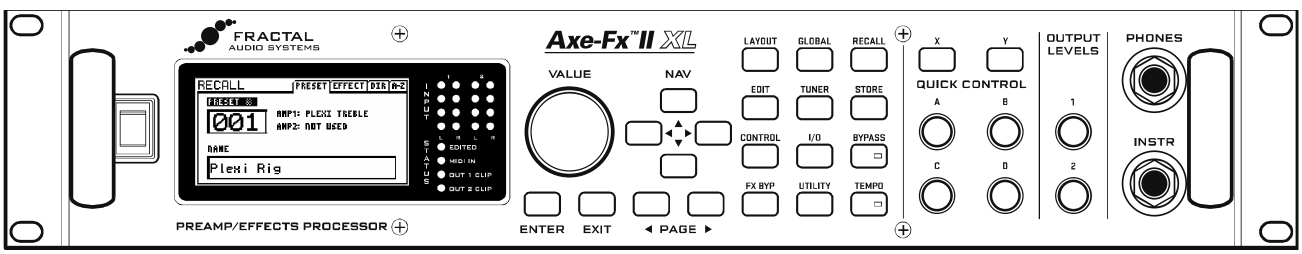
Axe-Fx™ II PREAMP/ FX PROCESSOR



OWNER'S MANUAL

Axe-Fx II XL/XL+

ALSO COVERS AXE-FX II ORIGINAL & MARK II MODELS

FRACTAL
AUDIO SYSTEMS

Legal Notices

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

Doc Q7.0

February 12, 2017

Certificate of Conformity

Fractal Audio Systems, USA, hereby declares on its own responsibility that the following products:

Axe-Fx II Digital Guitar Preamplifier and Effects Processor
Axe-Fx II Mark II Digital Guitar Preamplifier and Effects Processor
Axe-Fx II XL Digital Guitar Preamplifier and Effects Processor
Axe-Fx II XL+ Digital Guitar Preamplifier and Effects Processor

That are covered by this certificate, and marked with CE label, conform to following standards:

EN60065 (IEC 60065)	Safety requirement for mains operated electronic and related apparatus for household and similar use.
EN 55103-1	Product family standard for audio, video, audio-visual, and entertainment lighting control apparatus for professional use. Part 1: Emission.
EN 55103-2	Product family standard for audio, video, audio-visual, and entertainment lighting control apparatus for professional use. Part 2: Immunity.

with reference to regulations in following directives: 73/23/EEC, 89/336/EEC.

June 2014
Clifford Chase, President
Fractal Audio Systems

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.

About the Author

Matt Picone is a music technology product specialist, sound designer, creative director, and musician with over 35 years of experience spanning guitars, amps, effects, synthesizers, software, and beyond. He has worked with artists including Dweezil Zappa, Adrian Belew, Steve Vai, John Petrucci, the Edge, Neal Schon, Periphery, Animals As Leaders, Metallica, Scott Appleton (Def Leppard/Rush/etc.) and more. This work is based extensively on the original Axe-Fx manual by Fractal Audio founder and Axe-Fx creator Cliff Chase. Many thanks to our team of dedicated beta-testers, preset creators, copy-editors and proofreaders.

You may report manual corrections or suggestions in our forum at http://forum.fractalaudio.com

Foreword

Thank you for purchasing an Axe-Fx II, one of the most powerful musical instrument processors ever produced. Please take the time to read through this manual to become acquainted with the Axe-Fx II.

Thinking back to a date when the first Axe-Fx units rolled off the line back in 2006, it would have been a challenge to predict the scale of what was to follow… that the product would be such a worldwide success that we would have a hard time keeping it in stock; that musicians would rally around the unit, from online “Axe-evangelists” to the world’s most celebrated pro players; that we’d soon be writing the foreword to a manual for the sequel: the Axe-Fx II.

Nevertheless, the Axe-Fx II is here. Advances in technology and knowledge, along with the shared insights of our community, have allowed us to design and produce a next-generation product that represents a giant step forward. If you owned a Standard or an Ultra, we think you’ll be very impressed with all the updates, additions and improvements. If you’re new to the Axe-Fx family, this is an incredible place to start.

It has been said that the Axe-Fx “restored digital to its rightful place as the superior solution to musical effects processing.” Every aspect of the Axe-Fx II has been designed to deliver the latest word in this commentary. It has twice the power of the Axe-Fx Ultra (while even the older “Standard” still has more horsepower under the hood than any competitor). For the player, this means better sound, smarter features, and improved performance.

In 2015 we introduced the Axe-Fx II XL+, with upgraded/expanded memory and peripherals. We think it is the best of our products ever, and we hope you agree. Now, with the new releases in our latest generation *Quantum firmware* series, every Axe-Fx II sounds equally great (and better than ever!) with accurate amp models, masterfully created virtual speaker cabs, and effects that truly stand apart. We thank you all for your many contributions, and for choosing Fractal Audio Systems!

—Fractal Audio Systems, January 2017

Table of Contents

What's New

Years of R&D at Fractal Audio Systems have yielded our next-generation product, the Axe-Fx II. With twice the power of our former flagship the Ultra, the Axe-Fx II unveils new state-of-the-art algorithms and an innovative array of great hardware and software features and improvements. This all-in-one preamp/effects processor recreates complete signal chains—stompboxes, amps, cabs, mics, studio effects, and more—with unprecedented power, flexibility, and control. The Axe-Fx II takes "real amp tone and feel" to the next level, offering the latest word on restoring digital to its rightful place as the superior solution for guitar processing.

Twice the Processing Power

Our philosophy is never to cut corners on processing power. Our state-of-the-art algorithms require a powerful platform on which to operate, so the Axe-Fx II features two 600 MHz dual-core Analog Devices TigerSHARC™ Digital Signal Processors working in tandem. One is devoted solely to amp modeling while the other handles effects and system tasks. Mated to the processors is double the RAM of previous Axe-Fx products. The Axe-Fx II is by far the most powerful instrument processor ever created, with more raw, real-time audio processing horsepower than anything available at any price. Yet, unlike power-hungry PCs, it consumes less than 40W.

Quantum™ Amp Modeling™ with Virtual Vacuum Tube™ Technology and MIMIC™

All this power would be useless without superior algorithms to take advantage of it. Years of research have yielded **Quantum Amp Modeling Technology**, comprising major breakthroughs in both preamp and power amp modeling. Quantum is the pinnacle of our multi-generational amp modeling and a complete departure from the static waveshaping technology used by other products. It entails digital replicas of vacuum tubes, complete with time, frequency, and level dependencies. This creates a level of dynamic realism in a class of its own. Like our "G3" modeling before it, **Quantum™** also models the entire power amp, including the phase inverter, power tubes, output transformer, power transformer, choke, filter caps, and more. The results are amazing: tight bass, powerful midrange, silky highs, plus highly expressive touch sensitivity.

Our amp models are the result of thousands of hours of incredibly detailed analysis of the actual amps that inspired them. We spent a small fortune searching out and purchasing vintage and modern amplifiers to add to our reference collection. **Quantum™** is a significant advancement in amplifier simulation unmatched by any other product at any price point.

Improved Speaker Simulation, On-board IR Capture and UltraRes™

The Axe-Fx II cabinet simulator supports our patent-pending UltraRes™ Impulse Response format, with 170+* factory IRs including creations by Fractal Audio Systems, ML Sound Lab, OwnHammer, RedWirez, Jay Mitchell, James Santiago, TheAmpFactory, and John Petrucci of Dream Theater — plus 1024* "USER CAB" memory locations and built-in tools to capture your own IRs from a real speaker cab. UltraRes™ is a proprietary technique that enhances the spectral resolution of an IR without adding CPU burden or storage requirements.

Tone Matching™ to "Clone" any Tone

The Axe-Fx II features a Tone Matching block with the capability to match the sound of a real or recorded amp. It does this by analyzing the difference between the sound of your preset (the "local" signal) and a "reference" (usually the signal from one or more mics/preamps on a real amp, or a high quality recording.) Tone matching eliminates guesswork to create exact "clones" of your favorite tone. A separate Tone Match Mini-manual is provided.

*The Axe-Fx II Mark 2 has only 130+ factory IRs and 100 user IR slots.

Easier-to-Use Front Panel Features

A new, custom-designed 160x80 backlit LCD provides improved readability and more spacious screen layouts. In addition to the main VALUE knob, new QUICK CONTROL knobs provide hands-on access to four additional on-screen parameters. Many block types including Amp, Cab, Chorus, Drive, Delay, Flanger, Phaser, Pitch, Reverb, and Wahwah—are now equipped with two fully independent parameter sets called "X" and "Y." The X/Y switching feature allows one block to have all of its settings switched at the touch of a button (during editing) or via MIDI remote control (during a performance). The X and Y buttons also double as user-definable "quick jump" keys that can be set up to open the EDIT menus of any two blocks without going through the grid.

Axe-Fx II/Computer Integration with Onboard USB

The new onboard Audio Class 2.0 compliant USB interface provides great capabilities for recording and computer integration. You can record high quality 48k/24-bit audio from the Axe-Fx II directly to the computer, play or process audio tracks from the computer through the Axe-Fx II, and use two-way high-speed MIDI without a 3rd-party interface. On USB 2.0 or better systems, you can simultaneously record both the main processed stereo outs and a pair of dry channels for easy re-amping.

New I/O Capabilities and Less Noise

All rear analog inputs are now balanced, as are the onboard XLR outputs. The ¼" unbalanced outputs feature our **Humbuster**™ technology, which senses and subtracts the ground noise of equipment connected when you use a special Humbuster™ cable. This can provide up to 20 dB reduction in ground noise without resorting to dangerous "cheater plugs" or expensive isolation transformers.

We designed the Axe-Fx II with the "Four-Cable Method" in mind. Special analog processing keeps the noise floor even lower on outputs designed to be connected to the front of an amplifier.

The front panel input uses a proprietary circuit and dedicated A/D converter for astonishingly low noise. The original Axe-Fx was hailed for its low-noise performance; the Axe-Fx II provides an almost 10 dB SNR improvement with the same pristine quality. The XL+ has the lowest noise levels yet. A high-quality headphone jack is also provided.

Designed for Unity Gain

The Axe-Fx II uses digitally controlled potentiometers to operate as a unity-gain device irrespective of the input trim controls. Simply set the input trims with the LED input meters and you are done. Another benefit of this technique is that Amp and Drive blocks are unaffected by trim settings.

Improved Digital I/O

In addition to its USB interface, the Axe-Fx II sports SPDIF and AES input and output connectors. 7-pin MIDI In and a selectable MIDI Out/Thru jack are provided for interconnection with other MIDI-controllable equipment.

Built for MFC-101 and FASLINK™

Both the XL and the **Axe-Fx II Mark II** feature an EtherCON port for connecting an MFC-101 MIDI Foot Controller via network cables (the original Axe-Fx II has an Ethernet port only). The Axe-Fx II XL and XL+ also feature an onboard **FASLINK**™ port for connecting an MFC-101 Mark III over a standard XLR cable. FASLINK™ carries power (without the need for a wall-wart) and bi-directional communication between the two units. Don't worry if you own an MFC-101 original or Mark II—an optional FASLINK™ adapter allows you to use this new connection standard. A MIDI port for use with 7-pin phantom power or 5-pin MIDI cables (for connecting 3rd party MIDI interfaces or pedalboards, for example) is also provided.

New FX Processing Features and Enhancements

The effects-processing capabilities of the Axe-Fx II have been vetted and endorsed by some of the most discriminating players in the world. The sound and features of our effects provide extremely authentic representations of many classic originals, plus the range to take you where no tone has gone before. Now, with a TYPE control to instantly set all other parameters, it is easier than ever to dial in classic settings on many effect blocks. "Types" include tape and analog "bucket brigade" delay effects, "script" and "block" logo phasers, "dimension" chorus, jet flanger, vibe phaser, many classic wah pedals, whammy, and too many more to list here.

Share Settings Across Multiple Presets with Global Blocks

Those familiar with "Global Amps" from previous Axe-Fx products will appreciate Global Blocks. Those new to Axe-Fx products will appreciate how this feature allows centralized control of a preset collection. You can save any effect "block" to a special memory area, then load it into multiple presets with a "link" to keep all copies synchronized to the master. You can even update the master from any linked instance. Should you choose to remove a link, this leaves both the original and the new copy fully independent of each other.

Axe-Fx II XL/XL+ New Features

This manual covers both the **Axe-Fx II Mark II** and the newer **Axe-Fx II XL and XL+**. All Axe-Fx II units have the same DSP and amp modeling capabilities, but the XL offers the additions of expanded memory and peripherals as detailed below:

- Built-in FASLINK™ port for connection to MFC-101 Mark III over conventional XLR cables.
- Dedicated MIDI IN, OUT, and THRU jacks (vs. shared OUT/THRU in the Mark II).
- Two onboard PEDAL jacks (vs. one in the Mark II).
- Primary VALUE entry via optical encoder with a lifespan of 1,000,000+ rotations.
- "Secret Sauce III" instrument input features an even lower noise floor.
- 128 Mb of non-volatile Super-FLASH memory allows for storing 768 presets and 1024 user cabinets, plus copious memory reserves for future expansion (384 presets and 100 cabs on the original/Mark II)
- Double-capacity preset size allows for expanded functionality including X/Y switching on more blocks and possible future implementation of more effect instances per preset.
- Built-in backup firmware allows recovery in the event of complications during update.
- Backward compatibility with Axe-Fx II Mark I/II presets via Axe-Edit software.
- The XL+ has an even lower noise floor and a sharper, brighter display.

Improvements and Enhancements, plus More to Come...

Aside from the many features covered in this brief introduction, there are many more things in the Axe-Fx II that we hope will make it one of the most exciting and rewarding products you have ever owned. The upgradeable firmware of our products means that we are able to offer free updates in the form of easy-to-install downloads. These include exciting improvements to amp modeling quality, new models, cabs, effects, and more.

1 Introduction

1.1 What is the Axe FX II?

The Axe-Fx II is an advanced digital preamp and effects processor for guitar, bass, and other musical instruments. It replaces amps, speakers, microphones, stompboxes, studio processors, and more. It is an all-in-one, end-to-end great tone solution in a single black box.

Inside, a virtual environment allows you to build your dream rig (hundreds of them, in fact). Choose from an inventory of hundreds of classic and innovative components. Select and arrange things any way you like, limited only by the unit's ample CPU resources and your imagination. "Dial in" your signature sound using basic controls, or go deeper with advanced parameters, then save presets for instant recall when playing, performing, or recording.

The sound is of uncompromising quality, due both to extremely high standards of hardware design and to our advanced proprietary software algorithms. The Axe-Fx II, like its predecessors, asserts that digital has reclaimed its birthright as the superior solution for musical instrument processing. Words fall short. You only need to plug in and play to realize that this is "the real deal."

A Word on Modeling

You may have noticed that the Axe-Fx II is not typically described as a "modeler." This is not to diminish its debt to heritage; on the contrary, we've done thousands of hours of deep analysis of the greatest amps, cabs, and effects of all time. In fact, amps and pedals, their vacuum tubes and other components, plus speaker cabs and many effects, are painstakingly replicated to perform exactly like the originals. But while the unit includes emulations based on specific product types, it goes well beyond simply presenting models—with their limited controls, features, and sounds—to offer a do-it-yourself modeling platform. If it's models you want, we can give them to you, but why stop there?

The Axe-Fx II removes limits instead of recreating them. Take our Wahwah effect, for example. You might just plug in and start cryin', or you could tune the pedal sweep, tweak the resonance, overdrive the circuit, and tailor the sound to your exact wishes. Try the Plexi. Dial it in just right. Then open things up and hear what happens when you drop in the tonestack from a modern Rectifier (all it takes is one turn of a knob to make the change). There are hundreds more ways that the sound can be customized. Rediscover your all-time favorites, or go crazy creating sounds you only wished someone would put in a product. And you don't need to be an engineer to do so, as the unit is extremely user-friendly.

You are not alone in the quest for tone, either. The Fractal Audio online community has amassed a wealth of knowledge and is ready to share expertise on every subject, from the deviling details of differently dated diodes in dilapidated distortions to how to set up your favorite artist's "*exact* gear used in the *exact* order with the *exact* settings for the first half of the second bridge of the third bonus track off the re-master."

In comparison to its predecessors, the Axe-Fx II has some great new capabilities as an effects modeler. Just as our previous products allowed you to select, for instance, a "TYPE" of Amp, Cab, or Drive, now the Chorus, Delay, Flanger, Phaser and other effects include a control to automatically dial in all-time favorites—settings like "dimension chorus", "tape delay", "analog flanger", "script 90 phaser", and many more. Once you make a selection, however, you can go beyond the model. With deep recreations of the intricacies and interactions behind great tone, we create not only a sample or profile but a multidimensional whole enchilada. Again, just plug in and hear it for yourself.

To re-pose the original question then, what is the Axe-Fx II?

The Axe-Fx II is the new flagship processor from Fractal Audio, with far more power and many more capabilities than the former heavyweight champion, Axe-Fx Ultra—twice the DSP in fact, allowing it to deliver far more detailed amp modeling, plus numerous other upgrades. It contains our best-ever guitar amplifier simulation and effect technology—state-of-the-art algorithms designed to sound and feel like the real thing. It is a fully routable, fully programmable, real-time controllable, multi-effects processor offering the utmost in sound quality, with unrivaled flexibility and control options. It is a **modeling platform** upon which you can create any number of incredible guitar tones—able to replace entire rigs of traditional gear with a single black box. Let's take some of these concepts a bit further:

Routable: Place effects freely in any order and layout—series, parallel, or complex networks, including feedback loops and external send and return, at any point in the signal chain.

Programmable: Every effect has a full complement of parameters offering desirable features and tremendous range. Gone are the limits of processors with restrictive options or little-to-no depth.

Controllable: Many parameters—including all the usual expression pedal suspects and every effect bypass switch—can be operated remotely via MIDI, offering great real-time performance control capabilities. You can map control curves, assign multiple parameters at once, tap powerful global controllers, and much more.

Multi-Fx: The Axe-Fx II offers all the classic effects plus a few new ones. The massive "effects inventory" allows any preset to use two or more of almost every effect block type, so you can build huge virtual rigs. In addition, many effects now include X/Y states so you can instantly swap one set of settings for another without changing presets. Almost all the effects in the Axe-Fx II process in full stereo.

Utmost Quality: Sound quality is our first-and-foremost criterion for the success of the Axe-Fx II. This shows in the hardware design and in every detail of our proprietary natural processing software algorithms. Many of these replicate patterns that occur in nature (thus our company name of *Fractal* Audio Systems). The amp simulations use unique, dynamic, non-linearity generators that produce smooth, even-ordered harmonics, giving a depth to the sound that other processors lack. Our effects have been vetted and championed by some of the world's most demanding and discriminating players.

Rig Replacer: Having everything in one box has some great advantages, especially when that box is as powerful and versatile as the Axe-Fx II. In addition to being able to replace big rigs outright, this tightly integrated, unified system offers certain fringe benefits. No longer does changing a drive pedal mean fighting with cables half-an-inch too short. No longer must you labor over deciding which amps your tour's budget will or won't allow you to ship or handle. Gone are the headaches and hassles of systems of so many boxes strung together with so many wires, prone as they are to failure and noise. And let's say a small meteor hits the stage and obliterates your Axe-Fx II: you can literally restore to a new unit during the intermission and be up and running again for the next set.

Finally, after you've replaced your entire rig, the Axe-Fx II lets you continually re-invent it without ever touching Velcro, rack screws, or your credit card.

1.2 The Inventory/Grid Concept

In the real world, we are limited by the equipment we own and by the fact that building a rig requires making commitments. On the Axe-Fx II, these limitations are lifted, with the ability to tap a vast inventory of virtual amps, cabs, effects, mixers, and more. You have the freedom to set them up, dial them in, save them as presets, and then do it all again, as often as you like.

Axe-Fx II presets are created by selecting components—like amps, cabs, or effects—from an inventory, and placing them as “blocks” into the slots of a 12×4 “grid.” As with their real-world components, blocks must be connected together using “cables”—virtual ones in this case. Blocks in adjacent columns may be connected directly together, with splits and merges as needed. Passive “shunts” carry signal through otherwise empty grid spaces.

The Inventory: Hundreds of Amps, Cabs, and Effects

Components are removed from the inventory and placed into the grid as “blocks.” There, they can be interconnected and edited to create a preset.

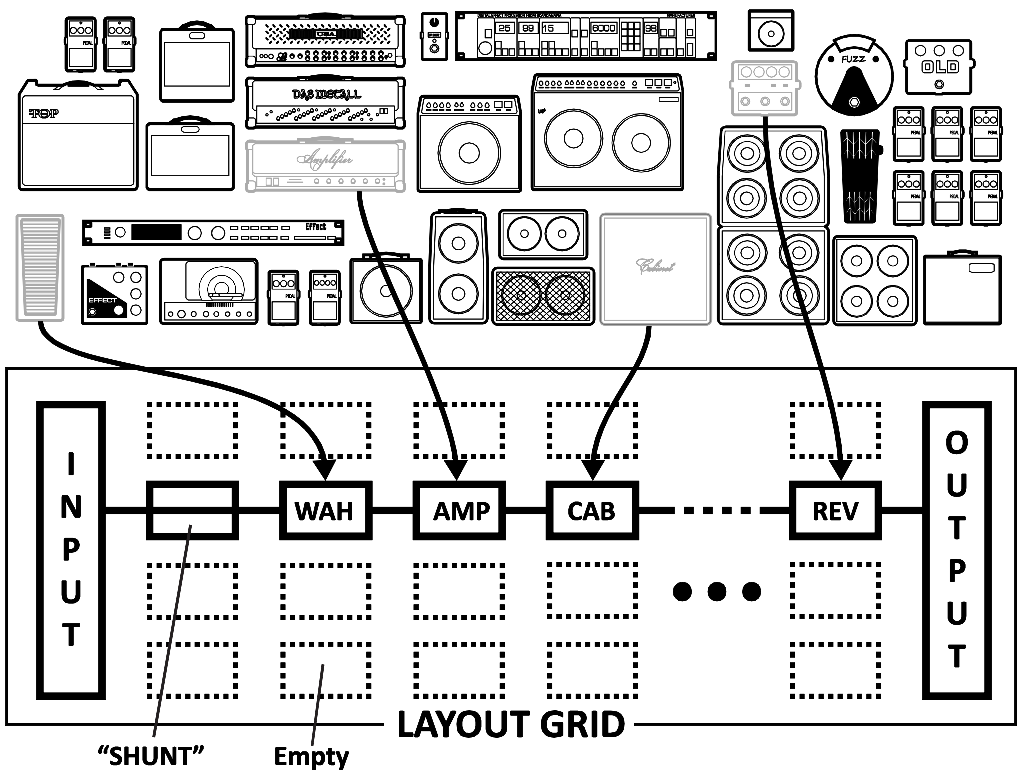


Figure 1-1 – The Inventory/Grid Concept

(Note: Seven empty columns were removed from the illustration and are represented by •••)

The figure above presents a stylized example of an Axe-Fx II preset. The INPUT is routed through a SHUNT to feed a “WAH” block. (The shunt has no effect on the sound and is shown only to introduce the concept of its use.) The WAH block is connected to an “AMP” block (we might set its type to “Plexi Normal”), which in turn feeds a “CAB” (one of the many “4x12” options, perhaps). This is routed to a reverb (“REV”) and then to the OUTPUT.

The size of a preset is limited only by the grid structure, block inventory, and total processing power or “CPU”. You’ll be pleased to discover that there is enough CPU power to allow large and complex creations.

The subject of creating and modifying presets on the grid is covered in detail in section **4**: **Basic Operation and Editing** (p.**27**). The inventory of blocks available to every Axe-Fx II preset is listed below:

Amp (×2)	**Filter (×4)**	**Mixer (×2)**	**Reverb (×2)**
Cab (×2)	**Feedback Return**	**Multiband Compressor (×2)**	**Ring Modulator**
Chorus (×2)	**Feedback Send**	**Multi-Delay (×2)**	**Rotary (×2)**
Compressor (×2)	**Flanger (×2)**	**Tremolo/Panner (×2)**	**3-Voice Synth (×2)**
Crossover (×2)	**Formant**	**Parametric EQ (×4)**	**Tone Matching**
Delay (×2)	**Gate/Expander (×2)**	**Phaser (×2)**	**Vocoder**
Drive (×2)	**Graphic EQ (×4)**	**Pitch Shifter (×2)**	**Volume/Pan (×4)**
Effects Loop	**Looper**	**Quad Chorus (×2)**	**Wahwah (×2)**
Enhancer	**Megatap Delay**	**Resonator (×2)**	**Shunt (36)**

In addition to the blocks listed above, each preset also includes a programmable **Input Noise Gate** (p. **125**) and an **Output Mixer** (p. **128**). Of course, having components on the grid is just the beginning. Every block can be edited, with parameters representing all the basic knobs you would expect to find, and advanced menus for deep control. See the **Effects Guide** (p. **39**) for more detail on blocks and their parameters.

A powerful new feature of the Axe-Fx II allows you to maintain your own collection of Global Blocks (p.**131**) that can be inserted and then kept synchronized across multiple presets.

Twenty-two different **Modifiers & Controllers** (p. **136**) are provided to automate or remotely control various parameters in any preset. These are LFO 1, LFO 2, ADSR 1, ADSR 2, Envelope, Pitch Detector, Sequencer, Manual A/B/C/D, and External 1–12.

1.3 Connectivity and More

The grid and effects inventory may be the centerpiece of the Axe-Fx II, but it is the powerful connectivity and companion features that allow the system to be so much to so many. The hardware itself is covered in **Section 2: Overview** (p. **8**), which also details the new USB features for **Computer Integration** (p. **12**). Rig design is covered in **Section 3: Connections** (p. **15**), where many diagrams are included.

Configuration and connectivity on the Axe-Fx II are managed with a number of user-specifiable options, listed and described in **Section 8: Global Parameters** (p. **145**), and **Section 9: Input/Output Parameters** (p. **148**). Meanwhile, **Section 10: Utilities** (p. **155**) can fill out your understanding of functions related to general use and maintenance.

Finally, **Sections 11** through **14** cover the **Tuner** (p. **157**) and **Tempo** (p.**158**) functions, plus the basics of **Backing Up and Restoring** (p. **160**) and **Firmware Updates** (p. **162**).

My Head Hertz…

As even this introduction shows, the Axe-Fx II contains a world of possibilities. Only the precise terminology of audio engineering allows the very communities of musicians, engineers, and others to use and enjoy this powerful device. In case you want to familiarize yourself with pro audio terms and jargon, the **Appendix** is filled with useful material, including a **Glossary** (p. **184**). It is followed by **Specifications** (p. **198**) and your **Warranty** (p. **195**).

2 Overview

Review the following to familiarize yourself with the hardware features of the Axe-Fx II.

2.1 The Front Panel

The front panel shown is that of the **Axe-Fx II XL**. It is functionally identical to that of the **Axe-Fx II Mark II**.

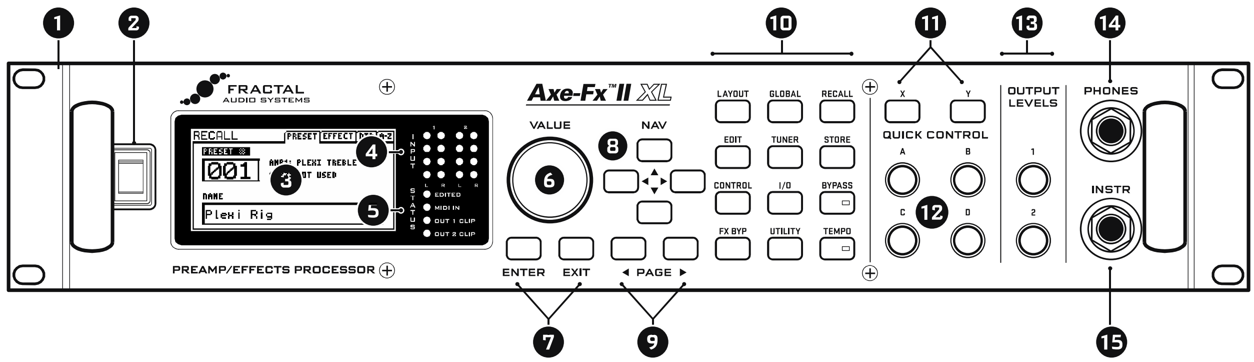


Figure 2-1 – The Axe-Fx II Front Panel

1. The Axe-Fx II is housed in a powder-coated steel enclosure with an anodized aluminum faceplate.
Dual front handles allow easy rack mounting and removal.

2. The POWER Switch turns the unit on or off.

3. The 160 × 80 pixel LCD is where all menu and function screens are displayed.

4. INPUT 1 and INPUT 2 LED meters display the levels of incoming signals. See p. **15** for more detail.

5. STATUS LEDs communicate important events:

- EDITED – This LED is lit when any change has been made to the current preset.
- MIDI IN – This LED is lit while data is received at the MIDI IN port.
- CLIP 1, CLIP 2 – These flash briefly whenever the signal level at the corresponding output causes the D/A converter to clip. Section **3.1** on p. **15** has more information on **Setting Levels**.

6. In RECALL mode, the VALUE wheel selects and loads presets as it is turned.
In edit or menu screens, it changes the **value** of the selected parameter.

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

8. In RECALL mode, the four NAV buttons select and load presets. Up = +1; Down = -1; Left=-10; Right =+10.
In edit or menu screens, these are used to **select** between on-screen parameters or options.

9. The PAGE buttons step through menu pages, shown as tabs at the top of the display.

10. The 12 main front panel **menu/function buttons** are listed below.

- LAYOUT – This menu contains four pages: EDIT, MOVE, GATE, and MIX.
  - **EDIT** contains the grid where presets are created by inserting blocks and cables (p. **27**).
  - **MOVE** has various utilities for moving preset components on the grid (p. **32**).

- **INPUT/GTE** contains parameters for the Noise Gate and Instrument input impedance (p. **125**).
- **OUTPUT** page contains a mixer for overall level control of a preset (p. **128**).

- EDIT – Select any block on the layout grid and press this button to open its **EDIT menu**.
  Press repeatedly to cycle through EDIT menus of all blocks in the preset (top-to-bottom, left-to-right).
- CONTROL – This menu contains pages for seven of the internal controllers available to every preset, plus a Modifiers overview screen. See **Modifiers & Controllers** on p. **136** for details.
- FX BYPASS – This button toggles the bypass state of the currently selected block (p. **37**).
  Double click BYPASS in any block EDIT menu to access SAVE/LOAD GLOBAL BLOCKS (p. **131**).
- GLOBAL – This menu contains four pages: CONFIG, OUT1, OUT2, and SCALES. See p. **145** for details.
  - **CONFIG** contains parameters that globally affect the sound of all presets.
  - **OUT1** and **OUT2** each hold a 10-band graphic EQ and master GAIN control for the given outputs.
  - **SCALES** allows the creation of custom harmonies for use with the pitch shifter block.
- TUNER – Engages the tuner (p. **157**) and displays its menu. Press EXIT or RECALL to close.
- I/O – Contains six pages used to configure the various input and output options of the Axe-Fx II.
  See p. **148** for details.
- UTILITY – This menu contains various utility functions. See p. **155** for details.
- RECALL – Enters RECALL mode, the main operating mode for use during musical performance.
  The Axe-Fx II always defaults to RECALL mode when you power it on.
- STORE – Enters the STORE menu where you can save, rename, or swap presets. See p. **37** for details.
- BYPASS – Bypasses the Axe-Fx II, routing the input directly to the output, defeating all processing, lighting the **BYPASS LED** and showing a flashing on-screen warning. Press again to un-bypass.

Double-clicking the BYPASS button in any block EDIT menu **initializes** that block to default settings.

- TEMPO – Flashes the current tempo on its built-in LED. You can also tap this button once to enter the TEMPO menu, or two or more times to set a new tempo. The tempo can also be entered using a remote switch or set via MIDI. See p. **158** for more information about Axe-Fx II tempo control.

11. X/Y – On the Axe-Fx II, certain effects offer two fully independent sets of settings, called “X” and “Y”. Think of them like two different “channels” for a given amp or effect. You can switch between X and Y to access different sound settings without changing presets (see p. **36**).
X and Y buttons perform other functions as well. See section **4.4** on p. **36**

12. QUICK CONTROL – The four Quick Control knobs, A, B, C, and D, perform different functions depending on what Axe-Fx II screen or menu is shown. See section **4.3.1** on p. **36** for more information.

13. OUTPUT LEVEL – These controls set the output levels of OUTPUT 1 and OUTPUT 2 (“FX Send”). See section **3.1** on p. **15** for information on setting levels. OUTPUT 1 also controls headphone jack levels.

14. HEADPHONES – Connect stereo headphones here to monitor OUTPUT 1 L+R.

15. INSTR – Plug your instrument into this **Instrument Input Jack**, designed specifically for use with electric, acoustic, and bass guitars. Plugging a line-level device into this input may cause clipping of the input amplifier and is not recommended.

MK2 The Axe-Fx II Mark II features “Secret Sauce II” instrument input for a low noise floor.

XL The Axe-Fx II XL/XL+ features “Secret Sauce III” instrument input for an even lower noise floor.

2.2 The Rear Panel

The following section details the rear panel of the Axe-Fx II. Please note differences between the Mark II and XL models in numbered areas 22 (MIDI), 24 (MFC and FASLINK™) and 25 (PEDALS).

