. Now let's set the values for our Scene Controller. Press **HOME** to show the home page.
9. Press the **CONTROLLERS** button (Push-knob **D**).
10. Use the **PAGE** buttons to locate the **Scene Ctrl** page of the **Controllers** menu.
11. Reading down, the **Controller 1** column shows eight different values. These will set the value for our target parameter in each of the eight scenes.
12. We'll want to change scenes as we dial in these values, so press the **MORE 1 of 2** button (Push-knob **E**) to show the Scene/Channel selectors on the "**MORE 2 of 2**" strip.
13. Notice the that the current scene is shown at the top of the page. If not, use the scene buttons to select it now.
14. Dial in the **Scene Controller 1** value for Scene 1. Let's set it to about 20% for a "cleanish breakup" type of sound. Use your ears!
15. Use the **Scene +** button (Push-knob **B**) to switch to **Scene 2**.
16. **NAV Down** to select the **Scene 2** row in the **Controller 1** column. Turn **VALUE** to set the value for Scene 2. Let's set it to about 45% for a nice crunch. Again, use your ears.
17. Using the process in the previous step, set the value for Scene 3 to 75% and Scene 4 to 100% (or as desired).
18. Press **HOME** to return to the **Home** page. Test your first four scenes using the **NAV UP/DOWN** buttons.
19. Press **STORE, ENTER, ENTER** to save your work. If desired, enter **Scene Names** as outlined in "Saving Changes" on p. 40.
20. Feel free to assign other **Scene Controllers** to other parameters in your preset as desired.

10 TEMPO

Tempo is used in electronic music for synchronizing different rates and times, whether inside one machine or across several. Tempo on the Axe-Fx III allows for both internal and external synchronization, providing effects with a central BPM clock that can either run autonomously or follow the tempo of a connected device outputting MIDI Beat Clock. The Axe-Fx III does not transmit MIDI Clock itself. The Tempo can be set to any whole number value in the range 20–250 BPM. The Axe-Fx III flashes its current tempo on a front-panel LED.

THE GLOBAL TEMPO

The Global Tempo value is a system setting outside of any presets, scenes or channels. You can change it in any of several ways:

1. Tap two or more times on the front panel **TEMPO** button.
   *NOTE: By default, the tempo averages across ten taps, but you can set it to use only two taps with an option found under **SETUP: Global: Config: Tap Tempo**.*
2. Tap the **TEMPO** button once and use the **Tempo** knob (**B**).
3. Use an external MIDI device to transmit **MIDI Clock** to the **MIDI IN** port of the Axe-Fx III.
4. Use a MIDI CC# or external switch assigned to **Tempo Tap**, located on the **Other** page of the **MIDI/Remote** menu under **SETUP**. (See NOTE under #1 above).

Be aware that whenever you change the Global Tempo, you are also changing the current Preset Tempo, which will be saved if you store the current preset (see below). Be on guard that you don't accidentally over-write saved tempos!

PRESET TEMPO

A given preset can *ignore* the Global Tempo and use its own saved **Preset Tempo** instead. In fact, this is the default setting. Notice that a preset's **Tempo** page contains two parameters: one is an actual **Tempo** (BPM) value, and the other is a setting called **Tempo To Use**, which determines what will happen when that preset is loaded.

When you load a preset whose **Tempo To Use** is "PRESET," the saved **Preset Tempo** takes over: the tempo LED flashes accordingly and all tempo-dependent time/rate parameters are re-calculated. In fact the Global Tempo has not changed; it remains in the background and will be used when you load a different preset with **Tempo to Use** set to "GLOBAL".

Be aware that whenever you alter a preset's own Tempo, you are also updating the Global Tempo.

Remember that each of the four channels of the **Controllers** menu has its own settings for both **Tempo** (BPM) and **Tempo to Use**. You can use this to have some Scenes or Channels change the tempo while others do not.

TO SET "TEMPO TO USE"...

1. Load the preset.
2. Tap once on the **TEMPO** button.
3. Change **TEMPO TO USE** as desired to either "GLOBAL" or "PRESET".
4. STORE the preset.

SYNCHRONIZING SOUND PARAMETERS

Rates and times in a preset can be made to sync rhythmically to the Tempo by setting their corresponding **Tempo** parameters. This is done by selecting from a list of values, ranging from 1/64-note-triplets to double whole notes, with 76+ options in all. For example, to set the **Time** of a Mono Delay block to follow the quarter note pulse of the tempo, find the **Tempo** parameter on the **Config** page of the Delay block's **Edit** menu and set this value to "1/4."

The moment you assign a value for Tempo (other than "NONE"), its associated rate or time parameter is overridden and may not be changed manually — as indicated by its appearance in parentheses. To regain control of an overridden parameter, set its corresponding Tempo parameter back to "NONE".

Almost every LFO rate and delay time parameter on the Axe-Fx III can be synced to the tempo.

TO SYNCHRONIZE A DELAY TO THE TEMPO...

- Navigate to any type of Delay block and press **EDIT** to show its Edit menu.
- Find the **Tempo** parameter and set it to "1/4". Feel free to choose/explore other options. "1/8th dot" is also very useful and popular.
- STORE the preset.

Remember that once a **Tempo** has been set, you will be unable to adjust **Time** manually or with a Modifier.

TO SYNCHRONIZE A TREMOLO OR OTHER RATE TO THE TEMPO...

- Navigate to the effect and press **EDIT** to show its Edit menu.
- Find the **Tempo** parameter and set it as desired. A typical value here might be "1/8th" or "1/16th". Feel free to choose/explore other options.
- STORE the preset.

Remember that once a **Tempo** has been set, you will be unable to adjust **Rate** manually or with a Modifier.

Note that syncing an LFO to the tempo does not align the phase of that LFO to the Tempo LED, to MIDI song position pointer, or to other synchronized LFOs. You can synchronize various LFOs together by setting their RATE knobs fully counterclockwise to "LFO1 SYNC" and then syncing the **LFO1 Controller** to the **Tempo** (p. 149).

11 THE TUNER

The Axe-Fx III has a built-in Tuner—essential for the performing or recording musician. The tuner is easy to use and has high-resolution automatic pitch detection, a calibration control, offsets for modified tuning schemes, and the option to mute audio while tuning. Find the Tuner on the **Home** page by pressing **TUNER** (Push-knob **B**).

The tuner features a note name display, a "zero-center" meter, a spinning "strobe" type display and triangular indicators for sharp and flat.

ADVANCED TUNER FUNCTIONS

The actual tuner is on the **Tune** page of the **Tuner** menu.
The **Config** and **Offsets** pages provide options for advanced tuner functions.

CONFIGURATION PARAMETERS

PARAMETER	Description.
Input	Selects which physical input the tuner should connect to.
Calibration 430.0 – 450.0 Hz	Calibrates the tuner by setting the frequency of A4 (in the octave above middle C). The tuner defaults to A440.
Mute OFF/INPUT/OUTPUT	Determines how the tuner mute function works. ▪ OFF: No mute. All signal is passed as usual when the tuner is engaged. ▪ OUTPUT: The signal is muted at the output. Tails are silenced. ▪ INPUT: The signal is muted at the noise gate. Tails ring out.
Use Offsets OFF/ON	Determines whether the **Offset** settings (see below) are applied or ignored.

OFFSET PARAMETERS

PARAMETER	Description
E1, B2, G3, D4, A5, E6 +/- 25.00 Cts	Offsets allow the tuner to be calibrated so individual notes diverge from standard concert tuning by a defined amount. Use this for example for Buzz Feiten tuning.

MINI TUNERS

For convenience, mini-tuners appear throughout the Axe-Fx III. These consist of two green triangles to indicate when a note is flat (left lit), sharp (right lit) or in tune (both lit).

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12 SETUP MENU

Open the menu by pressing the **SETUP** button (Push-knob **E**) on the **Home** page. The current **Firmware Version** is shown at the top of the page. To enter a menu, select it with **NAV** and press **ENTER**. All changes take effect immediately without needing to be stored. The settings for Setup parameters are included in a backup of the System (see "Backing Up & Restoring" on p. 172).

THE FOOT CONTROLLERS MENU

The Foot Controllers menu is covered in its own Section on p. 167.

THE GLOBAL SETTINGS MENU

☞ Reminder: You do not need to store settings in the **SETUP** menu. **EXIT** twice when finished.

GLOBAL SETTINGS: CONFIG PAGE

PARAMETER	Description
Power Amp Modeling ON/OFF	This parameter globally enables or disables power amp simulation in all Amp blocks in all presets. This capability is provided for cases when the Axe-Fx III is used with a guitar-oriented/tube power amp that contributes significantly to tone and dynamics. Having these characteristics applied twice to the sound—once in the virtual power amp and once in the real power amp—would result in an over-processed sound. All those Amp block parameters that are NOT part of the virtual power amp continue to work normally while Power Amp Modeling is turned off. Presence and Depth are disabled, and Master Volume becomes a simple volume control. Remember that like a real amp, the Axe-Fx III has separate preamp and power amp sections. When you disable Power Amp Modeling, the Amp block still models the preamp, which includes distortion, the tone stack, and more.
Cabinet Modeling ON/OFF	This parameter conveniently enables or disables Cab block processing in all presets. The Cab block will not physically appear to be bypassed, but acts exactly as if it had been replaced a shunt. Use this when you are using the Axe-Fx III with an amp powering a traditional guitar speaker cabinet, or the sound will be muffled, muddy, and boomy.
Spillover OFF/DELAY/REVERB/BOTH	Allows delay and reverb "tails" to ring out or spill over across Preset changes. You can select whether "DELAY", "REVERB", or "BOTH" effects will spill over. Setting to "OFF" causes effect tails to be cleared upon preset change. Spillover when switching Scenes or using "IA" Switches is not affected by this parameter. See also "Spillover" on p. 180.
Reverb Mix +/- 50%	This boosts or cuts the MIX for ALL Reverb blocks in all presets. Note that this offset is NOT reflected in the value shown for the actual Reverb block MIX parameter. This feature is provided because certain dead or lively performance spaces may require more or less reverb across all presets. Remember that MIX generally applies only to Reverb blocks which are wired in *series*. (Parallel reverbs typically run with MIX at 100%.)

PARAMETER	Description
Effects Mix +/- 50%	Boosts or cuts the MIX of all blocks for which the GLOBAL MIX parameter is set to “ON”. This switch must be enabled on a per-block/per-preset basis and is available on the Mix page of the Edit menu for these block types: **Chorus** **Formant** **Pitch** **Ring Mod** **Delay** **Multitap Delay** **Plex Delay** **Rotary** **Flanger** **Phaser** **Reverb** **Ten-Tap Delay** This feature is provided because certain performance spaces may require more or less of certain effects across all presets.
Noisegate Offset +/- 40.00 dB	Globally raises or lowers the THRESHOLD of the Noise Gate for all Input blocks. If the THRESHOLD for a given preset is set to “OFF”, the global Offset will have no effect. See “Noise Gate Parameters” on p. 52.
Prompt on Edited Preset Change	When set to ON the unit will prompt before changing presets if the current preset has been edited. This spares you from accidentally losing your edits. NOTE: Be sure to change this to OFF before performing!
Display Offset 0, 1	(Also appears in the MIDI/Remote menu) Causes presets numbers to begin at 001 instead of 000. This only offsets the display, meaning that it does not change which preset is actually loaded by a given footswitch or MIDI message.
Default Scene AS SAVED, SCENES 1–8	When set to “As Saved” the scene selected when recalling a preset is the scene that was active when the preset was last saved. When set to a particular scene value that scene will always be selected when any preset is recalled.
Tap Tempo AVERAGE, LAST TWO	Determines how the tempo changes when tapping the front panel TEMPO button or an external tap tempo controller (see p. 165). “AVERAGE” sets the tempo based on the average of ten taps, meaning taps are more forgiving but changes are more gradual. “LAST TWO” considers only the time interval between the last two taps, which means taps must be more precise but changes occur more quickly.
Value Knob Push Function	By default, pushing the main “VALUE” knob shows the layout grid. To disable this feature, change the setting of this parameter to “NONE”.
AC Line Frequency	The “Intelligent” Noise Gate type found in the **Input Block** provides smart EMI filtering which reduces hum and buzz. For best results, set this parameter to match the power line frequency of your country (i.e. 60 Hz for North America, 50 Hz for EU, etc.).

GLOBAL SETTINGS: OUT 1 EQ & OUT 2 EQ PAGES

Two separate pages for **OUT 1** and **OUT 2** each provide a 10-band graphic EQ, plus a master GAIN adjustment control**.** Use these to make global adjustments to the tone or level of ALL presets. This is convenient when you need to compensate for using a different amp or speakers, or if a particular performance space seems overly bright or boomy. The master GAIN fader should be used with care as it can cause clipping, and lowering it can affect the signal-to-noise ratio. These EQs affect ONLY the Analog outputs. AES/SPDIF/USB are not affected.

Output 1 Global EQ settings apply to all instances of the Output 1 mix signal, including the Output 1 balanced jacks, unbalanced jacks, headphones jack, and signal which might be routed to USB or digital outs.

Output 2 Global EQ settings apply to all instances of the Output 2 mix signal, including the Output 2 balanced jacks, and signal which might be routed to USB or digital outs. When **Output 2 | Copy Output 1** is enabled, Output 1 Global EQ settings are NOT applied to signals at Out 2. Instead, the Output 2 Global EQ settings apply as usual.

GLOBAL SETTINGS: CUSTOM SCALES PAGE

The **Scales** page is used to configure scales for the **Custom Shifter** type in the **Pitch** block (p. 118).

PARAMETER	Description
Custom Scale Number 1–32	Selects from among the 32 global custom scales available to edit using the 12 parameters which follow.
___ SHIFT (+/- 24)	These 12 parameters are used to set the shift amount for each of the 12 steps of the chromatic scale. Range is +/-24 semitones (+/- two octaves). To set up a custom scale, select its number in the field above and then set each of the 12 pitch values as desired. Changes take effect immediately, with no need to STORE. Soft-buttons provide options to reset a selected value or all values in the current scale to the default value of +/- 0 semitones.

THE I/O MENU

☞ Reminder: You do not need to store settings in the **SETUP** menu. **EXIT** twice when finished.

I/O: INPUT PAGE

The **Input** page of the **I/O** menu contains four parameters that scale the input to the A/D converters: **Input 1/Instrument Level, Input 2 Level, Input 3 Level, Input 4 Level.**

Remember that, except at very low settings (5% or less), input level adjustments do not affect gain levels or what you hear. This is because as you adjust the level to the input of the A/D converter, the output of the converter compensates accordingly, so your guitar signal level remains exactly the same when it reaches the grid and any virtual amps or effects.

I/O: AUDIO PAGE

PARAMETER	Description
Input 1 Source ANALOG USB [CHANNELS 5/6] SPDIF/AES	This selects which signal source should be routed through the Input 1 block on the grid. “ANALOG” selects the front or rear Input 1 Instrument input jack. When “USB (CHANNELS 5/6)” is selected, the Axe-Fx III will process signals sent from Axe-Fx III Outputs 5 and 6 of a connected computer—perfect for re-amping without custom presets. When “SPDIF/AES” is selected, signal from the currently selected digital input will be used (see SPDIF/AES Select, below).
Word Clock INTERNAL SPDIF/AES IN	This selects the clock source for the A/D and D/A converters as follows: “INTERNAL”: uses the internal clock. “SPDIF/AES IN”: uses the clock recovered from the current digital input. A valid 48 kHz data stream must be present at the AES or SPDIF port for the Axe-Fx III to operate properly.
SPDIF/AES Select SPDIF, AES	This switch selects whether the SPDIF or AES digital inputs and outputs are active. Only one format may be active at any time.
SPDIF/AES Source Output 1 or 2, Input 1 or 2	This allows you to specify which signal should be transmitted at the SPDIF or AES digital output. Options include Output 1, Output 2, Input 1 or Input 2.
USB Buffer Size 8–256	Set this to lower values for less latency with USB Audio, set to higher values if you are experiencing distorted audio. Stop USB audio streaming when changing this value so as to allow the buffer to reset properly. Streaming can be stopped by closing the application sending data to the Axe-Fx III or by disconnecting the USB cable.

PARAMETER	Description
Input 2 Mode **Input 3 Mode** **Input 4 Mode** LEFT ONLY L+R SUM STEREO	For each of the three stereo input pairs, this setting determines globally how the Axe-Fx III handles signals received at the corresponding input jacks, determining whether they should be processed in stereo or mono—and, if mono, whether "LEFT", "RIGHT", or "SUM L+R" should be used. The outputs of the connected device and the nature of the source material will determine which setting is best. It is critical that you choose the setting which corresponds to your actual physical connections. Setting an input to "STEREO" and then not connecting one of the two jacks will result in reduced levels!
Output 1 Mode **Output 2 Mode** **Output 3 Mode** **Output 4 Mode** STEREO, SUM L+R COPY L>R	These four parameters determine how signals at the corresponding output ports will be processed. This control makes it easy to use the same presets in a variety of stereo and mono performance or recording environments. The decision between the mono options "SUM L+R" or "COPY L>R" should be based on the source material. See "Mono vs. Stereo" on p. 6 for more information on these options. USB Audio goes directly to the converters and is NOT affected by this setting.
Output 1 Level **Output 2 Level** -10dBV +4 dBu	These two parameters set the nominal levels of their respective outputs. Use "-10 dBV" for consumer-grade equipment including many guitar products. Use "+4 dBu" with professional audio equipment. The product manual for your connected device should indicate whether it operates at +4 dBu or -10 dBV (Default). (Some devices are switchable.) USB Audio levels ARE affected by this setting.
Output 1 Phase **Output 2 Phase** **Output 3 Phase** **Output 4 Phase** NORMAL, INVERT	These four parameters determine whether signal at the corresponding output ports will be normal or phase-inverted relative to its state at the output of the grid. This lets you compensate for unwanted inversions elsewhere in the signal chain. USB Audio signals ARE affected by this setting.

PARAMETER	Description
Output 2 | Copy Output 1 OFF/ON	Enabling this option creates a copy of the Output 1 signal at Output 2. Use this for the convenience of having an extra copy of the Output 1 mix with separate front panel level control—without the need to insert an additional output block. This setting works only when the Output 2 block is NOT present on the grid for the current preset!
Output 3 | Copy Input 1 **Output 4 | Copy Input 1** OFF/ON	Enabling either of these options creates a copy of the Input 1 signal at Output(s) 3 or 4. Use this for the convenience of having an extra copy of the Input 1 signal (for all-analog re-amping, for instance) without the need to insert additional output block(s) on the grid. This setting works only when the corresponding output block is NOT present on the grid for the current preset! For example, Output 3 | Copy Input 1 will NOT copy Input 1 when the Output 3 block is on the grid.
Output 3 Boost/Pad **Output 4 Boost/Pad** 0/6/12 dB	These settings engage a boost/pad which can help lower the noise floor of Output 3 or 4. This may be useful when outputting to a tube amp or modeler or with the Four-Cable Method. Boost/Pad helps optimize D/A performance without affecting levels, since a boost at the converter's input is paired with a corresponding cut at its output. Be watchful, however, as boosting makes it easier to clip the converters. Watch the meters and if clipping occurs, reduce levels within your preset or turn this setting down.

I/O: PEDAL PAGE

The **Pedal** page of the **I/O** menu contains parameters to set up and use an expression pedal or a switch through the Pedal jacks on the rear panel of the Axe-Fx III. Expression pedals should have a linear resistance taper and max resistance of 10kΩ to 100kΩ. The Fractal Audio Systems EV-1 and EV-2 expression pedals are recommended. Expression pedals must be used with Tip-Ring-Sleeve (TRS) cables.

An external switch may also be used, as long as its contacts make and break the connection between tip and sleeve. A regular 1/4" guitar cable can be used with switches.

PARAMETER	Description
Pedal 1 Type **Pedal 2 Type** CONTINUOUS LATCHING MOMENTARY	Set this according to the physical type of the pedal or switch that is connected. Use "CONTINUOUS" for expression pedals and "MOMENTARY" or "LATCHING" for switches.
Calibrate PEDAL 1 **Calibrate PEDAL 2**	Calibrate expression pedals connected to an on-board Pedal jack. First select this menu choice, then: ▸ Press **ENTER**. ▸ Move the pedal through its full range of motion several times. ▸ Press **ENTER** again when finished. Switches, unlike pedals, do not need to be calibrated.

THE IR CAPTURE UTILITY

The IR Capture utility is covered in **Section 14**.

THE MIDI/REMOTE MENU

The **MIDI/Remote** menu contains global MIDI-related settings, and allows you to make assignments for Global Controller functions including Block Bypass, Block Channel, the Looper, External Controllers, and more.

☞ Reminder: You do not need to store settings in the **SETUP** menu. **EXIT** twice when finished.

MIDI/REMOTE: GENERAL PAGE

PARAMETER	Description
MIDI Channel 1–16, OMNI	Sets the channel on which the Axe-Fx III will receive MIDI messages. "OMNI" causes the unit to respond to incoming messages on *any* channel.
Display Offset 0, 1	(Also appears in the **Global** menu) Causes presets numbers to begin at 001 instead of 000. This only offsets the display, meaning that it does not change which preset is actually loaded by a given footswitch or MIDI message.
Scene Revert ON/OFF	Selects between two different ways for scenes to work when you change them via a Footswitch or MIDI: "OFF" (Default): Scene edits are RETAINED across Scene changes as long as you do not change PRESETS. So if you tweak Scene 1, switch to Scene 2, then switch back to Scene 1, your tweaks will still be intact. "ON": Scene edits are LOST if you change the Scene without saving. So if you tweak Scene 1, switch to Scene 2, then back to Scene 1, Scene 1 will have reverted to its previously saved state. This makes Scene changes feel more like traditional preset changes.
Effect Bypass Mode VALUE/TOGGLE	This setting determines how MIDI messages set the Bypass State of a block. When set to "Value" the bypass state is controlled by the CC value (0–63=OFF, 64–127=ON). When set to "Toggle" the bypass state toggles whenever the CC message is received, regardless of the value (0–127).
Program Change ON/OFF	Determines whether the Axe-Fx III will process or ignore incoming MIDI Program Change messages.
Ignore Redundant PC ON/OFF	This setting determines whether the Axe-Fx III should re-process or ignore a Program Change message that corresponding to the current preset. With this setting "OFF", a preset will be reloaded—all changes discarded—when it is selected again via PC. This allows you, for instance, to load a preset, use various "Instant Access" switches to bypass effects, and then stomp the footswitch that originally selected the preset to have it revert to its saved state. With this setting "ON", redundant PC messages are ignored. When PC mapping is used, if the current Scene has changed, the preset will not reload but the Scene will still revert to that set in the map.
Send MIDI PC 1–16 OMNI OFF	Determines whether or not the Axe-Fx III will automatically transmit a MIDI Program Change message at its MIDI OUT port when a new preset is loaded. This is the easiest way for most people to operate a single connected MIDI device while changing presets on the Axe-Fx III. Any custom MIDI mapping is left to the downstream device. To use this feature you simply select which channel you want the message to be transmitted on. The "OMNI" setting transmits the message on *all* channels.