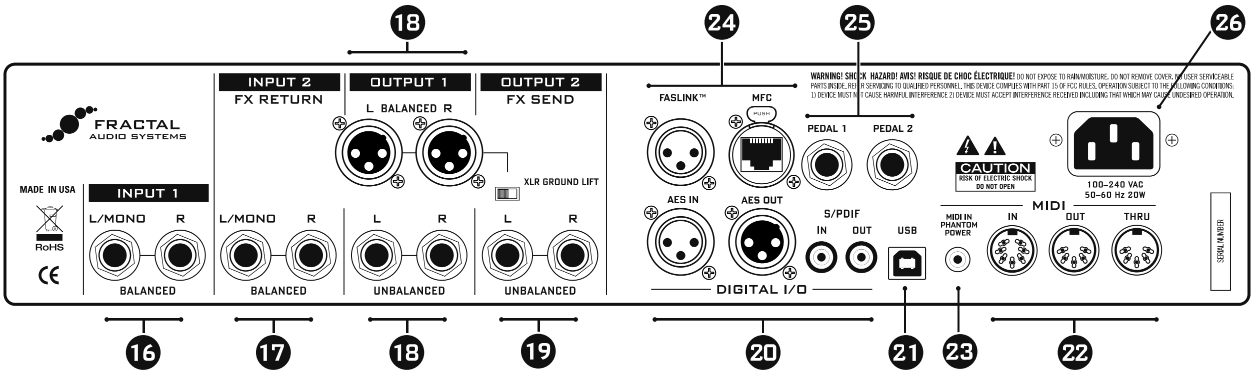


Figure 2-2 The Rear Panel of the Axe-Fx II XL+

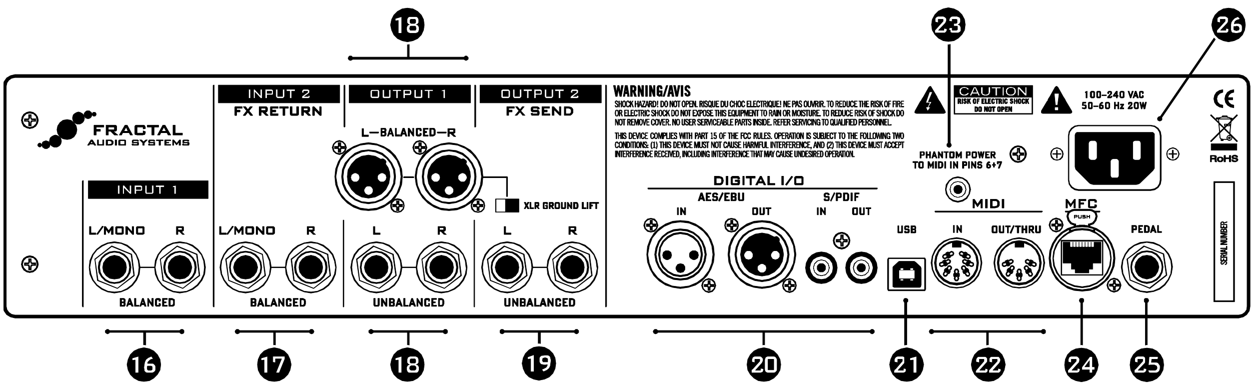


Figure 2-3 The Rear Panel of the Axe-Fx II Mark II

16. INPUT 1 **Left/Mono and Right, Balanced (1/4" Tip-Ring-Sleeve) Jacks** – Connect line-level input sources to these jacks, being sure to set the **INPUT 1 LEFT SELECT** to REAR in the I/O menu (p. **148**).

17. INPUT 2 – **Left/Mono and Right, Balanced (1/4" Tip-Ring-Sleeve) Jacks** ("FX RETURN") – Connect to the output(s) of outboard equipment when using the **FX Loop** block (p. **70**). You can also use this as an auxiliary input to any point in the signal chain of any preset using the **FX Loop** block.

18. OUTPUT 1 – This section includes the **Left and Right Output 1 unbalanced (1/4") Jacks**, **Balanced (XLR) jacks**, and **XLR Ground Lift Switch**. The main processed output of the Axe-Fx II appears at these jacks. Use the XLR jacks to connect to balanced inputs, employing the provided ground lift switch if necessary to reduce unwanted 60-cycle hum. Use the 1/4" unbalanced outputs to connect to unbalanced inputs, such as those on some guitar power amps or other devices.

19. OUTPUT 2 – **Left/Mono and Right, Unbalanced (1/4" Tip-Sleeve) Jacks** ("FX SEND") — The output from Connect to he inputs(s) of outboard equipment when using the **FX Loop** block (p. **70**). You can also use this as an auxiliary output to tap any point of the signal chain for any preset using the **FX Loop** block. New **Humbuster**™ technology, featured on **Left and Right Output 1 and Output 2 unbalanced (1/4") Jacks** uses a simple TRS-to-TS cable to significantly reduce ground hum. See Section **16.11** on p. **178.**

20. DIGITAL I/O – This includes both **S/PDIF** and **AES/EBU** format **Input and Output Jacks**. Only one or the other pair of jacks can be active at any time depending on the setting of the **SPDIF/AES SELECT** parameter in the I/O:AUDIO menu (p. **148**). These jacks transmit and receive at a fixed rate of 48k.

21. USB – This provides the means to connect the Axe-Fx II to a PC or Mac, enabling a host of two-way audio and MIDI capabilities. See section **2.3** on p.**12**. Like the digital i/o, USB Audio operates *only* at 48k.

22. MIDI jacks – The Axe-Fx II XL/XL+ has separate dedicated MIDI IN, MIDI OUT and MIDI THRU jacks.

XL Unlike a shared/soft MIDI THRU, the XL's dedicated MIDI THRU jack adds no latency, but it is "hard-wired" to the MIDI IN PORT, and therefore does not pass signals from other MIDI inputs. For THRU functionality when using an MFC-101 at the FASLINK™ or MFC port of an Axe-Fx II XL/XL+, MIDI OUT can be made to work as a soft THRU by setting MFC ECHO TO MIDI OUT to "ON" (on the MIDI page of the I/O menu).

MK2 The Axe-Fx II Mark II has a MIDI Out/Thru combo jack that transmits or forwards MIDI signals to another device. MIDI THRU is disabled by default but can be enabled on the MIDI page of the I/O menu).

23. MIDI PHANTOM POWER **Jack** – When using the MFC-101 MIDI Foot Controller over a 7-pin MIDI cable, connect the supplied AC Adapter to this jack to provide power to the floor unit via pins 6+7. Some other MIDI controllers also support the use of phantom power on pins 6+7.

WARNING! Do not connect an AC adapter with a rating higher than 1A to the Phantom Power jack. Doing so will damage your Axe-Fx II.

24. MFC **Control Port** – This RJ45 jack allows you to use a standard (non-crossover) CAT5/Ethernet cable to connect the Axe-Fx II to a Fractal Audio Systems MFC-101 MIDI Foot Controller. The cable used to connect the Axe-Fx II and MFC-101 carries two-way data communication and phantom power ***without*** needing an external "wall wart" adapter. High quality Ethernet/EtherCON cables are available via http://www.fractalaudio.com/cables. (Axe-Fx II "Original Version" supports only Ethernet, not EtherCON.)

WARNING! DO NOT connect the MFC jack to an Ethernet device such as a computer, hub, switch, or router, as damage to one or both units could occur! Damage of this type leaves tell-tale signs on the motherboard and is NOT covered under warranty.

Always ensure that the Axe-Fx II power is OFF before inserting/removing Ethernet/EtherCON cables.

Be careful not to insert other types of connectors such as USB or guitar cables into the MFC port of the Axe-Fx II, as doing so can damage your unit, leaving tell-tale damage NOT covered under warranty.

XL **The FASLINK™ connector** of the Axe-Fx II XL/XL+ allows you to connect to the FASLINK™ port of an MFC-101 Mark III. The standard XLR cable used for a FASLINK™ connection powers the MFC ***without*** an external "wall wart" adapter, and carries 2-way communications. An optional **XA-2 FASLINK™ Adapter** allows you to connect the Axe-Fx II XL/XL+ to an earlier MFC-101.

MK2 The optional **XA-1 FASLINK adapter** adds a FASLINK port to the Axe-Fx II Mark II.

Find FASLINK™ adapters at http://shop.fractalaudio.com

25. PEDAL **Jack** – This jack is used to connect an external expression pedal or switch to control various functions of the Axe-Fx II. See p. **16** for more information on this function.

26. **Main Power Input** – Insert the supplied power cable and connect the other end to a grounded AC power receptacle.

2.3 Computer Integration

USB provides every Axe-Fx II with a host of great features.

2.3.1 Minimum Requirements

Windows Minimum Requirements:

- **OS**: Windows 8.x, Windows 7 SP1, Windows Vista SP2 (All versions compatible with x86 or x64).
- **CPU:** Intel Core 2 @1.6 GHz or better, or AMD equivalent
- **Memory**: 1GB minimum
- **USB 2.0 support required**

Mac Minimum Requirements:

- **OS X**: 10.6.8 for MIDI over USB (Fractal-Bot, Axe-Edit, Cab-Lab, etc.)
  10.9 or later required for USB audio. An issue in older versions causes audio pops.
- **CPU**: Intel Processor
- **Memory**: 512MB minimum
- **USB 2.0 Support required**

2.3.2 Software Installation

Although the Axe-Fx II is fully class-compliant, software installation is still required on all platforms. Without drivers installed, USB capabilities will not work correctly. Both Mac and Windows versions can be downloaded from our web site at http://www.fractalaudio.com/support. Step-by-step instructions are included with the installer.

2.3.3 Capabilities

The USB 2.0 class-compliant driver provides two channels of 48k/24-bit audio from the computer to the Axe-Fx II, up to four channels from the Axe-Fx II to the computer, and two-way MIDI-over-USB. All features can be used simultaneously. Note that even though the Axe-Fx II is fully class-compliant, you must still install the drivers found at http://fractalaudio.com/support .

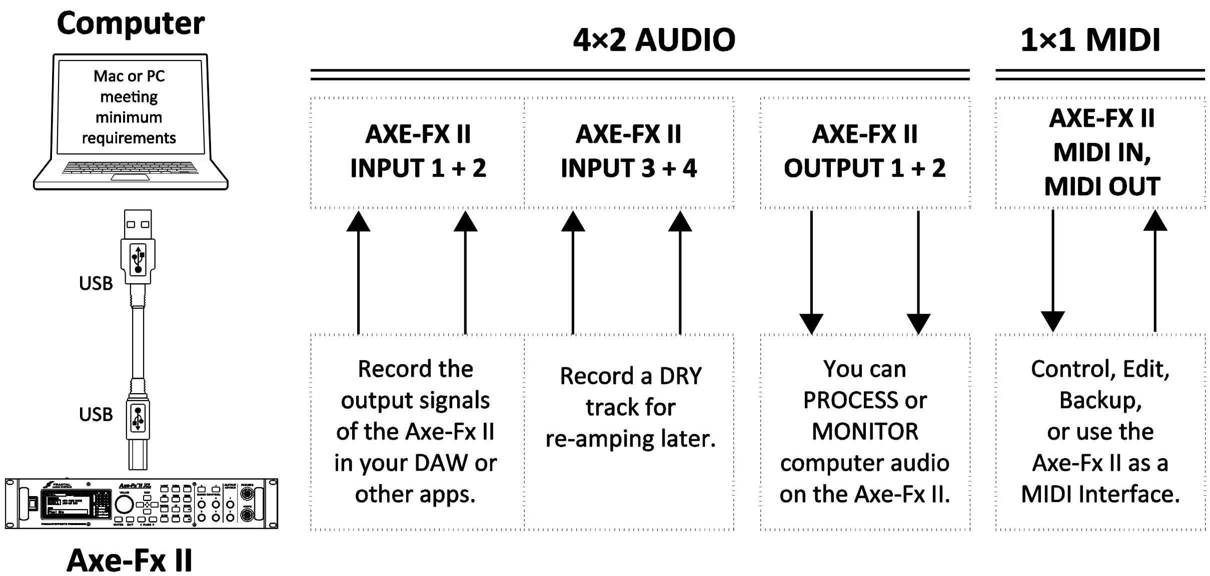


Figure 2-4 – USB Features

Audio and MIDI ports (shown in the top dotted-outline boxes above) have different names on different systems. In many applications, you can also assign “friendly names” to audio and MIDI ports.

Two Simultaneous Channels of 48k/24-bit Audio from the Computer to the Axe-Fx II

Two output channels allow audio to be sent from the computer to the Axe-Fx II where it can be processed by the unit or simply played back to OUTPUT 1.

To process computer audio with the onboard effects, set the **MAIN INPUT SELECT** (p. **148**) to “USB.” Computer audio will be routed to the grid INPUT. This allows you to re-amp a dry track, for example, or use the Axe-Fx II to process other audio or plugin tracks. It is possible to simultaneously record the processed output on the computer using the Axe-Fx II audio inputs (0/1).

To pass unprocessed computer audio through the Axe-Fx II, set the **MAIN INPUT SELECT** (p. **148**) to “ANALOG (IN1)” (the default setting) or “SPDIF/AES.” Computer audio is mixed with the regular output of the Axe-Fx, allowing you to play along with backing tracks or use the Axe-Fx II as a high quality “soundcard.”

Four Simultaneous Channels of 48k/24-bit Audio from the Axe-Fx II to the Computer

Four outputs, typically numbered 1–4, allow audio to be routed from the Axe-Fx II to the computer and recorded, processed, or monitored.

USB/DIGI OUT SOURCE in the I/O:AUDIO menu (p. **148**) determines what is sent to the first pair of USB outputs:

- Selecting **OUTPUT 1 L+R** sends the Axe-Fx II main output to the computer. The same signal is, of course, still routed normally to the rear XLR and unbalanced jacks. Use this to record your fully-processed guitar.
- Selecting **OUTPUT 2 L+R** routes output 2—typically the FX loop block (p. **70**) —to the first two outs.
- Selecting **MAIN INPUT** routes the main input (see **MAIN INPUT SELECT** p. **148**) to the first two outs.
  - A similar result can be obtained by using OUTPUT 1 with a "shunts-only" preset (p. **29**). Remember to turn off the Input Noise Gate and set Output levels to 0.0
  - Switching **INPUT 1 LEFT SELECT** (p. **148**) to "REAR" allows you to record the **line level** outputs of a microphone preamp, keyboard, or any other sound source via the rear INPUT 1 L/R jacks.

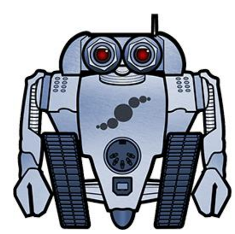
The second pair of USB Outputs always outputs the raw, dry, unprocessed signal from the main input (i.e. the front INSTR jack or the rear INPUT1 L/MONO jack, depending on both what is selected under **INPUT 1 LEFT SELECT** (p. **148**)) and the rear INPUT1 R jack). These require USB 2.0 support (see **Minimum Requirements** above).

> ***Warning:*** *As with all input/output systems, certain routing configurations can result in audio feedback loops. Please exercise care not to route active outputs to active inputs, or damage could occur to connected amps, speakers, or your hearing.*

Two-way High-Speed MIDI Communication

MIDI-over-USB enables the Axe-Fx II and the computer to communicate back and forth. You can edit, perform updates, send program changes from a sequencer, synchronize the tempo, automate sound changes, and more. MIDI-over-USB is considerably faster than "legacy" MIDI, and allows two-way communication with the computer over a single cable.

2.3.4 Fractal-Bot



Fractal-Bot, available from http://www.fractalaudio.com/fractal-bot.php is a powerful, lightweight MIDI Utility for the Axe-Fx II. Use Fractal-Bot for firmware updates, backing up your Axe-Fx, restoring backups, managing User Cabs, and more.

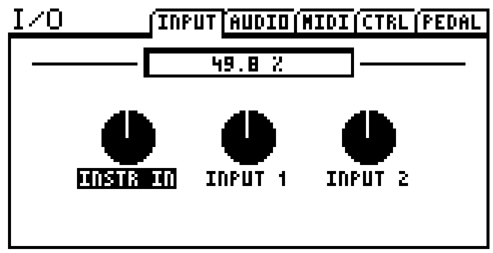
2.3.5 Axe-Edit

Axe-Edit is a full-featured software editor for the Axe-Fx II. It provides a large, easy-to-use graphical interface that allows you to create and edit sounds. Learn more at http://www.fractalaudio.com/axe-edit

3 Connections

Before making connections, be sure to turn down the volume of your amps and switch off all power. Take extreme care NEVER to connect the SPEAKER outputs of an amplifier to any jack on the Axe-Fx II as this will damage one or both devices. If you're not sure, don't do it!

3.1 Setting Levels



For the Axe-Fx II to work properly, it is important that input and output levels be configured correctly.

INPUT LEVELS are set with "soft-knobs" on the INPUT page of the I/O menu. Adjust according to the level of input source material until "hot" signals "tickle" the red LEDs on the front panel INPUT meters. The red LED lights at -6 dB (below clipping). Some sources may not reach ideal levels but can still be used with no problems.

Changing input levels will NOT affect *what you hear*. Inputs are *compensated*, meaning that as you lower trims to optimize signal-to-noise ratio going into the converters, the output of the converters is adjusted inversely so "what you hear" (and what reaches the signal processor) always *remains the same*.

Each input has its own dedicated analog-to-digital converter. The INSTR input jack parallels the rear inputs for improved signal-to-noise performance.

The front panel OUTPUT LEVELS 1 and 2 knobs independently control the volumes at the corresponding rear panel jacks. The output 1 knob simultaneously controls OUTPUT1 XLR and ¼-inch jacks, and the unit's headphone jack level. Optimal levels will depend on what the Axe-Fx II is connected to.

To operate with unity gain, set the Output Level knobs to *maximum*. If you then route shunts from the input to the output you will get out exactly what you put in. (If you're not using the Axe-Fx in the loop of a tube amp, unity gain is likely not relevant.)

If levels result in clipping of attached equipment, turn down the front panel OUTPUT LEVELS1/2 knobs. At the minimum setting, volume is reduced but may not be silent.

***NOTE:** The Axe-Fx II uses digital potentiometers to adjust the output levels. These actually contain hundreds of tiny resistors and switches. As such, some noise may be generated while adjusting the knobs.*

If the OUT1 or OUT2 CLIP LEDS light while you use the Axe-Fx II, the problem is not trim settings but levels in the digital domain. Chances are that the effects in your preset—many of which can increase gain significantly—are simply too hot. Reduce the output of one or more blocks (the AMP block **OUTPUT LEVEL** is usually a good starting point) or adjust the main GAIN slider of the preset's output mixer (see p. **128**).

When you need to adjust the level of all of your presets at once because some are clipping, you can also use the GAIN slider on the OUT1 or OUT2 graphic equalizers to make a global adjustment. (p. **146**)

Clipping can also be caused if you have increased the BOOST/PAD setting for one of the converters and can be reduced if you adjust this setting to be closer to 0 dB (see p. **148**). Block or preset adjustment may still be required.

3.2 The PEDAL Jack(s)

Note: the Axe-Fx II XL+ and XL have **two** onboard PEDAL jacks. The Axe-Fx II original and Mark II models have **one**.

Connect an expression pedal or footswitch to the **pedal** jack to control sound functions. You must first configure the **TYPE,** and perform a simple **calibration** routine when using an expression pedal. See section **9.5** on p.**154** for how to use the PEDAL page of the I/O menu.

Any type of external switch can be used, as long as its contacts make and break the connection between tip and sleeve on a regular 1/4” guitar cable. Expression pedals should have a linear resistance taper and max resistance between 10kΩ and 100kΩ, and must be used with Tip-Ring-Sleeve cables.

To control sound parameters, you must first assign the PEDAL jack to an “EXTERNAL CONTROLLER” and then set up a “MODIFIER.” This topic is covered in section **7**: **Modifiers & Controllers** on p. **136**, with a section specifically about External Controllers on p. **144**.

3.3 System Parameters

As you will discover, the Axe-Fx II has a range of flexible input/output options. The hardware descriptions cover the basics, but seeing a few complete setups can also be helpful as you decide how to connect other equipment. The following section illustrates some typical setups, some of which require specific “System Parameter” settings. For example, when using a real, physical guitar cab, the global “Speaker Simulation” switch should be set to “OFF.”

In addition to the short introductions provided with the diagrams below, system parameters are detailed in section **8**: **Global Parameters**, and section **9**: **Input/Output Parameters**.

Many of the basic setups shown in the following pages can be easily combined or expanded. Make a mono setup stereo. Add an MFC-101 MIDI Foot controller for intelligent remote control. Connect a computer to run **Axe-Edit**, our free software editor. Combine multiple Axe-Fx II units into a single rig. Countless combinations are possible, and creativity and experimentation can be as valuable here as when you are creating tones or dialing in effects.

3.4 Connection Diagrams

This overview will help familiarize you with the various inputs, outputs, and control connections of the Axe-Fx II. The diagrams that follow in sections **3.4.1** through **3.4.10** illustrate several real-world applications.

Note that the Axe-Fx II Mark II has a shared MIDI OUT/THRU, and that MIDI THRU must be enabled in the I/O:MIDI menu. The Mark II also does not have a FASLINK™ port, so an MFC-101 would be connected via MIDI, Ethernet/EtherCON or a FASLINK™ adapter.

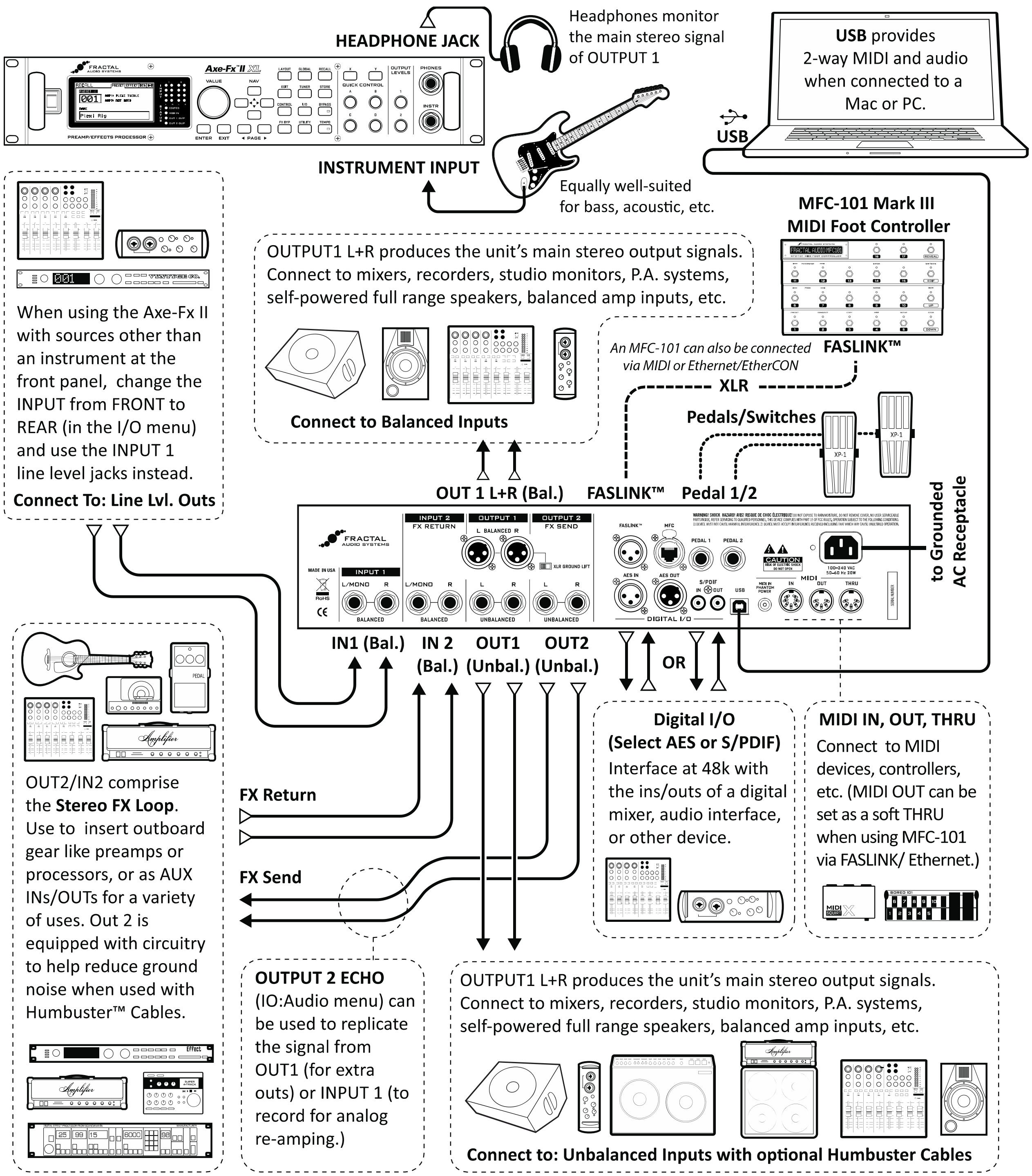
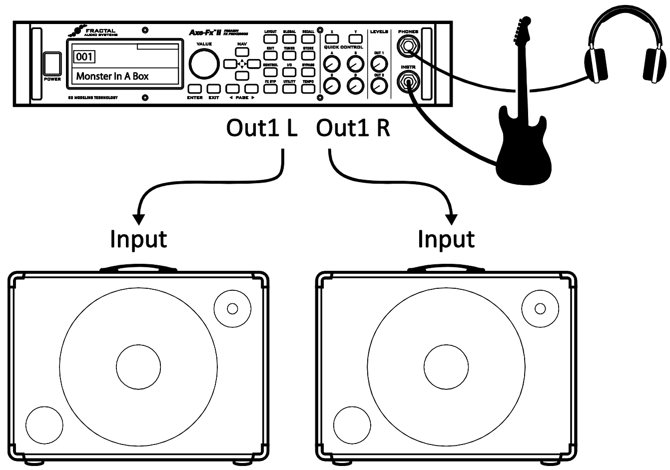


Figure 3-1 – I/O Overview

3.4.1 Axe-Fx II into Self-Powered Full-Range Speakers

COMPONENTS:

- Guitar
- Axe-Fx II
- Self-Powered Full Range Speaker(s)
- Headphones (opt.)



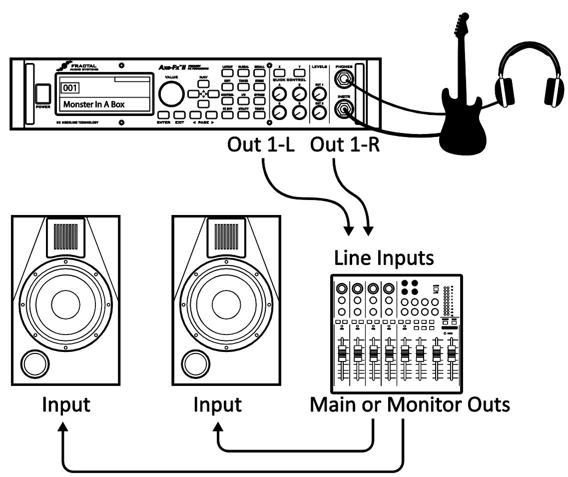
Global Settings: Default **I/O Settings:** Default

Notes: With its built-in amp and speaker simulations, the Axe-Fx II can be played directly into P.A. or other self-amplified, full-range, flat-response (FRFR) speakers. Externally amplified (passive) FRFR speakers are equally well-suited. This might just as easily be a Front-of-House system with floor or in-ear monitors. In this configuration, the Axe-Fx II creates all aspects of the end-to-end guitar chain for the ultimate in tonal flexibility—stompboxes, amps, cabs, post effects, and more. Optional headphones provide an alternate way to listen when cabs are not present or are switched off. Balanced (XLR) or unbalanced (1/4”) jacks and cables can be used to connect the Axe-Fx II, with the former providing the advantage that less noise will be picked up over longer cable runs. If only one speaker is used, set **OUT 1 MODE** in the I/O menu (p.**148**) to one of the mono options.

3.4.2 Axe-Fx II into Studio Monitors

COMPONENTS:

- Guitar
- Axe-Fx II
- Mixer (opt.)
- Studio Monitors
- Headphones (opt.)



Global Settings: Default **I/O Settings:** Default

Notes: This is essentially identical to the diagram shown above for Self-Powered FRFR speakers. The point to take away here is that any system intended for full-range monitoring or sound reinforcement—from small computer speakers to the P.A. at a huge arena—represents a suitable counterpart to the capabilities of the Axe-Fx II.

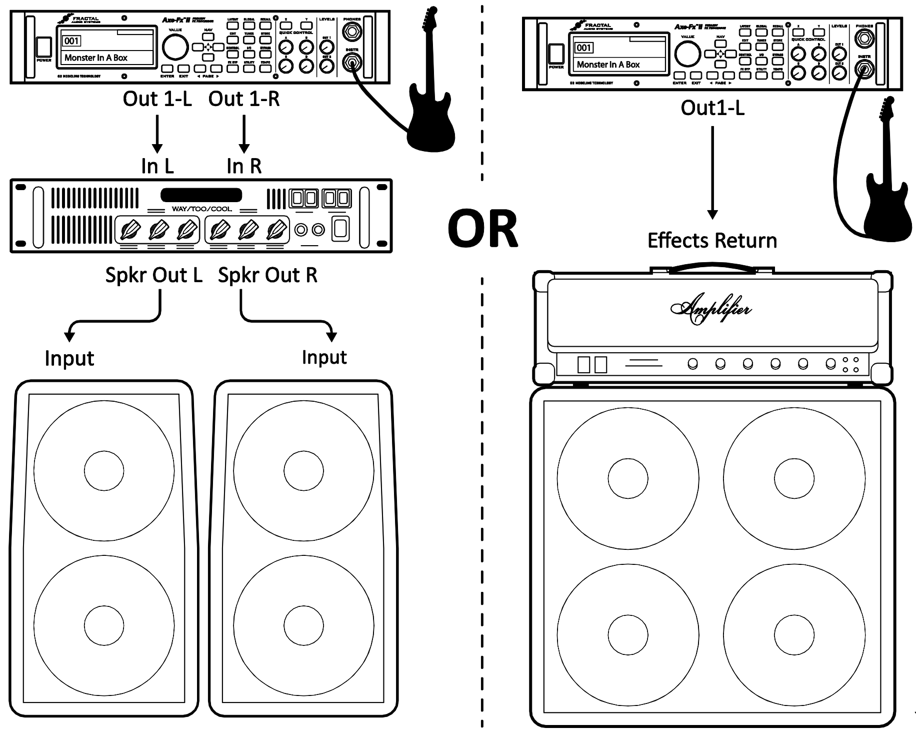
3.4.3 Axe-Fx II with Power Amp and Guitar Speakers

COMPONENTS:

- Guitar
- Axe-Fx II
- Power Amp and Guitar Speakers

  -OR-

  Amp Head/Combo with FX RETURN jack (power amp input) and Guitar Speakers



Global Settings: Power Amp Simulation **ON** or **OFF** (see below), Speaker Cabinet Simulation **OFF**

I/O Settings: Set **OUT1 MODE** (p.**148**) as required for stereo or mono.

Notes: Depending on the character of the amplifier being used, Power Amp Simulations may need to be ON or OFF for this type configuration.

- Power Amp simulations should be turned ON when using a "neutral-sounding" (Solid State) power amp that does NOT significantly affect the tone or feel.
- Power Amp simulations should be turned OFF when using the RETURN on a tube head or combo, or a guitar-oriented tube power amp—one that significantly affects to the tone and feel.
- In either case, it is completely safe and reasonable to try both settings to see which you prefer.

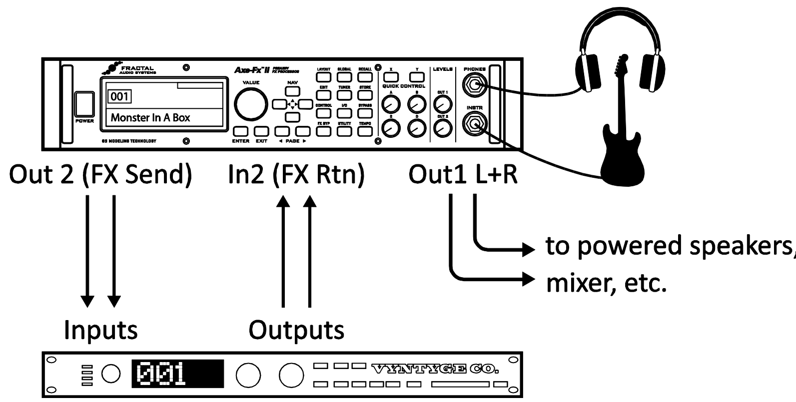
Whenever you use the Axe-Fx II with traditional guitar speaker cabinets (whether open back or closed, small or large, alone or in pairs) it is best to turn Speaker simulations OFF in the CONFIG page of the GLOBAL menu (p. **145**). Guitar speakers differ from full-range speakers in that they are voiced to focus on traditional electric guitar sounds: mids tend to be prominent, highs rolled off, etc.

The settings required for this type of operation are not ideally suited for monitoring through the Axe-Fx II headphone jack, as what you hear through headphones will not sound like what you will hear through the speakers. Switching Power Amp and Speaker Sims on and off, however, is simple and can be done as needed to switch back and forth.

3.4.4 Axe-Fx II Effects Loop

COMPONENTS:

- Guitar
- Axe-Fx II, connected as desired to monitors/mixers/amps/ etc. (see other diagrams for setup ideas)
- Outboard Processor or Preamp



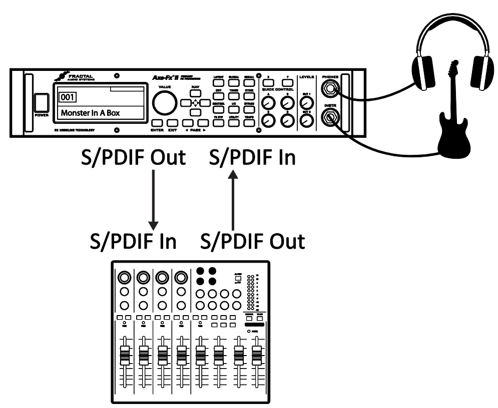
Global Settings: See Below **I/O Settings:** See Below

Notes: The Axe-Fx II has a stereo FX Loop that allows you to insert outboard gear such as preamps or processors at almost any point in the signal chain of any preset. You'll learn more about editing presets in section **4**, and the Effects Loop block is detailed on page **70**. For now, just be aware that custom presets are required for the Effects Loop to be used. The Effects Loop configuration above is compatible with other setups that do not utilize INPUT2 or OUTPUT2 for other purposes. Global and I/O options should be set accordingly.