PARAMETER	Description
MIDI PC Offset	Adds a specified value to all incoming MIDI Program Change requests before they are processed. This makes it possible, for instance, to address alternate presets in different "registers". You might use the same MIDI messages to access presets 1–16 for a gig with one guitar, and —by specifying an offset of +16— use presets 17–32 for a different guitar without re-programming your controller.
PC Mapping	PC Mapping determines whether incoming MIDI Program Change messages load presets 1-for-1, or load other presets and scenes instead. With PC Mapping turned "OFF", presets are loaded 1:1 based on incoming MIDI bank Select and Program Change messages. With custom PC Mapping enabled, incoming Program Change messages are re-mapped according to the values in the table on the Custom page of the MIDI/Remote menu (see below).
INITIAL VALUE: External Control 1–16	These sixteen parameters specify the initial value to be used for each of the 16 External Controllers (p. 150) when the Axe-Fx III is powered on. This value persists until data is received from the MIDI controller. This is especially useful when an external MIDI controller is absent. For example, if you normally use an expression pedal to control the volume in your presets, a missing pedal might make the preset get "stuck" in a muted position. Setting an initial value of 100% for the External Controller mapped to that pedal would ensure that when the pedal is not connected, the volume will stay at 100% instead of 0%. This setting is only for controllers with a MIDI CC# assignment. It does not apply to local or FC pedals or switches.

MIDI/REMOTE: MAPPING PAGE

With PC Mapping enabled (see above), incoming Program Change messages are re-mapped according to the values in the table in this section.

PARAMETER	Description
Map to Preset **Map to Scene**	When PC Mapping (above) is turned "ON", the parameters in this table specify which preset and scene are loaded for each incoming MIDI PC message. See p. 46 for additional info on PC Mapping.

MIDI/REMOTE: BYPASS PAGE

The **Bypass** page of the **MIDI** menu allows you to map MIDI CC messages—or connected pedals and switches— to bypass or engage blocks. CC messages can be sent by a MIDI controller or computer connected to the MIDI IN jack, or via the USB connection of the Axe-Fx III. Each of these settings is global. In every case, the method for mapping a controller to one of these functions is the same:

1. Use the **NAV** knob to select the desired function.
2. Use the **VALUE** knob to assign a controller to the function:

- **1—127** to assign a MIDI CC#.
- **PEDAL 1** or **PEDAL 2** to assign a pedal connected to the on-board expression jacks of the Axe-Fx III.
- **FC__ PEDAL 1–4** to assign a pedal connected to the corresponding **pedal** jack of the designated FC controller.
- **FC__ SW 1–4** to assign a switch connected to the corresponding **switch** jack of the designated FC controller.
- **NONE** to remove all assignments from the selected item.

MIDI/REMOTE: CHANNEL PAGE

The **Channel** page works exactly like the **Bypass** page except the entries are used to set the Channel of each block instead of its bypass state. The *value* of the CC message sets the channel as follows:

0 = Channel A 1 = Channel B 2 = Channel C 3 = Channel D

The series continues, repeating A, B, C, D for values up to 127 (D).
For additional information, see "Channel Select" on p. 45.

MIDI/REMOTE: EXTERNAL PAGE

Controller assignments for the sixteen **External Controllers** used in Modifiers (See Section 9: Modifiers).

MIDI/REMOTE: LOOPER PAGE

Controller assignments for the functions of the Looper.

MIDI/REMOTE: OTHER PAGE

Controller assignments for various other functions of the Axe-Fx III.

PARAMETER	Description
Tempo Tap	Provides the ability to tap the tempo using an external controller. Remember that tempo is calculated using either the last two taps OR an average, as set in the global Tempo configuration (p. 152). Learn more about Tempo in **Section 10**.
Tuner	Provides a way to remotely enter or exit the Tuner (**Section 11**).
Preset Inc/Dec **Preset Inc/Dec Start/Stop**	These allow you to step up or down through a sequence of presets with pre-defined start and end/wrap points. Preset mapping and offsets are ignored.
Scene Select	Selects specific Scenes. The controller *value* (NOT the CC number) determines which Scene is loaded: **0** = Scene 1 **2** = Scene 3 **4** = Scene 5 **6** = Scene 7 **1** = Scene 2 **3** = Scene 4 **5** = Scene 6 **7** = Scene 8 The series continues, cycling through scenes 1–8 for values up to 127 (Scene 8). For more, see "Scene Select" on p. 45 or "CC Value to Scene" on p. 185
Scene Increment **Scene Decrement**	The Scene Increment and Decrement functions allow you to step up or down through scenes. These functions are triggered by CC data values above 63.
Input 1,2,3,4 Volume **Output 1,2,3,4 Volume**	Eight individual parameters allow you to globally control the levels of the corresponding Input or Output blocks.
Output 1 Volume Increment **Output 1 Volume Decrement**	These two options provide a convenient way to permanently increase or decrease the MAIN output volume for the Out 1 Block in the currently loaded preset. Each time **Volume Increment** is triggered, the MAIN is increased by 1.0 dB and the preset is saved. This is the same for **Volume Decrement**, except it decreases volume. ⚠ IMPORTANT! Any unsaved changes such as altered effect parameters or bypass states will also be stored when either of these functions is triggered! ⚠ These functions are designed for use with momentary footswitches set up to send a CC# value of 127 for "ON" and 0 for "OFF". Do not use an expression pedal or you may change levels +/-20 dB with a single sweep!

THE UTILITIES MENU

UTILITY: DISPLAY PAGE

Provides a slider to adjust contrast/viewing angle for the built-in LCD display.

UTILITY: ADC LEVELS

Shows Analog-to-Digital Converter levels for the four front panel Level knobs and rear panel Expression jacks for troubleshooting. You can set the output level here at an exact percentage.

UTILITY: RESET PAGE

This page provides a tool to set parameters in the **SETUP** menu back to factory default values. **Reset system parameters** is one of the first steps of troubleshooting. It never deletes or modifies your presets. You will be prompted to press ENTER to confirm.

Be very careful with this option. Resetting system parameters is irreversible!

A second utility provides the ability to **Clear all Presets.** You will be prompted to press ENTER to confirm.

Be very careful with this option. Erasing presets is irreversible!

UTILITY: USB PAGE

This page meters the USB input and output FIFO buffers. Ideally these will be at around 50%. If the buffer overflows or underflows, **USB Buffer Size** (p. 159) should be increased. Remember that you need to quit all audio applications or remove and reconnect the USB cable to reset the buffer.

UTILITY: FIRMWARE PAGE

This page indicates which firmware version the Axe-Fx III is running.

USB and FPGA firmware versions are also shown.

For more information, see “Tap Tempo” on p. 157.

13 FC PROGRAMMING

This section will detail programming the Axe-Fx III for new Fractal Audio Systems F-6 and FC-12 Controllers.

14 IR CAPTURE

Whether part of a combo or housed in its own cabinet, the **speaker** is a major part of the sound of any guitar or bass amp. When you hear great tone on a recording or at a live show, this means someone has done a great job of selecting and placing **mics** (even if the player is already using a Fractal Audio product!)

The Axe-Fx III uses Impulse Responses ("IRs") to faithfully reproduce the sound of a **mic'd speaker**, with over 2,048 **Factory Cabs** provided, and 2,048 more memory slots for **User Cabs**. (See "The Sound of IMPULSE RESPONSES" on p. 68 for an introduction to IRs). Our store offers many professionally produced Cab Packs, but you can also create your own IRs using the **IR Capture utility** in the Axe-Fx III. The process is easy and requires only some common pro-audio gear.

DECIDE ON A CAPTURE METHOD

There are two different methods of IR capture. Both methods require microphone(s) and preamp(s), and both methods produce excellent results.

Method 1 requires a flat, neutral, full-range **power amp**. This will typically be a solid-state amp designed for pro-audio applications (e.g. *not* a guitar-oriented amp, which would add tonal coloration).

Method 2 adds a **speaker-level DI box** — such as the **Fractal Audio X-Load LB-2** — to "factor out" the power amp, so you can use *any amp*—even a regular guitar amp.

MONITOR YOUR AXE-FX III

When creating IRs, you'll probably want to test-drive the results right away. Connect your guitar to your Axe-Fx III and set up whatever you'd normally use for direct, **full-range monitoring** (headphones, studio monitors, powered FRFR speakers, etc.). The capture process uses **OUTPUT 2 LEFT**, so please ensure that **Output 2 | Copy Output 1** is set to "OFF" on the **Audio** page of the **I/O** menu under **SETUP**.

MIC THE SPEAKER

- Before you can mic a speaker, you'll need to play through it. You can plug right in to the actual guitar amp itself as you normally would, or you can use an Axe-Fx preset (with the Cab block bypassed or circumvented) into a neutral power amp, into the speaker. One advantage of the latter method is that once you capture the IR, you can use the same exact amp for testing, which should produce results remarkably close to the original.
- Once you've got a good guitar tone coming through the speaker, it is time to set up the mic(s). Recording studios have control rooms isolated from the "live room", making it easy to hear what the microphone is picking up without interference from a loud amp. Having an assistant or robot to adjust the mic is convenient. If this option isn't available to you, try using headphones to let you focus on the mic'd tone. It might be loud, so take care not to damage your hearing.[1] You can also record the amp and listen back to make sure you are satisfied with the tone.
- If the cab is loaded with multiple speakers, you should listen to each one to determine your favorite(s), and capture any and all that sound good to you.

1 https://www.osha.gov/SLTC/noisehearingconservation/

- The tone you get from a mic'd speaker depends on the type(s) and position(s) of the microphone(s). Select a mic that is well-suited for guitar cab recording and position it according to your preference. Good information is widely available about mic placement, distance, angle, etc., but don't be afraid to experiment.
- Each mic needs to be connected to a preamp to bring the signal up to line level for processing. The preamp also contributes to the tone, so a high-quality preamp (like an API or Neve) is generally preferred, but even the preamp of an inexpensive mixer can produce excellent results.
- With an optional mixer, multiple microphones can be combined to produce and capture a tone that blends their characteristics. Do not apply dynamics processing (e.g. compression) as this will misrepresent the response.