3.4.5 Axe-Fx II Digital Audio Interconnection

COMPONENTS:

- Guitar
- Axe-Fx II
- Mixer, recorder, computer, etc. with digital (S/PDIF or AES/EBU) inputs and/or outputs



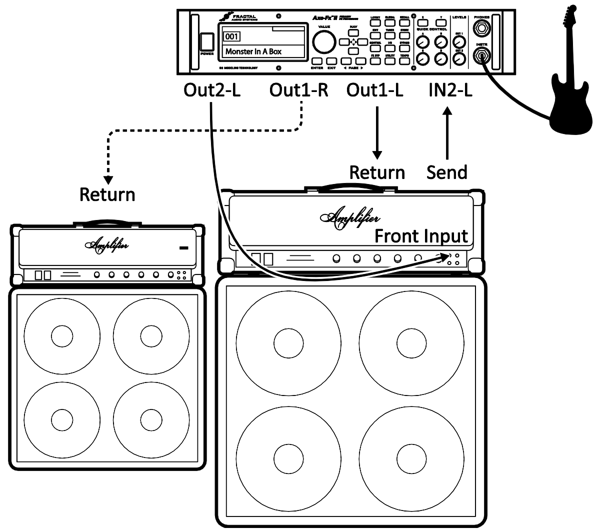
Global Settings: Default **I/O Settings:** See below.

Notes: The digital connector jacks on the Axe-Fx II allow it to interface with a variety of devices with S/PDIF or AES/EBU I/O. You can connect to the inputs of a digital mixer, for example, or process the signal from another unit with a digital output, avoiding an unnecessary D/A:A/D round trip. The required clock rate for both input and output signals is 48kHz. The Axe-Fx must be the master clock. "SLAVE" your 3rd party device by setting its clock source to AES or S/PDIF. The onboard digital outs can monitor OUTPUT1, OUTPUT2, or the main INPUT, as specified in **USB/DIGI OUT SOURCE** on the AUDIO page of the I/O menu (p. **148**). The **MAIN INPUT SOURCE** must be set to "SPDIF/AES" for digital inputs to be connected to the grid. A valid signal must be present at the selected digital input or the Axe-Fx II will display "NO INPUT CLOCK!"

3.4.6 Axe-Fx II Four Cable Method (“4CM”)

COMPONENTS:

- Guitar
- Axe-Fx II
- Guitar Amp with a series effects loop[1] and built-in or separate Guitar Speaker(s).
- Second Amp for Stereo (opt.)



Global Settings: Defaults OK, since typical 4CM presets will not have AMP or CAB blocks!

I/O Settings: Set **INPUT** and **OUTPUT MODES** as required for stereo or mono (Remember: even in a stereo rig, when you are connecting a guitar to the front INSTRUMENT input of the Axe-Fx II, **INPUT 1 MODE** must be set to “LEFT”.) Adjust the two **OUTPUT BOOST PAD** parameters to lower the noise floor. See p.**148** for details on these parameters.

Notes: This highly integrated setup places the Axe-Fx II simultaneously “in front of” an amplifier’s preamp section, where it replaces traditional stompboxes, *and* in its effects loop, where “post” effects like delay and reverb have a totally different sound than when used “out front”. (Although a head and separate cab are shown, many combo amps also offer an onboard effects loop and can be used as well.)

To use the 4CM, you will need to create special presets where AMP and CAB blocks are replaced by the FX LOOP block (p. **70**). Signal hits the Axe-Fx II first, into any effects that you want in front of the amp—compressor, drive, wah, and the like. Then, the FX LOOP block is used to “insert” the preamp of the actual amplifier on the grid. Output 2 of the Axe-Fx II has an all-new, *extremely* low-noise design that is well suited for feeding the front of an amplifier. The signal makes a “round trip” to the amp’s preamp and back to the grid, where it is then processed by additional blocks—your post FX: maybe chorus, delay, reverb, etc. The final routing is via Axe-Fx II OUT1/L to the Return (Power Amp Input) of the amplifier which powers your speaker. To extend this configuration for optional stereo, connect OUT1/R to the RETURN of a second amplifier, bypassing that unit’s preamp altogether.

The new Boost/Pad functions are designed to help run the OUT1 and OUT 2 D/A converters at optimum levels, padding their outputs for even lower noise. To find the right setting, adjust either of these controls upwards until you light the respective OUT CLIP LED on the front panel, then back off a few dB to prevent further clipping. You’ll actually hear the noise floor drop as you make these adjustments.

Humbuster™ technology (p. **178**), featured on all 1/4" outputs of the Axe-Fx II, can also provide a significant reduction of ground hum when simple stereo-to-mono cables are used to connect to an amp or other device.

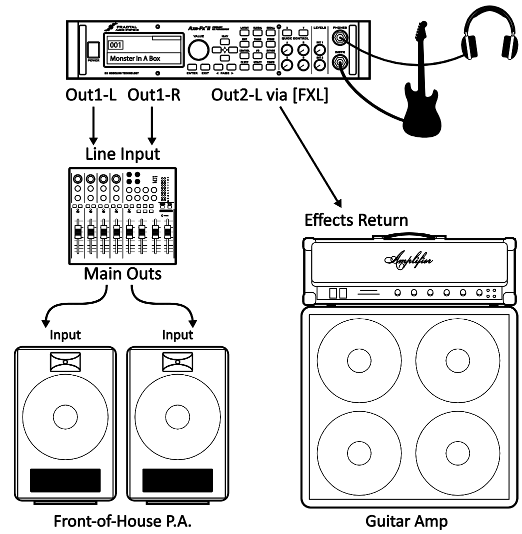
The no-amp/no-cab presets required for this setup are not well suited for headphone monitoring, as what you hear in the headphones will not include any power amp or speaker simulations.

[1] Note: If your amp’s effects loop is the PARALLEL type, the chain of post-FX-loop blocks must be set to output a wet-only signal.

3.4.7 Direct to FOH plus Real Amps on Stage

COMPONENTS:

- Guitar
- Axe-Fx II
- Guitar Amp
  with Guitar Speaker
- Front-of-House P.A.



Global Settings: Default **I/O Settings:** Set OUT1 MODE and OUT 2 MODE as required for stereo or mono

Notes: This setup is similar to others in which the Axe-Fx II is used direct into full-range speakers. Here, however, a special preset feeds both the direct and power amp/cab rig by placing an FX LOOP block between the AMP and CAB, so signal is passed to Output 2 before it is processed by the CAB block and passed to Output 1.
Note: A template is provided in preset memory location 381: "**OUT1->FOH OUT2->CAB**"

Another possibility is to use a preset with two totally separate signal paths. In the example below, the top (direct) signal path contains a fully simulated guitar chain, while the second (backline) does not use a CAB sim.

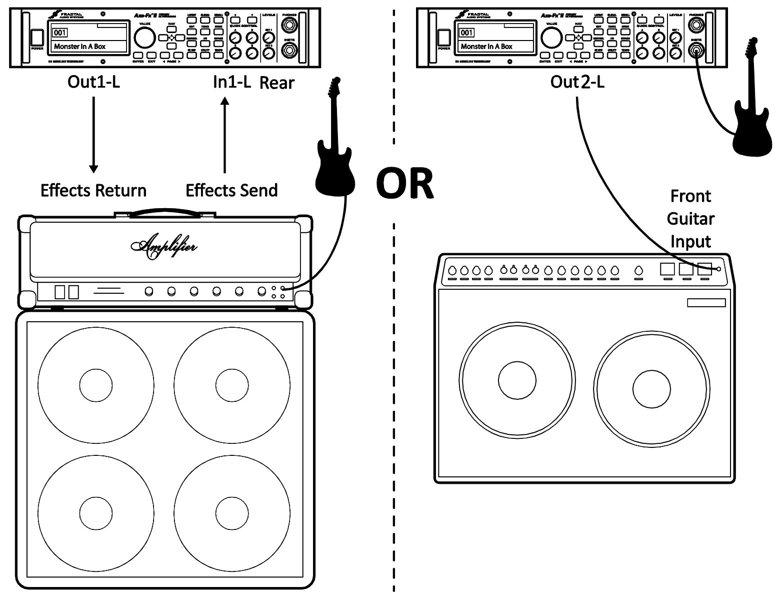
The backline chain differs from the direct chain in several key ways. First, its AMP block has the "SAG" parameter set to "0.00" to disable power amp simulation *in that block only*. Since real guitar speakers and a power amp are being used at Output 2, no CAB block is inserted. *Duplicates* of effects following the amp are placed. This is useful because different settings may be required to "sweeten" each system. The backline chain ends in an FX LOOP block, which is routed via OUTPUT 2 to a power amp. If a very "neutral" amp is used, the SAG of that second AMP block might be set higher for simulated tube power amp tone and dynamics.

Figure 3-2 – A Dual Chain Preset

3.4.8 Axe-Fx II as Effects Processor Only (with Guitar Amps)

COMPONENTS:

- Guitar
- Axe-Fx II
- Guitar Amplifier with built-in or separate speaker cabinet



Global Settings: Defaults OK, since special presets required should not have AMP or CAB blocks!
I/O Settings: Change **INPUT 1 LEFT SELECT** to "REAR" if using the Axe-Fx II in an effects loop as shown above, left. Leave the default setting of "FRONT" if you are connecting a guitar to the Axe-Fx II, as shown above, right.

Notes: Although it was designed for complete guitar signal path simulation, the Axe-Fx II can also be put to superb use as a standalone FX processor. As such, it can be placed between guitar and amp to replace stompboxes or in an amp's effects loop where a rack processor would normally appear.

To use either setup, you will need to create custom presets without AMP or CAB blocks. For operation inside an amp's effects loop, use the REAR input(s) and set **INPUT 1 LEFT SELECT** (section **9.2**) accordingly. Presets in this case will likely contain only those effects that sound best after a preamp: chorus, EQs, delays, reverbs, certain types of pitch shift and modulation, et al.

For the best-case scenario when running in front of an amp, recognize that Output 2 of the Axe-Fx II has an all-new, *extremely* low-noise design that is well suited for this purpose. Effects-only presets should terminate in an FX LOOP block to route signal to OUTPUT2. (In this case, nothing would be connected to the FX RETURN of the Axe-Fx II.)

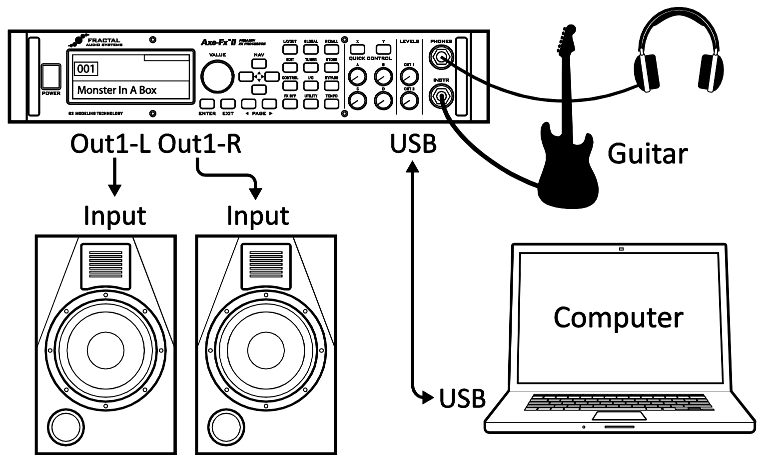
The new Boost/Pad functions are designed to help run the OUT1 and OUT 2 D/A converters at optimum levels, padding their outputs for even lower noise. To find the right setting, adjust either of these controls upwards until you light the respective OUT CLIP LED on the front panel, then back off a few dB to prevent further clipping. You'll actually hear the noise floor drop as you make this adjustment. Humbuster™ technology (p. **178**), featured on all 1/4" outputs of the Axe-Fx II, can also significantly reduce ground hum when simple stereo-to-mono cables are used to connect to an amp or other device.

It is easy enough to extend each of the above configurations for use in stereo. With the Axe-Fx II in a loop, connect OUT1 R to the RETURN of a second amp, bypassing that unit's preamp altogether, or run fully within the loops of two separate amps using IN1 L/R and OUT1 L/R. With the Axe-Fx II between guitar and amp, connect a second amplifier to OUT2 R. The no-amp/no-cab presets required for the scenarios in this section are not well suited for headphone monitoring.

3.4.9 Axe-Fx II as a Computer Audio Interface

COMPONENTS:

- Guitar
- Axe-Fx II
- Computer meeting minimum requirements (p. **12**)
- Powered Monitors
- Headphones (opt.)



Global Settings: Default **I/O Settings**: See below

Notes: The Axe-Fx II offers great features when connected via USB to a computer. Pair this system with headphones, studio monitors, or any other full-range listening system.

- Simultaneously record stereo processed guitar and dry tracks for reamping.[2]
- Route stereo audio from the computer through the Axe-Fx II while simultaneously processing your guitar so you can play along with tracks.
- Send audio from the computer to the Axe-Fx II for processing (and send it back to record the result).
- Use the Axe-Fx II rear inputs to record other line-level audio sources, with or without Axe-Fx II processing.
- Use two-way, high-speed MIDI over USB for control/automation or Axe-Edit, the companion Editor Librarian for the Axe-Fx II.

See section **2.3** for full details on USB audio and MIDI capabilities.

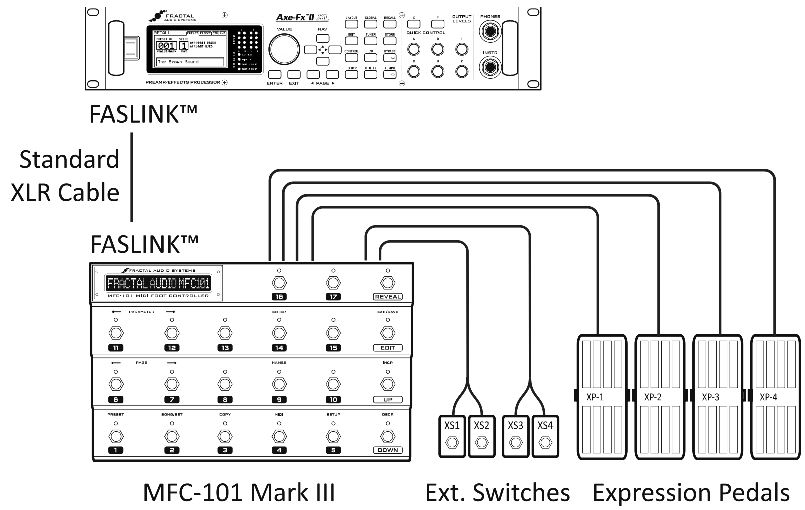
The Axe-Fx II can also be used in conjunction with any 3rd-party audio/MIDI interfaces.

[2] USB 2.0 required for more than two simultaneous channels from the Axe-Fx II to the computer.

3.4.10 Axe-Fx II XL/XL+ and MFC-101 Mark III

COMPONENTS:

- Guitar
- Axe-Fx II
- XLR Cable (for FASLINK™)
- MFC-101 Mk III MIDI Foot Controller



Global Settings: Determined by how the Axe-Fx II is to be used with other connected amps/speakers/etc.

I/O Settings: Determined by how the Axe-Fx II is to be used with other connected amps/speakers/etc.

Notes: The Axe-Fx II XL/XL+ provides a significant advantage over its predecessors in that it can be connected directly to the Fractal Audio MFC-101 Mark III MIDI foot controller via **FASLINK™**. FASLINK™ offers several advantages over traditional MIDI or Ethernet/EtherCON connections. A standard XLR cable—easy-to-find and built to withstand abuse on stage—carries bidirectional communications to support an MFC-101 running in “Axe-Fx Mode,” with automatic preset name and tuner display on the floor, “smart” tri-state Instant Access Switch LED support, TotalSync (where changes made at the Axe-Fx also happen on the MFC) and more.

When FASLINK™ is used, the MFC-101 still transmits/receives standard MIDI data to/from the Axe-Fx II XL/XL+, which can then forward messages to other devices (Note: The MIDI THRU port is hardwired to MIDI IN… You must instead use the MIDI OUT port and enable MFC ECHO TO MIDI OUT in the MIDI page of the I/O menu of the XL. This is not required on a Mark II, but you must ENABLE MIDI THRU in the same menu on that unit.)

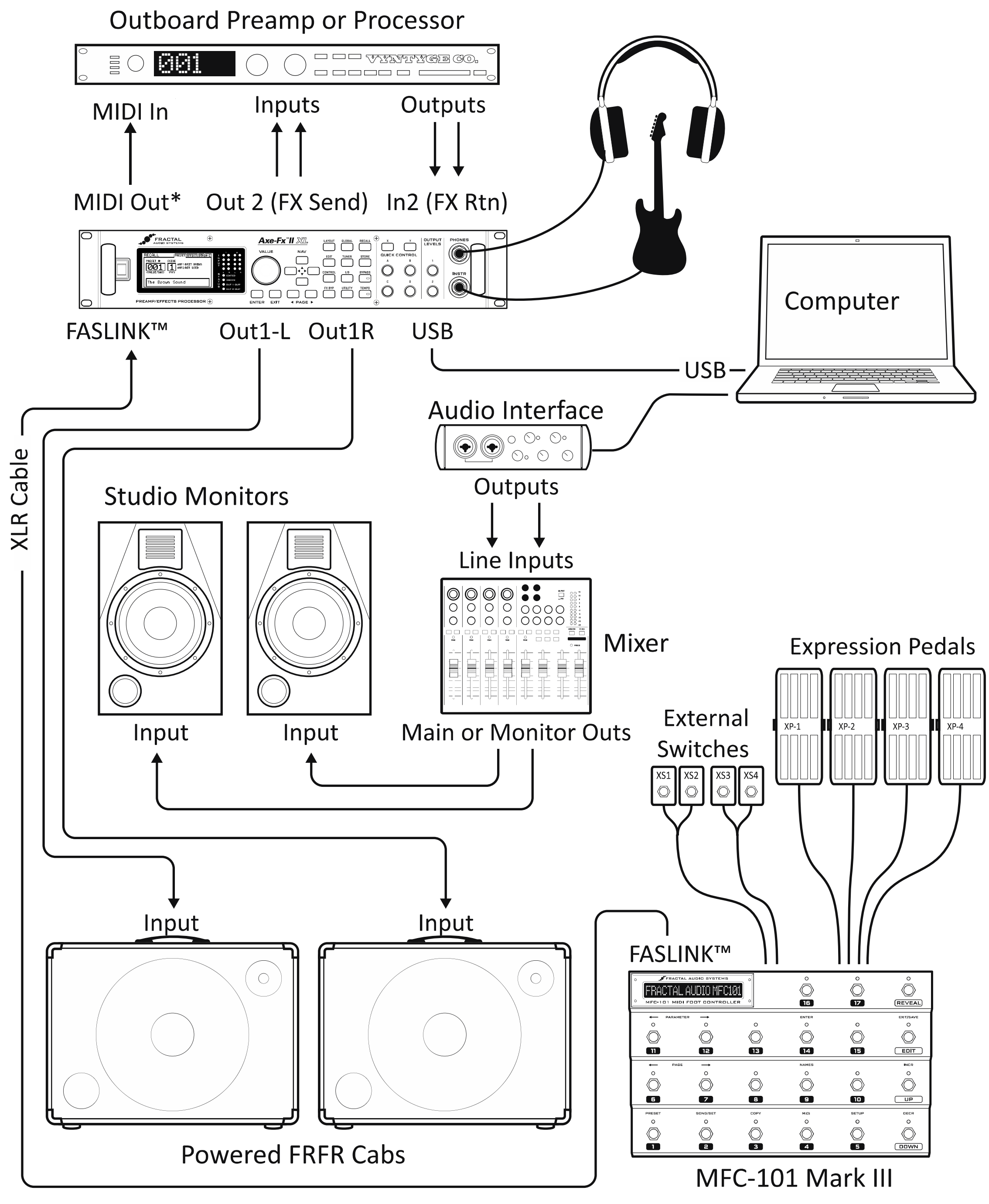
Though the XL and Mk III are shown above, you can in fact use *any* Axe-Fx II with *any* MFC-101. For a complete list of connectivity options, please see the MFC-101 Quickstart Guide, available on the support page of fractalaudio.com.

High-quality XLR, Ethernet and EtherCON cables can be found at http://www.fractalaudio.com/cables

WARNING: Do not connect the MFC-101 power adapter to the MFC-101 or to the Phantom Power jack of the Axe-Fx II while using FASLINK™ or an Ethernet/EtherCON cable. The Axe-Fx II provides required power to the MFC from its internal power supply.

Always ensure that the Axe-Fx II is powered OFF before connecting or disconnecting the MFC-101.

3.4.11 Axe-Fx II XL/XL+ & MFC-101 Mark III: One Possible "Big Rig"



Here the Axe-Fx II XL/XL+ is the centerpiece of a "big rig," combining the capabilities of several other diagrams shown above. The main outs feed a pair of powered FRFR cabs, so you can design and monitor sounds through the speakers you'll perform with. Meanwhile, USB provides all of the usual computer audio and MIDI integration features, with full-range studio monitors connected to a 3rd-party audio interface for monitoring and playback. We've added an outboard processor in the FX LOOP (so you can clone its settings and put it on eBay ;)

*Note that **MIDI OUT** is used for MIDI THRU on the XL when an MFC-101 is connected via FASLINK™ or EtherCON. Enable **MFC ECHO TO MIDI OUT** (In I/O:MIDI). On an earlier Axe-Fx II, the MIDI OUT/THRU would be used.

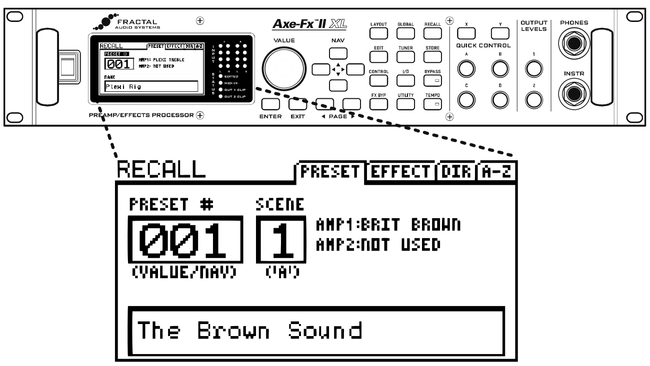
4 Basic Operation and Editing

Once you have set up your Axe-Fx II with speakers/amps/monitors or a pair of headphones, you can begin to audition the factory preset sounds and learn to make changes by following this detailed guide to basic operation. For a super-condensed version, see the **60-Second Edit Guide** on p. **172** of the Appendix.

4.1 Presets

Each factory preset is a complete “end-to-end” guitar sound with amps, cabs, and effects—perfectly mixed and ready to perform. In the sections that follow, you’ll learn to view, edit, and create presets from scratch. To begin, let’s look at the ways to load the presets stored in the Axe-Fx II.

Presets are accessed from the PRESET page of the RECALL menu. There are 4 ways to change the preset:

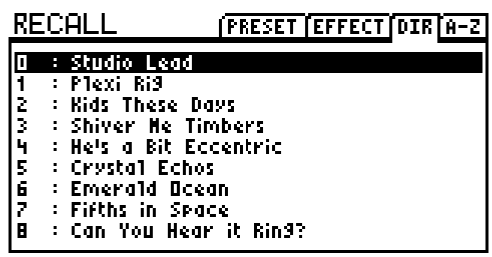


1. The **VALUE** knob selects Presets sequentially.

2. The UP/DOWN **NAV** buttons change by +/- 1

3. The LEFT/RIGHT **NAV** buttons change by +/- 10

4. Turn the **PAGE** to “DIR” for a **directory** of all presets by number. The “A-Z” page lists all presets by name. Use the wheel to navigate and press **ENTER** to load the selected preset.



The Axe-Fx II Mark II has 384 presets. The XL has 768.

Presets can also be loaded using MIDI program change commands from external devices, such as an MFC-101 or 3rd party MIDI Interface. Although it isn’t indicated in the display, presets are organized into banks. **Bank A** contains presets 0–127, **B** contains 128–255, and **C** contains 256–383. On the Axe-Fx II XL/XL+, presets 384– 511 are in Bank D, presets 512–640 are in Bank E, and presets 641–768 are in Bank F. (See p. **192** for a cross-reference on CC#0 bank/program changes and preset numbers.)

An onscreen message will be displayed in the main RECALL screen for any preset containing one or more links to Global Blocks (p. **131**).

Each Axe-Fx II preset also contains eight **SCENES**. The current Scene is shown on the recall screen and can be changed with the “A” QUICK CONTROL knob. For more on setting up SCENES, see p. **184**.

4.2 The Grid

Press **LAYOUT** for the grid
RECALL to go back

The grid, located on the EDIT page of the LAYOUT menu, is a 12 × 4 “matrix” where effect “blocks” are inserted and connected to build presets. The INPUT appears at the left, the OUTPUT at the right. The display can fit a 5 × 4 section of the 12 × 4 grid at any one time, with the ability to scroll to off-screen areas using the **NAV** buttons. A bottom scrollbar indicates where you are in the overall left-to-right layout.

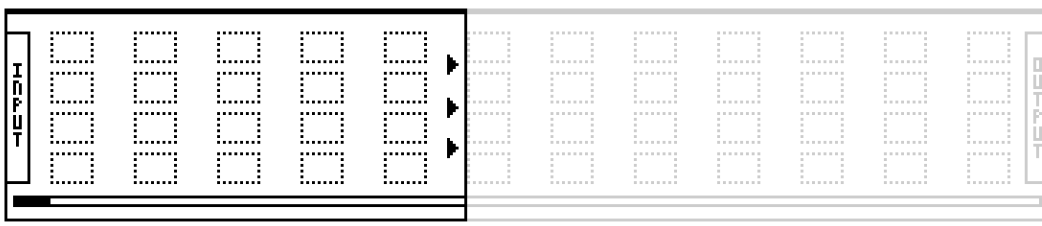
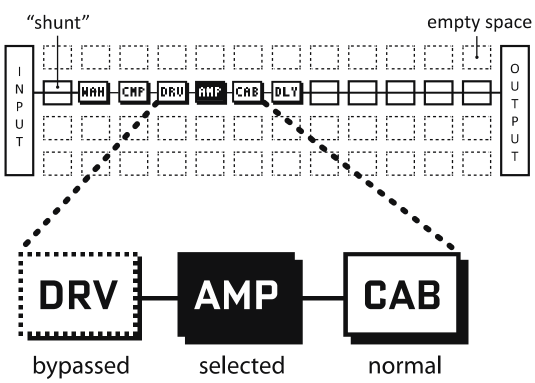


Figure 4-1 – In this image of an empty grid, off-screen areas are grayed.

4.2.1 Inserting and Removing Blocks

As explained in “The Concept” on page **6**, the Axe-Fx II grid must be populated with blocks—components pulled from a large inventory of amps, cabs, stompbox, studio effects, mixers, and more. The insertion, modification, or removal of blocks happens at the grid cursor, a filled rectangle controlled with the **NAV** buttons.



The **NAV** buttons select grid spaces or blocks.

Turning **VALUE** displays the list of available blocks, plus options for SHUNT (see below) and NONE.

Press **ENTER** to confirm or **EXIT** to cancel changes.

Press **FX BYP** to bypass or engage the selected block.

Figure 4-2 – Grid Operations

<u>To INSERT a BLOCK into an empty space…</u>

- Use the **NAV** buttons to select the desired empty grid location.
- Turn the **VALUE** wheel. Available block names (AMP 1, CAB 1, etc.) will be displayed on the screen in alphabetical order, and the selected block (if visible) will flash.
- When the desired item is found, press **ENTER** to insert it into the grid. To cancel, press **EXIT** instead.

As detailed above in **The Inventory/Grid Concept** (p. **6**), every preset draws from its own complete inventory of available blocks. As you place them on to the grid, they are removed from the inventory.

The total number of blocks you can insert in any one preset is dictated by the fact that CPU utilization must not exceed around 94%. Each block has a "cost," and when the sum of all blocks reaches the limit, a warning message prevents you from adding additional blocks. The Axe-Fx II is extremely powerful, and most presets do not come close to the limit. See **Understanding Preset Size Limits** on p. **173** for more on this subject.

To CHANGE the type of an existing BLOCK…

- Use the NAV buttons to select the desired block.
- Turn the VALUE wheel. Available block names (AMP 1, CAB 1, etc.) will be displayed on the screen in alphabetical order, and the selected block (if visible) will flash.
    - The Axe-Fx II offers multiple instances of most block types (ex: 2 Amps, 4 Graphic Equalizers, etc.) To keep the list manageable, only the "next-in-line" instance is shown as you cycle through the insertion menu, so AMP 2 is hidden until you have placed AMP 1.
- When the desired block is shown, press ENTER. To cancel without making changes, press EXIT instead.

To REMOVE an existing BLOCK…

- Use the NAV buttons to select the block you wish to remove.
- Turn the VALUE wheel until **NONE** is displayed and press ENTER, or press EXIT to cancel.

Shortcut! To remove any block except a Shunt, select it and press EXIT, ENTER, EXIT, ENTER

4.2.2 Shunts

A shunt is a passive connector—a utility block that carries signal through otherwise empty grid locations. Shunts are required because, although you may only wish to have a few components in your chain, signal does NOT flow through an Axe-Fx II preset unless you completely connect the INPUT to the OUTPUT. A single Axe-Fx II preset can contain up to 36 shunts.

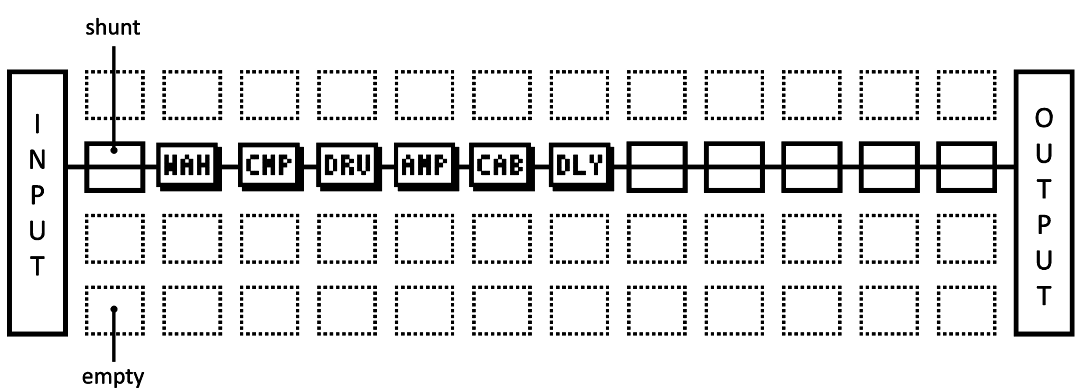


Figure 4-3 – Shunts are used to span distances between INPUT, BLOCKS, and OUTPUT.

In the figure above, a shunt connects the Input to the WAH, and five shunts connect the Delay [DLY] to the Output. Empty grid locations are indicated by dotted outlines and the absence of a three-letter abbreviation or thru-line.

To INSERT a SHUNT into an empty grid location…

- Use the **NAV** buttons to select the desired empty grid location.
- Turn the **VALUE** wheel once to the right. "SHUNT" will be displayed in a popup, and the selected grid space will flash if not hidden.
- Press **ENTER**. To cancel without inserting, press **EXIT** instead.

To CHANGE any other block type into a SHUNT…

- Select the desired block.
- Press **EXIT**. "SHUNT" will be displayed in a popup, and the selected space will flash if not hidden.
- Press **ENTER**. To cancel without inserting, press **EXIT** instead.

4.2.3 Connector Cables

As mentioned above, a preset's INPUT must be connected to its OUTPUT in order for that output to produce any sound. Blocks (including shunts) create the *components* of a chain, but these still need to be *connected* to one another for signal to "flow." This is done using routing connectors, commonly referred to as "cables."

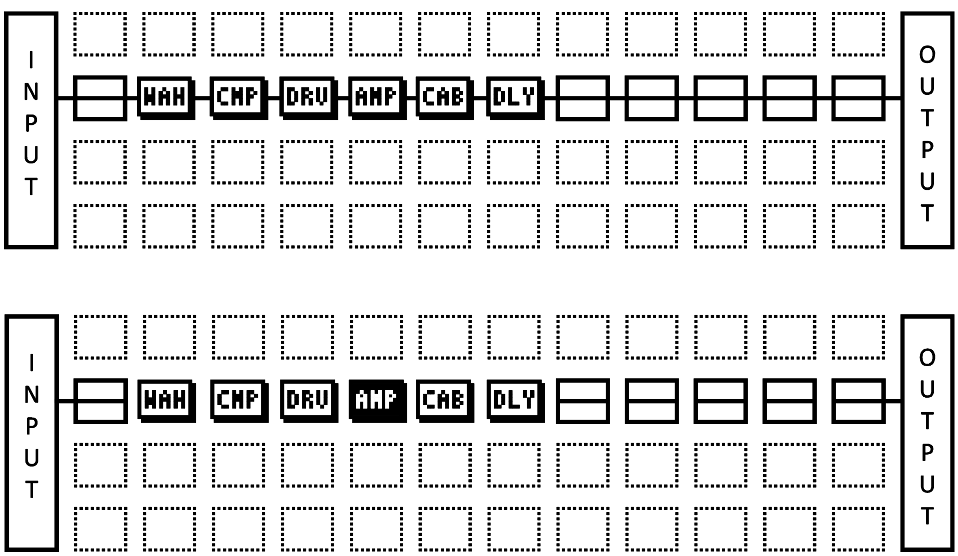


Figure 4-4 – Two otherwise identical presets shown WITH and WITHOUT connector cables between the blocks.

The second example above would produce NO SOUND as nothing is "wired up" to pass signal to the output!
If a preset is unexpectedly silent, inspect it carefully for one or more missing cables.

! A key concept to understand is that every connector/cable is *stereo*. The grid allows 4 full stereo paths, and most blocks are stereo in/stereo out. Even blocks which process audio in mono (like AMP or DRIVE output in stereo and have a left-right BALANCE control. The Effects Guide (p. **39**) covers the mono/stereo capabilities of each block type in detail. For now, remember that *every connector/cable is stereo* in and of itself.

The Rules of Axe-Fx II Cables

- No cables = No sound. Even one missing link will break the entire chain.
- Signal always 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 ★ 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.

- Cables are created AUTOMATICALLY between the INPUT and any blocks in the first column.
- Cables are created AUTOMATICALLY between the OUTPUT and any blocks in the last column.
- You may freely SPLIT or MERGE the signal up to four ways at any point. This is sonically transparent and there is zero risk of inherent phase or other problems in the split/merge itself. CROSSING is also possible.

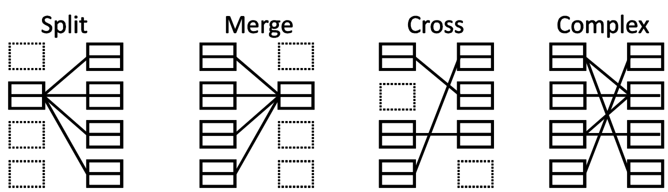


Figure 4-5 – Cables can CROSS one another without issue. Routings can be as COMPLEX as required.

To Create a Connector Cable…

- Use the **NAV** buttons to select the block where you wish the cable to BEGIN.
  - You can't start from an EMPTY grid space!
  - The first and last grid columns are automatically connected to the INPUT and the OUTPUT.
- Press **ENTER**. The selected block and its neighbor to the right will alternately flash as "selected."
- Use the UP or DOWN **NAV** buttons to select the desired destination block.
  - Remember that you will be prevented from trying to select blocks in other columns.
  - It is possible to select an empty location, but a shunt will be added automatically at the destination if you complete the cable.
  - Be sure to select a destination that is not already connected to the origin block, 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.

- Select the block where the cable begins.
- Press **ENTER**. The selected block and its neighbor to the right will alternately flash as "selected."

- Use the UP or DOWN NAV buttons to select the "other end" of the cable you wish to remove.
- Press ENTER. To cancel without removing, press EXIT instead.

A Shortcut for Spanning Empty Spaces

This shortcut allows you to span multiple empty grid columns with a series of automatic shunts and cables. This technique is especially useful when you have placed your last effect block and need to wire it to the output or if you want to fill an empty row with shunts.

- Select any block that is followed by a series of empty spaces.
- PRESS and HOLD the ENTER button. The intervening spaces will be automatically filled with shunts and connected with cables.

You can also use this shortcut for small "runs" to connect any blocks with one or more columns of empty space between them, but be careful: any existing cables encountered along the way will be REMOVED by the process!

A Word on Shunts and Cables

In the real world, cables and connectors can have an impact on the tone of a guitar rig. In the Axe-Fx II, nothing could be further from the truth. Shunts and connectors, whether long, short, split, merged, or crisscrossed in a huge mess, have *absolutely no sonic properties whatsoever*. They do not impart color, add latency, suck tone, create load, invite hum, develop shorts, or get tangled up in the road case. A word of advice: merging identical copies of a split signal will result in additive level increase and should be avoided in favor of simply increasing one or another gain or level parameters. And remember… *both shunts and cables are stereo!*

4.2.4 Moving Blocks on the Grid

The LAYOUT menu also includes a **MOVE** page with tools to move individual blocks or entire rows or 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 can 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.

- Press LAYOUT.
- Use the PAGE buttons to select the **MOVE** tab.
- Select a function with the VALUE wheel:
  - EFFECT LEFT/RIGHT/UP/DOWN
  - COLUMN LEFT/RIGHT
  - ROW UP/DOWN
- Use the NAV buttons to select the target effect **block** or **row/column** you wish to move. Look for the SOLID square(s) in the grid representation on the screen.
- Press ENTER to execute the move.
  - Repeat this step to move the same target again in the same direction.

4.2.5 Example Presets on the Grid

Four sample presets are shown below as visualized in Axe-Edit, the companion software editor to the Axe-Fx II. Review the diagrams to get a sense of how presets are constructed and how they appear on the grid.

EXAMPLE 1: Simple Amp Tone – In this extremely simple preset, an AMP and a CAB are combined for a straight-ahead tone. Shunts and connectors (which appear in Axe-Edit almost as one continuous cable), connect the INPUT to the AMP, the AMP to the CAB, and the CAB to the OUTPUT.

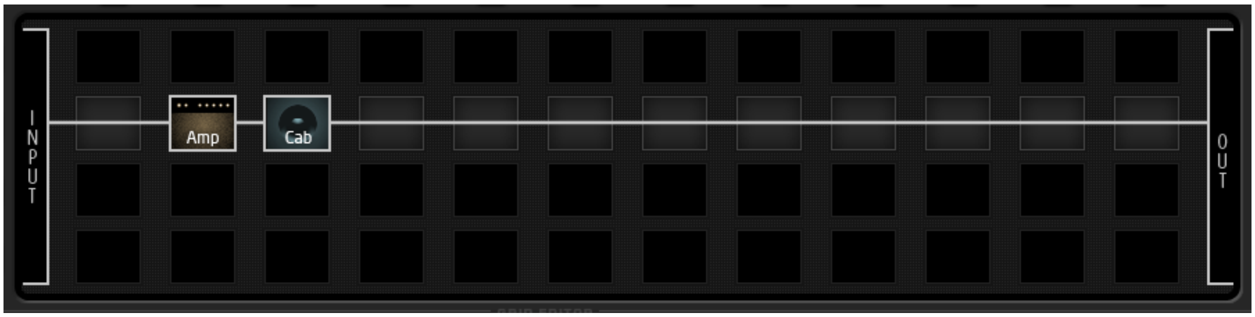


Figure 4-6 – Amp/Cab Preset

EXAMPLE 2: Long Pedal Chain – Here, effects are strung together to create a giant virtual rig. Compressor, Volume, Wah, Whammy, Tremolo, Overdrive, Phaser, and Flanger are all connected to an AMP/CAB combination, followed by Delay and Reverb. As you can see, the Axe-Fx II supports a great number of simultaneous effects; this preset uses just a portion of the unit's total CPU power, leaving considerable room for more effects on other rows.

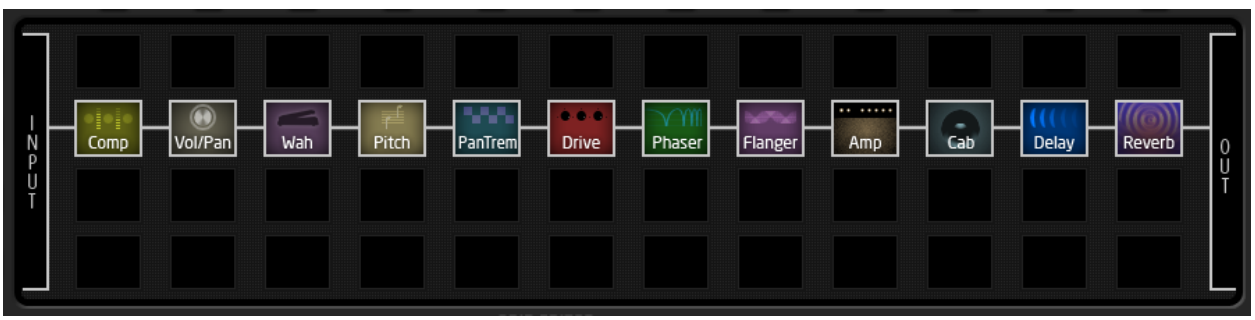


Figure 4-7 – A Pedal Chain

EXAMPLE 3: Dual Amp Preset – This preset shows a dual-amp rig. A series of effects (Wahwah, Drive, Phaser) begins the chain, then the signal is split into two amps and two cabs. Signal is panned hard left and right with CAB block **BALANCE** controls and then merged to feed stereo post effects. A Multiband Compressor helps balance dynamics, while spaces in the chain leave plenty of room for future expansion.

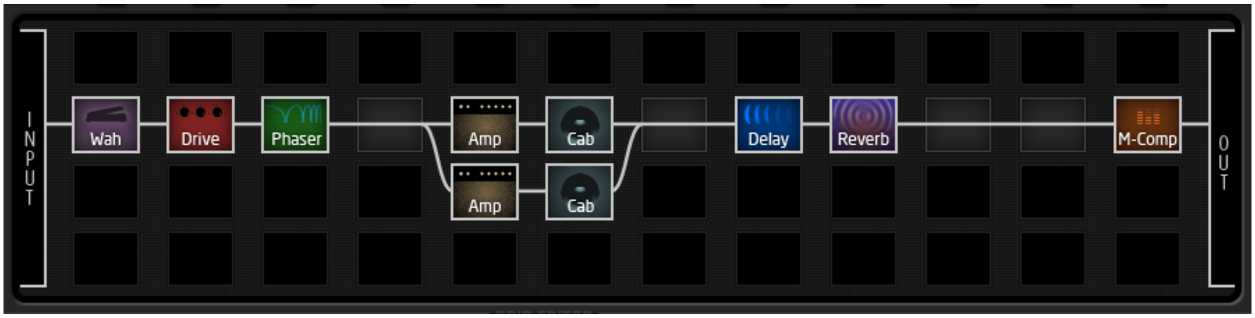


Figure 4-8 – Dual Amp Preset

EXAMPLE 4: Complex Routing – The program below demonstrates a more intricate routing scenario. The signal is split and merged at various points between the input and the output, with various effects appearing before and after two independent amps. Notice the complex feedback-loop routing in parallel with the “dry” signal feeding the amps, and the reverb at the end of the chain running in parallel. (Reverb level is controlled by an envelope for a “ducking” effect).

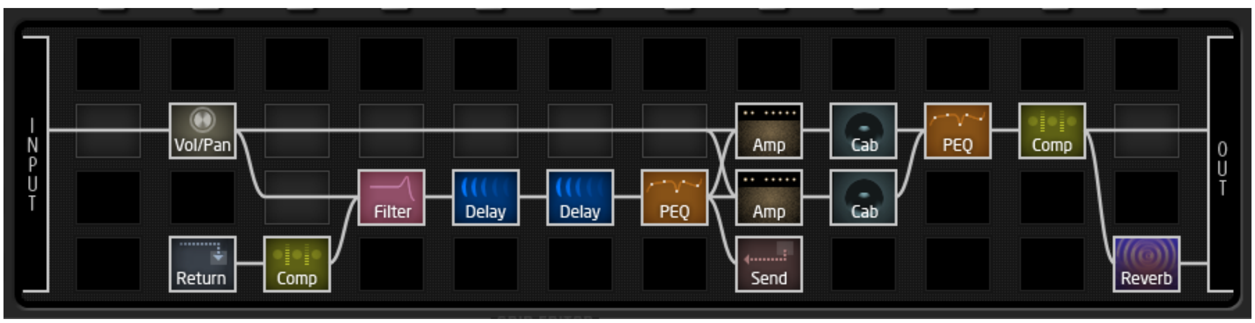


Figure 4-9 – Complex Preset

MORE EXAMPLES…

The presets in this section represent only four examples of the nearly limitless possibilities that can be created using the GRID, BLOCKS, and CONNECTORS of the Axe-Fx II. In addition to reviewing these diagrams, you can gain valuable insight and ideas by exploring the factory presets or by discussing techniques with other members of the Fractal Audio community. Visit forum.fractalaudio.com and join the discussion.

You can download additional presets for your Fractal Audio products at http://axechange.fractalaudio.com

4.3 Editing Sounds

The blocks of the Axe-Fx II represent diverse types of real-world equipment. In the same way that such hardware devices are equipped with different controls, blocks also typically have many adjustable settings called **parameters.** Parameter settings determine precisely how an effect will sound. These are arranged on PAGES in the display of the Axe-Fx II. The system of parameters and pages for any block is referred to as its **EDIT menu**.

To Open the EDIT Menu for Any Effect Block…

With a preset loaded in normal RECALL mode…

- Press LAYOUT to display the GRID.
  - If the GRID is not shown at first, press ◂ PAGE to reach the EDIT tab.
- Use the NAV buttons to select the desired block.
- Press EDIT to display the EDIT menu.
  - NAV buttons select parameters on the page.
  - The name of the selected parameter will be displayed in **inverse** with "VAL" shown above if it is a knob, or ">" shown to its left in text menu pages.
  - In scrolling menu pages, the sidebar has an indicator to show relative position.

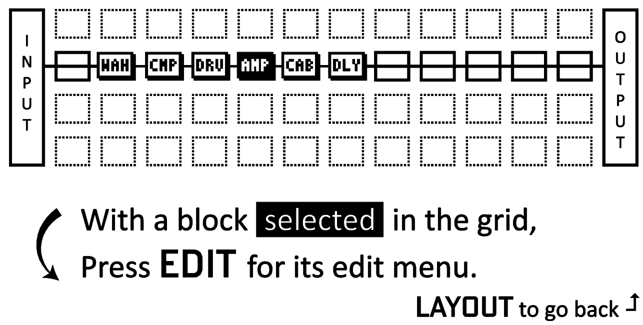
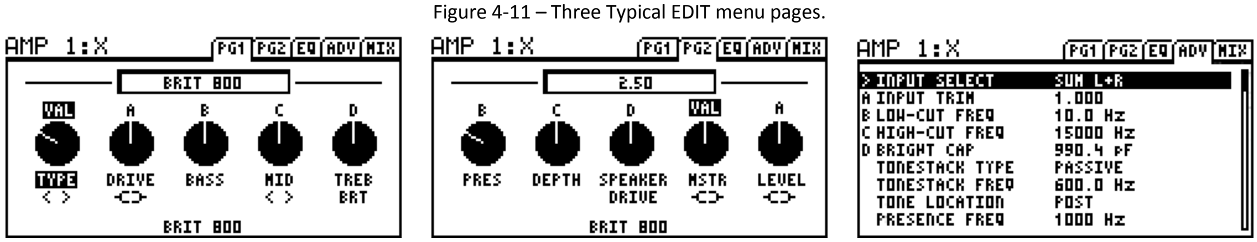


Figure 4-10: Press EDIT with any block selected on the Grid

- The VALUE wheel changes the selected parameter.
  Changes can be auditioned in real-time by playing while you edit.
- For effect blocks with multi-page menus, the PAGE buttons select between them.
- Press LAYOUT to return to the grid, or press RECALL to go back to play mode.

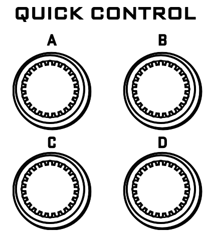
Pressing EDIT repeatedly will step through the EDIT menus for all blocks in a preset, sparing you a return trip to the grid. You can also press EDIT without entering the grid to go directly to editing a selected block in any preset.

Figure 4-11 – Three Typical EDIT menu pages.



The above figure shows three pages of typical EDIT menus (from the AMP block). Notice the tabbed labels near the top: PG1, PG2, EQ, ADV, and MIX. Select between these using the ◂ PAGE ▸ buttons. The third example shows a text-only menu. Some blocks, like the Graphic EQ (p. **78**), Parametric EQ (p. **97**), and others, have specialized graphical displays designed to provide rich, intuitive editing experiences.

4.3.1 Quick Control



The functions of the **QUICK CONTROL A**, **B**, **C** and **D** knobs depend on which Axe-Fx II menu or function is selected.

- In the main **RECALL** screen, the A knob selects SCENES. See section **16.15** on p. **184.**
- In every EDIT menu[1], A, B, C and D are dynamically mapped to four on-screen parameters for easy editing of multiple controls without the **NAV** keys. Mapping is indicated by a small A, B, C or D above or to the left of on-screen parameters.
- When editing Amp **TYPE**, A, B, and C are conveniently mapped to **DRIVE**, **MASTER**, and **LEVEL**.
- While on the MANUAL page of the **CONTROL** menu, A, B, C and D work as "Manual knob" modifier sources, allowing control of assigned parameters. See p. **144**.
- While STORING a preset, A, B, C and D are used as shortcuts to enter text. See p. **38**.

4.4 X/Y Switching

One of the new features of the Axe-Fx II is X/Y switching, available on many different effect block types:

Axe-Fx II Mark II: Amp, Cab, Chorus, Delay, Drive, Flanger, Pitch Shifter, Phaser, Reverb, and Wahwah.
Axe-Fx II XL/XL+: All of the above, plus Compressor, Gate/Expander, Graphic EQ, PEQ, Trem/Pan, and Mixer.

Each instance of each of these blocks is equipped with two fully independent sets of parameters—"X" and "Y"—making it possible to switch between two completely different sound settings using only one block. You can change between X and Y instantly at the touch of a front panel button (while editing) or by using MIDI remote control (during performance). When you save a preset, the current X/Y state of each X/Y block is saved as well, so the preset loads with your choices already selected. The X/Y state of each X/Y block is also stored in every scene (see **Scenes** on p.**184**).

The benefits of X/Y Switching are considerable. You might simulate amp channel switching (X as American Clean and Y as British Crunch). *This won't require the CPU overhead of two separate amp blocks!* Another application is channel-switching effects. Instead of painstakingly tuning modifiers for morphing (p. **136**), you can just dial in X, dial in Y, and switch between them at will. Imagine a delay that can be switched at will from pristine dotted-eighth-note echoes with light feedback to one with saturated, modulated quarter-note echoes and heavy feedback.

X/Y switching is generally seamless, but the AMP block requires a moment to reload when you change XY.
It is important to understand that **MODIFIERS** (see p. **136**) are SHARED between X and Y states.

To Use X-Y Switching

- Open the desired block for editing.
- All blocks default with the X state selected. Dial in all parameters for the X state.
- Press **Y** and dial in all parameters for the Y state.
- Save the preset (see p. **38**).

Each available X/Y switch has its own dedicated global MIDI CC# assignment for remote switching[2]. This option is found in the I/O:CONTROL menu. See p. **152** for details.

[1] Exceptions: Quick control functions work in PEQ blocks, but letters are not shown. Mixer type blocks do not support Quick Control.

X-Y / Y-X Copying

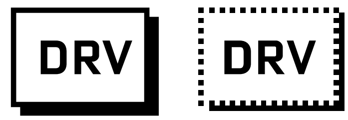
You can copy all of the settings from X to Y, or Y to X, by double tapping the button for the one you want to copy *to,* and then pressing ENTER to confirm. So, to copy X to Y, double-tap Y and then press ENTER. If you accidentally double tap when you don't want to copy, press EXIT to cancel.

4.4.1 X/Y Quick Jump

The X and Y buttons also double as user-definable Quick-Jump keys. These allow quick access to the EDIT menus of your two favorite blocks from almost any screen without needing to "drill down" through the grid. You might, for instance, set "X" to jump to **AMP1**, while "Y" might be assigned to bring up the **PHASER1** menu.

Quick-Jump is disabled if you are already in the EDIT menu of another block (including Global Blocks or Modifier sub menus), or if you are in the process of SAVING (where X and Y are used for quick character entry).

Quick Jump settings are assigned in the X/Y page of the I/O menu, detailed on p. **154**.



Engaged Bypassed

4.5 Bypassing a Block

Can you imagine a stompbox that can't be stomped? Neither can we.
Any effect[3] on the Axe-Fx II can be bypassed (or engaged) in three different ways:

1. Press the front panel FX BYP button while the block is either selected in the grid or open to edit. *Note that accidentally double tapping or holding FX BYP will enter the Global Blocks menu instead (see p.* **131***). Press* EXIT *and try again if this happens.*
2. Use a MODIFIER (section **7**) attached to the block's **BYPASS MODE** parameter.[4]
3. By remote control via the block's global remote bypass function, which may be set to "PEDAL" or to any MIDI CC# on the CTRL page of the I/O menu (p. **148.**)

A bypassed block is shown on the grid with a dotted outline.
When you bypass any block, one of several things can actually happen depending on how its **BYPASS MODE** parameter is set. For more on this, see the section on **Common Mix Parameters** beginning on p.**128**.
Every SCENE stores the bypass state for every block in a preset. See section **16.15** on p. **184.**

4.6 Loading Effects from another Preset

The **Recall Effect** function makes it possible to load block settings directly from one preset to another. To perform this operation, select the EFFECT tab of the RECALL menu using the PAGE buttons. Select the preset and block that you want to load *from* and press ENTER. "EFFECT RETRIEVED!" will be displayed, and the "local" block (whether or not it is already on the grid in the current preset) will be updated with all settings from the "retrieved" block. All settings for both X and Y are recalled for those effects that support X/Y. Modifiers (p. 136) are also recalled. If the block you are trying to load is not found in the preset you try to load it from, "EFFECT NOT FOUND" will be shown instead. Note that you can also recall NOISE GATE, MAIN OUT, and CONTROLLER (LFO1, LFO2, etc.) settings.

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[2] As noted in the section which covers I/O parameters, X is in fact the "ON" state for these CC switches (64-127) and Y is the OFF (0-63).
[3] The MIXER, FB SEND, and SHUNT blocks have no bypass functions.
[4] The ENHANCER has no **BYPASS MODE** parameter.

4.7 Saving Changes

After making various changes, you will undoubtedly want to save the results of your edits.

To store a sound in place, without changing its name or location…

- Press STORE to show the STORE screen.
- Press ENTER to initiate the process, and ENTER again to confirm.

The message "STORED!" is displayed when the operation completes.

To store a sound to a new location or with a new name…

The Axe-Fx II has hundreds of numbered preset memory locations. It is possible at any time to save any preset into any location. It is also possible to change the NAME of a preset before you store it.

- Press STORE to show the STORE screen.
- Use the NAV buttons to select between the two available functions:
  - The **LOCATION** parameter selects where the preset will be stored:
    - The VALUE wheel selects numbered memory locations. NAV left/right skip by 10.
  - Navigating down to the **NAME** parameter allows you to edit preset names before saving:
    - Turn the VALUE wheel to select from all available characters.
    - The "A" knob selects UPPER CASE letters.
    - The "B" knob selects LOWER CASE letters.
    - The "C" knob selects NUMBERS.
    - NAV left/right or the "D" knob will move the cursor position.
    - The "X" button inserts a character at the cursor position.
    - The "Y" button deletes a character at the cursor position.
    - You can use up to 23 characters in a preset name.
- Press ENTER to Store, then press ENTER *again* to Confirm.

The message "STORED!" is displayed when the operation completes.

4.7.1 Swapping the Locations of Two Presets

The Axe-Fx II has a new feature that allows you to SWAP the locations of two saved presets. This is useful, for instance, if you want to re-order without overwriting factory presets, or if you need to move a "keeper" preset to a different location so you can overwrite its previous location with a newer entry. To SWAP two presets:

- Press STORE and PAGE to the right to the SWAP page.
- Use NAV keys and the VALUE wheel to select the two presets whose locations you want to switch.
- Press ENTER to Store.
- Press ENTER to Confirm. The message "SWAPPED!" will briefly appear when the operation completes.

5 Effects Guide

The Axe-Fx II offers 34 different basic block types that can be combined freely up to the limit of available DSP resources to create your own presets. An alphabetical listing of block types follows.

5.1 Amplifier [AMP]

The Amp block reproduces the sounds of an impressive array of vintage and modern guitar and bass amplifiers, with 256+ different "types" based on stock, custom, and hybrid models. It uses our newest Quantum Amp Modeling technology, including proprietary multi-stage nonlinearity generators to create ultra-realistic distortions. Separate virtual preamp and power amplifier stages create rich, cascaded drive tones that cannot be obtained using simpler modeling methods.

You can achieve great tone using only the basic amp controls on the first page or two. Should you desire to dig deeper, you'll find many exciting parameters that allow you to tweak and adjust the deepest aspects of your amp's sound. These are detailed below. Axe-Edit also makes using the amp block easier, with a few simple pages organized with common parameters grouped together.

Note: The **Cab** block (p. **50**) is vital in creating an overall sound. If you're not getting the tones you're expecting from an amp, try different cab settings!

The heart of the **Quantum™** Amp block is its capacity for ultra-realistic distortions, created using our proprietary multi-stage non-linearity generators, with virtual **preamp** and **power amp** stages emulating the different types of distortion generated by a tube preamps and power amps. The models also offers accurate behavior for the drive and tone stack controls over the full range of operation. This means you can set onscreen controls to match those of the original amps to dial in familiar settings and generally achieve expected results.

Each Axe-Fx II preset can use two fully independent **Amp** blocks. Presets that use a single amp block run in **high resolution**, providing the utmost in fidelity and resistance to aliasing. This mode is automatic and is selected whenever there is only one amp block in the layout grid. Adding a second amp block will revert both to **normal resolution** (in which the modeling is still of *extremely* high quality.) Note that switching between high and normal resolution presets there can be an additional switching latency as amp blocks need to be soft-reset.

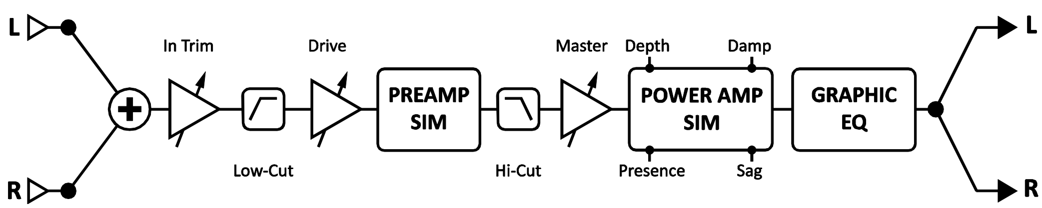


Figure 5-1 – A simplified diagram of the AMP block. The AMP block processes audio in mono.
INPUT SELECT and BALANCE parameters allow flexibility when combining the AMP with stereo effects.

The Amp block supports X/Y Switching. See p. **36** for more information.

5.1.1 Amp Type

Amp types are presented in an alphabetical list. A complete table appears on p. **165**. You can adjust **DRIVE**, **MASTER,** and **LEVEL** parameters directly from the **TYPE** page by using the A, B, and C knobs.

5.1.2 Amp Preamp Page ("PRE")

INPUT DRIVE – (aka "Drive") sets the amount of preamp gain/distortion. Used in conjunction with the **MASTER** (see below), **INPUT DRIVE** determines whether the sound will be clean, slightly broken up, moderately overdriven, or completely distorted.

Our modeling faithfully reproduces the sound of the treble peaker circuit on the **INPUT DRIVE** control found on many amps. This can be heard as the low frequencies are reduced more than the highs when the **INPUT DRIVE** is turned down (and vice versa).

For amps that have no **MASTER VOLUME**, the **INPUT DRIVE** also functions as the amp's VOLUME control.

> *NOTE: Amp models that simulate "jumpering" the inputs of a 4-hole amplifier (e.g. PLEXI 50W JUMP, HIPOWER JUMPED, etc.) have separate TREBLE DRIVE and NORMAL DRIVE controls, which behave as Controls for their respective channels.*

OVERDRIVE – The **OVERDRIVE** control appears only for certain amp types. Note that **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.

INPUT TRIM – Amps without **OVERDRIVE** will display the **INPUT TRIM** instead. This allows you to adjust for more or less preamp gain than the actual circuit being modeled. This is different than the Input **DRIVE** control because **DRIVE** interacts with the surrounding circuitry, changing frequency response as it is varied.

BOOST – Toggles an additional 12 dB of gain at the input of the amp sim. (For amp types that have an **OVERDRIVE** control (see above,) **BOOST** appears only on the ADVANCED page.)

BASS, MID, TREBLE – While other modelers use simple filters to approximate amp tone controls, the Axe-Fx II replicates exactly the frequency and phase response of a classic passive tonestack. In most instances, matching knob settings between the Axe-Fx and the original amp will recreate the same tones.

Some of the original amps simulated on the Axe-Fx II do not have all of the tone controls offered on our models. Some, for example, have no mid control. To faithfully simulate the configuration of the original, set any superfluous controls to noon (or "0.00" if you are using the "ACTIVE" tonestack type; see below). Of course, you may still adjust to achieve tones the original amp does not have.

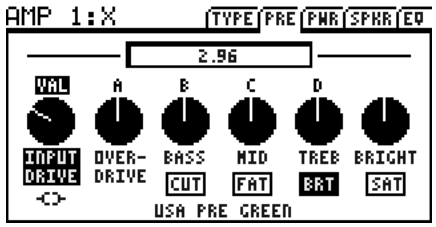
❗ Extreme tone and high gain settings can cause pickup squealing or excessive noise. This is especially true with the **TONESTACK TYPE** (p. 54) set to "ACTIVE."

CUT SWITCH – (engaged by a switch beneath **BASS**) reduces the amount of low frequencies into the amp simulation. This can be used to achieve a "tighter" tone or to reduce low-end "flub".

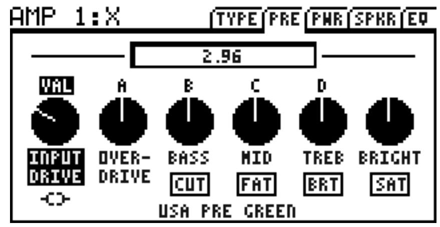
FAT SWITCH– In the same way the **TREBLE** knob (above) also controls **BRIGHT**, the MID operates as a "Fat" switch, emphasizing midrange "body" by shifting the tonestack center frequency down.

BRIGHT SWITCH – Many amplifiers contain a "treble peaker," included as a pull or toggle switch, or even hard-wired. Every amp **TYPE** on the Axe-Fx II 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 (p.54). 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 **BRIGHT** state is ON.

To turn the Bright, Fat, or Cut switches ON or OFF, use the NAV keys to select the knob above them and press ENTER. The label beneath the knob will fill to indicate that the circuit is engaged.



BRIGHT switch ON



BRIGHT switch OFF

BRIGHT (knob) – This is a high treble filter between the preamp and power amp, useful to darken or brighten the tone in a unique way. This control accurately replicates the "Presence" control on the original version of the "USA Pre" type amps. (Not to be confused with the "**BRIGHT SWITCH"** (see below) which engages/disengages a capacitor across the drive pot.

5.1.3 Amp Power Amp Page ("PWR")

PRESENCE/HI-CUT – The Presence control boosts (or cuts) the upper frequencies from the power amp simulator by varying the negative feedback frequency response. Increased presence can help a sound cut through a heavy mix.
Amps with no negative feedback circuits in their design cannot utilize a presence circuit. Therefore, for amp types of this type (or for any type if you manually set **NEGATIVE FDBK** to 0.00) the PRESENCE control is replaced with the "HI-CUT" control. This high-frequency filter allows you to control the tone the power amp. When changing to a model with no negative feedback (i.e. Class-A, Mr.Z, Recto Red), be sure to check your presence settings as settings higher than zero may darken the sound undesirably.

When certain "USA" amp models are used, a **PRESENCE SHIFT** switch appears beneath the Presence knob. This replicates the behavior when the Presence knob is pulled out on these amps. Note that the behavior of this switch is authentic and may result in volume reduction when active since the negative feedback is increased which lowers the loop gain.

DEPTH – Boosts low frequencies from the power amp simulation by varying the negative feedback frequency response. The **DEPTH** control is set by default to an appropriate value when the amp **TYPE** is selected, but this setting may be overridden.

TUBE TYPE – The virtual power amp in the Axe-Fx includes modeling of the plate impedance of the power tubes. Plate characteristics are adjustable via **DYNAMIC DAMPING**, an advanced parameter. The **TUBE TYPE** parameter sets **DYNAMIC DAMPING** automatically for you, allowing you to select from common power tube types instead of selecting a number. EL34, EL84, 6L6, 6V6, KT66, KT88, 6550, 6973, 6AQ5 and 300B (triode) are offered, as well as an ideal tetrode and ideal pentode. The power tube type defaults to the appropriate type when the amp type is selected but may be changed freely.

NEG FDBK – 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 then be changed as desired. For example, you might dial in some negative feedback on a "Top Boost" to give the power amp a more "American" sound while still retaining the preamp voicing.

MASTER VOLUME – The almighty **Master Volume!** This is a very important control. It determines the distortion and dynamics characteristics of the power amp simulator, and its setting is central to an amp's sound. As the Master 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 on your favorite Axe-Fx II amps, you will learn to appreciate the different **DRIVE** and **MASTER VOLUME** settings and how to enjoy different 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 for **MASTER VOLUME** is 10.0.
- At high Master settings, less drive is usually required, especially for high-gain types.
- Amps designed for preamp distortion will typically sound great with the **MASTER VOLUME** set low to prevent the tone becoming muddy or noisy. This includes the USA Lead types, SOLO 100, and others.
- Amps with negative feedback (damping greater than zero) tend to have a "crunchier" power amp distortion, which can get "raspy" if driven too hard. You can experiment with the interactivity of **DAMPING** (see **Advanced Parameters**, below) and **MASTER VOLUME** to achieve desired distortion timbres.
- Setting **SAG** (see below) to zero will *disable* Power Amp simulation, at which point the **MASTER** becomes a simple level control with 40 dB of range.
- If more power amp gain is desired, the **MASTER VOLUME TRIM** parameter (See section **5.1.7**) can be used for increased range.

OUTPUT LEVEL – This is a copy of the **LEVEL** control on the MIX page for easy volume adjustment without page turning. This only affects volume. It has no effect on tone! For many people, this is the "go to" parameter for setting the output level of a preset.

5.1.4 Amp Speaker Page ("SPKR")

These parameters shape the virtual speaker impedance curve and the resulting resonances in the virtual power amp. Amp/Speaker interaction elicits an increase in power amplifier response at certain frequencies, affecting the tone. Note that the power amp frequency response will not equal the speaker impedance if the **NEGATIVE FDBK** is greater than "0" (because negative feedback *flattens* the response curve).

LOW FREQ, LOW Q, LOW RES – Guitar loudspeakers have a low-frequency resonance, typically about 100 Hz. This shifts up slightly when the speaker is mounted in an enclosure.

HI FREQ, HI RES – 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.

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

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 the 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 parameter simulates distortion caused by pushing a speaker too hard. It interacts with the **MASTER,** which determines how hard the actual power amp is pushing.

XFRMR DRIVE – Controls how hard the virtual output transformer is driven. Higher values simulate a smaller, more easily saturated transformer.