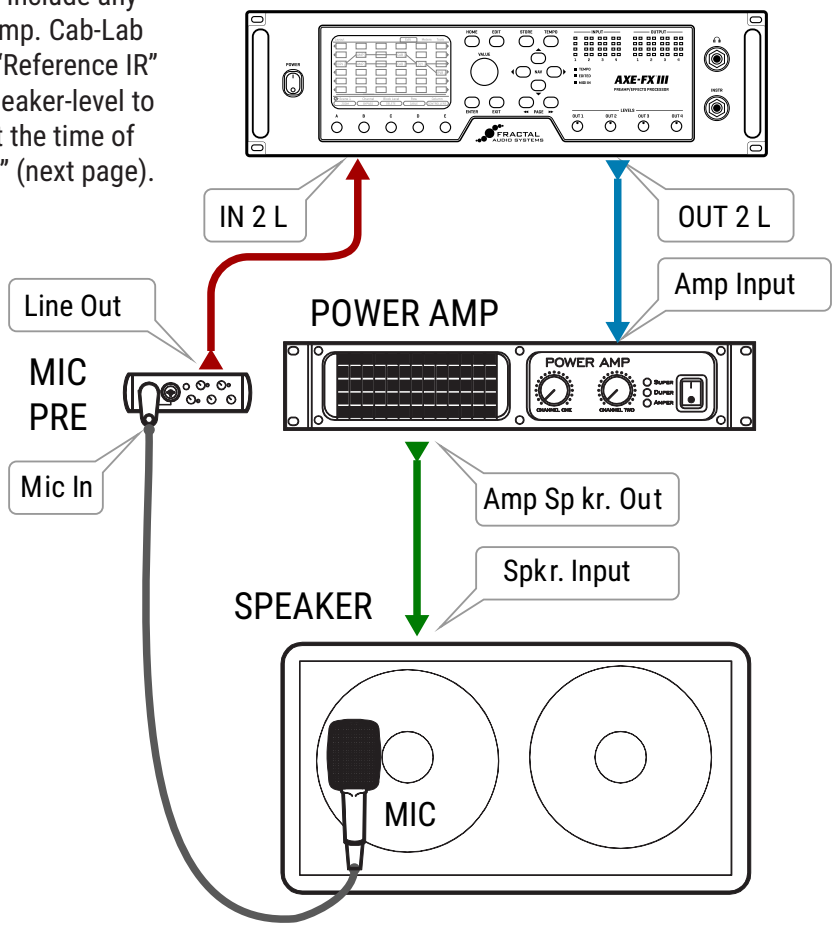
METHOD 1 SETUP: MIC ONLY

Method 1 ("MIC ONLY") requires a microphone, a preamp, and a flat, neutral power amp.

- Set up a preset with your preferred **Amp** block, and set its **Cab** block to "SCRATCHPAD 1" for instant auditioning.
- Connect the **mic** to the input of the preamp.
- Connect the **output** of the preamp (or mixer) to **INPUT 2 LEFT** of the Axe-Fx III.
- Connect **OUTPUT 2 LEFT** of the Axe-Fx III to the **input** of your power amp.
- Connect the power amp's **speaker out** to your speaker.
- On the **Config** page of **IR Capture** utility in the Axe-Fx III, set **Capture Method** to "MIC ONLY".

NOTE: IRs created using this method will include any tonal coloration imparted by the power amp. Cab-Lab can remove this effect using a separate "Reference IR" of just the power amp created using a speaker-level to line-level DI, but if you have this DI-box at the time of capture you might as well use "Method 2" (next page).

Follow the capture instructions on p. 171 to perform the capture.

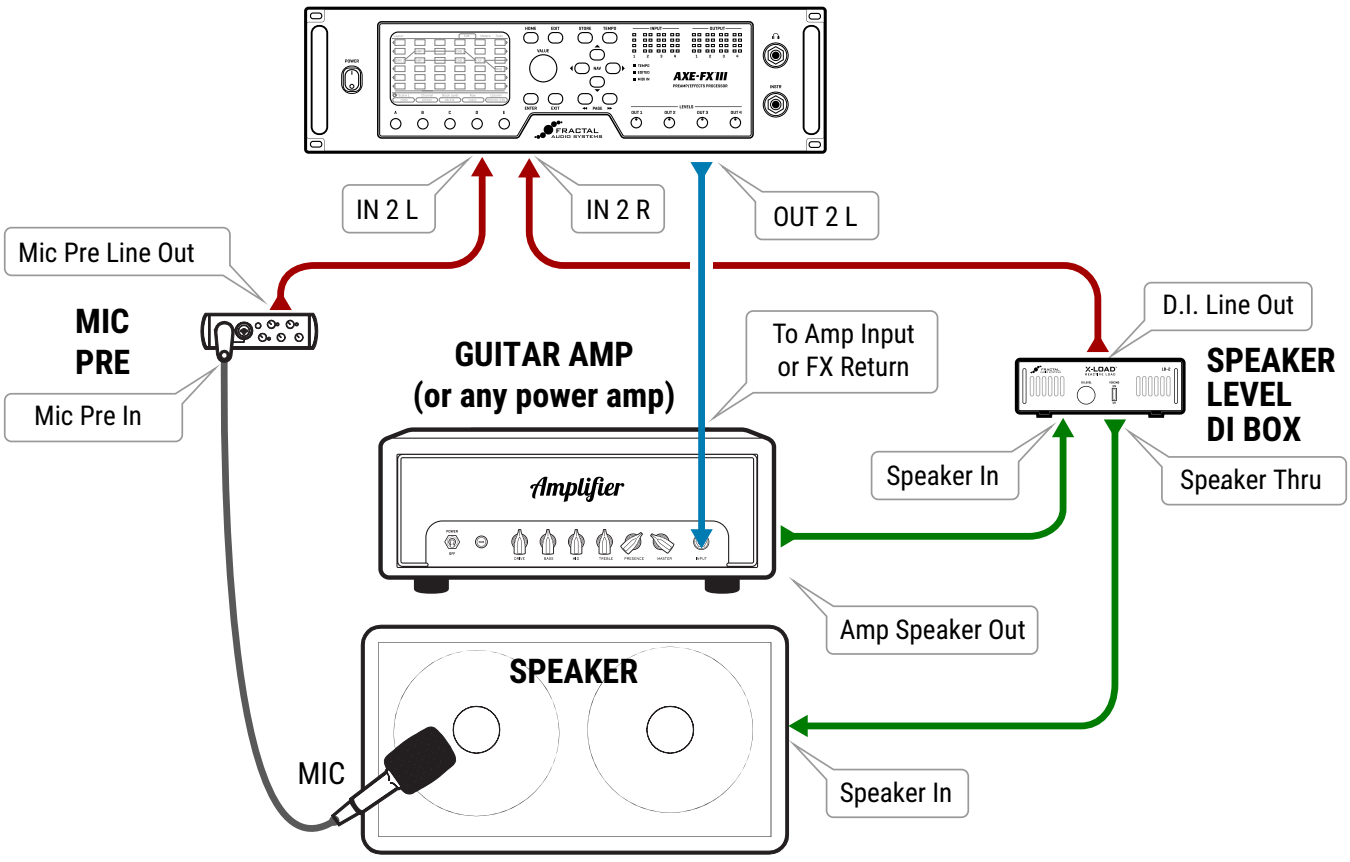


METHOD 2 SETUP: MIC + DI

Method 2 (“MIC + DI”) requires a microphone, a preamp, and a DI Box with the special capability to convert speaker level to line level. Our own load box, the Fractal Audio X-LOAD LB-2, is ideal for this application. The “MIC + DI” setup completely “factors out” the tonal influence of the power amp, allowing almost any amp to be used. A neutral power amp is still preferred, but even a guitar amp will produce excellent results. If you do use a guitar amp, you should connect to its Effects Return if possible, but even the front input can be used, as the Axe-Fx III is able to remove distortion, completely separate the speaker response, and produce a pure and accurate IR.

- Connect the **mic** to the input of the preamp.
- Connect the **output** of the preamp (or mixer) to **INPUT 2 LEFT** on the Axe-Fx III.
- Connect **OUTPUT 2 LEFT** of the Axe-Fx III to the **input** of your power amp. The Effects Return of a guitar amp can be used, or even the instrument input with a very clean setting.
- Connect the amplifier’s **speaker output** to the **speaker level input** of the DI box. Be sure to use a DI box that is specifically designed to work with speaker level signals and one that does NOT have any on-board cabinet emulation. You can also use the DI output of your amp if it has one, as long as it is not a speaker-emulated output.
- Connect the **Speaker Thru** jack of the DI box to the **input** of your speaker.
- Connect the **Line Level Output** of the DI box to **INPUT 2 RIGHT** on the Axe-Fx III.
- On the **Config** page of **IR Capture** utility in the Axe-Fx III, set **Capture Method** to “MIC + DI”.
- On the **Audio** page of the **I /O** menu under **SETUP**, ensure that **Input 2 Mode** is set to “STEREO”.

Follow the capture instructions on p. 171 to perform the capture.



CAPTURING IMPULSE RESPONSES

USING CAB LAB

Cab-Lab is an IR Mixer and Toolbox, available at https://www.fractalaudio.com/cab-lab-3.

It is strongly recommended that you use Cab-Lab to capture IRs. The interface is easier to use, and you will also be able to save the raw IR data to your computer for later remixing or re-processing.

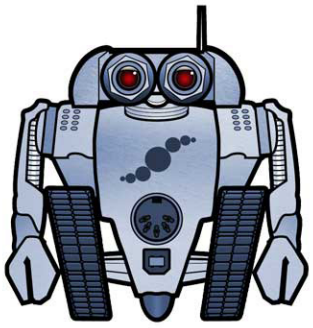
FROM THE AXE-FX III FRONT PANEL

This section covers capturing from the Axe-Fx III front panel:

1. Make the connections for your chosen setup as detailed on the previous pages of this section.
2. In the Axe-Fx III, create a preset with an **Amp** block set to an appropriate model and a **Cab** block with Cab 1 set to Bank: "SCRATCHPAD", Number: "1". In that same Cab block, set Cabs 2, 3 and 4 to "MUTED". IRs are automatically sent to this special location upon capture for instant auditioning.
3. Open **SETUP**, **NAV** to the **IR Capture** menu, and press **ENTER**.
4. On the **Config** page, ensure that **Capture Mode** is set to "ULTRA-RES" (unless for some odd reason you wish to capture in the older resolution format). **Capture Method** should be set to "MIC ONLY" or "MIC+DI" depending on which setup you are using.
5. **IMPORTANT!** While you are on the **Config** page of the **IR Capture** utility, select a **Bank** and a **Slot** where you want the finished IR to be saved. Enter a **Name** using the functions on the **B**,**C**, **D** and **E** knobs.
6. Press the **TEST** button (Push-knob **A**) to generate a test sweep for setting levels.
7. The **OUTPUT LEVEL 2** knob on the front of the Axe-Fx III sets the level of the sweep signals sent to the power amp. Turn it — and your power amp — up until you hear the test through the speaker. It should be loud enough to push the speaker to "optimal" levels and eliminate background noise interference, but not loud enough to be uncomfortable or cause distortion.
8. The front panel **INPUT 2** LED meter of the Axe-Fx III indicates the signal level coming back to the Axe-Fx III from the mic and preamp or mixer. Adjust the preamp output levels and use **INPUT 2** level on the **Input** page of the **I/O** menu to ensure that the signal is nice and hot without clipping. It is OK to "tickle the red" LED, as this indicates a level of -6 dB, below actual clipping.
9. Once levels are set correctly, it is time to capture. Press the on-screen **CAPTURE** button (Push-knob **B**) and wait quietly for the capture to complete (two sweeps, ~5 seconds).
10. If you set up the test preset in Step 2 correctly, you should hear the IR automatically and be able to audition it at once.
11. Press **SAVE** to store your capture to the location you selected in Step 5.
12. Be sure to select a new **location** and **name** before performing any additional captures with the same setup.

15 ADDITIONAL TOPICS

FRACTAL-BOT



Fractal-Bot is required for **Backing up and Restoring** sounds and settings (below), **Updating Firmware** (p. 173) and can be used for **Transmitting Presets,** or **Installing User Cabs** (p. 175).

The program is self-explanatory and has instructions built in.

To be able to use Fractal-Bot, Windows users will need to install a driver to enable communication between the computer and the Axe-Fx III. The Windows driver is available at https://www.fractalaudio.com/axe-fx-iii-downloads.

Computers running OS X do not require a driver to communicate with the Axe-Fx III.

Download and install **Fractal-Bot** from our web site at http://fractalaudio.com/fractal-bot.php.