Amp EQ Page

The amp block includes a built-in graphic equalizer, eliminating the need to use a separate block for tone-shaping. You can change the number of bands to a variety of types. Press ENTER to reset all EQ bands to flat.

TIP: You can also change the Amp EQ type with the UP and DOWN nav buttons.

See the **Graphic Equalizer [GEQ]** block on (p.79) for additional detail on bands, Q, etc., but remember that the GEQ in the amp block is limited to a maximum of eight bands.

5.1.5 Power Amp Dynamics Page ("PWR DYN")

SUPPLY SAG – This controls 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. As the power amp is pushed and draws more virtual current from its virtual power supply, the **SAG** control will have more effect. (Note this same parameter appears in the ADVANCED menu as **MAINS IMPEDANCE (SAG).**

❗ **IMPORTANT:** Turning **SUPPLY SAG** fully counterclockwise defeats power amp simulation for an individual AMP block, allowing it to be used into an external (real) tube power amp <u>without</u> globally disabling power amp simulation. (See section **8.1** on p. **145** or the "FOH + Real Amps" diagram on p. **22** for more.) In this mode, **MASTER** works as a simple volume, **DEPTH** is deactivated, and **PRESENCE** turns into a simple shelving filter.

❗ **B+ TIME CONSTANT** (in the Advanced menu) interacts with the Sag control because it makes the power supply response slower or faster. When the supply is fast it 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 displays the supply voltage in dB, relative to the idle voltage.

TUBE BIAS – (This is the same parameter as the **PWR TUBE BIAS** control on the Advanced page). This sets the controls the quiescent operating current of the virtual power tubes. This powerful parameter allows you to fine-tune power amp distortion characteristics to your particular style. The higher the value, the *less* crossover distortion. When bias is reduced the amp sags and bounces more, with tighter bass and edgier tone.

XFRMR MATCH – Transformer Match is an extremely powerful parameter that sets the relative output transformer primary impedance to determine how easily the power tubes are driven into clipping. Higher **MASTER** Volume settings result in a more pronounced effect. Increasing **XFRMR MATCH** causes power tubes to clip *sooner*. Decreasing **XFRMR MATCH** causes power tubes to clip *later* and therefore the phase inverter and grid clipping becomes more predominant. At higher settings, the resonance settings on the SPEAKER page of the AMP block will be 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.

XFRMR Grind – Transformer Grind accurately simulates the effects of dynamic core losses and leakage inductance in the virtual transformer. Higher values result in more high frequency response and a more "open" sound. Very high values can yield a raspy, spitty tone common in vintage and/or low wattage amps. Modern "big iron" amps tend to have low values. Note that the amount of grind you will hear is dependent upon how hard the virtual power amp is driven and is more noticeable as the **MASTER** is increased. In real amps this effect is highly dependent on the speaker. Some speaker/transformer combinations exhibit significant high frequency dynamic boost while other combinations yield almost none. As always use your ears as the final determinant of the settings that will work best for you.

OUT COMP –The Output Compressor parameter controls the ratio of a compressor specifically tailored to reduce the output dynamics of the Amp block. A bar graph beneath the knob shows gain reduction.

COMP TYPE – (Output Comp Type) - Sets the type of Output compression. The "**Output**" type compresses the block's output. The "**Feedback**" type also applies dynamics to the input of the block, so you will get more distortion as you play harder and less when you play softer or roll back the volume.

The Output Compressor also uses **OUT COMP THRESHOLD** and **OUT COMP CLARITY** in the Advanced menu.

5.1.6 Amp Preamp Dynamics Page ("PRE DYN")

PREAMP COMP ("Preamp Compression") – Determines the amount of compression in the virtual cathode follower. You can also set the attack time and ratio of this compressor (in the Advanced menu using the **CF TIME** and **CF RATIO** parameters) or change its **TYPE** (below)

COMP TYPE ("Preamp Compression Type") – Selects between "Authentic", which accurately models the compression in a tube amp, and "Ideal", which is an idealized distorting compressor. The idealized type is more focused and has tighter bass whereas the authentic type is bolder and looser. High gain players may prefer the ideal type due to its tight character.

DYNAMICS – Sets the strength of an input dynamics processor that can be used to alter the response of the amp. When set below zero the amp *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 and clipping.

CRUNCH – Adds crunch, as in Cap'n...

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). This value is set automatically when amp **TYPE** is changed, but can be altered any time as desired.

HARMONICS – Not the type you play with a soft touch, but the type that occur naturally inside an amp as tubes interact. Higher values increase the interaction between virtual tubes, yielding "softer" distortion.

5.1.7 Amp Dynamic EQ Page ("DYNEQ")

DYNAMIC PRESENCE – This models output transformer leakage inductance that results in a brightening of the tone when the power amp is pushed. This control is set to a default value when the model is selected corresponding to the real amp, if applicable. Increasing this value results in a brighter response as 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 hard. This control can also be used to help "dial in" the sweet spot of an amp model. As the **MASTER** is increased an amp becomes more liquid, compressed and easier to play. However, the highs may get overly compressed, causing the amp to sound too dark. The Dynamic Presence control allows you to get the desired power amp drive and liquid feeling and then bring the highs back without affecting the rest of the spectrum.

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

CHARACTER TYPE, CHARACTER FREQ, CHARACTER Q, CHARACTER AMOUNT – These parameters control a powerful "inverse homomorphic" filter which adjusts tone dynamically in a very musical way. When playing softly these dynamic filters have little effect on the sound. As the amount of distortion increases, the influence of these filters increases. **CHARACTER FREQUENCY** and **Q** set the center frequency and width of the filter, while **CHARACTER AMOUNT** sets how pronounced the effect is.

This control is similar to **DYNAMIC PRESENCE** and **DYNAMIC DEPTH** but the frequency is adjustable. For example, for a tone that darkens when you play harder, set **CHARACTER FREQUENCY** to 10000 Hz and the **CHARACTER AMOUNT** to -5. For the reverse, set amount +5 and the tone will brighten when you play hard. **CHARACTER AMOUNT** defaults to zero whenever a new amp **TYPE** is selected.

CHARACTER TYPE determines the type of filter type used, shelving, peaking or dynamic. The "Dynamic" type can be used to fatten or scoop the tone as a function of picking strength. For example, set the **TYPE** to Dynamic, **CHARACTER FREQUENCY** to 450.0, **CHARACTER Q** to 0.7 and **CHARACTER AMOUNT** to 4.0 for a tone that gets fatter and thicker as you play hard but is not "honky" when you play soft.

5.1.8 Amp Advanced Page ("ADV")

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).

MODELING VERSION – **(XL/XL+ ONLY!)** Selects which version of Quantum modeling the currently selected Amp block will use. (See also "Force Default Version" in the Global Config menu on p. 145.)

BOOST – Toggles an additional 12 dB of gain at the input of the amp block.

INPUT TRIM – Allows you to adjust the relative gain of the preamp. Increasing the value will cause the amp to have (more or less) gain. It is simply a linear gain applied at the input to the block. You can use it to give a typically clean amp a bit more oomph or decrease the gain of a very high-gain amp. Note that this is different than the Input Drive control because the Drive control interacts with the surrounding circuitry and changes the frequency response as it is varied.

MSTR VOL TRIM – Allows you to adjust the range of the **MASTER**. Increasing this value above 1.0 will cause more gain in the virtual power amp, while values below 1.0 will result in less gain.

MSTR VOL CAP – ("Master Volume Capacitor") sets the value of the bright cap across the Master Volume.

MSTR VOLUME LOCATION – ("Master Volume Location") – Sets the location of the Master Volume. Most amps have the Master Volume before the phase inverter ("Pre PI"). On some amps (like the "Class-A" types) the Master Volume is after the phase inverter ("PI"). A third option, "pre-triode," is the default for amp types based on Hiwatt® models.

BRIGHT – This switches appears on the PRE page, but is offered here with a **modifier** option.

BRIGHT CAP – Sets the value of a virtual capacitor to determine the sonic effect of the **BRIGHT** switch (above). Increasing this will make the preamp brighter and vice versa.

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

SAT DRIVE – Controls the amount of **SAT SWITCH** saturation. The default value differs for each model.

LOW-CUT FREQ – This control allows you to reduce the amount of low-frequency content at the input to the amp simulator. This parameter defaults to a value for each type but can be overridden if desired.

HIGH-CUT FREQ – This control sets the cutoff frequency of a low-pass filter at the very end of the preamp simulation. It defaults to a preset value for each amp type but can be overridden if desired. 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 or decrease for a brighter or darker tone.

DYNAMIC DAMPING – The Axe-Fx virtual power amp models the plate impedance of the power tubes. They give tight bass and warm highs at higher **MASTER** settings, with "3-dimensional" tone. The plate characteristics are adjustable via the Dynamic Damping parameter. The value of this control is changed automatically when you set the **TUBE TYPE** parameter on the PWR page.

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

TONESTACK TYPE – The **BASS**, **MID** and **TREBLE** controls operate by default as "passive" controls. That is, they simulate exactly the frequency and phase response of the classic passive tonestacks found in the original amplifiers our simulations are based on. The **TONESTACK TYPE** control lets you change this behavior from PASSIVE to ACTIVE, or to substitute the passive tonestack of another amp type.

- 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, and less extreme settings still achieve pretty extreme sounds. 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.
- 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, or a modern German tonestack in a British Preamp.

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. However, if you subsequently change the **TONESTACK TYPE**, the **TONESTACK FREQUENCY** will not necessarily be correct anymore.

TONESTACK LOCATION – This control lets you change the location of the tone stack. "PRE" places the tone stack at the input to the 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). Defaults to an appropriate value whenever you change the amp **TYPE**.

EQ TYPE – This determines the number of bands for the amp block's built-in graphic equalizer (from 3 to 8) and whether it will be variable Q, constant Q, passive or console type.

PRESENCE FREQ – This multiplier alters the center frequency of the amp's **PRESENCE** control, which is naturally determined based on the current selection for amp **TYPE**.

DEPTH FREQ – Alters the center frequency of the amp's **DEPTH** and **DYN DEPTH** controls. This parameter defaults to an appropriate value whenever you change the amp **TYPE**, but can then be changed as desired.

POWER AMP BIAS SHIFT – Controls the amount of phase inverter bias shift. Note that some real amps are "spitty" in nature due to PI bias shifting, and the new algorithm is designed to replicate that behavior accurately. If you find this undesirable reduce **PI BIAS SHIFT** as desired although this will reduce authenticity.

POWER AMP GRID BIAS – (aka "Power Tube Bias") Sets the bias point of the virtual power amp. Lower values approach pure Class-B operation. Higher values approach pure Class-A operation.

POWER AMP 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.

BIAS EXCURSION – The higher this 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.

PA 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.

POWER 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 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.

AC VOLTAGE (VARIAC) – This sets the relative AC line voltage into the amp simulation implementing a virtual "Variac". Note that normally the volume would vary with the Variac setting in a real amp but the simulation compensates for this.

MAINS IMP. (SAG) – This is a duplicate of SUPPLY SAG on the amp's PWR DYN page.

PREAMP SAG –Turning this ON replicates the behavior of an integrated tube head or combo amp as described above. Turning this OFF replicates the preamp sag behavior of separate preamp and power amp. NOTE: Preamp modeling uses screen voltage from "power amp in" calculations rather than operating independently. This improves feel as preamp voltage drops with power amp sag. The effect is more noticeable as **SAG** is increased. Note that preamp sag has a long time constant and, as such, the initial pick attack is relatively unaffected while sustained sounds undergo compression. This results in a "chewier" sensation.

B+ TIME CONSTANT – Controls the rate of change in the power tube plate supply. Lower values give a bouncier feel, while higher values give a tighter feel. TIP: Remember that you can also monitor B+ when supply sag is selected on the Power Amp Dynamics page ("PWR DYN").

TRIODE1 PLATE FREQ, TRIODE2 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.

CATHODE SQUISH, SQUISH TIME – The Axe-Fx II has "cathode squish modeling" for cathode biased power amp models. **CATHODE SQUISH** sets the amount of bias shift due to cathode voltage rise and **SQUISH TIME** sets the time constant of the cathode network. These parameters are set to default values upon selection of an amp **TYPE**. (Setting **CATHODE SQUISH** to zero defeats the cathode squish modeling.)

OUTPUT COMP, COMP THRESHLD – These control the output compressor (detailed previously on the DYNAMICS page.) The **THRESHOLD** can only be set here, on the ADVANCED page.

BIAS EXCURSION — The Axe-Fx II accurately models grid conduction and resulting bias excursion. This results in a more dynamic, thicker and bouncier tone. **BIAS EXCURSION** controls how much the grid voltage droops when the grids conduct.

CF (PREAMP) COMP – a duplicate of the Preamp Comp parameter on the Preamp Dynamics page.

CF TIME, CF RATIO – These parameters determine the attack time and ratio used for CF (Preamp) Comp.

CF HARDNESS – The Quantum firmware series improved modeling for tubes driving cathode followers. In models that use cathode followers, this results in warmer distortion with smoother decay. Set the shape of cathode follower distortion using this parameter.

OUTPUT COMP THRESH – Sets the threshold of the Output Compressor (found on the Power Amp Dynamics page ("PWR DYN").

OUTPUT COMP CLARITY – Used in conjunction with the other Output Compressor parameters, this adjusts the bass response of the compressor and can be used to add clarity to the bass.

PREAMP TUBE TYPE – This selects a tube type for the virtual preamp from the following options: 12AX7A JJ, 12AX7A RCA, 12AX7A Syl(vania),12AX7A, 12AX7B, 7025, ECC83, ECC803S, EF86. This parameter is set automatically to an appropriate type whenever a new amp model is selected.

PREAMP HARDNESS – Controls the asymmetry of the triode mode to determines how sharply they enter saturation, simulating "softer" or "harder" tubes. This subtle effect is most apparent at edge of breakup. Lower values give softer saturation with less even and more odd harmonics. Higher values give a more aggressive breakup. The default is set when an amp Type is selected but can be changed any time.

PREAMP BIAS – This is a duplicate of the PREAMP BIAS parameter on the amp's DYN PRE page.

POWER AMP HARDNESS – Controls the hardness of the virtual power tube grid clipping. The lower the value the softer the distortion, but this often is not noticeable because negative feedback around the power amp makes the distortion harder. Another factor which controls power amp hardness is Transformer Match: turn it up, and turn down Negative Feedback for softer power amp distortion.

POWER AMP BIAS – Adjusts the offset voltage of the virtual power amp to vary the symmetry of the clipping of the virtual power amp. A value of zero produces nearly symmetrical clipping with very little even harmonics. Higher values are increasingly asymmetrical which increases even harmonics. Small amounts of even harmonics can make the power amp distortion sound "warmer" and more bell-like while higher amounts will give a "fuzzier" tone. Most amps have some amount of offset and the amp models will default to a typical value. Note that this parameter is only applicable for push-pull power amp types. For single-ended power amps the Power Tube Bias parameter sets the symmetry (as always).

PICK ATTACK – Controls a sophisticated dynamic range processor that operates on leading edge transients. Negative values reduce pick attack while positive values enhance it.

AMP Trem/Mix Page

The **Amplifier** block also has a **MIX** page with **LEVEL**, **BALANCE,** and **BYPASS MODE** parameters.
See **Common Mix Parameters** on p. **128** for more information.

TREM FREQ, 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, and more. It is also "self-ducking" and decreases as you play harder. On some amp types, extreme bias **TREM 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.

5.2 Cabinet [CAB]

The **Speaker Cabinet Simulator** (“Cab” for short) recreates the tonal characteristics of any number of speaker cabinet configurations. The Axe-Fx II XL/XL+ contains over 150 built-in “factory” cabinet simulations, plus 1024 memory locations for loading custom “User Cab” files. (The Mark II contains 130+ factory and 100 user cabs). The Cab block also offers room and microphone simulations (including sub-millisecond delays), plus basic tone controls.

Factory cabs include custom creations by Fractal Audio Systems, plus selections from 3rd-party libraries by Mikko Logrén of ML Soundlab, Buddy Gill, RedWirez, OwnHammer, TheAmpFactory, and contributions from Fractal Artists John Petrucci, James Santiago, and loudspeaker design engineer Jay Mitchell.

The cab block supports both standard (2048) resolution IRs, and newer UltraRes™ format IRs. UltraRes™ is a proprietary format that enhances the resolution of an IR without added CPU burden or storage requirements.

Each Axe-Fx II preset can use two fully independent **Cab** blocks.

XY The Cab block supports X/Y Switching. See p. **36** for more information.

Cab Parameters

CAB (TYPE) – Sets the cabinet type by selecting from “FACTORY” and “USER” IRs. Cabinet types are listed in the table in section on p.169. The four SCRATCHPAD locations found at the end of the list are designed to allow you to “audition” cabs before committing them to a memory location. This capability is especially useful when user cab memory is full, or when you are using **Cab-Lab** (available from http://shop.fractalaudio.com). Please note that the contents of the SCRATCHPADs are cleared every time you restart the Axe-Fx.

MIC (TYPE) – Selects the microphone simulation type used. There are ten different types based on classic guitar cabinet microphones.

Manufacturer and product names mentioned below are trademarks or registered trademarks of their respective owners, which are in no way associated or affiliated with Fractal Audio Systems. The names are used only to illustrate sonic and performance characteristics of the Axe-Fx II MIC TYPES.

57 DYN (based on the Shure® SM57®)
58 DYN (based on the Shure® SM58®)
421 DYN (based on the Sennheiser MD 421 II®)
87A COND (based on the Shure® Beta 87A®)
U87 COND (based on the Neumann® U87®)
E609 DYN (based on the Sennheiser® e609® Silver)
RE16 DYN (based on the Electro-Voice® RE16®)
R121 COND (based on the Royer Labs® R-121®)
D112 DYN (based on the AKG® D112®)
67 COND (based on the Neumann® U67®)

“NULL” is a perfectly transparent mic with a **PROXIMITY** control (below.)
“INVERT” is perfectly transparent and enables **PROXIMITY**, but also phase inverts the signal.
“NONE” disables microphone processing in the CAB block.

PROXIMITY – Simulates the classic proximity effect, causing an increase in bass or low frequency response as proximity is increased. The **PROXIMITY** control has no effect when **MIC TYPE** is set to “NONE.”

DELAY – This short delay (0.000-1.000 ms) provides the ability to simulate microphone distance as employed to create interesting phase or comb filter effects. You’ll need to have two parallel cabs with *different* delay settings to hear this. Furthermore, the effect is most pronounced when the cabs are summed to mono.

SPKR SIZE – This control “scales” the IR to simulate shrinking or enlarging the virtual speaker. This effect can be used to shift where the tone “sits” in a mix, or to create dramatic effects. Subtle settings (0.9-1.1) will sound most natural. **SPKR SIZE** is not offered in STEREO modes or when the selected IR is running in UltraRes.

INPUT SELECT – For use in the MONO cab modes. This determines how incoming stereo signals will be processed. Options include inputting only LEFT or RIGHT channels, STEREO or SUM L+R. This can be used, for example, to run two Cab blocks in parallel for stereo processing by setting one to Left and the other to Right.

MODE – Offers “HI-/ULTRA-RES,” “STEREO ULTRA-RES” “NORMAL RES” and “STEREO” (normal res) modes.

Fractal Audio Systems Cab IRs come in two different formats: **Standard Res,** and **UltraRes**™. UltraRes IRs use a patent-pending proprietary technology to deliver enhanced sonic resolution without high latency or CPU load.

When you use UltraRes™ IRs and set the Cab block mode to UltraRes, the highest quality sound is achieved. If you don’t use Ultra-Res data, the IRs will load in “high” resolution.

Selecting one of the “Normal” options causes all IRs (even UltraRes™ IRs!) to load in standard resolution, without the benefits of UltraRes™.

“STEREO” modes offer two Ultra-Res or two Standard Res impulses to be loaded in a single cab block. Fully independent left and right side parameters appear when you select this option.

Figure 5-2 –HI-/ULTRA-RES and NORMAL RES modes.

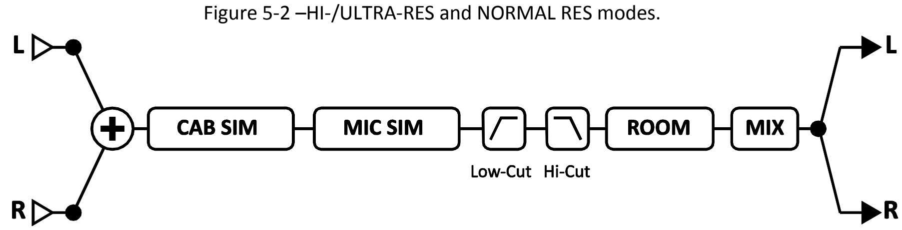
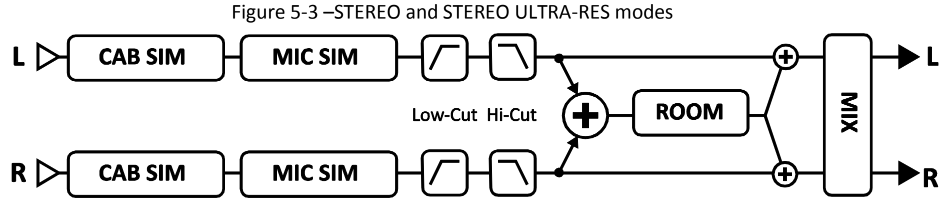


Figure 5-3 –STEREO and STEREO ULTRA-RES modes



To use a stereo Cab block with two amps, connect both amps to the cab. Then set the **BALANCE** control for one Amp fully left, the **BALANCE** for the second amp block fully right, and set the Cab block **MODE** to “STEREO.”

See Stereo Cab Mode Parameters (below) for more.

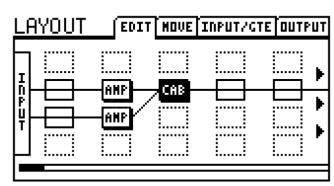


Figure 5-4 Two Amp Blocks into a STEREO Cab

DEPHASE – This parameter controls a sophisticated process that removes the “phasiness” from IRs and can yield a more “amp in the room” experience.

LOW-CUT/HI-CUT/FILTER SLOPE – Adjusts the cutoff points of high-pass and low-pass filters. Increase the low-cut to reduce bass boom. Decrease the high-cut to darken the tone. Slope allows you to select between first-order (6 dB/octave) or second-order (12 dB/octave) filters.

ROOM LEVEL, ROOM SIZE – These controls determine the level and virtual size of a room reverb simulation that is built into the cab simulator block. Increase to add room ambience to the sound.

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

PROXIMITY FREQ – This allows tuning the frequency range over which the proximity effect occurs.

STEREO LINK – Available only when the Cab **MODE** is set to “STEREO”, LINK turns the LEFT channel parameters into master controls, which set identical values for LEFT and RIGHT parameters. You can still override the right channel parameters values if desired.

Stereo Cab Mode Parameters

When the **MODE** of the CAB block is set to “STEREO”, independent instances of the following parameters appear for left and right.

- **CAB (TYPE) L/R**
- **MIC L/R**
- **PROXIMITY L/R**
- **LEVEL L/R**
- **PAN L/R** – Pan Parameters appear *only* when cab MODE is stereo.
- **DELAY L/R**

SPKR SIZE is not offered when CAB **MODE** is set to “STEREO”.

Cab Preamp Simulation Parameters

A microphone on a guitar speaker is subject to the pleasing musical distortion generated by the mic preamp. This might range from a subtle “warming up” to a full on “nasty-fying.” Preamps also offer their own tone controls which change the sound of the mic’d speaker. The Cab block includes several controls to simulate these effects.

PREAMP MODE – (on “PG2”) offers “High Quality” and “Economy” modes which use more or less CPU.

PREAMP TYPE – Selects from a number of highly musical preamp types including Tube, FET, Transformer, Tape, etc. Select the type which sounds best to you.

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 up **DRIVE** and the VU meter approaches or exceeds the 0 dB marker you will begin to overdrive the preamp.

SAT– The Saturation control controls the ratio of even/odd harmonics in preamp distortion.

BASS, MID, TREBLE – These adjust the tone of the virtual mic preamp.

Cab Mix Parameters

The **Cab** block **MIX** page also has **LEVEL, BALANCE,** and **BYPASS MODE** parameters.
See **Common Mix Parameters** on p.**128** for more information.

The following parameters also appear on the MIX page:

MOTOR DRIVE – This models the effect of high power levels on the tone of the speaker. The Motor Drive parameter controls the relative drive level and, therefore, the intensity of the effect.

AIR, AIR FREQ – Adds "air" and sets the cutoff frequency to determine if it is dark or bright sounding.

5.2.1 User Cabs

In addition to the onboard "Factory" cabs, the Axe-Fx II allows you to store User" Impulse Response ("**IR**") files onboard—1024 (on the XL/XL+) or 100 (on the Mark II/original).

Here's how it works: first, you need an IR file. An great source for free IRs is http://axechange.fractalaudio.com, our online repository of presets and cabs. Fractal Audio Systems also offers professionally produced **Cab Packs** at http://shop.fractalaudio.com. Next, you'll need to transmit your IR to the Axe-Fx II. Our various software applications are ideal: Fractal-Bot, Cab-Lab, and Axe-Edit. Before transmitting the IR, you'll select a location on your Axe-Fx II where it is to be stored. Once you transmit the file, the sound of that cab is then available on your unit. For step-by-step instructions on loading User Cab IRs, see section **1.1** of the Appendix on p. **170**.

The results of Tone Matching (p. **121**) can also be saved into a user cab memory.

You can also capture your own User Cab IRs, using a built in utility and a mic'd cab. See **IR Capture** on p. **156**.

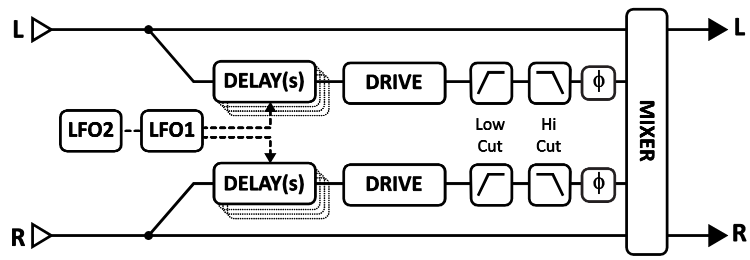
As you scroll through user cabs by their numbers in the Cab **TYPE** parameter, NAMES will appear in the bottom of the display. These names come from within the data of the user cab SysEx file. Names can be changed before a Cab IR file is loaded into the Axe-Fx II, but not after.

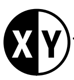
5.3 Chorus [CHO]

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 ambient and liquid, while more extreme settings can produce a vibrato or "Leslie" effect. The Axe-Fx II offers a high-quality, multi-voice stereo chorus capable of producing anything from exceptionally smooth ensemble effects to a wildly detuned warble.

Each Axe-Fx II preset can use two fully independent **Chorus** blocks.

Figure 5-5 – The Chorus Block




The Chorus block supports X/Y Switching. See p. **36** for more information.

Basic Chorus Parameters

TYPE – This control instantly sets other Chorus 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, and DIMENSION. All "ANALOG" types use an algorithm which models the behavior of classic bucket brigade (BBD) units. VINTAGE TAPE uses a unique algorithm to models the behavior of a tape delay used as a chorus unit. Rate, Depth, Level, Balance, Bypass Mode, and Global Mix are not affected when you change TYPE.

NUMBER OF VOICES – Each stereo channel in the chorus can have from one to four voices. Increasing the number of voices increases the fullness of the effect. Use two voices for a vintage chorus effect, or use up to eight for a lush, multi-layered ensemble.

RATE – This 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 chorus LFO to global LFO1. When **RATE** is shown in parenthesis, it is being set automatically by the **TEMPO** parameter (see below). Set the **TEMPO** to "NONE" for manual control.

DEPTH – Sets the 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 precise control of depth, turn the AUTO DEPTH parameter on the ADVANCED page to OFF.*

MIX – Sets the ratio of wet and dry (duplicated from the MIX page). A setting of 50% produces the most prominent effect. Try setting the mix to 100% for vibrato effects.

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

Advanced Chorus Parameters

DELAY TIME – Adjusts the minimum delay time from 0.0–50.0 ms. Lower values create a more unified sound while higher values go beyond double tracking towards "slap back."

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 LFO waveforms, which creates a noticeable effect on the stereo width of the chorus.

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 be changed 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 section **16.7** on p.**174** for more information on LFO waveform shapes and phase.

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 of the Dimension with no tonal coloration.
- Med: Classic Dimension processing buttons 1-3. Set **RATE** and **DEPTH** to taste.
- High: Classic Dimension processing button 4. Set **RATE** and **DEPTH** to taste.

The Chorus block also has a **MIX** page with **LEVEL, BALANCE,** and **BYPASS MODE** parameters.
See **Common Mix Parameters** on p.**128** for more information.

5.4 Compressor [CMP]

A compressor reduces the difference between loud and soft sounds by reducing the level of—or compressing—loud signals. The reduction is triggered when the input 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 in front of high-gain distortion can increase noise or squealing). In the recording studio, a Compressor is typically placed towards the end of a signal chain to smooth irregular levels. The Axe-Fx II provides both pedal and studio-type compressors (detailed below).

XY XL XL+ The Compressor block supports X/Y Switching on the XL/XL+ only. See p. **36** for more information.

Each Axe-Fx II preset can use two fully independent **Compressor** blocks.

Pedal-Type and Common Compressor Parameters

TYPE – The Axe-Fx II contains three different compressor types: STUDIO, PEDAL 1, PEDAL 2 and DYNAMICS. The STUDIO type simulates the behavior of popular high-end “Feed Forward” stereo studio compressors. The PEDAL types simulate classic stompbox “Feedback” compressors. “PEDAL 2” uses a smoother detector which pumps less. Dynamics. The Dynamics algorithm allows compression or expansion with a single control.

Figure 5-6 – The Compressor Block's "Pedal" Type

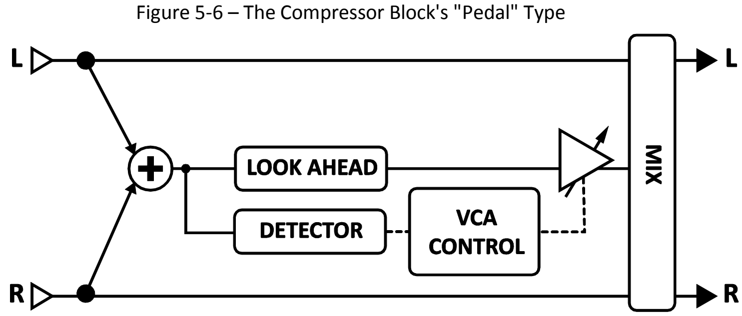
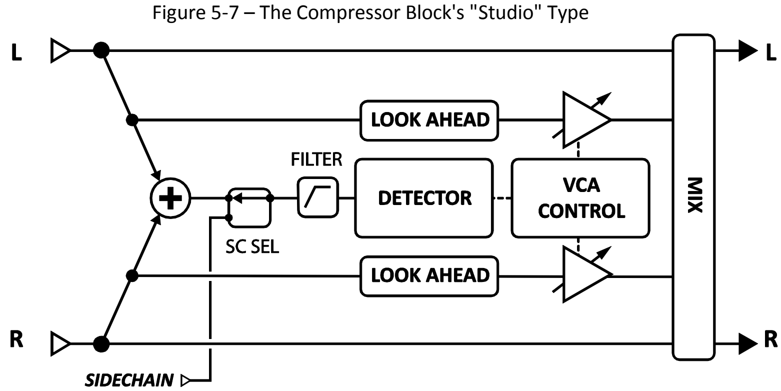


Figure 5-7 – The Compressor Block's "Studio" Type



AUTO – Turns the Dynamic Attack filter on or off. Turning this switch ON automatically varies the ATTACK rate according to the program material; the compressor will respond to faster transients with a faster attack.

LOOK AHEAD – Despite fast attack times, a compressor can fail to "catch" very fast transients. *Look Ahead* introduces a short audio delay so the compressor's gain control stage has sufficient time to respond to the detector, which is side-chained with *no* delay. Look ahead can reduce "popping," especially when heavy compression is used on very percussive sources.

MIX – Sets the ratio of wet (compressed) and dry sounds. This should normally be set to "100%."

THRSH – (Threshold) Sets the level at which automatic volume reduction will occur. When input power exceeds the threshold, the compressor reduces output volume as set by the RATIO. (When the TYPE is set to "PEDAL", the **THRESHOLD** is hidden and automatically set to "minus infinity").

SUSTAIN/RATIO – In "PEDAL" type compression, **SUSTAIN** increases compression amount as the knob is turned clockwise. In "STUDIO" type compression, **SUSTAIN** is replaced by **RATIO**, which sets the input-to-output ratio for signals above the THRESHOLD. A ratio of 2.00 (2:1) means that an input that is 10 dB over the threshold will increase the output by only 5 dB. A ratio of 10.00 (10:1) means that an input that is 10 dB over the threshold is reduced to a mere 1 dB above. Setting the **RATIO** to "INFINITY" turns the compressor into a "limiter," reducing any level above the threshold *to* the threshold, applying a sort of "ceiling" or "brick-wall limiting" above which nothing can rise.

ATT – Attack rate sets how fast the Compressor reduces the volume once the threshold is exceeded. For guitar, a fast attack rate often works best.

REL – Release rate determines how quickly the output volume returns to normal once the input level falls below the compressor's threshold. Fast release rates can produce a snappy attack, but a setting that is too fast can cause distortion if used in conjunction with fast attack times and high compression ratios. 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 rate so the decay observed is slightly faster than the natural decay of the instrument.

EMPH – **EMPHASIS** creates a cool effect similar to using a filter on the detector. It boosts highs at the compressor input and then cuts them back to normal levels at the output. Apply this to prevent thumpy lows from causing your compressor to "pump."

LEVEL – Sets the output level of the compressor.

Studio Compressor Parameters

When **TYPE** is set to "STUDIO", the following additional parameters appear:

KNEE – 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.

MAKEUP – Automatic Makeup gain, when turned ON, compensates the output level to maintain perceived loudness at the current threshold and ratio. The LEVEL control can then be used for fine control.

DETECT – Selects whether the compressor will use RMS (“Root Mean Square”), FAST RMS (mimicking classic rack-mount compressors), 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.

FILTER – Sets the frequency of a high-pass filter on the input of the compressor’s detector stage. Raising the filter frequency can help prevent low frequencies from “pumping” the entire mix. Does NOT filter the output.

SCSEL – **SIDECHAIN SELECT** determines which signal is used to feed the compressor’s detector. “NONE” is the normal setting and selects the compressor’s input (sum of the rows feeding the block). You can also use the input from a designated row, or either of the main inputs. The BLOCK L and BLOCK R options are useful when the compressor follows an effect with one side out of phase (delay, chorus, enhancer).

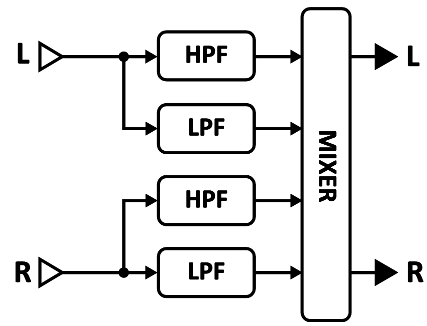
Dynamics Processor Parameters

When **TYPE** is set to “STUDIO”, the following additional parameters appear:

DYNAMICS –When this is set below zero, compression occurs and dynamics are de-emphasized, when set above zero, expansion occurs and dynamics are exaggerated.

Other controls for the “Dynamics” type work just like those described above for the Compressor.

5.5 Crossover [XVR]



This two-way stereo **crossover** contains 4th-order Linkwitz-Reilly filters. Each Axe-Fx II preset can use two crossover blocks. You can create a three-way crossover by feeding one output of the first to the input of the second.

Each Axe-Fx II preset can use two fully independent **Crossover** blocks.

XOVER FREQ – Sets the crossover frequency of the filters.

FREQ MULTIPLIER – When set to “×10,” the crossover frequency is multiplied by ten.

LEFT LOW LEVEL – Sets the level of the left input low-pass filter.

LEFT HI LEVEL – Sets the output level of the left input high-pass filter.

RIGHT LOW LEVEL – Sets the output level of the right input low-pass filter.

RIGHT HI LEVEL – Sets the output level of the right input high-pass filter.

LEFT LOW PAN – Sets the panning of the left input low-pass filter.

LEFT HI PAN – Sets the panning of the left input high-pass filter.

RIGHT LOW PAN – Sets the panning of the right input low-pass filter.

RIGHT HI PAN – Sets the panning of the right input high-pass filter.

Crossover Mix Parameters

The **Crossover** block also has a **MIX** page with **LEVEL, BALANCE,** and **BYPASS MODE** parameters. See **Common Mix Parameters** on p.**128** for more information.

5.6 Delay [DLY]

The Axe-Fx II **Delay** block lets you create classic, modern, and innovative echo effects. A "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.

Each Axe-Fx II preset can use two fully independent **Delay** blocks. Don't forget the additional two **Multi-Delay** blocks (p. **86**), the **Megatap Delay** block (p. **82**), and the **Looper** block (p. **80**).

The Delay block supports X/Y Switching. See p. **36** for more information.

Delay Common Parameters

TYPE – The Delay TYPE control sets various parameters of the delay block to achieve popular delay effects instantly. See the table below for a listing of DELAY block types.

CONFIG – The delay configuration determines which one of several base delay algorithms is used. Depending on the configuration you select on PG1 of the delay block, PG2 is configured with different parameters. The details and parameters of each configuration are listed in the subsections that follow.

Table 1: Delay Block Types and Configurations

TYPE		Configuration
Digital Mono	Full-range, a pristine modern delay (Default Mono).	**MONO DELAY**
Analog Mono	Frequency response and character of an analog delay.	
Vintage Digital	Uses bit-depth reduction for a lo-fi vibe.	
Deluxe Mind Guy	Recreates the sound of a classic delay pedal.	
Mono BBD	Simulating a vintage bucket brigade delay pedal	
2290 w/ Mod	Based on a former industry-standard unit.	
Mono Tape	A config with MOTOR SPEED and other tape controls.	**TAPE DELAY**
Lo-Fi Tape	The same, but very low fidelity.	
Digital Stereo	Full-range, a pristine modern delay (Default Stereo).	**STEREO DELAY**
Analog Stereo	Frequency response and character of an analog delay.	
Stereo Tape	Frequency response and character of a tape delay.	
Ambient Stereo	Ultra-wide echoes.	
Stereo BBD	Simulating a vintage bucket brigade delay pedal in stereo.	
Ducking Delay	"Ducking" automatically lowers delay volume when you play harder, resulting in a less "cluttered" mix.	
Dual Delay	A default setting for the dual delay.	**DUAL DELAY**
Ping-Pong Delay	A default setting for the ping-pong delay.	**PING PONG**
Sweep Delay	A default setting for the sweep delay.	**SWEEP**
Reverse Delay	A default setting for the Reverse delay.	**REVERSE**

INPUT GAIN – Sets the input level into the delay lines. This lets you to attach a controller (e.g. pedal) to the delay level input level for operation similar to that of an "Aux Send." In other situations this control should be set at 100%.

MSTR FDBK – Master Feedback scales any and all feedback parameters on PG2 of the Delay. Note that the range of this control is 0–200%, making it possible (easy, in fact) to "overload" the feedback loop.

MIX – This is a copy of the **MIX** control on the MIX page, placed here for easy adjustment of the wet/dry balance without page flipping.

LEVEL – This is a copy of the **LEVEL** control on the MIX page, placed here for easy adjustment of the overall volume without page flipping.

5.6.1 Mono Delay

The **Mono Delay** can be used for a variety of great sounding standard and exotic delays. This configuration sums the inputs into a single delay line.

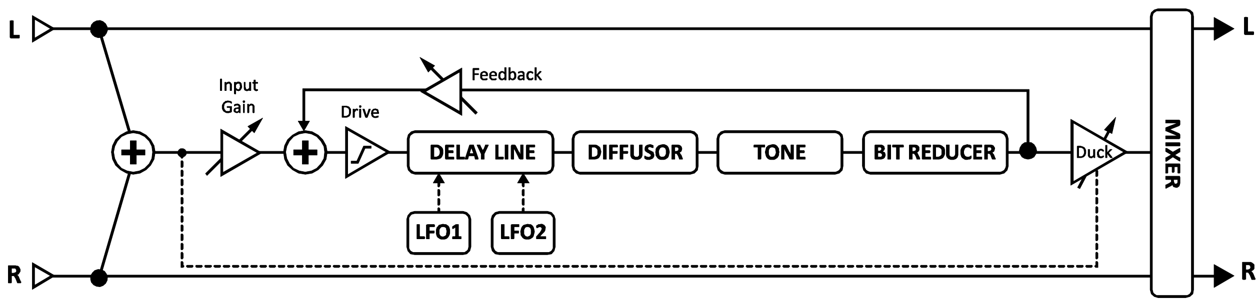


Figure 5-8 – The Mono Delay Block

TIME – Sets the time of the delay in milliseconds When **TIME** is shown in parenthesis, it is being set automatically by the **TEMPO** parameter (see below). Set **TEMPO** to "NONE" to regain manual control.

FEEDBK – 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 – Controls the placement of the "wet" signal (the echoes) in the stereo image. Note that this is different than the MIX page BALANCE control, which acts on the mix of both wet and dry.

REPEAT HOLD – This switch defeats the inputs of the delay and "captures" the current feedback loop, which plays infinitely, as long as the **REPEAT HOLD** switch remains ON.

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

DRIVE – Determines the amount of distortion created by a drive model 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 rolloff.

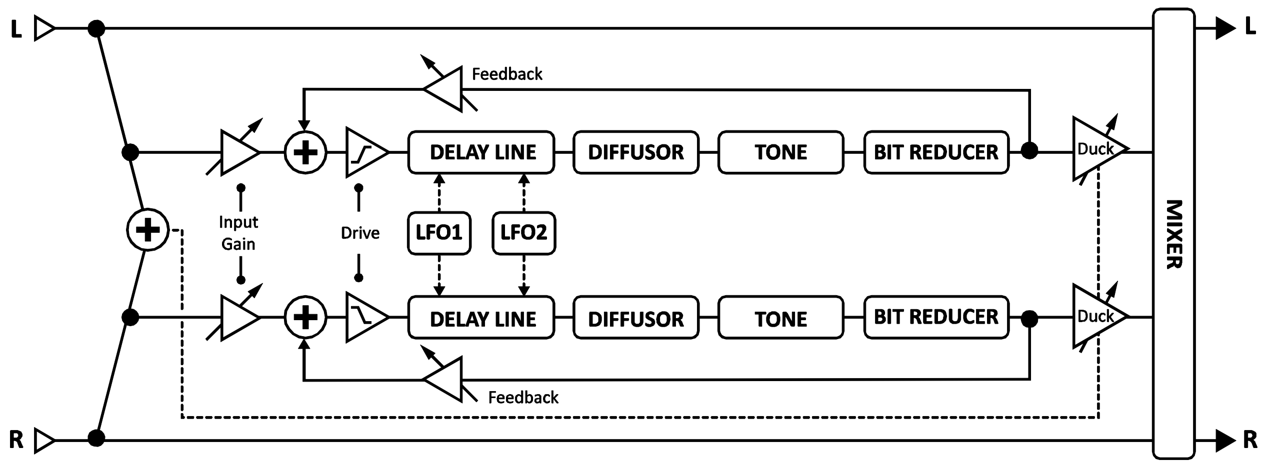
RIGHT POST DELAY – Widens echo sounds by adding 0–100 milliseconds of delay at the right (wet) output.

Please be aware that because the MONO delay contains only one delay line, the two **LFO PHASE** parameters on its MOD page have no effect. Similarly, the **LFO TARGET** parameters must be set to "LEFT" or "BOTH" for modulation to occur.

5.6.2 Stereo Delay

This stereo-in/stereo-out delay has the convenience of common controls for most L-R parameters.

Figure 5-9 – The Stereo Delay Block



TIME – Sets the left delay time in milliseconds. When **TIME** is shown in parenthesis, it is being set automatically by the **TEMPO** parameter (see below). Set **TEMPO** to "NONE" for manual control.

RATIO – Sets the right channel time as a percentage of the left. 100% results in both channels having equal delay time. Settings close to 100% (e.g. 99.6%) will subtly 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 interesting "grooves."

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

REPEAT HOLD – Defeats the inputs of the delay and "captures" the current feedback loop.

FEEDBACK L – Sets the amount of feedback for the left channel to determine the number of repeats.

FEEDBACK R – Sets the amount of delay feedback for the right channel. To preserve "tail" balance, this control will be adjusted automatically when RATIO is changed. You may override automatic settings by setting a new value here manually. Negative feedback values phase invert the signal in the feedback loop.

TEMPO – Locks the TIME parameter in rhythmic relation to the global tempo.

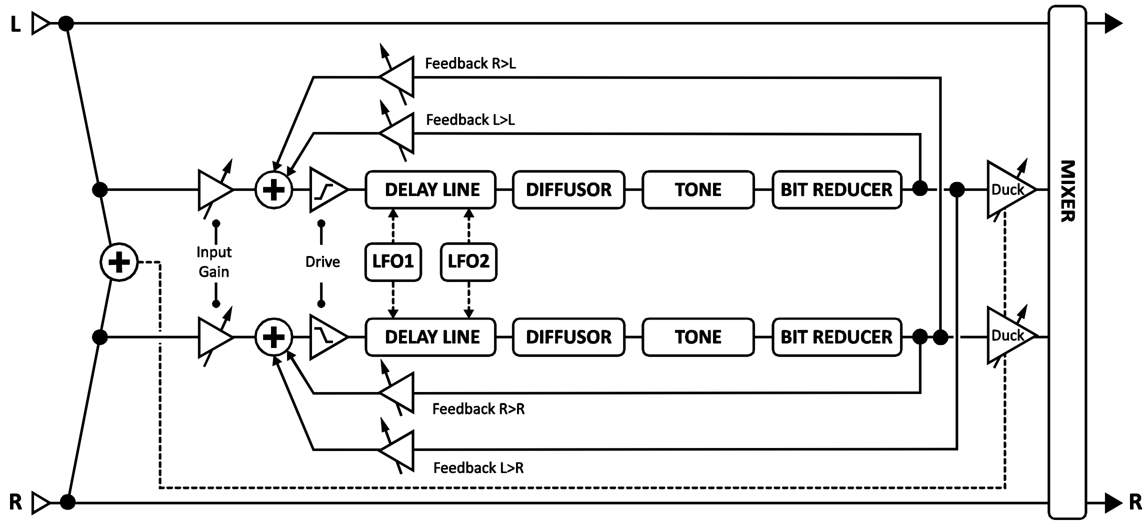
DRIVE – Sets the amount of distortion in the delay path.

BIT REDUCTION – Sets the number of bits to be subtracted (from 24 bits), enabling lo-fi effects.

5.6.3 Dual Delay

This is a stereo-in/stereo-out delay with fully independent controls for most L-R parameters.

Figure 5-10 – The Dual Delay Block



TIME L, **TIME R** – Dual parameters to set the time of the left and right delay lines. When **TIME** is shown in parenthesis, it is being set automatically by the **TEMPO** parameters (see below). Set **TEMPO** to "NONE" to regain manual control.

LEVEL L, LEVEL R – Dual parameters to independently set the volume levels of the dual delay lines.

MASTER PAN – The panning of each voice is multiplied by this value. A value of 100% will result in each voice being panned as set in the individual pan controls. A value of 0% will result in both voices being panned to center. A value of -100% will reverse the position of the voices. You can use a modifier on this parameter to move the voices around the stereo field in real time.

R TIME RATIO – Allows the right delay line time to be reduced. Very for subtle **TEMPO** offsets!

TEMPO L, TEMPO R – Dual parameters to lock the independent **TIME L** and/or **TIME R** parameters in rhythmic relation to the global tempo. See the **TEMPO** section under the MONO DELAY config (p.61) for more about the relationship between BPM and delay time in milliseconds.

FEEDBK L->L, FEEDBK R->R – Dual parameters to independently set the amount of feedback for the left and right channels, determining the number of repeats heard.

FEEDBK L->R, FEEDBK R->L – Dual parameters to independently set the amounts of cross feedback for the delay lines. This controls how much of the left delay line is fed back into the right and vice versa.

Negative values phase invert the signal in the feedback loop.

PAN L, PAN R – Dual parameters to independently set the pan positions of the dual delay lines.

DRIVE – Sets the amount of distortion in the delay path.

BIT REDUCTION – Sets the number of bits to be subtracted (from 24 bits), enabling lo-fi effects.

5.6.4 Ping-Pong Delay

The echoes of this easy-to-use **Ping-Pong Delay** alternate between left and right channels in stereo. The Ping-Pong Delay uses the same algorithm as the Stereo Delay (p. 66), except the ECHO PAN parameter is replaced by SPREAD.

SPREAD – Controls stereo width by setting the pan position of the delay outputs from hard pan (100%) to mono (0%) to swapped hard pan (-100%).

RATIO – Allows altering the time difference between the two echoes of the ping-pong.

5.6.5 Sweep Delay

The **Sweep Delay** uses the same algorithm as the Stereo Delay (above, p. 67), 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 an "intensity" control.

SWEEP TYPE – Sets the waveform of the LFO that controls the sweeps. See section **16.7** on p. **174** for more information on LFO waveform shapes and phase.

SWEEP RATE – Sets the speed of the sweeps.

SWEEP TEMPO – Locks the SWEEP RATE parameter in rhythmic relation to the global tempo.

SWEEP PHASE – Adjusts the phase differential between left and right sweep LFO waveforms.

5.6.6 Reverse Delay

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 backwards sound, make sure **MIX** is set to "100%."

The Reverse Delay uses the same layout as the Mono Delay (**5.6.1**, above) except as noted below:

TIME – Sets the length of time that the delay line will "record" before reverse playback begins. When **TIME** is shown in parenthesis, it is being set automatically by the **TEMPO** parameter (see below). Set **TEMPO** to "NONE" for manual control.

FEEDBK – Sets the amount of feedback to add additional repeats to the reversed snippets.

ECHO PAN – Controls the placement of the wet (reverse playback) signal in the stereo field. Note that this is different than the MIX page BALANCE control, which affects both wet and dry.

RUN – When this is turned ON, the reverse playback process is active and can be heard. Turning RUN to OFF will mute playback (though any samples in the buffer will still silently run out). This switch can be remotely operated with a modifier (attached, for example, to a footswitch) to stop and start playback.

TRIG RESTART – When this is set to “ON” the reverse playback restarts when triggered via the RUN control. If set to “OFF,” playback continues from the current position. The combination of RUN+TRIG RESTART can be used to precisely align reversed passages to certain moments in a performance or to re-align tempo-based reversing to the groove.

Tip: If you’re working with a sequencer, assign an EXTERNAL controller and re-trigger this every few bars to keep sync.

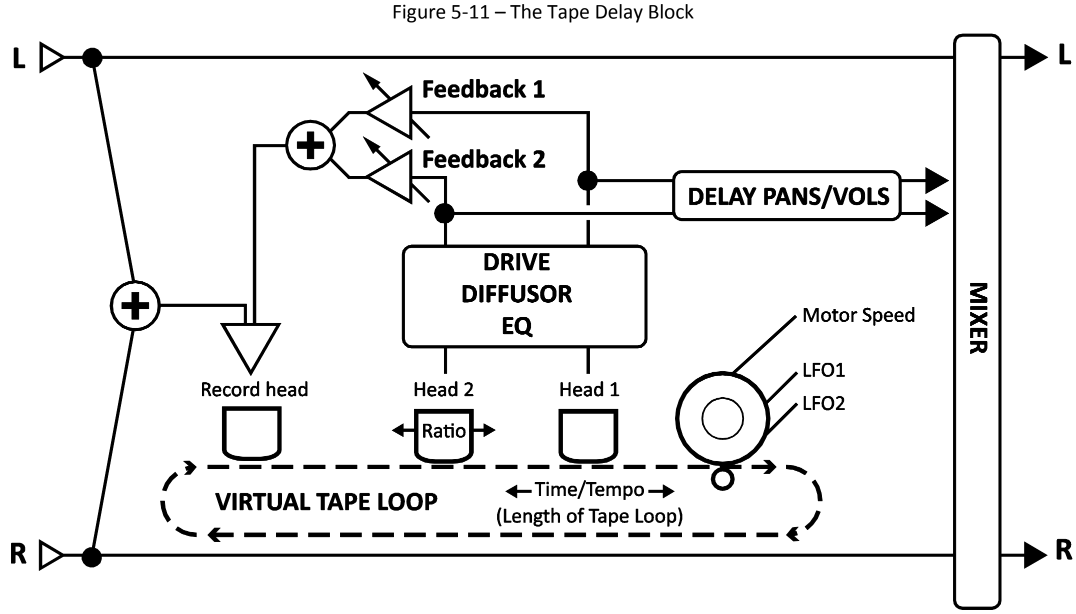
TEMPO – Locks the TIME parameter in rhythmic relation to the global tempo. See the **TEMPO** section under the **MONO DELAY** type (above) for more about the relationship between BPM and delay time in milliseconds.

XFADE TIME – Sets the crossfade time between reverse audio snippets. When the playback position approaches the delay time, a new snippet begins playback at time zero. The crossfade time controls how long it takes for the old snippet to fade out and the new one to fade in. You can achieve interesting and rhythmic variations by setting long crossfade times. For classic reverse delay sounds, set this at or near minimum value.

5.6.7 Tape Delay

The **Tape Delay** simulates a tape echo with two heads and motor speed control. It is ideal for vintage tape echoes, but can also create vintage digital or analog delay effects. Built in EQ and modulation make it warm and warbly.

Figure 5-11 – The Tape Delay Block



MOTOR SPEED – Sets the relative speed of the tape motor from 50% to 200%. This parameter can be modified in real time, making it possible to “time warp” delayed material.

NOTE: The effect of MOTOR SPEED is cumulative with that of the onboard LFOs (see Delay Common Parameters on p. 65). To accommodate maximum flexibility for these individual controls, it is possible that extreme settings may “clip” modulation by pushing time excursion beyond the “legal limit.”

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 global tempo. See the **TEMPO** section under the **MONO DELAY** config (p. 66) for more information on tempo and time.

HEAD 2 RATIO – The Axe-Fx II 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. 95%) can widen the echo sound, while values expressing whole number relationships like 2:8 (87.5%), 3:4 (75%), 2:3 (66%) or 1:2 (50%) create rhythmic grooves.

LEVEL 1, LEVEL 2 – These set the output level of each of the playback heads.

FEEDBACK 1, FEEDBACK 2 – These set the amount from each playback head to be routed back to the recording head to create feedback or "regeneration". Higher values will 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 dangerously loud. Increase feedback settings slowly, and watch the front panel clip LED as a warning indicator. Lowering **MASTER FEEDBACK** first can also help. Negative values phase invert the signal in the feedback loop.

PAN 1, PAN 2 – These place the outputs from each head within the stereo listening field.

Note that the "MONO TAPE" **TYPE** uses the "TAPE" **CONFIG**, but the "STEREO TAPE" uses the "STEREO" **CONFIG**.

5.6.8 Delay Common Parameters

Delay Modulation Parameters

Modulation systematically changes delay time, resulting in Doppler-like changes to the speed and pitch of the echoes. This can create chorus effects, the "wow and flutter" of a worn tape delay, or extreme "Ray Gun" sounds.

LFO1 TYPE, **LFO2 TYPE** – Sets the "shape" of the modulation. See section **16.7** on p. **174** for more information on LFO waveform shapes. 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, PINGPONG, and REVERSE configurations use only the LEFT delay line).

LFO1 RATE, **LFO2 RATE** – Sets the delay time modulation speed. When any **RATE** is shown in parenthesis, 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 global tempo. For example, if the tempo is set to "1/4" and the global tempo is 120 BPM, the LFO rate will automatically be set to 2 Hz (BPM/60 = Hz). To ignore the global tempo, set these controls to NONE.

LFO1 DEPTH, **LFO2 DEPTH** – Sets the depth of delay time modulation.

LFO1 DEPTH RANGE, **LFO2 DEPTH RANGE** – Sets the range of delay time modulation to LOW or HIGH.

LFO1 PHASE, **LFO2 PHASE** – Sets the LFO phase offset for the right delay line. See section **16.7** on p. **174** for more information on LFO phase. Has no effect on MONO, REVERSE, and TAPE configs.

The MOD 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 0.0 to defeat the ducker.

DUCKER THRSHLD – 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 REL TIME – Sets how long it takes for the delay signal to return to normal when the input goes below the threshold. A short value here causes the ducked echoes to return to full volume the moment you stop playing. Longer times cause the levels to swell back more gradually.

Finally, the MOD page contains some other cool parameters.

DIFFUSION – Sets the amount of echo diffusion. This causes the echoes to get "blurry" and can be used to smooth the sound.

DIFF TIME – Sets the delay time for the diffuser.

PHASE REV – Allows the Left, Right, or Both delay line outputs to be phase inverted.

Delay EQ Parameters

The delay features an equalizer inside the loop (before the feedback tap) to shape echo tone over time.

LOW CUT – Sets the frequency of the low-cut filter. Increase for thinner sounds.

HIGH CUT – Sets the frequency of the high-cut filter. Decrease for darker sounds.

SLOPE – Sets the filter slopes, in dB per octave, of the high and low-cut filters.

Q – Sets the resonance of the high and low-cut filters. High values create boosted peaks at the cutoff points.

FREQ 1, GAIN 1, Q 1 – Controls for one of two peaking filters. Select the frequency to boost or cut, and set Q to determine the width of the effect.

FREQ 2, GAIN 2, Q 2 – Controls for the second peaking filter.

Delay Mix Parameters

The **Delay** block has a **MIX** page with **LEVEL, BALANCE,** and **BYPASS MODE** parameters.
See **Common Mix Parameters** on p.**128** for more information.

Delay MIX works differently than other MIX parameters. The dry signal stays constant at unity until Mix reaches 50% then decreases linearly to zero. Conversely the wet signal starts at zero and then increases linearly to unity when Mix reaches 50%.

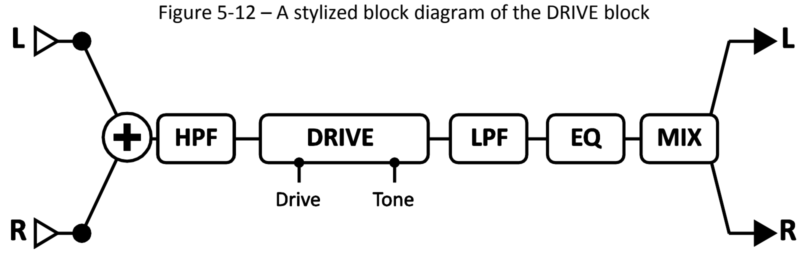
A Word on "Spillover"

The Axe-Fx II Delay is capable of "spillover," which means that effect tails ring out when the effect is bypassed or when you change presets. For more on this subject, please see **Setting Up Spillover** on p. **181**.

5.7 Drive [DRV]

The **Drive** block replicates 29 different, classic stompbox effects, ranging from sublime to intense. The overdrive ("OD" or "DRIVE") types are based on a cold-cathode tube model and give a warm, mellow overdrive tone. The BOOST types don't distort much unless the drive is set quite high. BOOST types are primarily tone-shaping devices useful for "pushing" an amp. Distortion types ("DIST") are based on a variety of tube and solid-state models and give classic distortion tones. The fuzz ("FUZZ") types are based on a hard-clipping distortion and give a raspy sound. Drive effects include the basic controls you will find on their real world equivalents: tone, drive amount, and level, plus advanced controls like **SLEW**, **BIAS**, **CLIP TYPE,** and more, enabling you to create drives with custom gain and tone-shaping. Each Axe-Fx II preset can use two fully independent **Drive** blocks.

Figure 5-12 – A stylized block diagram of the DRIVE block



XY The Drive block supports X/Y Switching. See p. **36** for more information.

Drive Parameters

TYPE – Selects the type of drive pedal or effect. A complete list follows.

Manufacturer names and product names mentioned below are trademarks or registered trademarks of their respective owners, which are in no way associated with or affiliated with Fractal Audio Systems. The names are used only to illustrate sonic and performance characteristics of the Fractal Drive TYPES, which have been created by incredibly detailed analysis of the actual amps that inspired them.

TYPE	NOTES
BB Pre	Based on the Xotic® Pedals BB Preamp®.
Bender Fuzz	Based on the classic Tonebender circuit.
Bit Crusher	Based on a black box we found lying in the trash outside *Studio Harshclip.*
Blues OD	Based on the Marshall™ Bluesbreaker®.
Esoteric ACB	Based on the Xotic® Pedals AC Booster®.
Esoteric RCB	Based on the Xotic® Pedals 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™ Metalzone™, popular for extreme gain settings.
Master Fuzz	Based on the Maestro Fuzztone, aka Satisfaction fuzz.
Micro Boost	Based on MXR's 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 ProCo™ 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
Shred Dist	Based on the Marshall™ Shredmaster®.
Super OD	Based on the Boss™ Super Overdrive.
T808 MOD	Captures the most popular TS overdrive mods.
T808 OD	Based on the Ibanez™ TS-808® Tube Screamer overdrive.
Tape Dist	Simulates the clipping of an overdriven reel-to-reel tape deck.
Timothy	Based on a Paul Cochrane "Timmy"
Treble Boost	Based on a classic Treble Booster.
Tube Drv 3-knob	Based on the Chandler™ Tube Driver that actually contained a 12AX7 (3-knob version)
Tube Drv 4-knob	Based on the Chandler™ Tube Driver (4-knob version)
Zen Master	Based on Hermida® Zen Drive.

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

Note: A high-gain drive before a high-gain amp can cause noise and squealing.

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

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

MIX – Controls the ratio of dry to wet. This should normally be set to "100%."

BAL – Sets the left/right output balance of the block.

LOW CUT – Controls the frequency of the input high-pass filter. Increase to prevent "flubby" distortion.

HIGH CUT – Controls the frequency of the output low-pass filter. Lower this for a darker sound.

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 SHAPE – 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.

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.

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.

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.” Tip: It is *supposed* to sound nasty!

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

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 (the default).

BYP MODE – Sets the bypass mode of the block to MUTE or THRU. See **Common Mix Parameters** on p.**128**.

BASS/TREBLE – These adjust the low-end and high-end of the built-in equalizer from +/- 12 dB.

MID, MID FREQ – Sets mid-boost or cut (+/- 12 dB) and frequency for the built-in equalizer.

5.8 Effects Loop [FXL]

The Axe-Fx II has a full-stereo effects loop that can be used to insert outboard hardware anywhere in the signal chain of a preset. Any signal at the input of the [FXL] block is passed to the physical OUTPUT 2 (“FX SEND”) on the Axe-Fx II. Any signal received at the physical INPUT 2 (“FX RETURN”) appears at the outputs of [FXL].

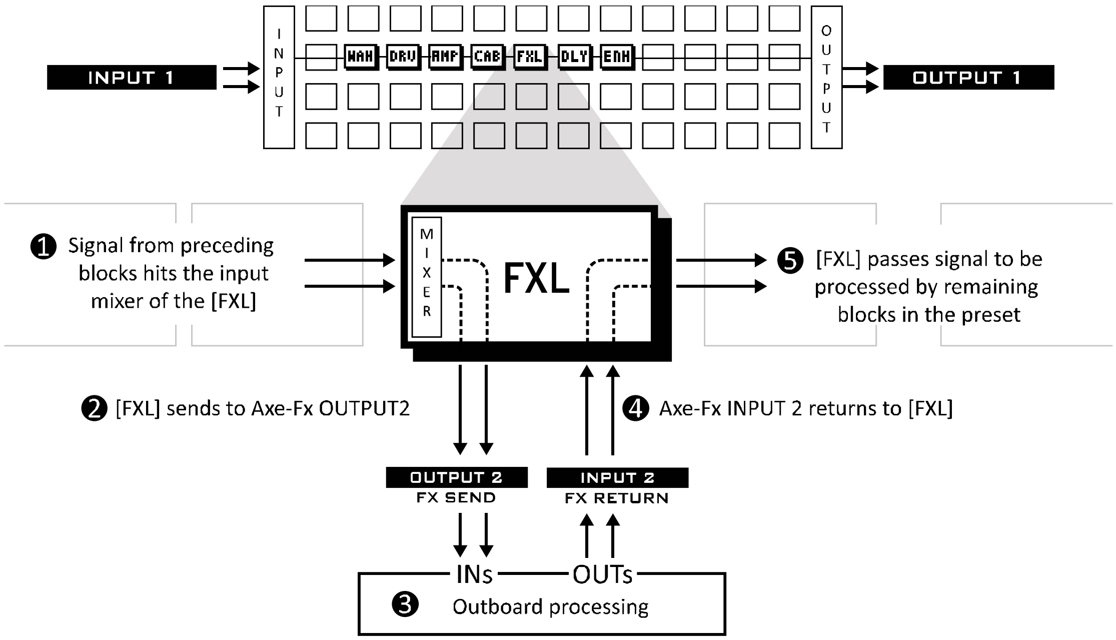


Figure 5-13 – Effects Loop Routing

The Effects Loop block is frequently used to insert the preamp section of a head or combo. The main output of the Axe-Fx is then routed to the effect loop input of the real amp. See the diagram on p. **21** for more on "the four cable method" of setting up the Axe-Fx II.

Alternate Use as an Auxiliary Output or Input

The **FX Loop** Block doubles as a means to provide an auxiliary output or Input. Feeding signal to the [FXL] block routes signal directly to the OUTPUT 2 jacks. This is useful, for instance, to send a fully processed mix to the front-of-house while simultaneously feeding a real power amp and speaker cabs. See the example on p. **22** for more details. Alternatively, you might use the INPUTS and not the OUTPUTS so the [FXL] block allows a second input signal to be "injected" on to the Axe-Fx layout grid at any point. It is possible to process (discretely) up to four channels of input in this way and have them mixed to a single pair of outputs.

The SEND page of the **FX Loop** is a standard Axe-Fx mixer. See page **Mixology** on p. 177 for details. The FX LOOP's setting for **MAIN** is stored per scene (see **SCENES** on p. 169).

The **FX Loop** block has a **MIX** page with **LEVEL, BALANCE,** and **BYPASS MODE** parameters. See **Common Mix Parameters** on p. **128** for more information.

Each Axe-Fx II preset can use one **FX Loop** block.

5.9 Enhancer [ENH]

The **Enhancer** offers two modes to increase “spatialization” or stereo separation of a signal.

Modern Enhancer Parameters

The **Modern Enhancer** creates a widening effect through frequency-based separation of left and right channels. In comparison to the **Classic Enhancer**, it does not introduce the risk of phase cancellation issues when the signal is later summed to mono, and therefore poses less of a risk for use in presets.

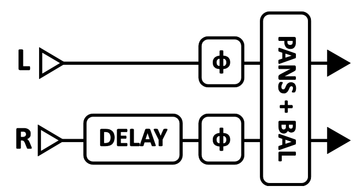
WIDTH – Determines the character of the effect, shaping the bands of frequency separation.

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

The **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.



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.

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

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

BAL – The Balance control changes the relative volumes of the left or right outputs. Sometimes the enhancer effect will cause a shift in the apparent stereo placement of sound. The balance control may be used to compensate for this.

The Enhancer has no mix parameters or bypass mode modifier switch. Both types offer a **LEVEL** control.

Each Axe-Fx II preset can use one **Enhancer** block.

5.10 **Feedback Send [SND] & Return [RTN]**

The **Feedback Send** and **Feedback Return** blocks allow you to route signal from any point to any other point, bending the rule that it must only flow from input to output. No connection will be visible between the two blocks, but signal nonetheless flows from the output of the SEND to the input of the RETURN. Both blocks must be used for either to function. The primary function of the Send and Return blocks is to allow effects to be inserted inside the feedback loop of a delay. The MIX of the delay in the loop is normally set to "100%," since direct signal recirculation results in instability.