BACKING UP & RESTORING

Once Fractal-Bot is installed, you can use it for backing up or restoring or installing files.

BACKING UP

Fractal-Bot automates the backup process. Select the **RECEIVE** tab and follow the built-in instructions. Here are some things to consider when using Fractal-Bot:

- The Axe-Fx III doesn't emphasize this distinction, but its **Presets** are actually divided into banks of 128. To backup all of your presets, backup all of the banks: **A** (0–127), **B** (128-255), **C** (265–383) and **D** (384–511).
- A backup of the **SYSTEM** includes all of the custom settings in the **Setup** menu: **Global**, **I/O**, **MIDI/Remote**, and **Tuner** settings.
- **User Cabs** are backed up in two individual banks.
- In Fractal-Bot, you will need to specify a location for the backup files. It is wise to prepare this in advance. Many pros we work with use a consistent naming system for their backup folders: "**yymmdd - Axe-Fx III**" (two digit year, two digit month, two digit day). In addition to the backup files, create a text file to remind you what firmware version was installed when the files were created.

RESTORING

Fractal-Bot is also used to restore backup files to your Axe-Fx III. In this case, the process is less automated but still very easy. You will need to send over all of the files from your backup set individually. Once the first file completes, send the next file over, and so on, until they are all finished. Remember to *reboot* the Axe-Fx III immediately if a **System** file has been transmitted to it! (Fractal-Bot will remind you.)

You can use the same process to install presets or cab files you download.

MIDI

On the Axe-Fx III, backup and dumping are not supported over 5-pin MIDI. You can transmit backup files to the unit this way, but because the transfer rate is so slow, this can take a LONG time and is advised against.

FIRMWARE UPDATES

Our products are upgradeable, and we are committed to continual improvement. These highly antipated updates add new amps, cabs, effects and features, fix issues, and more. We recommend keeping the firmware in your Axe-Fx III up to date as new versions are released. Firmware is technically software — the "operating system" of your Axe-Fx III — tracked using version numbers: 1.00, 1.01, 2.00, etc. Every Axe-Fx III is shipped from the factory with the latest version installed. You can check your version any time by pressing the **SETUP** button (Push-knob **E**) on the **Home** page. The version is displayed at the top left of the **Setup** menu.

UPDATING

Updating is safe, easy, and quick with Fractal-Bot (see previous page). Before you begin, please QUIT any other applications which might interfere, including Axe-Edit and any DAW software or audio/MIDI applications.

1. **Download** the latest firmware for your Axe-Fx III from https://www.fractalaudio.com/axe-fx-iii-downloads.
2. **Unzip** the download file. The archive may contain various documents in addition to the actual firmware file, which is a MIDI System Exclusive or "SysEx" file. Extract the **.syx** file to a location you can easily find when required. **Do not double-click the .syx file**. Fractal-Bot will prompt you for its location when it is needed.
3. Please read the **Release Notes** included with the firmware download. They'll let you know what's new and alert you to anything you may need to be aware of before updating to the new version.
4. Before performing a major firmware update, it is not unwise to **back up** your Axe-Fx III (see previous page).
5. Launch **Fractal-Bot** and proceed through three self-explanatory steps.
6. Firmware updates will occasionally be accompanied by new versions of the Axe-Fx III Factory Presets. These will be available on our web site support page and can also be installed using Fractal-Bot.

FIRMWARE Q&A

Q: How do I know the upgrade worked?

A: Check your firmware version anytime by pressing the **SETUP** button (Push-knob **E**) on the **Home** page. The version is displayed at the top of the **Setup** menu.

Q: Can I skip versions to go from a very old firmware version to a much newer one?

A: You can upgrade from any version to any other version. When you skip versions it is recommended that you also read the Release Notes for all interim versions, all of which are included with every firmware release.

Q: Will updating my firmware erase, modify or "upgrade" my factory presets?

A: Firmware updates do not erase customized presets. However, firmware updates may alter the *sound* of existing presets. Always read Release Notes before updating. Saving presets after a firmware update can render them incompatible with previous versions. It is always wise to create a full back up using Fractal-Bot before a major update.

Firmware updates do *not* upgrade Factory Presets. Updated Factory Presets, when offered, are released as a separate update on our web site support page. Fractal-Bot can be used to transmit this update in the same way it is used to update firmware.

Q: The firmware update failed mysteriously. What should I do?

A: If the Axe-Fx still boots normally, just try the update again. If successive failures occur, please delete and re-download the update file before trying again. Trying a different USB port or cable can also solve issues.

Q: After updating, my Axe-Fx will no longer boot normally.

A: See "Recovery" on the next page.

RECOVERY

PROBLEMS DURING/AFTER FIRMWARE UPDATE

The Axe-Fx III has a built-in recovery system known as the "Emergency Boot Loader" to protect you against mishaps during firmware update. To boot from the emergency boot loader:

1. Power down the unit and wait five seconds.
2. Power on holding **both** **PAGE LEFT** and **PAGE RIGHT** buttons until the **Emergency Utility** page appears.
3. Update as normal using **Fractal-Bot**.

PROBLEMS WITH A SINGLE PRESET

If your Axe-Fx III will not boot normally, the problem may just be the current preset. You can force the unit to load an empty initialized preset as follows:

4. Power down the unit and wait five seconds.
5. Power on holding **HOME** until the boot-up progress bar first appears.

The Axe-Fx III will boot with an empty initialized preset in location "000". You can STORE this to any other location to overwrite a problematic preset.

PROBLEMS WITH GLOBAL SETTINGS OR PARAMETERS

In the very unlikely event that a problematic system parameter prevents the Axe-Fx III from booting normally, you may use the following procedure to recover:

1. Power down the unit and wait five seconds.
2. Power on holding **EDIT** until the boot-up progress bar first appears.
3. Enter **SETUP**, open the **Utilities** menu, and page to **Reset.**
4. Execute **Reset System Parameters** as indicated on-screen (Push-knob **A** followed by **ENTER**).
   WARNING! Do NOT accidentally perform CLEAR ALL PRESETS!!!
5. Follow the on-screen instructions to execute. Press HOME when finished.



You can in fact hold both **HOME** and **EDIT** at once to reset the current preset *and* prevent the loading of global settings and parameters.

GETTING HELP

Our forum is a source of great help ranging from product Q&A to tutorials and more. Fractal Audio staff participate in the conversation, and response times can be very fast. Find it at https://forum.fractalaudio.com.

Our wiki, maintained by members of the Fractal Audio community, is also an excellent resource: https://wiki.fractalaudio.com.

You can get support directly from Fractal Audio Systems at: https://support.fractalaudio.com
or internationally via our dealers listed at https://www.fractalaudio.com/international-ordering.

LOADING USER CABS

In addition to thousands of onboard cabs in its factory banks, the Axe-Fx III allows you to store up to 2,048 cabs in user memories. These "User Cabs" allow you to personalize your Axe-Fx III with unique tones.

User Cabs — also know as "Impulse Responses (IRs)" — may be transferred to the Axe-Fx III as follows:

1. First, you'll need an impulse response file in SysEx format (.syx).
   - Fractal Audio Systems offers various professionally produced user Cab Packs at https://shop.fractalaudio.com. These include creations by Fractal Audio as well as third-party producers.
   - Axe-Change, our file sharing site, is a great resource for FREE cabs. https://axechange.fractalaudio.com
   - Don't confuse the newer ".IR" file format, which is solely used for re-mixing in Cab-Lab (see below).
2. Using **Fractal-Bot** or **Axe-Edit III**, transmit the file to your Axe-Fx III, noting which **User Bank** and which numbered **Slot** you are sending it to. (Example: User 1, #215.)
   - **Fractal-Bot** – This basic utility can send individual cabs to any location or transmit entire banks. To import a single IR, open a Cab block on the grid and set **Cab 1** to the **Bank** and **Number** you want to transmit a cab to. Then send the sysex file to the Axe-Fx III using Fractal-Bot and the unit will save the IR to that location.
   - **Axe-Edit III** – The **Axe-Manage Cabs** tool allows you to drag and drop user cab files into memory slots. Don't forget to press **SAVE**! You can also manage entries already in the memory of the Axe-Fx III with operations like copy, paste, rename, and drag-and-drop re-ordering. When you purchase a Cab Pack, this is the best and easiest way to load multiple IRs at once into your Axe-Fx III, audition the results, and organize your favorites.
3. Once the IR has been transmitted to your Axe-Fx III, select the **Cab** block in your preset and press **EDIT**.
4. Change the cab to the **Bank** and **Number** that your IR was transmitted to. (Example: "USER 1, #215").

You can even create your own cabs if you have some basic equipment. See "IR Capture" on p. 168.

NOTE: In comparison to older Fractal Audio systems products, user cabs are NOT included in a backup of the SYSTEM area of the Axe-Fx III. Conversely, restoring a System backup will never overwrite user cabs.



Cab-Lab is a full-featured 8-channel IR mixer and toolbox in standalone and plug-in formats.

For decades, artists, producers and engineers have crafted tone using a mixer to blend the sounds of different mics or speakers. While the Axe-Fx III provides four mixer slots, Cab-Lab allows up to 8, and has additional options and tools, with the ability to export Cab files and save mix sessions.

Learn more at:

www.fractalaudio.com/cab-lab-3

AXE-CHANGE

Axe-Change is the official source for sharing Preset and Cab files for your Axe-Fx III and other Fractal Audio Systems products. You can upload your own creations or browse what others have contributed including some high profile artists (though regrettably many artists treat their Axe-Fx presets as guarded secrets).

Find Axe-Change at https://axechange.fractalaudio.com.



FREQUENTLY ASKED QUESTIONS

Q: Why all the technical terminology?

A: The language of the Axe-Fx III is for the most part the universal language of professional audio. This allows the Axe-Fx III to serve the very diverse communities of casual and professional players, producers, engineers, and beyond. The terminology and concepts you will use and learn are not unique to the Axe-Fx III either. Understanding them will help you to master the craft of sound and to communicate with other pros. At the same time, the Axe-Fx III is easier than ever, with dedicated controls and a clear interface that doesn't distract or disrupt the creative flow.

Q: What is "FRFR"?

A: FRFR stands for "full-range, flat response." This acronym is used to describe a "neutral" speaker or speaker system that is designed to reproduce the entire audible spectrum of 20 Hz – 20kHz without emphasis. Examples of FRFR systems would include high-quality studio monitors and properly designed PA systems or monitors. Many manufacturers are also now offering FRFR systems designed specifically for direct guitar applications.

Q: How do I upgrade the firmware of my Axe-Fx III?

A: Use Fractal-Bot, available from www.fractalaudio.com.

Q: Can I load my presets from the Axe-Fx II, AX8, or FX8 into the Axe-Fx III (or vice versa)?

A: No, presets are not cross-compatible, but you can generally transfer parameter settings by hand without unexpected results. As of this writing, all of the same amp models are present, and "factory" speaker cabs of the Axe-Fx II are present in the "Legacy" bank on the Axe-Fx III.

Q: My expression pedal isn't working. What should I do?

A: Expression pedals need to be calibrated and assigned to a parameter, controller or remote function. See "Connecting Pedals Directly" on p. 12 for a basic overview. Are you using a TRS cable? Is it connected to the correct port on the Axe-Fx III? Is it actually an expression pedal? Did you set up a Modifier as described in **Section 9**?

Q: My Axe-Fx III seems much quieter than my Axe-Fx II with similar settings.

A: By default, analog outputs 1 and 2 are now set to -10dBV. To change these to +4dBu, use the **Output Level** parameters for Out 1 and Out 2 on the **Audio** page of the **I/O** menu under **SETUP**. (See p. 160.)

Q: My Axe-Fx III won't start up correctly.

A: Please see "Recovery" on p. 174.

Q: Can I use a computer or external MIDI controller to remote control the Axe-Fx III?

A: Yes. The Axe-Fx III has a rich MIDI spec which allows it to be remote controlled. Using a computer, you can use the built-in USB connection for "MIDI-over-USB", rather than relying on a third-party MIDI interface. See "The MIDI/Remote Menu" on p. 163 and "MIDI Implementation" on p. 188 for more on this.

Q: I'm hearing click and pops.

A: First, check all cables. You'd be surprised by how often a short can be found in a brand new or trusted cable.

Then, check to see whether you are clipping the Axe-Fx III inputs or outputs (p. 5).
Excessive CPU usage may also be to blame. Is the CPU meter at or near 90%? If so, you've overloaded the current preset. Try removing one block and review "Preset CPU Limits" on p. 41 for useful tips.

Q: One or more of my presets produces no sound.

A: This might be any one (or several) of a number of things. Is every other component in your rig working correctly? Most of the time, the problem is a faulty or disconnected cable. Checking the Axe-Fx III with headphones can help rule this possibility out. Using the DELETE button (Push-knob **C**), try changing each block to a SHUNT, ending with the Amp and Cab blocks. Do some presets work? If so, have you double-checked to ensure that each has a complete path from the input to the output? Does every preset begin and end with Input and Output blocks corresponding to inputs and outputs connected to other equipment? Is there a Modifier assigned to a volume or level control while the pedal or external switch is not present? You may simply need to change the INITIAL VALUE for an external controller from 0% to 100% (see p. 164). Does the preset require a USER CAB which is not loaded? Try changing the Cab block to a Factory cab. See also http://wiki.fractalaudio.com/axefx2/index.php?title=No_sound.

Q: Why would I place a certain effect *before* or *after* an Amp and Cab?

A: Sonically speaking, the main reason to care about effect placement is that a given effect will sound different when placed before or after distortion.

How does this difference sound? If you've ever switched the sequence of traditional drive and wah pedals, you've heard an excellent example. In the case of wah before overdrive, the resonant filter of the wah "excites" the overdrive in a cool way while still retaining a natural overall tone. When the wah follows distortion, you might hear a more dramatic sweep that almost sounds synth-like and might be considered less "classic." It's not surprising then that wah would traditionally be a run as a "pre" effect between guitar and amp. The amp's distortion follows the wah. Many other effects fall in this same category.

A different example is found in overdrive with reverb or delay. In the natural world, reverb and echo occur because of open spaces around your guitar amp – like a club or concert hall. These effects would therefore NOT be heard before a distorted amp, but *after* it. Recording studios often add these kinds of effects "post", i.e. at the console—after the mic has picked up the distorted sound from the guitar amp. If you wanted to simulate this natural sounding reverb or delay these effects would likely be run "post." This is not to say that delay or reverb before distortion is a "no-no." Many "legendary" tones came from echo units in front of an amp—but this effect is very different from "post" delay—not only tonally, but also in terms of dynamics.

The good news is that the Axe-Fx III allows you to experiment easily and find what combinations of pre- and post-effects work best for you. Creativity begins where conformity ends.

SHORTCUTS

The Axe-Fx III has several shortcuts and hidden features. These are summarized below.

IN GENERAL

- To show the Layout Grid for the current preset, push the **VALUE** wheel.
- Press **EDIT** to jump to the Edit menu for the currently selected block.

ON THE HOME PAGE

- Press **ENTER** to show the Layout Grid for the current preset.
- Use **NAV LEFT/RIGHT** to select presets and press **NAV UP/DOWN** or turn knob **A** to select scenes.
- Press Push-knob **A** to jump to the Edit menu of the AMP 1 block.

IN THE AMP BLOCK

- On the Output EQ page, press **ENTER** to flatten the EQ.
- On the Output EQ page, press **NAV UP/DOWN** to change the number of bands.

IN THE CAB BLOCK

- **NAV** to the Cab Number field and press **ENTER** to enter the Cab Picker for the selected.

ON THE GRID

- With any block selected, **press-and-hold ENTER** to create a series of shunts and cables to bridge empty space to the right. This will also CLEAR existing connectors between a series of blocks.

IN THE EDIT MENU OF ANY BLOCK

- Press **EDIT** to skip to the Edit menu of the next block, or multiple times to step through all blocks.

TO OPEN THE CONTROLLERS MENU

- Press the **TEMPO** button once.

ON THE SEQUENCER PAGE OF THE CONTROLLERS MENU

- With any **Stage** selected, press **ENTER** to randomize the values of all stages.

SPILLOVER

Spillover allows delay and reverb tails to “ring out” when an effect is bypassed or when you change channels, scenes, or presets. This section covers how to set up spillover in different scenarios.

WHEN BYPASSING A BLOCK (“IA” SWITCHES)...

Block spillover is easy and requires only a particular setting in the block. For tails to ring out when an individual block is bypassed, set that block’s **Bypass Mode** to “MUTE FX IN”. If an effect is running in parallel, use “MUTE IN” instead. Be aware that different channels share effect memory, so changes to time, size, etc, may cause a “sweep” in the tail.

WHEN SWITCHING SCENES...

Switching Scenes provides one of the best ways for sound changes to have perfect spillover. Since Scenes simply bypass or engage blocks one-by-one or in groups, just refer to the instructions above for all blocks in your preset.

WHEN CHANGING PRESETS...

Spillover across different presets is a bit more involved. The first step is to open the **Settings** page of the **Global** menu under **SETUP** and set the **Spillover** parameter to determine whether “DELAY”, “REVERB”, or “BOTH” will spill over when you change presets.

You must also ensure that the exact same Delay or Reverb blocks exist in each of them. These need to be the same block *and* the same number (i.e. **Delay 1** spills over only through **Delay 1**, **Delay 2** through **Delay 2**, etc.).

For spillover to work perfectly, the pair(s) of blocks in both “starting” and “landing” presets must have similar settings and placement on the grid. The moment you change to a new preset, the parameter settings for its blocks will “take over” the duties of processing any tails. So, for example, if you change from a preset where Delay 1 has a time of 500 ms to one where the same value is 100 ms, the tails will suddenly be heard as 100 ms echoes instead. You would likewise hear quite a sudden difference in the tail, for instance, if a delay were placed *after* a clean amp block in one preset, and *before* an overdriven amp block in another! Bypass states and Bypass Mode settings must also be considered.

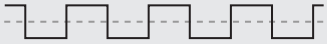
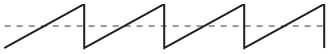
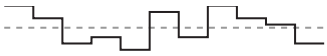
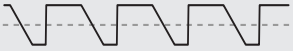


For a simple preset spillover experiment, create a preset, then save an exact copy to a new location and test spillover. Then begin making changes as needed to settings outside of those blocks that you want to spill over. Axe-Edit III also makes it easy to copy and paste a block from one preset to another.

LFO WAVEFORMS & PHASE

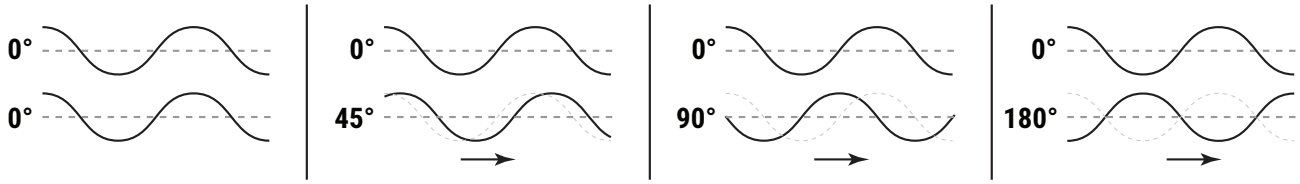
All of the modulation effects (chorus, flanger, delays, phaser, tremolo, etc.) and the two global LFOs, share a common set of waveform types—"shapes" that define the way values change over time. These are represented below, together with an indication of how the **Duty** parameter controls wave symmetry.

Remember that in cases where an LFO modulates the delay time (chorus, flanger, and any delay blocks) it is the slope, rather than the actual LFO "value," that determines the pitch offset at any moment. A triangle with a constant up/down slope will "sound" like the square wave "looks." A square waveform with no effective slope will produce only a series of clicks unless used with some "damping".

Type	50% Duty Cycle (Normal)	0% Duty	100% Duty
Sine		NA	NA
Triangle			
Square			
Saw Up		NA	NA
Saw Down		NA	NA
Random		NA	NA
Log		NA	NA
Exponential (Exp)		NA	NA
Trapezoid			

LFO PHASE

Phase adjustments shift the alignment of the "RIGHT" or "B" LFO output. At 0° (below, far left), the two channels are in phase; at 180° (below, far right), the two signals are phase-opposite, so while one is swinging from 0 to 1, the other is swinging from 1 to 0 (and vice versa). Any interim setting is also allowed.
Phase has no effect on the RANDOM waveform.



TEMPO CROSS REFERENCE

The tables below list the rhythmic values available in every Tempo parameter on the Axe-Fx III. The first lists these in the order they appear as you scroll through the Tempo list, with their equivalent in Beats.

The second table (cross-referenced to the first by the INDEX) lists tempo time values from shortest to longest.

Values in Actual Order, with Beat Equivalents

INDEX	VALUE	BEATS
1	1/64 trip	0.042
2	1/64	0.063
3	1/64 dot	0.094
4	1/32 trip	0.083
5	1/32	0.125
6	1/32 dot	0.188
7	1/16 trip	0.167
8	1/16	0.25
9	1/16 dot	0.375
10	1/8 trip	0.333
11	1/8	0.5
12	1/8 dot	0.75
13	1/4 trip	0.667
14	1/4	1
15	1/4 dot	1.5
16	1/2 trip	1.333
17	1/2	2
18	1/2 dot	3
19	1 trip	2.667
20	1	4
21	1 dot	6
22	2	8
23	3	12
24	4	16
25	4/3	5.333
26	5/4	5
27	5/64	0.313
28	7/64	0.438
29	9/64	0.563
30	10/64 (5/32)	0.625
31	11/64	0.688
32	13/64	0.813
33	14/64 (7/32)	0.875
34	15/64	0.938
35	17/64	1.063
36	18/64 (9/32)	1.125
37	19/64	1.188
38	20/64 (5/16)	1.25
39	21/64	1.313
40	22/64 (11/32)	1.375
41	23/64	1.438
42	25/64	1.563
43	26/64 (13/32)	1.625
44	27/64	1.688
45	28/64 (7/16)	1.75
46	29/64	1.813
47	30/64 (15/32)	1.875
48	31/64	1.938
49	33/64	2.063
50	34/64 (17/32)	2.125
51	35/64	2.188
52	36/64 (9/16)	2.250
53	37/64	2.313
54	38/64 (19/32)	2.375
55	39/64	2.438
56	40/64 (5/8)	2.5
57	41/64	2.563
58	42/64 (21/32)	2.625
59	43/64	2.688
60	44/64 (11/16)	2.75
61	45/64	2.813
62	46/64 (23/32)	2.875
63	47/64	2.938
64	49/64 (49/64)	3.063
65	50/64 (25/32)	3.125
66	51/64	3.188
67	52/64 (13/16)	3.25
68	53/64	3.313
69	54/64 (27/32)	3.375
70	55/64	3.438
71	56/64 (7/8)	3.5
72	57/64	3.563
73	58/64 (29/32)	3.625
74	59/64	3.688
75	60/64 (15/16)	3.75
76	61/64	3.813
77	62/64 (31/32)	3.875
78	63/64	3.938