Warning: Use these blocks with caution, as you can easily program an unstable loop and cause internal clipping and/or very high sound levels, which may damage your hearing or connected equipment. With the RETURN block **MIX** at 100%, set its **LEVEL** control to **minus 80 dB** and bring it up slowly. If you start to hear squealing or other signs of runaway feedback, return the **LEVEL** control to minimum and analyze your routing for possible causes of instability. See **Using Send and Return** on p. **182** for creative application ideas.

The **Feedback Send** block has **SEND LEVEL** and **OUTPUT LEVEL** controls. The latter controls the amount of signal that passes through the block. Each Axe-Fx II preset can use one **FB Send** and one **FB Return** block.

The **Feedback Return** block has **MIX, LEVEL, BALANCE,** and **BYPASS MODE** parameters. See **Common Mix Parameters** on p. **128** for more on these.

5.11 Filter [FLT]

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) booster with greater programmability than the ones in the **DRIVE** block. The Filter block is equipped with individual left and right pan controls. These allow you to adjust the placement of the left and right output signals in the stereo field. You can use these controls to turn a stereo signal into two mono outputs (set both to 0.0) or to reduce the stereo separation, or you can use them as general stereo manipulation tools.

The filter is stereo-in/stereo-out. Each Axe-Fx II preset can use four fully independent **Filter** blocks.

Parameters

TYPE – Selects between Null, Lowpass, Bandpass, Highpass, Low Shelf, High Shelf, Low Shelf 2 (passive type), High Shelf 2 (passive type), Peaking, Peaking 2 (Variable Q type), Notch, or Tilt EQ. The NULL type has no effect on frequency response but allows gain, phase, pan and other adjustments to be made. Note: Low and High Shelf "2" filters recreate the quirly analog shelving filters found on classic mixing consoles. These exhibit "overshoot" which gives them a certain musical quality. Set the Q between 0.5 and 0.707 to recreate those classic sounds or experiment with the Q for different effects.

FREQ – Sets the center frequency of the filter.

ORDER – Selects different filter slopes. 2nd = 12 dB/ octave, 4th = 24 dB/ octave

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.

LEVEL – Sets the output volume level of the block.

BAL – Sets the output balance of the block.

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.

BYP – Sets the bypass mode of the block. See **Common Mix Parameters** on p.**128** for more information.

5.12 **Flanger [FLG]**

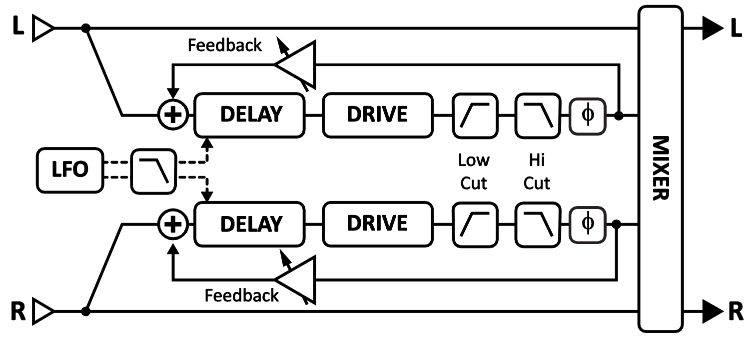


Figure 5-14 - The Flanger Block

The sound of a **Flanger** can range from subtle chorusing, to swooshing 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). Great examples of tape flanging can be heard in "Itchycoo Park" by the Small Faces (hi Justin!) or the "Listen to the Music" by the Doobie Brothers. The flanger effect has evolved through countless 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 sweep. A regenerative flanger is a real attention-getter. Each Axe-Fx II preset can use two fully independent **Flanger** blocks.

XY The **Flanger** block supports X/Y Switching. See p. **36** for more information.

Basic Parameters

TYPE – This control instantly sets other Flanger parameters for useful sound settings. Types include: DIGITAL MONO, DIGITAL STEREO, ANALOG MONO, ANALOG STEREO, THRU-ZERO, and STEREO JET.

TIME – Adjusts the delay time from 0—10ms. This changes the character of the effect. Low values give a phaser-like sound whereas high values are more ringy and metallic. Adjust to taste.

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 Flanger LFO to global LFO1. When **RATE** is shown in parenthesis, it is being set automatically by the **TEMPO** parameter (see below). Set the **TEMPO** to "NONE" for manual control.

DEPTH – Sets the maximum delay variation. Higher depths increase the amount of detuning. Usually, the rate and depth settings should be varied inversely, so an increase in rate warrants a decrease in depth. Unique sounds can also be obtained by ignoring this convention and using other combinations of rate and depth.

FDBK – Feedback sets the amount of wet signal fed back to the input. Extreme values give the flanger a more intense quality as it produces sharp resonances in the frequency response. With negative **FEEDBACK** values, the wet signals are out of phase with the dry, creating sounds with a different character than those created using positive feedback.

Note that extreme feedback at minimum flanger **DRIVE** settings will cause a siren-like ringing oscillation.

MIX – Sets the ratio of wet and dry (duplicated from the MIX page).

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

Advanced Parameters

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.

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.

HIGH CUT – This filters the wet portion of the effect signal, gently rolling off treble at the set frequency with a slope of 6db. Set to lower values for a “darker” flanging sound.

LOW CUT – Adjusts the cutoff frequency of a high-pass filter in the flanger’s feedback loop, gently removing bass frequencies as the value is increased for a “thinner” flanging sound.

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 flanger effects. Set to zero for “pristine clean.”

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.

LFO TYPE – Sets the “shape” of the modulation waveform.

*See section **16.7** on p.**174** for more information on LFO phase and waveform types.*

LFO HICUT – Lowering this control filters the LFO waveform, rounding sharp turns in its shape. Certain waveform types (saw, square, random) otherwise have “discontinuities,” which can cause clicks or pops as their values jump from one extreme to another. Lowering the LFO HICUT frequency will mitigate this.

LFO BYPASS RESET – This allows you to synchronize the start point of the flanger effect by setting the LFO to start from a set position (0°, 90°, 180°, 270°) when the effect is engaged. With the default value of “Off” the Flanger LFO will cycle freely, even when the effect is off.

AUTO DEPTH – Scales **DEPTH** to create a consistent sound at any **RATE**. This control simplifies dialing in “musical” results, but for precise control or wild sounds you may wish to turn it OFF.

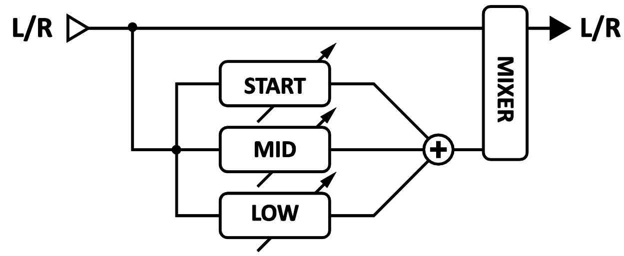
STEREO SPREAD – Controls stereo width by setting the pan position of the two delays from hard panned (100%) to dead center (0%).

DRY DELAY SHIFT – When **THRU ZERO** (above) is engaged, this allows shifting the cancellation point from the center of the waveform to the edge or anywhere in between.

Flanger Mix Parameters

The **Flanger** block also has a **MIX** page with **LEVEL, BALANCE,** and **BYPASS MODE** parameters.
See **Common Mix Parameters** on p.**128** for more information.

5.13 Formant [FRM]



Although the wah effect was originally intended to mimic the sound of the 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 considerably more unwieldy than a wah pedal. The Axe-Fx II **Formant Filter** makes "talk-box" effects possible without such a fussy apparatus.

A formant filter is an extension of the wah principle but operates with a far more *vocal* quality. Formants represent particular resonances of instruments, or in this case, of the human vocal tract. The human vocal tract generates a handful of formants to produce the vowel sounds we recognize. For example, the vowel sound "eee" can be reproduced with a bank of narrow bandpass filters with various frequencies and amplitudes.

The Axe-Fx II Formant Filter can be set statically or blend dynamically between **START**, **MID,** and **END** vowels. The **CONTROL** knob sweeps across this range, gradually changing from one vowel to the next between positions. For example, we can program the Formant Filter to go "III – AAA – OOO" for a “yoww” sound as a pedal is moved.

The Formant Filter usually sounds best when placed after distortion, although there are no hard and fast rules.

Each Axe-Fx II preset can use one **Formant** block.

Parameters

START – Sets the start vowel sound.

MID – Sets the mid vowel sound.

END – Sets the end vowel sound.

RES – Sets the resonance of the filters. Higher resonance can yield a more dramatic effect.

CTRL – Controls morphing between 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.

Formant Mix Parameters

The **Formant** block has a **MIX** page with **MIX, LEVEL**, **BALANCE**, **BYPASS MODE,** and **GLOBAL MIX** parameters.

See **Common Mix Parameters** on p.**128** for more information.

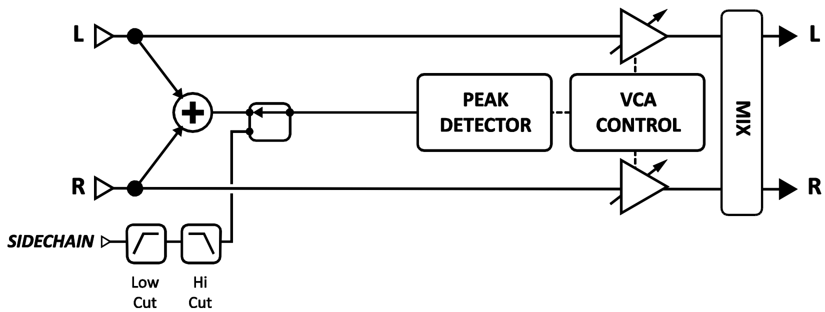
5.14 **Gate/Expander [GTE]**

The downward **Expander** module is sort of a "reverse compressor" that increases the difference between loud sounds and soft 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**.

The Gate/Expander block supports X/Y Switching on the XL/XL+ only. See p. **36** for more information.

Each Axe-Fx II preset can use two fully independent Gate/Expander blocks.

Figure 5-15 – The Gate/Expander Block



THRSH – (Threshold) Sets the level below which automatic volume reduction will occur. When input level is lower than the threshold, the expander reduces output volume as set by the RATIO.

RATIO – Sets the gain expansion ratio to determine how greatly signals below the threshold will be reduced. For example, when a ratio of "2" is chosen, for every dB the input signal falls below the threshold, the output signal will drop by 2 dB.

ATT – Attack time. Sets how quickly the Gate/Expander restores the gain once the threshold is exceeded.

REL – Release time. Sets how quickly the Gate/Expander reduces the gain once the signal has fallen below the threshold.

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

SCSEL – Selects the sidechain input source. NONE is the normal setting and selects the block input (sum of the rows feeding the block) as the sidechain source. The other settings allow isolating a single row or main input as the sidechain input. The other rows are summed as usual. By using a row or main input as the sidechain input, you can use the **Gate/Expander** as a ducker or de-esser. You can also choose either of the main inputs as sidechain sources, i.e. Input 1 or Input 2.

LOWCUT/LOCUT – 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 Parameters

The **Gate/Expander** block has **LEVEL**, **BALANCE,** and **BYPASS MODE** parameters, detailed on p.**128**.

5.15 Graphic Equalizer [GEQ]

The **Graphic Equalizer** is a multi-band equalizer capable of running in 5,7, 8, or 10 band configurations.

Each instance of the Graphic EQ block stores two fully independent sets of parameters called **X** and **Y**. Selecting between these allows you to change all block settings—instantly—at the touch of a switch or button (excluding any **Modifier** assignments). See **X/Y Switching** on p. **36** for more information.

The GEQ block supports X/Y Switching on the XL/XL+ only. See p. **36** for more information.

Each Axe-Fx II preset can use four fully independent **Graphic Equalizer** blocks.

The **TYPE** determines the number of bands.

	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	2500	5000	10k	--	--
7 Band	100	200	400	800	1600	3200	6400	--	--	--
5 Band	80	240	750	2200	6600	--	--	--	--	--

Each band can boost or cut up to 12 dB. The outermost bands are shelving filters.

The above modes can be run as Constant Q or Variable Q. In a variable Q equalizer, the bandwidth varies as a function of boost/cut levels, such that lower boosts/cut levels result in a lower Q.

MASTER Q adjusts the Q of all bands. A value of 1.0 sets the Q to the default value (typically one octave). Lower values increase the bandwidth and overlap of each band, higher values decrease the bandwidth.

Two **Passive EQ** types are also offered: 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 characteristic sound of vintage analog gear.

For any type, press ENTER while on PAGE 1 of the block's edit menu to reset all bands to flat.

The **AMP** block (p.**39**) has a built-in graphic equalizer at its output, making it unnecessary to follow it with a separate GEQ. There is also a global, 10-band graphic equalizer on each output (p. **146**), which can be used to modify the sound of *all* presets at once.

The **Graphic Equalizer** is stereo-in/stereo-out. Its **LEVEL**, **BALANCE,** and **BYPASS MODE** parameters are covered in additional detail on p.**128**.

5.16 **Looper [LPR]**

The Axe-Fx has a full-featured **Looper** which allows you to create multi-layered performances in real time. The maximum loop time varies from 15 to 60 seconds, depending on your choices for mono/stereo/undo. Looper functions can be operated from the Axe-Fx front panel or remotely via MIDI. Fractal Audio Systems' MIDI Foot Controller product, the MFC-101, offers a dedicated LOOPER CONTROL MODE for the Axe-Fx II.
Looper CC# assignments can be changed on the CONTROL page of the I/O menu (p. 152).

Looper Basic Controls

RECORD – When you press **RECORD,** the Looper starts recording (if you hadn't guessed as much we have to wonder…) Here's something that's not as obvious: press record again to stop recording and instantly start playback. This prevents you needing to hunt around for more than one footswitch to start a loop. If the record length reaches the maximum allowed for the current mode, recording will end and playback will stop automatically.

PLAY – This switch has two functions: use it to stop recording and begin playback, or use it to instantly stop playback that is already rolling. A momentary switch can be assigned to **PLAY** for stutter effects.

ONCE – Toggles "auto-stop at loop-end," so that if playback is running, the looper will stop automatically when it reaches the end of the loop. If playback is already stopped, **ONCE** will start playback, play through the loop and then stop. You cannot go directly from **RECORD** to **ONCE**.

STACK – Overdubs audio on an existing loop. Pressing it again stops the addition but playback continues. Existing audio is faded every time through the loop based on the setting of the **DUB MIX** parameter on the Looper's MIX page.

UNDO – This removes the most recently recorded overdub. Undo removes a given "take," defined as everything between when you pressed **STACK** to start overdub recording and when you turned it off.

REV – Pressing this reverses the direction of the Looper. Playback and **STACK** recording are supported.

HALF – Slows the Looper's speed to half. Playback and normal or overdub recording are supported at half speed. For double speed playback, record in half speed and then switch back to normal. *Note: Halving the speed slightly reduces high-end frequency response.*

Each Axe-Fx II preset can use one **Looper** block.

Looper Advanced Controls

MODE – Selects the Looper mode, determining mono/stereo, looper length, and undo. Whenever you change the mode, the looper memory will be entirely cleared.

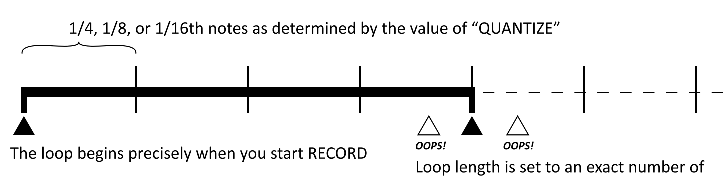
MONO – Recording and playback are in mono. Maximum loop length is 60 seconds. Undo is not possible.

STEREO – Recording and playback are in stereo. Maximum loop length is 30 seconds. Undo is not possible.

MONO UNDO – Recording and playback are in mono. Maximum loop length is 30 seconds. Undo works.

STEREO UNDO – Recording and playback are in stereo. Maximum loop length is 15 seconds. Undo works.

QUANTIZE, RECORD BEATS –
With **QUANTIZE** set to any value



except “OFF”, loop length is forced to a whole number of beats/subdivisions of the GLOBAL TEMPO (p. 158). If you stop recording too soon or too late, it will be extended or trimmed to the nearest quantize “tick” value. **RECORD BEATS** allows you to pre-determine the number of units. Recording ends and playback automatically begins when you reach the designated length (unless **PLAY IMMEDIATELY** is set to “OFF”; See below.)

THRESHOLD – The Axe-Fx Looper can start recording automatically when the input exceeds a given level.

THRESH LEVEL – sets the level that must be exceeded (at the input of the LOOPER) for recording to begin.

DUB MIX – Actually located on the Looper’s MIX page, this determines how much layers are reduced when you **STACK** new layers. If you never want old layers to decay, set this to 100%, but be aware that adding additional layers in this way can result in excessive signal and clipping.

PLAY IMMEDIATELY – Turning this OFF disables automatic playback upon completion of recording.

Looper Trim Parameters

The "Trim" **feature** allows **you** to trim the start and end points of the loop. Use **NAV** keys to select either Start or End and then turn the value wheel to adjust the trim. Modifiers can be attached to START or END by pressing **ENTER** when either is selected.

Looper Mix Parameters

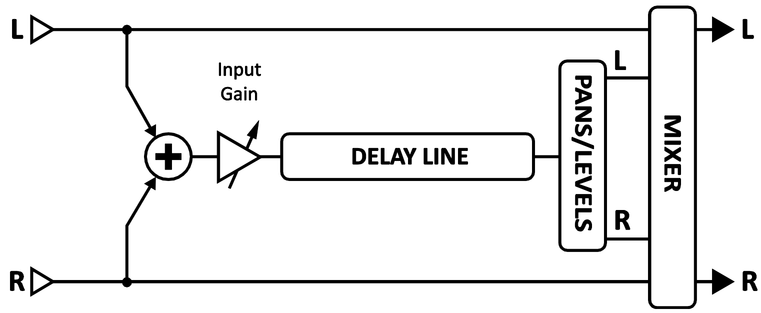
The **Looper** has a **MIX** page with **LEVEL, BALANCE,** and **BYPASS MODE** parameters.
See **Common Mix Parameters** on p. **128** for more information.

5.17 **Megatap Delay [MGT]**

The **Megatap Delay** is a 2.5 second, 40-tap delay line with parametric control of time, amplitude, and panning. This effect can be used to create interesting sonic patterns or to increase "density" before reverberant effects.

Each Axe-Fx II preset can use one **Megatap** block.

Figure 5-16 – The Megatap Delay Block



Parameters

INPUT GAIN – Sets the input level to the effect. The primary purpose of this is to allow you to attach a controller (e.g. pedal) to the delay level input level for operation similar to that of an "Aux Send." In other situations, this control should be set at 100%.

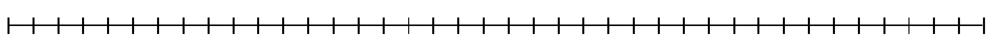
MASTER LEVEL – Controls the overall level of the delay.

TIME – Sets the delay time of the last tap. The echoes will be distributed between zero and this time.

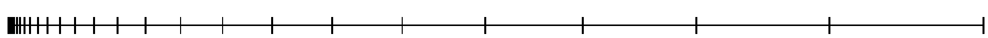
NUMBER OF TAPS – Sets the number of taps (repeats) on the delay line.

TIME SHAPE – Specifies how the time between taps changes as they progress.

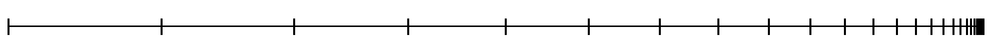
- **CONSTANT** – The time between taps will not change, regardless of the TIME ALPHA setting.



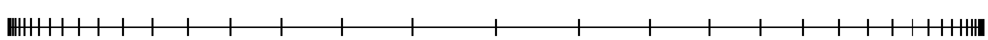
- **INCREASING** – The time between taps will increase.



- **DECREASING** – The time between taps will decrease.

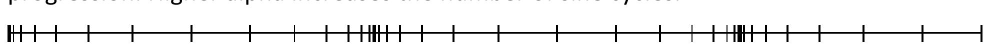


- **UP / DOWN** – The time between taps will increase and then decrease.



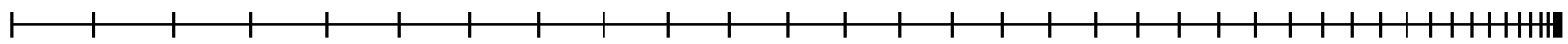
- **DOWN / UP** – The time between taps will decrease and then increase.

- **SINE** – The time between taps will decrease and then increase repeatedly in a sinusoidal progression. Higher alpha increases the number of sine cycles.

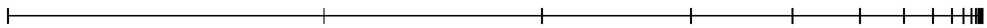


TIME ALPHA – Sets the acceleration of the rate of time change across the taps. A setting of 0% results in no effect, while 100% results in an extreme effect.

Ex: Decreasing, Moderate Alpha



Ex: Decreasing, Higher Alpha



AMPLITUDE SHAPE – This specifies how the volume increases or decreases from tap to tap.

AMPLITUDE ALPHA – Sets the acceleration of the rate of volume change across the taps.
A setting of 0% results in no effect, while 100% results in an extreme effect.

PAN SHAPE – This specifies how the pan changes from tap to tap as they progress.

PAN ALPHA – Sets the acceleration of the rate of pan change across the taps.
A setting of 0% results in no effect, while 100% results in an extreme effect.

TIME RANDOMIZE – Controls how much the tap spacing is randomized.

Megatap Mix Parameters

The **Megatap** block has a **MIX** page with **MIX**, **LEVEL, BALANCE**, **BYPASS MODE**, and **GLOBAL MIX** parameters.
See **Common Mix Parameters** on p. **128** for more information.

5.18 **Mixer [MIX]**

The **Mixer** block contains a simple, linear mixer able to combine up to four stereo signals into either a stereo or mono mix. Each pair of gain and balance controls corresponds to a **row** of the grid. For a more thorough description of how Axe-Fx II mixers work, see section **16.10**, Mixology, on p. **177**.

The Mixer block supports X/Y Switching on the XL/XL+ only. See p. **36** for more information.

Page 1 Parameters

GAIN 1 – Sets the level for the incoming signal from a block in **row 1** of the column left of the mixer.

BAL 1 – Sets the balance between left and right signals of the block in **row 1** of the column left of the mixer.

GAIN/BAL 2, 3, 4 – These pairs of controls respectively set the level and balance of incoming signals from the blocks in rows 2, 3, and 4 of the column left of the mixer.

Page 2 Parameters

LEVEL – Sets the level of the output mix.

OUTPUT MODE – Specifies whether the outgoing mix should be stereo or summed to dual mono.

Each Axe-Fx II preset can use two **Mixer** blocks. Note that on the Mark II, the mixer cannot be BYPASSED.

5.19 **Multiband Compressor [MBC]**

The Axe-Fx II contains a three-band compressor that is great for mastering or compressing a mix. It also works as a tone-shaping tool, providing independent level and dynamics control over low, mid, and high frequencies.

The basic principle of the **Multiband Compressor** is that the input is divided into three component signals using a crossover. Compression is applied to the bands individually before they are recombined. The MBC lets you isolate frequency bands of the input material and apply different types or amounts of compression to each. Multiband compression is a de-facto mastering tool and can greatly improve a final mix or a complex guitar sound.

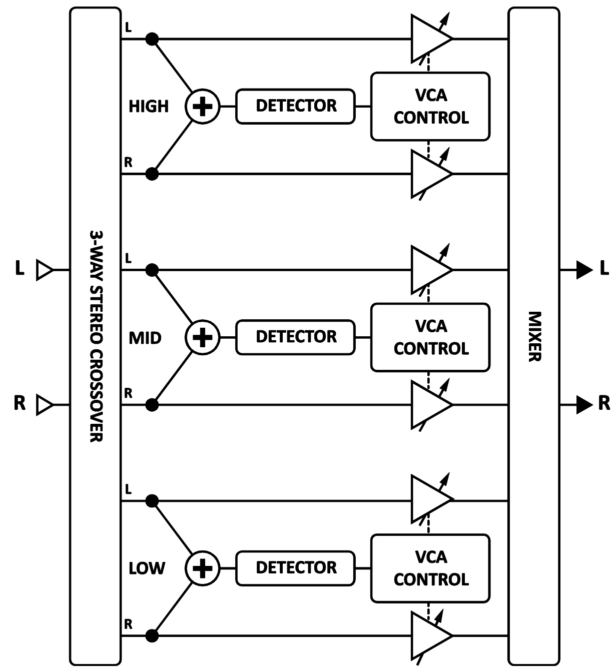


Figure 5-17 – Stylized view of the Multiband Compressor

Each Axe-Fx II preset can use two fully independent **Multiband Compressor** blocks.

Parameters

FREQ1 – Sets the crossover frequency between bands 1 and 2 from 50–500 Hz.

FREQ2 – Sets the crossover frequency between bands 2 and 3 from 1000–10000 Hz.

Each compressor section has its own menu page with the following parameters:

THRSH – Sets the threshold above which output compression starts to 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.

ATT – Attack rate. Sets how long it takes for compression to occur once the signal exceeds the threshold. Slower values allow more of the loud signal to punch through before the compressor can reduce it.

REL – Release rate. Sets how long it takes for the level to return to normal after the signal falls beneath the threshold. Slower times can cause the compressor's gain reduction to remain in effect even after a loud signal has given way to a quieter section.

LEVEL – Sets the output level of the selected band.

DET – Selects whether the compressors will use RMS ("Root Mean Square"), FAST RMS (mimicking classic rack-mount compressors), PEAK detection, or RMS+PEAK detection. RMS is "smooth" and useful to even out program material over longer periods of time. Peak detection, commonly used with guitar, is useful for fast limiting. RMS+Peak responds quickly to transients but also considers RMS for overall smoothness.

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.

5.20 Multi-Delay [MTD]

The **Multi-Delay** is a rhythmic multi-tap delay block. Each Axe-Fx II preset can use two fully independent **Multi-Delay blocks**, each of which may be set to any one of ten sub-algorithms: Quad Tap, Plex Delay, Plex Detune, Plex Shift, Band Delay, Quad Series, Ten-Tap, Rhythm Tap, Diffusor, and Multi Tape Delay. These are detailed below.

TYPE	
Quad-Tap	A stereo delay line offering four taps, each with its own parameters. A truly creative delay experience.
Plex Delay	Four delay lines interacting in a matrix of feedback, with overall feedback set as **DECAY TIME.**
Plex Detune	Same as the above, but with a pitch detune control for each line. Gorgeous ambience!
Plex Shift	Extending shifter range to +/- 24 results in sounds from sweet and shimmery to scary and surreal.
Band Delay	Similar to the quad-tap, with bandpass filters at the output of each tap.
Quad-Series	Four delays in series interact through a very cool and unique feedback structure.
Ten-Tap Delay	Create cool rhythms with ten independent taps on a ten-second delay and an innocative “decay” control.
Rhythm Tap	Enter LEARN mode and tap out the rhythm you want to hear.
Diffusor	Multiple delay lines in a matrix create lush reverb-like effects.
Quad Tape Delay	Similar to the TAPE config of the DELAY block, Quad-Tap offers four heads plus a MOTOR SPEED control.

Multi-Delay Common Parameters

Every Multi-Delay **TYPE** shares a common set of PAGE 1 parameters. The **TYPE** control selects which of the above-mentioned sub-algorithms to use, and **INPUT GAIN** determines the amount of signal fed to the effect.

Master Parameters

Most types have one or more “MASTER” parameters, summarized here. Not all MASTER parameters appear in every type; those that do are at the top of menu PAGE 2. MASTER parameters scale the effects of other controls and can be controlled with a modifier for interesting real-time changes.

MASTER TIME – Scales all the delay times in the block.

MASTER LEVEL – Scales the output levels of all taps at once.

MASTER PAN – Scales all tap pan amounts, essentially acting as a width or spread control. Negative values will reverse pan placement in left and right channels.

MASTER FEEDBACK – Scales the feedback amounts of all taps or diffusors.

MASTER FREQ – Scales the frequency values for the filters of all four taps from 0.316–3.162×. You can create dynamic filter effects by using a modifier to change this parameter in real-time, but be sure not to set Q values too high or too low, or the result will be difficult to hear.

MASTER PITCH – Scales the values for all shift parameters in the block.

MASTER DETUNE – Scales the values for all detune parameters in the block.

MASTER Q – Scales the Q values of all four taps from 0.1–10.0×

MASTER RATE – Scales the rate of all LFOs in the block.

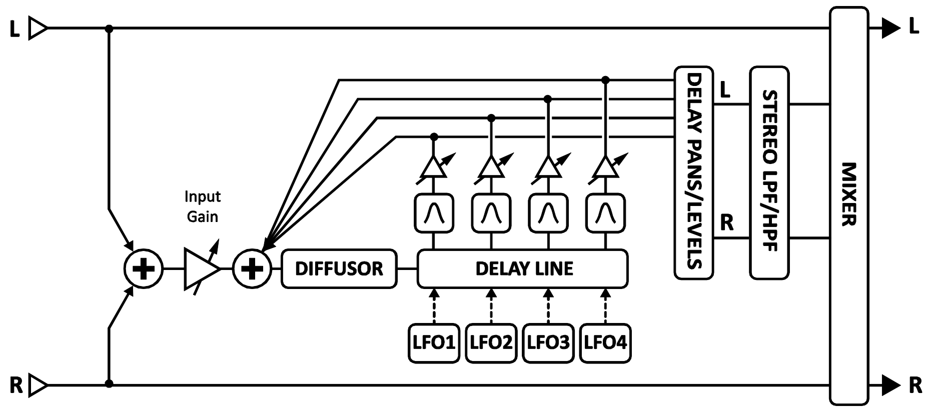
MASTER DEPTH – Scales all the depths of all LFOs in the block.

All **Multi-Delay** types share a common **MIX** page with **MIX**, **LEVEL**, **BALANCE**, **BYPASS MODE**, and **GLOBAL MIX** parameters. See **Common Mix Parameters** on p.**128** for more information on these controls.

5.20.1 Quad Tap Delay

The **Quad Tap Delay** offers four “taps,” each of which extracts a signal from any point in the delay line. It is useful for cool creative and rhythmic effects. Each tap has its own level and pan controls, plus a bandpass filter with adjustable frequency and Q. Four feedback controls are provided, but the sum of the four feedbacks may not exceed 100%. Notice that feedback from all four taps is summed at the input, so even if its output level is reduced to 0%, a tap with any feedback value greater than zero will still be heard the next time another tap plays.

Figure 5-18 – The Quad Tap Multi-Delay Block



Parameters

TEMPO 1,2,3,4 – Sets the corresponding TIME parameter in rhythmic relation to the global tempo. For example, if the global tempo is 120 BPM, and **TEMPO** is set to “1/4” (one echo per beat), time will be 500 ms. To ignore the global tempo, set to “NONE.”

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 parenthesis, it is being set automatically by the corresponding **TEMPO** parameter (see below). Set **TEMPO** to “NONE” for manual control.

LEVEL 1,2,3,4 – Sets the level at the output of the selected tap.

PAN 1,2,3,4 – Sets the pan for the selected tap in the stereo mix.

FEEDBACK 1,2,3,4 – These sets the overall feedback blend. If the total exceeds 100%, the total feedback is calculated and then normalized to 100%. Negative values are phase inverted.

FREQ 1,2,3,4 – Sets the center frequency of the bandpass filter for the selected tap.

Q 1,2,3,4 – Sets the width of the bandpass filter for the selected tap. Higher values result in a narrower range of frequencies passing through.

DIFFUSION – Sets the mix level of the diffuser block that precedes the delay line. Diffusion “smears” transients and can be used as a type of reverb for creating interesting ambience effects.

DIFFUSION TIME – Longer times will smear transients over a longer period.

DUCKER ATTEN – (**Ducker 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 0.0 to defeat the ducker.

DUCKER THRSHLD – (**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 REL 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 “pop back” into the sonic forefront the moment you stop playing. Longer times will cause the level to swell back in more gradually.

LFO 1 AS MASTER – Locks the rates of the LFOs for taps 2, 3, and 4 to the settings for LFO1.

LFO 1,2,3,4 RATE – Sets the rate of modulation for the selected tap. Remember that while LFO1 is set as the master, the controls for 2, 3, and 4 will have no effect on the sound. When any **RATE** is shown in parenthesis, it is being set automatically by the corresponding **TEMPO** parameter (see below). Set the **TEMPO** to “NONE” for manual control.

LFO 1,2,3,4 TEMPO – Synchronizes the rate of the LFO for the selected tap in relation to the global tempo. For example, if the tempo is set to “1/4” and the global tempo is 120 BPM, the LFO rate will automatically be set to 2 Hz (BPM/60 = Hz). To ignore the global tempo, set these controls to NONE.

LFO 1,2,3,4 DEPTH – Sets the modulation depth for the selected tap. Remember that while LFO1 is set as the master, the controls for 2, 3, and 4 will have no effect on the sound

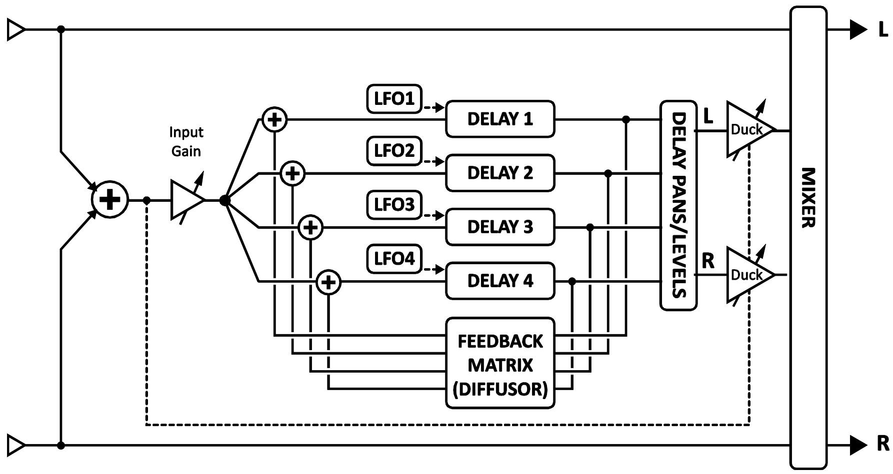
LOW CUT, HIGH CUT – These control highpass and lowpass filters which filter the feedback from all the delay lines—in contrast to the individual bandpass filters on each delay line. This allows quick and easy creation of bandlimited delays.