Values from Shortest to Longest

VALUE	AKA	INDEX
1/64 trip	1/96th	1
1/64		2
1/32 trip	1/48th	4
1/64 dot		3
1/32		5
1/16 trip	1/24th	7
1/32 dot	3/64th	6
1/16		8
5/64		27
1/8 trip	1/12th	10
1/16 dot	3/32nd	9
7/64		28
1/8th		11
9/64		29
10/64 (5/32)		30
1/4 trip	1/6th	13
11/64		31
1/8 dot	3/16th	12
13/64		32
14/64 (7/32)		33
15/64		34
1/4		14
17/64		35
18/64 (9/32)		36
19/64		37
20/64 (5/16)		38
21/64		39
1/2 trip	1/3rd	16
22/64 (11/32)		40
23/64		41
1/4 dot	3/8th	15
25/64		42
26/64 (13/32)		43
27/64		44
28/64 (7/16)		45
29/64		46
30/64 (15/32)		47
31/64		48
1/2		17
33/64		49
34/64 (17/32)		50
35/64		51
36/64 (9/16)		52
37/64		53
38/64 (19/32)		54
39/64		55
40/64 (5/8)		56
41/64		57
42/64 (21/32)		58
1 trip	2/3	19
43/64		59
44/64 (11/16)		60
45/64		61
46/64 (23/32)		62
47/64		63
1/2 dot	3/4	18
49/64 (49/64)		64
50/64 (25/32)		65
51/64		66
52/64 (13/16)		67
53/64		68
54/64 (27/32)		69
55/64		70
56/64 (7/8)		71
57/64		72
58/64 (29/32)		73
59/64		74
60/64 (15/16)		75
61/64		76
62/64 (31/32)		77
63/64		78
1	whole	20
5/4		26
4/3		25
1 dot	3/2	21
2	2 bars	22
3	3 bars	23
4	4 bars	24

MIDI REFERENCE TABLES

MIDI BANK & PROGRAM CHANGE

The following table lists the MIDI Bank and Program Change messages required to select Axe-Fx III presets.
MIDI Bank Select (CC#0) Value , Midi Program Change = Axe-Fx III Preset Number.

0, 0 = 0	0, 42 = 42	0, 84 = 84	0, 126 = 126	1, 40 = 168
0, 1 = 1	0, 43 = 43	0, 85 = 85	0, 127 = 127	1, 41 = 169
0, 2 = 2	0, 44 = 44	0, 86 = 86	1, 0 = 128	1, 42 = 170
0, 3 = 3	0, 45 = 45	0, 87 = 87	1, 1 = 129	1, 43 = 171
0, 4 = 4	0, 46 = 46	0, 88 = 88	1, 2 = 130	1, 44 = 172
0, 5 = 5	0, 47 = 47	0, 89 = 89	1, 3 = 131	1, 45 = 173
0, 6 = 6	0, 48 = 48	0, 90 = 90	1, 4 = 132	1, 46 = 174
0, 7 = 7	0, 49 = 49	0, 91 = 91	1, 5 = 133	1, 47 = 175
0, 8 = 8	0, 50 = 50	0, 92 = 92	1, 6 = 134	1, 48 = 176
0, 9 = 9	0, 51 = 51	0, 93 = 93	1, 7 = 135	1, 49 = 177
0, 10 = 10	0, 52 = 52	0, 94 = 94	1, 8 = 136	1, 50 = 178
0, 11 = 11	0, 53 = 53	0, 95 = 95	1, 9 = 137	1, 51 = 179
0, 12 = 12	0, 54 = 54	0, 96 = 96	1, 10 = 138	1, 52 = 180
0, 13 = 13	0, 55 = 55	0, 97 = 97	1, 11 = 139	1, 53 = 181
0, 14 = 14	0, 56 = 56	0, 98 = 98	1, 12 = 140	1, 54 = 182
0, 15 = 15	0, 57 = 57	0, 99 = 99	1, 13 = 141	1, 55 = 183
0, 16 = 16	0, 58 = 58	0, 100 = 100	1, 14 = 142	1, 56 = 184
0, 17 = 17	0, 59 = 59	0, 101 = 101	1, 15 = 143	1, 57 = 185
0, 18 = 18	0, 60 = 60	0, 102 = 102	1, 16 = 144	1, 58 = 186
0, 19 = 19	0, 61 = 61	0, 103 = 103	1, 17 = 145	1, 59 = 187
0, 20 = 20	0, 62 = 62	0, 104 = 104	1, 18 = 146	1, 60 = 188
0, 21 = 21	0, 63 = 63	0, 105 = 105	1, 19 = 147	1, 61 = 189
0, 22 = 22	0, 64 = 64	0, 106 = 106	1, 20 = 148	1, 62 = 190
0, 23 = 23	0, 65 = 65	0, 107 = 107	1, 21 = 149	1, 63 = 191
0, 24 = 24	0, 66 = 66	0, 108 = 108	1, 22 = 150	1, 64 = 192
0, 25 = 25	0, 67 = 67	0, 109 = 109	1, 23 = 151	1, 65 = 193
0, 26 = 26	0, 68 = 68	0, 110 = 110	1, 24 = 152	1, 66 = 194
0, 27 = 27	0, 69 = 69	0, 111 = 111	1, 25 = 153	1, 67 = 195
0, 28 = 28	0, 70 = 70	0, 112 = 112	1, 26 = 154	1, 68 = 196
0, 29 = 29	0, 71 = 71	0, 113 = 113	1, 27 = 155	1, 69 = 197
0, 30 = 30	0, 72 = 72	0, 114 = 114	1, 28 = 156	1, 70 = 198
0, 31 = 31	0, 73 = 73	0, 115 = 115	1, 29 = 157	1, 71 = 199
0, 32 = 32	0, 74 = 74	0, 116 = 116	1, 30 = 158	1, 72 = 200
0, 33 = 33	0, 75 = 75	0, 117 = 117	1, 31 = 159	1, 73 = 201
0, 34 = 34	0, 76 = 76	0, 118 = 118	1, 32 = 160	1, 74 = 202
0, 35 = 35	0, 77 = 77	0, 119 = 119	1, 33 = 161	1, 75 = 203
0, 36 = 36	0, 78 = 78	0, 120 = 120	1, 34 = 162	1, 76 = 204
0, 37 = 37	0, 79 = 79	0, 121 = 121	1, 35 = 163	1, 77 = 205
0, 38 = 38	0, 80 = 80	0, 122 = 122	1, 36 = 164	1, 78 = 206
0, 39 = 39	0, 81 = 81	0, 123 = 123	1, 37 = 165	1, 79 = 207
0, 40 = 40	0, 82 = 82	0, 124 = 124	1, 38 = 166	1, 80 = 208
0, 41 = 41	0, 83 = 83	0, 125 = 125	1, 39 = 167	1, 81 = 209

1, 82 = 210	2, 6 = 262	2, 58 = 314	2, 110 = 366	3, 34 = 418	3, 86 = 470
1, 83 = 211	2, 7 = 263	2, 59 = 315	2, 111 = 367	3, 35 = 419	3, 87 = 471
1, 84 = 212	2, 8 = 264	2, 60 = 316	2, 112 = 368	3, 36 = 420	3, 88 = 472
1, 85 = 213	2, 9 = 265	2, 61 = 317	2, 113 = 369	3, 37 = 421	3, 89 = 473
1, 86 = 214	2, 10 = 266	2, 62 = 318	2, 114 = 370	3, 38 = 422	3, 90 = 474
1, 87 = 215	2, 11 = 267	2, 63 = 319	2, 115 = 371	3, 39 = 423	3, 91 = 475
1, 88 = 216	2, 12 = 268	2, 64 = 320	2, 116 = 372	3, 40 = 424	3, 92 = 476
1, 89 = 217	2, 13 = 269	2, 65 = 321	2, 117 = 373	3, 41 = 425	3, 93 = 477
1, 90 = 218	2, 14 = 270	2, 66 = 322	2, 118 = 374	3, 42 = 426	3, 94 = 478
1, 91 = 219	2, 15 = 271	2, 67 = 323	2, 119 = 375	3, 43 = 427	3, 95 = 479
1, 92 = 220	2, 16 = 272	2, 68 = 324	2, 120 = 376	3, 44 = 428	3, 96 = 480
1, 93 = 221	2, 17 = 273	2, 69 = 325	2, 121 = 377	3, 45 = 429	3, 97 = 481
1, 94 = 222	2, 18 = 274	2, 70 = 326	2, 122 = 378	3, 46 = 430	3, 98 = 482
1, 95 = 223	2, 19 = 275	2, 71 = 327	2, 123 = 379	3, 47 = 431	3, 99 = 483
1, 96 = 224	2, 20 = 276	2, 72 = 328	2, 124 = 380	3, 48 = 432	3, 100 = 484
1, 97 = 225	2, 21 = 277	2, 73 = 329	2, 125 = 381	3, 49 = 433	3, 101 = 485
1, 98 = 226	2, 22 = 278	2, 74 = 330	2, 126 = 382	3, 50 = 434	3, 102 = 486
1, 99 = 227	2, 23 = 279	2, 75 = 331	2, 127 = 383	3, 51 = 435	3, 103 = 487
1, 100 = 228	2, 24 = 280	2, 76 = 332	3, 0 = 384	3, 52 = 436	3, 104 = 488
1, 101 = 229	2, 25 = 281	2, 77 = 333	3, 1 = 385	3, 53 = 437	3, 105 = 489
1, 102 = 230	2, 26 = 282	2, 78 = 334	3, 2 = 386	3, 54 = 438	3, 106 = 490
1, 103 = 231	2, 27 = 283	2, 79 = 335	3, 3 = 387	3, 55 = 439	3, 107 = 491
1, 104 = 232	2, 28 = 284	2, 80 = 336	3, 4 = 388	3, 56 = 440	3, 108 = 492
1, 105 = 233	2, 29 = 285	2, 81 = 337	3, 5 = 389	3, 57 = 441	3, 109 = 493
1, 106 = 234	2, 30 = 286	2, 82 = 338	3, 6 = 390	3, 58 = 442	3, 110 = 494
1, 107 = 235	2, 31 = 287	2, 83 = 339	3, 7 = 391	3, 59 = 443	3, 111 = 495
1, 108 = 236	2, 32 = 288	2, 84 = 340	3, 8 = 392	3, 60 = 444	3, 112 = 496
1, 109 = 237	2, 33 = 289	2, 85 = 341	3, 9 = 393	3, 61 = 445	3, 113 = 497
1, 110 = 238	2, 34 = 290	2, 86 = 342	3, 10 = 394	3, 62 = 446	3, 114 = 498
1, 111 = 239	2, 35 = 291	2, 87 = 343	3, 11 = 395	3, 63 = 447	3, 115 = 499
1, 112 = 240	2, 36 = 292	2, 88 = 344	3, 12 = 396	3, 64 = 448	3, 116 = 500
1, 113 = 241	2, 37 = 293	2, 89 = 345	3, 13 = 397	3, 65 = 449	3, 117 = 501
1, 114 = 242	2, 38 = 294	2, 90 = 346	3, 14 = 398	3, 66 = 450	3, 118 = 502
1, 115 = 243	2, 39 = 295	2, 91 = 347	3, 15 = 399	3, 67 = 451	3, 119 = 503
1, 116 = 244	2, 40 = 296	2, 92 = 348	3, 16 = 400	3, 68 = 452	3, 120 = 504
1, 117 = 245	2, 41 = 297	2, 93 = 349	3, 17 = 401	3, 69 = 453	3, 121 = 505
1, 118 = 246	2, 42 = 298	2, 94 = 350	3, 18 = 402	3, 70 = 454	3, 122 = 506
1, 119 = 247	2, 43 = 299	2, 95 = 351	3, 19 = 403	3, 71 = 455	3, 123 = 507
1, 120 = 248	2, 44 = 300	2, 96 = 352	3, 20 = 404	3, 72 = 456	3, 124 = 508
1, 121 = 249	2, 45 = 301	2, 97 = 353	3, 21 = 405	3, 73 = 457	3, 125 = 509
1, 122 = 250	2, 46 = 302	2, 98 = 354	3, 22 = 406	3, 74 = 458	3, 126 = 510
1, 123 = 251	2, 47 = 303	2, 99 = 355	3, 23 = 407	3, 75 = 459	3, 127 = 511
1, 124 = 252	2, 48 = 304	2, 100 = 356	3, 24 = 408	3, 76 = 460	
1, 125 = 253	2, 49 = 305	2, 101 = 357	3, 25 = 409	3, 77 = 461	
1, 126 = 254	2, 50 = 306	2, 102 = 358	3, 26 = 410	3, 78 = 462	
1, 127 = 255	2, 51 = 307	2, 103 = 359	3, 27 = 411	3, 79 = 463	
2, 0 = 256	2, 52 = 308	2, 104 = 360	3, 28 = 412	3, 80 = 464	
2, 1 = 257	2, 53 = 309	2, 105 = 361	3, 29 = 413	3, 81 = 465	
2, 2 = 258	2, 54 = 310	2, 106 = 362	3, 30 = 414	3, 82 = 466	
2, 3 = 259	2, 55 = 311	2, 107 = 363	3, 31 = 415	3, 83 = 467	
2, 4 = 260	2, 56 = 312	2, 108 = 364	3, 32 = 416	3, 84 = 468	
2, 5 = 261	2, 57 = 313	2, 109 = 365	3, 33 = 417	3, 85 = 469	

CC VALUE TO SCENE

When selecting Scenes using the global options found on the **Other** page of the **MIDI/Remote** menu under **SETUP**, the *value* of the designated CC# determines the Scene:

0. 1	16. 1	32. 1	48. 1	64. 1	80. 1	96. 1	112. 1
1. 2	17. 2	33. 2	49. 2	65. 2	81. 2	97. 2	113. 2
2. 3	18. 3	34. 3	50. 3	66. 3	82. 3	98. 3	114. 3
3. 4	19. 4	35. 4	51. 4	67. 4	83. 4	99. 4	115. 4
4. 5	20. 5	36. 5	52. 5	68. 5	84. 5	100. 5	116. 5
5. 6	21. 6	37. 6	53. 6	69. 6	85. 6	101. 6	117. 6
6. 7	22. 7	38. 7	54. 7	70. 7	86. 7	102. 7	118. 7
7. 8	23. 8	39. 8	55. 8	71. 8	87. 8	103. 8	119. 8
8. 1	24. 1	40. 1	56. 1	72. 1	88. 1	104. 1	120. 1
9. 2	25. 2	41. 2	57. 2	73. 2	89. 2	105. 2	121. 2
10. 3	26. 3	42. 3	58. 3	74. 3	90. 3	106. 3	122. 3
11. 4	27. 4	43. 4	59. 4	75. 4	91. 4	107. 4	123. 4
12. 5	28. 5	44. 5	60. 5	76. 5	92. 5	108. 5	124. 5
13. 6	29. 6	45. 6	61. 6	77. 6	93. 6	109. 6	125. 6
14. 7	30. 7	46. 7	62. 7	78. 7	94. 7	110. 7	126. 7
15. 8	31. 8	47. 8	63. 8	79. 8	95. 8	111. 8	127. 8

CC VALUE TO CHANNEL

When changing Channels using the global options found on the **Channel** page of the **MIDI/Remote** menu under **SETUP**, the *value* of the designated CC# determines the Channel:

0. . . . . A	16. . . . . A	32. . . . . A	48. . . . . A	64. . . . . A	80. . . . . A	96. . . . . A	112. . . . . A
1. . . . . B	17. . . . . B	33. . . . . B	49. . . . . B	65. . . . . B	81. . . . . B	97. . . . . B	113. . . . . B
2. C	18. C	34. C	50. C	66. C	82. C	98. C	114. C
3. . . . . D	19. . . . . D	35. . . . . D	51. . . . . D	67. . . . . D	83. . . . . D	99. . . . . D	115. . . . . D
4. . . . . A	20. . . . . A	36. . . . . A	52. . . . . A	68. . . . . A	84. . . . . A	100. . . . . A	116. . . . . A
5. . . . . B	21. . . . . B	37. . . . . B	53. . . . . B	69. . . . . B	85. . . . . B	101. . . . . B	117. . . . . B
6. C	22. C	38. C	54. C	70. C	86. C	102. C	118. C
7. . . . . D	23. . . . . D	39. . . . . D	55. . . . . D	71. . . . . D	87. . . . . D	103. . . . . D	119. . . . . D
8. . . . . A	24. . . . . A	40. . . . . A	56. . . . . A	72. . . . . A	88. . . . . A	104. . . . . A	120. . . . . A
9. . . . . B	25. . . . . B	41. . . . . B	57. . . . . B	73. . . . . B	89. . . . . B	105. . . . . B	121. . . . . B
10. C	26. C	42. C	58. C	74. C	90. C	106. C	122. C
11. . . . . D	27. . . . . D	43. . . . . D	59. . . . . D	75. . . . . D	91. . . . . D	107. . . . . D	123. . . . . D
12. . . . . A	28. . . . . A	44. . . . . A	60. . . . . A	76. . . . . A	92. . . . . A	108. . . . . A	124. . . . . A
13. . . . . B	29. . . . . B	45. . . . . B	61. . . . . B	77. . . . . B	93. . . . . B	109. . . . . B	125. . . . . B
14. C	30. C	46. C	62. C	78. C	94. C	110. C	126. C
15. . . . . D	31. . . . . D	47. . . . . D	63. . . . . D	79. . . . . D	95. . . . . D	111. . . . . D	127. . . . . D

16 SPECIFICATIONS

INSTRUMENT INPUTS

Connectors:	Front and rear 1/4" phone jacks, unbalanced *(front automatically overrides rear)*
Impedance:	1 MΩ (less if “Auto-Z” is active)
Max. Input Level:	+16 dBu

ANALOG INPUT 2

Connectors:	(2) XLR-F + 1/4" phone “combo jack” balanced
Impedance:	1 MΩ
Max. Input Level:	+20 dBu

ANALOG INPUT 3

Connectors:	(2) 1/4" phone jack balanced (TRS)
Impedance:	1 MΩ
Max. Input Level:	+20 dBu

ANALOG INPUT 4

Connectors:	(2) 1/4" phone jack balanced (TRS)
Impedance:	1 MΩ
Max. Input Level:	+20 dBu

A/D CONVERSION

Bit Depth:	24 bits
Sample Rate:	48 kHz
Dynamic Range:	114 dB
Frequency Response:	20 – 20kHz, +0 / -1 dB
Crosstalk:	< -60 dB over full bandwidth

ANALOG OUTPUT 1

Connectors:	(2) XLR balanced with Ground Lift (2) 1/4" phone jack unbalanced (hum-canceling) *Selectable between -10 dBV and +4 dBu in software*
Impedance:	600 Ω
Max Output Level:	+20 dBu

ANALOG OUTPUT 2

Connectors:	(2) XLR-M balanced with Ground Lift *Selectable between -10 dBV and +4 dBu in software*
Impedance:	600 Ω
Max Output Level:	+20 dBu

ANALOG OUTPUT 3

Connectors:	(2) 1/4" phone jack unbalanced (hum-canceling)
Impedance:	600 Ω
Max Output Level:	+20 dBu

ANALOG OUTPUT 4

Connectors:	(2) 1/4" phone jack unbalanced (hum-canceling)
Impedance:	600 Ω
Max Output Level:	+20 dBu

HEADPHONE OUTPUT

Connector:	1/4" stereo phone jack
Impedance:	35 Ω

D/A CONVERSION

Dynamic Range:	114 dB
Frequency Response:	20 – 20kHz, +0 / -1 dB

DIGITAL I/O

Connectors:	RCA Coaxial Type for S/PDIF, XLR for AES *(selectable in software)*
Format:	Uncompressed PCM
Sample Rate:	48 kHz fixed

USB AUDIO

Format:	USB 2.0 Class compliant
Channels:	16 *(8 in, 8 out, each mapped to a specific source or connector)*
USB Audio Clock:	48 kHz fixed

MIDI INTERFACE

Input Connector:	(1) 5-pin DIN
Output Connector:	(1) 5-pin DIN
Thru Connector:	(1) 5-pin DIN

PEDAL INTERFACE

Connectors:	(2) 1/4" TRS phone jack
Format:	Pedal: 10–100 kΩ max
	Switch: make/break; momentary or latching

FASLINK II INTERFACE

Connectors:	(1) XLR-F

WARNING: Connect ONLY to the FASLINK II connector on a Fractal Audio FC Series controller

GENERAL

Finish:	Powder coated steel chassis with anodized aluminum faceplate
Controls:	12 buttons, 10 knobs *(6 with additional "push" functions)*
Display:	800×480 high contrast color LCD
Dimensions:	19.00" W. × 5.16" H. × 11.53" D.
	482.6mm W. × 131.1mm H. × 292.87mm D.
	(including front handles and rear jacks)
Weight:	15 lbs 2.4 oz (6.87 kg)
Input Voltage:	90–264 VAC, 47 – 63 Hz *(universal input)*
Power Consumption:	<40 W
Backup Battery Life:	>10 years
Backup Battery Type:	CR-2032

ENVIRONMENTAL

Operating Temperature:	32 to 122 °F (0 to 50 °C)
Storage Temperature:	-22 to 167 °F (-30 to 70 °C)
Humidity:	Max. 90% non-condensing

MIDI IMPLEMENTATION

The Axe-Fx III features a robust MIDI implementation detailed below.

Function		Tx	Rx	Remarks
Basic Channel	Default	1	1	
	Changed	1-16	1-16	
Note Number	True Voice	X	X	
Velocity	Note ON	X	X	
	Note OFF	X	X	
After Touch	Keys	X	X	
	Channels	X	X	
Pitch Bend		X	X	
Control Change		X	O	Receivable CCs are globally soft-assigned to functions via the MIDI/Remote menu under SETUP. These include Input and Output volumes, Tap Tempo, Tuner, 16 “External Controllers” (assignable as modifiers to one or more parameters on a per-preset basis), some Scene functions, all Looper functions, and all block BYPASS and CHANNEL switches.
Program Change	True Number	O	O	The Axe-Fx III can transmit CC and PC messages.
	Bank Select	X	O	
System Exclusive	Fractal Audio	O	O	SysEx is used extensively for Axe-Edit III.
	Real time	O	X	
	Non-Real time	X	X	
System Common	Song Position	X	X	
	Song Select	X	X	
	Tune Request	X	X	
System Real time	Clock	X	O	Axe-Fx III Global Tempo syncs automatically to MIDI Beat Clock. Axe-Fx III does not transmit MIDI clock.
	Commands	X	X	
Auxiliary Messages	Local ON/OFF	X	X	
	All Notes OFF	X	X	
	Active Sense	X	X	
	Reset	X	X	

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