DRIVE –Increase drive to create the saturated sound of overdriven tape feedback.

5.20.2 Plex Delay

In terms of delay effects, a multiplexer, or "plex," is a feedback network through which each of several delay lines is fed back to itself and all the others. The result is a very smooth, reverb-like effect. When combined with modulation, the result is a huge and lush-sounding space effect that can have qualities of echo, reverb, and chorus all at once. The **Plex Delay** uses four delay lines.

Figure 5-19 – The Plex Delay Multi-Delay Type



Parameters

DECAY TIME – Sets the amount of time required for the echoes to fade by adjusting the coefficients of the feedback matrix. Use caution, as high decay times can result in instability.

DIFFUSION – Sets the amount of cross coupling between delay lines. Higher values increase the density of the echoes and result in a more reverb-like sound.

TEMPO 1,2,3,4 – Sets the corresponding TIME parameter in rhythmic relation to the global tempo. For example, if the global tempo is 120 BPM, and **TEMPO** is set to "1/4" (one echo per beat), time will be 500 ms. To ignore the global tempo, set to "NONE."

TIME 1,2,3,4 – Sets the time before the selected tap will be heard, from 0-2000 ms. Setting these values to prime numbers will result in a dense array of repeats. When any **TIME** is shown in parenthesis, it is being set automatically by the corresponding **TEMPO** parameter (see below). Set **TEMPO** to "NONE" for manual control.

LEVEL 1,2,3,4 – Sets the level at the output of the selected tap.

PAN 1,2,3,4 – Sets the pan for the selected tap in the stereo mix.

LOW CUT, HIGH CUT – Sets the cutoff frequency for gentle high and low-pass filters in the feedback loop of the delays. These controls affect all four taps simultaneously.

DUCKER ATTEN – (**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 0.0 to defeat the ducker.

DUCKER THRSHLD – (**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 REL 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.

LFO 1 RATE – Sets the rate of modulation. When **RATE** is shown in parenthesis, it is being set automatically by the **TEMPO** parameter (see below). Set the **TEMPO** to "NONE" for manual control.

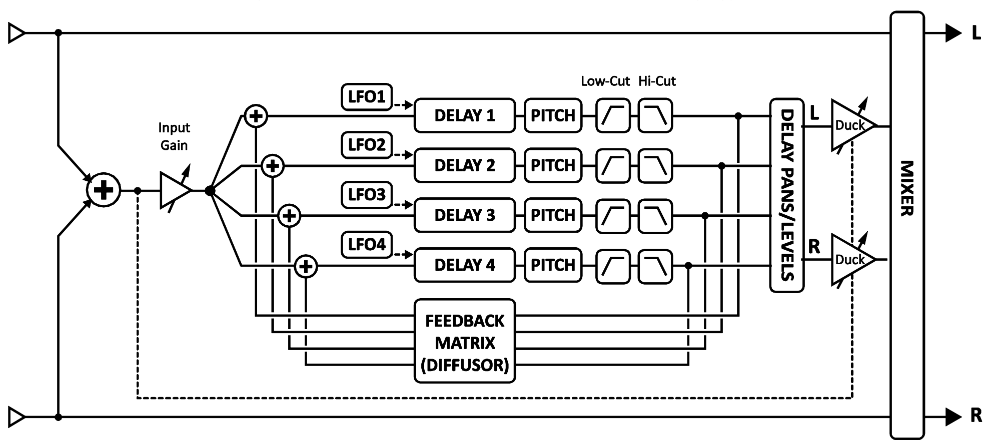
LFO 1 TEMPO – Synchronizes the rate of the LFO in relation to the global tempo.

LFO 1 DEPTH – Sets the modulation depth. Increasing modulation adds a chorus effect to the Plex Delay.

5.20.3 Plex Detune

The **Plex Detune** is based on the Plex Delay (**0** above) but adds four high-quality pitch shifters with a range of +/- 50 cents to the output of the delay taps. Like the LFOs of the Plex Delay, these shifters help create layered effect tails rich with pitch variations. With the following exceptions, the Plex Detune is identical to the Plex Delay.

Figure 5-20 – Plex Detune (and Plex Shift) Multi-Delay Type



CROSSFADE – Sets the amount of overlap used in the granules of the pitch shifters. Lower settings give a "grainy" sound, while higher values smooth the sound.

DETUNE 1,2,3,4 – Sets the amount of detune within a range of +/- 50 cents. Small values create a subtle shimmer; higher settings create descending or ascending cascades.

In comparison to the Plex Delay, the Plex Detune has no LFO or modulation parameters.

5.20.4 Plex Shift

The **Plex Shift** is nearly identical to the Plex Detune, which is itself very similar to the Plex Delay. Its pitch shifters, however, are granted a two-octave range with SHIFT parameters. This sub-algorithm has the same parameters as the Plex Detune (**5.20.3**, above) with two exceptions:

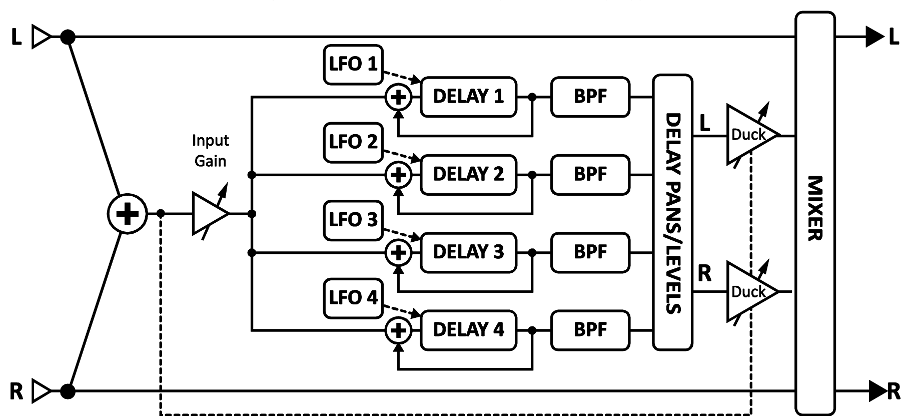
DIRECTION – This determines whether small granules of audio in the pitch shifter are played back forward or reversed. To understand how this works, imagine a word whose individual letters have been mirror-imaged but are still in the correct left-to-right order ("Axɘ-ꟻx"). In this case, the letters are 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 depends on the **TIME** setting of the tap.

SHIFT 1,2,3,4 – This sets the amount of pitch shift applied at the output of each tap within a range of +/- 24 semitones.

5.20.5 Band Delay

The Band Delay, shown below, creates filter sweep echoes with a bandpass filter at the output of each of four parallel delay lines.

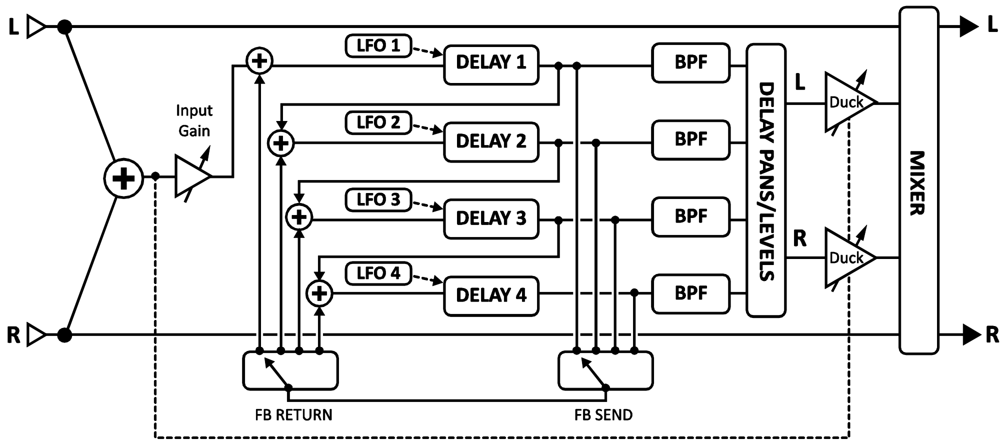
Figure 5-21 – The Band Delay Multi-Delay Type



5.20.6 Quad Series Delay

The delay lines of the **Quad Series Delay** 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 the output of any line can also be heard as it enters the next delay in the series. If you then set each delay time to 100 ms, you would hear echoes at 100, 200, 300, and 400 ms after the input.

Figure 5-22 – The Quad Series Multi-Delay Type



The parameters of the Quad Series Delay are identical to those of the Quad Tap delay (**5.20.1** above), except for the absence of the diffusor block controls, the **FEEDBACK SEND** and **RETURN** parameters, and the single **FEEDBACK** control.

FDBK SEND – Specifies which delay output (1–4) should be tapped to feedback to the input.

FDBK RET – Specifies which delay input (1–4) the feedback tap should be returned to.

FEEDBACK – Sets the amount of feedback from the send to the return.

5.20.7 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.

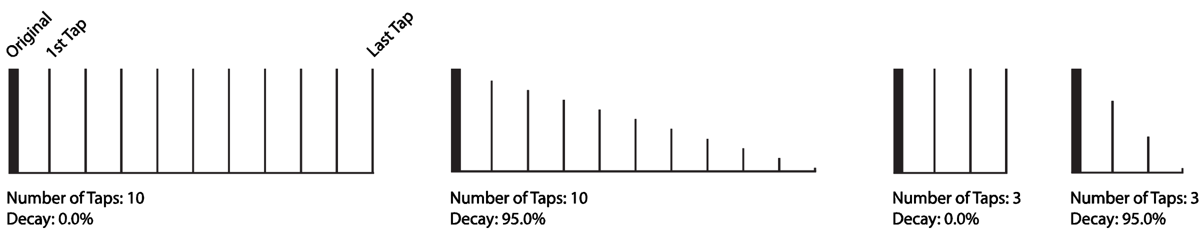
MONO/STEREO – Sets the mode of the Ten-Tap Delay. In mono mode, twice as much delay per tap is possible.

DELAY TIME – Sets the time between delay taps. When **TIME** is shown in parenthesis, it is being set automatically by **DELAY TEMPO** (below). Set **DELAY TEMPO** to "NONE" for manual control.

DELAY TEMPO – Sets the **DELAY TIME** in rhythmic relation to the global tempo.

NUMBER OF TAPS – Sets the exact number of repeats.

DECAY – Sets how rapidly the volumes of the repeats decay over time.



SHUFFLE – Sets the amount of time-offset for the odd 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 – Sets the ratio of the left-to-right delay time in stereo mode.

PAN SHAPE – Controls the shape of the panning as a function of tap number. The repeats can move slowly from one side to the other ("increasing" or "decreasing"), stay "constant," or move back and forth ("sine"). Dynamic pan effects are disabled if the Ten-Tap Delay is set to "STEREO" mode.

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 the PAN SHAPE to SINE and the PAN ALPHA to maximum.

LOW CUT – Sets the cutoff point of the high-pass filter. Higher values produce a thinner sound.

HIGH CUT – Sets the cutoff point of the low-pass filter. Lower values produce a darker sound.

TAP LEVEL (1–10) – Sets the relative level of the selected tap.

5.20.8 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 milliseconds between each tap and the previous.
2. By specifying some number of quantized time units (“divs”) between each tap and the previous.
3. By tapping a rhythm with the ENTER button and the **LEARN** function.

The parameters for the Rhythm Tap Delay include those of the Ten-Tap Delay (**0** above) plus the following:

FEEDBACK – Sets the feedback level from the final repeat to the input of the delay line. You can use this in conjunction with the decay to control the overall decay behavior. If you set decay to zero and feedback to a moderate value, the pattern will repeat, getting quieter each time through.

QUANTIZE – Quantizes the tap times to the entered note value. This can be used to aid in tapping in a rhythm. The tap times will be rounded to the nearest multiple of the note duration. You can change this value even after you’ve tapped in your rhythm.

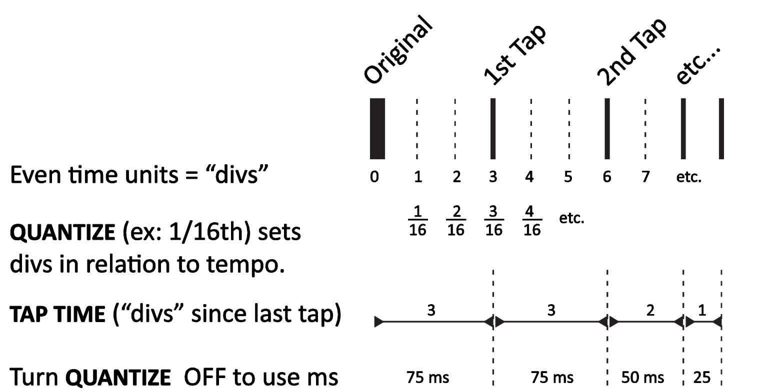


Figure 5-23 – The concept of Rhythm Tap "divs"

LEARN – Use this function to enter a rhythm by tapping. For **LEARN** to work, **QUANTIZE** must be turned OFF.

- Use the NAV keys to select the LEARN Parameter
- Turn the VALUE wheel clockwise until “<TAP ENTER>” is displayed.
- Tap the rhythm you want using the ENTER button. Be sure to include a tap for the original (dry) signal.
- When finished tapping, turn the VALUE wheel to “<DONE>”.

TAP TIME 1–10 – Sets the time of the tap (relative to the previous one) in ms or divisions (“divs”). Divs are units with the **QUANTIZE** value length. For example, if **QUANTIZE** is set to “1/16”, each div is 1/16^{th} note, and all tap times will sound some whole number of 16^{ths} 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.

5.20.9 Diffusor

A **diffuser** uses feedback delays to increase density, "smearing" transients to create interesting reverb effects. At certain time and feedback settings, taps can be heard individually, but the diffusor is typically used to create a lush blanket of sound. This algorithm chains four diffusors in series and controls the matrix with a single feedback parameter.

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 global tempo.

LFO 1 DEPTH – Sets the modulation depth to determine the intensity of time variations/chorusing.

TEMPO 1,2,3,4 – Sets the corresponding TIME parameter in rhythmic relation to the global tempo. For example, if the global tempo is 120 BPM, and **TEMPO** is set to "1/4" (one echo per beat), time will be 500 ms. To ignore the global tempo, set to "NONE."

TIME 1,2,3,4 – Sets the time of each diffusor from 0-2000 ms. When any **TIME** is shown in parenthesis, it is being set automatically by the corresponding **TEMPO** parameter (see below). Set **TEMPO** to "NONE" for manual control.

5.20.10 Quad Tape Delay

The **Quad Tape Delay** adds a **MOTOR SPEED** parameter to the "QUAD TAP" Multi-Delay type, also reducing the number of LFOs from four to two. Like the classic Space Echo effect, it can produce wild oscillating echoes in complex rhythmic patterns. For details on **MOTOR SPEED**, see section **5.6.7** on p.**65.**

5.21 **Tremolo/Panner [PAN]**

The **Tremolo/Panner** block, as its name suggests, has two uses. Tremolo varies the volume of a signal in a pulsing or chopping way, while the Panner (often referred to a “auto-pan”) varies left and right channel volumes to create the illusion of motion in the stereo field. Tremolo can be used to get classic "surf" sounds (add some spring reverb!) or to create stark, modern “chopper” effects (use a square wave LFO!) Panning covers anything from slow swings to psychotic shudders.

The **Tremolo/Panner** block supports X/Y Switching on the XL/XL+ only. See p. **36** for more information.

Each Axe-Fx II preset can use two fully independent **Tremolo/Panner** blocks.

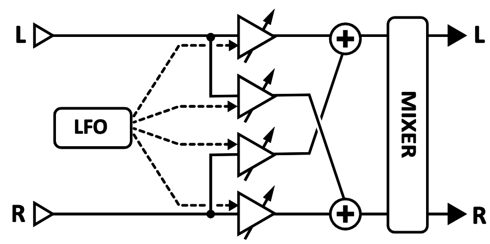


Figure 5-24 – The Pan/Tremolo Block

Parameters

EFF TYPE – Effect Type chooses between Tremolo and Panner.

RATE – Controls the speed of the tremolo or panner effect. Set fully counterclockwise to sync the chorus LFO to global LFO1. When **RATE** is shown in parenthesis, it is being set automatically by the **TEMPO** parameter (see below). Set the **TEMPO** to “NONE” for manual control.

DEPTH / WIDTH – Sets the intensity of the modulation. When **WIDTH** is set to more than 100%, the Panner uses psychoacoustic effects to pan beyond the boundaries of the normal stereo image.

TEMPO – Locks the rate to the global tempo. For example, if the global tempo is 120 BPM, and **TEMPO** is set to a “1/4,” the LFO rate will be 2 Hz (120 BPM / 60 seconds = 2). To ignore the global tempo, set the tempo control to NONE.

LFO TYPE – Selects the “shape” of the LFO waveform. Try experimenting with the Log or Exp waveforms.

DUTY – This varies the duty cycle—or “symmetry”—of the Triangle, Square, and Trapezoid waveforms.

LFO PHASE – Adjusts the phase angle of the LFO’s right waveform. With extreme settings, the Tremolo turns into a Panner and the Panner turns into a Tremolo!
See section **16.7** on p. **174** for more information on LFO waveform shapes, duty, and phase.

START PHASE – In tremolo mode, determines where the LFO will start at the moment the block is engaged.

PAN CENTER – In panner mode, this shifts the apparent center of the stereo image.

Tremolo/Panner Mix Parameters

The **Tremolo/Panner** block has a **MIX** page with **LEVEL**, **BALANCE,** and **BYPASS MODE** parameters.

See **Common Mix Parameters** on p.**128** for more information.

5.22 **Parametric EQ [PEQ]**

The 5-band **Parametric Equalizer** is one of the most precise and flexible tone-shaping tools in the Axe-Fx II. It lets you select the exact frequencies you want to focus on, adjust how much you want to boost or cut, and specify how the change should affect neighboring frequencies. The bands include a selectable-type low filter, three bell filters, and a selectable-type high filter. Select the desired band using the PAGE buttons and adjust parameters as required. A graphical display depicts the response, showing the effects of all five bands at once.

The PEQ block supports X/Y Switching on the XL/XL+ only. See p. **36** for more information.

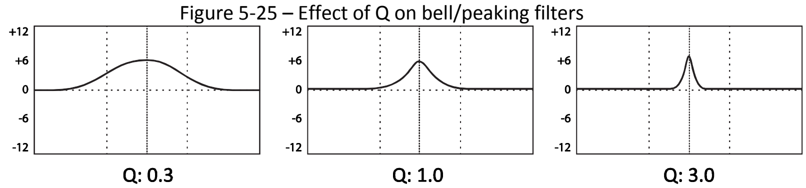
Each Axe-Fx II preset can use four **Parametric EQ** blocks.

Parameters

FREQ – Sets the center or cutoff frequency for the selected band.

Q – A measure of filter bandwidth around the center or cutoff frequency. Set a higher Q for narrower boosts or cuts. Extreme Q-values may cause output clipping even though the apparent volume is low. Reduce the Q, gain, or block output level if this occurs. Frequency and gain are kept constant as Q is adjusted in the three examples below (showing the constant Q “Peaking” type):

Figure 5-25 – Effect of Q on bell/peaking filters



Q exerts a different effect for BLOCKING or SHELVING EQ types, selectable for bands 1 and 5.

GAIN – Sets the strength of the filter through a range of +/- 12 dB.

TYPE – The first and last bands have a selectable filter type. This parameter selects between the three available types.

- **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 is a “Passive EQ” type, with all the very musical response you’d expect from a vintage piece, but without the typical signal-to-noise issues.
- **Peaking** – A peak filter cuts or boosts around a center frequency. When you boost or cut, neighboring frequencies are also affected somewhat, depending on the bandwidth, or Q, of the band. Bands 2, 3, and 4 are always set to this type.
- **Blocking** – The blocking filter is so named because it only allows frequencies above or below the cutoff frequency to pass through. Band 1 is selectable as a low-blocking type and Band 5 is selectable as a high-blocking type.
- **Shelving 2 –** This somewhat quirky type exhibits “overshoot” and a certain musical quality associated with passive equalization.

Parametric EQ Mix Parameters

The **Parametric EQ** block has a **MIX** page with **LEVEL**, **BALANCE,** and **BYPASS MODE** parameters.
See **Common Mix Parameters** on p.**128** for more information.

5.23 **Phaser [PHA]**

The Phase Shifter, or **Phaser**, works by cascading a series of all-pass filters and then mixing the processed signal with the input. This causes certain frequencies to be canceled or reinforced, creating notches and peaks. When phase is shifted using a low-frequency oscillator (LFO), these peaks and notches sweep up and down the frequency range to create the phaser's distinct, hollow, swooshing sound.

The phaser in the Axe-Fx II is extremely powerful. It allows two to 12 stages to be cascaded with positive or negative feedback and a flexible, stereo LFO. It also offers a special mode to recreate the classic 'vibe effects with astonishing accuracy.

XY The **Phaser** block supports X/Y Switching. See p. **36** for more information.

Each Axe-Fx II preset can use two fully independent **Phaser** blocks.

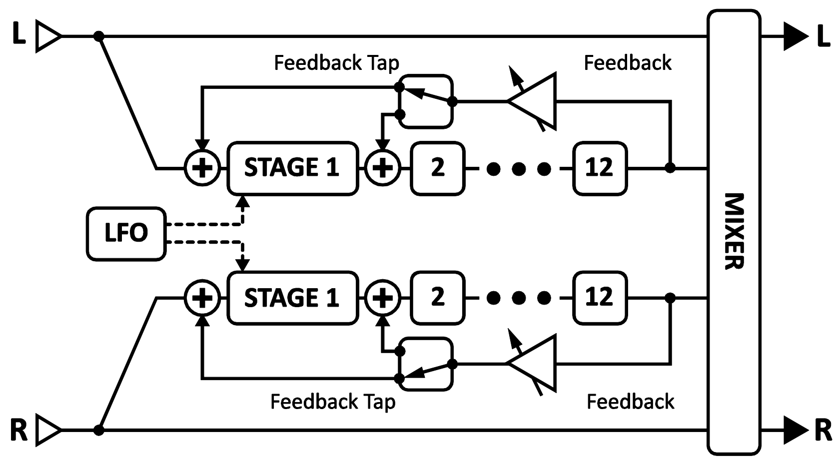


Figure 5-26 – The Phaser Block

Basic Parameters

TYPE – This sets other Phaser 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 Global LFO1 (p. **141**). When **RATE** is shown in parenthesis, it is being set automatically by the **TEMPO** parameter (see below). Set the **TEMPO** to "NONE" for manual control.

DEPTH – Sets the depth of the LFO that controls the "sweep." Set higher for more dramatic phasing effects.

FDBK – Feedback, also known as "regeneration" or "resonance," controls how pronounced the peaks and notches are. This is largely responsible for the iconic sound we associate with a Phaser.

TONE – A simple tone control for the Phaser effect (wet only).

FREQ – Sets the start frequency of the first stage filter. This, in combination with the depth control, controls the sweep range of the notches. This parameter is repeated as "START FREQ" on the Advanced page. (**FREQUENCY** is not used in the "BARBERPOLE" type.)

TEMPO – Synchronizes the rate of the Phaser LFO in rhythmic relation to the global tempo. For example, if the global tempo is 120 BPM, and **TEMPO** is set to a "1/4," the LFO rate will be 2 Hz (120 BPM / 60 seconds = 2). To ignore the global tempo set the tempo control to NONE. (Tempo is not used in BARBERPOLE type.)

Advanced Parameters

(All BASIC controls of the Phaser except **TYPE** are duplicated on the ADVANCED page for convenience).

ORDER – Sets the number of phase shifting circuits—or "stages"—in increments of two. Different settings have distinctly different sound qualities. For a more "pronounced" effect, increase the order.

LFO TYPE – Selects the "shape" of the LFO, determining how the sweep changes over time. SINE or TRIANGLE shapes will emulate classic phaser sounds. SAW shapes provide rising or falling effects, and Exponential/Logarithmic "shapes" create a more extreme "pulsing" effect.

LFO PHASE – Sets the phase difference of the right waveform of the phaser LFO. Values greater than 0° produce stereo phasing with 180°, reproducing the legendary "reverse sync" setting of "the world's largest phase shifter" (which incidentally employed six stages per side in this mode.)

*See section **16.7** on p.**174** for more information on LFO waveform shapes and phase.*

FREQ SPAN – Sets the span of the filters. Higher values separate the resulting notches by greater amounts.

VIBE MODE – Technically, this parameter sets the all-pass frequency spacing. But it would probably suffice to say that if Jimi, Robin, and David had a favorite Axe-Fx II phaser setting, this would be it. The **TYPE** parameter on the BASIC page can be used to get dial in classic 'vibe sounds instantly, or you can manually turn this switch ON to experiment with your own settings.

BULB BIAS – Allows you to control the "quiescent" current of the virtual light bulb used in Vibe Mode. Varying this parameter controls how "lumpy" the frequency sweep behaves. Unlike a real 'vibe, the Axe-Fx II compensates so that the center frequency doesn't change with the bias, allowing easier control of the sweep range. This parameter has no effect if Vibe Mode is switched OFF.

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" for **FEEDBACK TAP.**)

Phaser Mix Parameters

The **Phaser** block has a **MIX** page with **LEVEL**, **BALANCE** and **BYPASS MODE** and **GLOBAL MIX** parameters.
See **Common Mix Parameters** on p. **128** for more details on these.

5.24 **Pitch Shifter [PIT]**

Fractal Audio's Intelligent Maximum-Likelihood Adaptive Real-Time (IMART) technology provides superb mono or polyphonic pitch shifting. Pitch shift technology creates an incredible range of effects, from shimmering, detune-based chorus to one-guitar orchestrations of complex harmony, to dive-bombing whammy pedals, and beyond. The Axe-Fx II **Pitch Shifter** gives you all of these sounds and many more, with the following modes of operation:

- **Detune** – Creates chorus sounds with up to two detuned copies of the original signal.
- **Fixed Harmony** – Shifts two voices by a constant amount.
- **Intelligent Harmony** – Shifts two voices to another note in the selected key/scale.
- **Octave Divider** – Simulates the octave down effects of classic analog stomp effects.
- **Classic Whammy** – Shifts note(s) up and/or down 1-2 octaves with a control that can be assigned to a pedal or other controller.
- **Advanced Whammy** – Extends the Classic Whammy with custom range within +/- 2 octaves.
- **Crystals** – Creates exotic "crystal" shifting with long splice times and optional reverse.
- **Arpeggiator** – Shifts the pitch with a 16-step sequencer to create arpeggios or phrases from single notes.
- **Custom Shifter** – Employs "User Scales" for totally custom intelligent shifting.
- **Auto Pitch** – Turn your guitar or voice into Cher or T-Pain. Actually, please don't…we've *removed* the Axe-Fx Ultra's "Auto Pitch" algorithm from the Axe-Fx II ;)

Each Axe-Fx II preset can use two fully independent **Pitch Shifter** blocks.

XY The **Pitch** block supports X/Y Switching. See p. **36** for more information.

Common Parameters

The first menu page has several common parameters for the pitch block.

TYPE – Sets the sub-algorithm to use.

INGAIN – Sets the input level to the block for "Aux Send" type control even when the block is in series.

LOCUT FREQ, HICUT FREQ – Sets the cutoff frequency of hi-and lowpass filters at the output of the pitch shifter(s). Note that when the "OCTAVE DIV" type is selected, these controls have no effect.

PITCH SOURCE – The Pitch Shifter allows you to select a source for the shifter to use when making pitch calculations.

- GLOBAL – In this mode, the pitch information comes from the global pitch detector connected directly to the main inputs (L+R summed). The signal into this detector is unaffected by other effects in the current preset and is usually the best choice for single note usage.
- LOCAL – In this mode the 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.

Master Parameters

Several of the Pitch Shift types include MASTER parameters, detailed below.

MASTER PITCH – Scales the SHIFT setting for each of the voices. For example, with **VOICE 1 SHIFT** at "+12," **VOICE 2 SHIFT** set to "-12," and **MASTER PITCH** set to "50%," the shifts will be heard as **VOICE 1: +6, VOICE 2: -6.**

MASTER DELAY – Scales all delay times in the block.

MASTER FEEDBACK – Scales all feedback controls in the block.

MASTER PAN – The panning of each voice is multiplied by this value. A value of 100% will result in each voice being panned as set in the individual pan controls. A value of 0% will result in both voices being panned to center. A value of -100% will reverse the position of the voices. You can use a modifier on this parameter to move the voices around the stereo field in real time.

MASTER LEVEL – Multiplies the level values by this amount.

Pitch Source, Track, and Adjust

PITCH TRACK – Allows you to select from different styles of pitch shift tracking. When set to "MONO" the Pitch Shifter will track the pitch of incoming single notes note and adjust techniques accordingly for the best performance at any moment. "POLY" is best for shifting chords and lower amounts of shift. "OFF" causes the shifter to use fixed shifting techniques. With pitch tracking OFF, the sound can waver or flutter depending upon the note(s) played, but this effect can be desirable when simulating certain gear.

TRACK ADJ – This control allows fine-tuning of the pitch shifter "splice length." For large shifts, this control can help improve the quality of the shifted note.

Pitch Shifter Mix Parameters

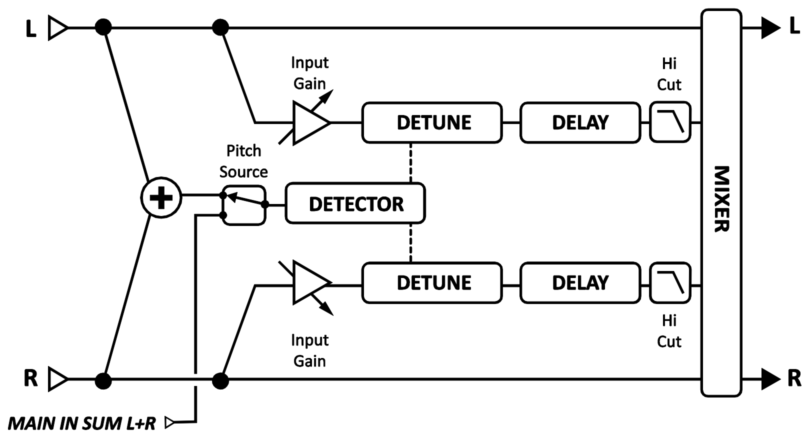
All **Pitch Shifter** types share a common MIX page with **MIX**, **LEVEL**, **BALANCE**, **BYPASS MODE,** and **GLOBAL MIX** parameters. See **Common Mix Parameters** on p. **128** for more information on these controls.

Details about the different Pitch algorithms appear on the following pages.

5.24.1 Detune

The **Detune** sub-algorithm creates two voices that are detuned between -50 and +50 cents (one quarter step) from the input. This mode is useful for creating double-tracked sounds or chorus-like effects.

Figure 5-27 – The Detune Pitch Shifter Type



INPUT MODE – Determines whether the inputs are stereo or summed.

VOICE 1 DETUNE, VOICE 2 DETUNE – Sets the detune amount for each voice. Attach an LFO to create modulating detune chorus effects.

VOICE 1 LEVEL, VOICE 2 LEVEL – Sets the volume level of the selected voice.

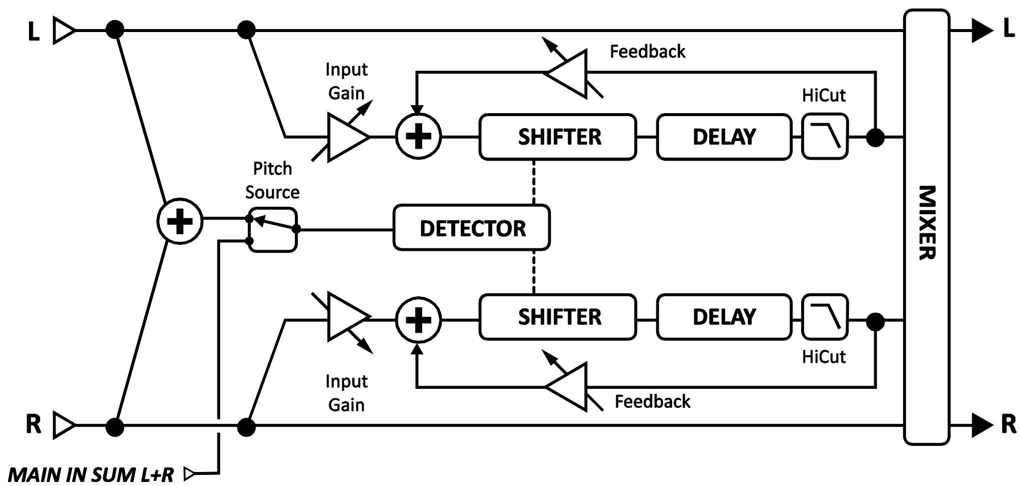
VOICE 1 PAN, VOICE 2 PAN – Sets the panning of the selected voice.

VOICE 1 DELAY, VOICE 2 DELAY – Sets the delay time of the selected voice.

5.24.2 Fixed Harmony

The **Fixed Harmony** mode creates two voices at fixed intervals from the note played, with the possibility of using feedback and/or delay to create cascades of upward/downward shifting.

Figure 5-28 – The Fixed Harmony Pitch Shifter Type



INPUT MODE – Specifies whether the incoming signal should be processed in stereo (as shown in the block diagram above) or summed to mono and then sent to both voices.

VOICE1 DETUNE, VOICE 2 DETUNE – Sets the detune amount of the voice in a range of +/- 50 cents.

VOICE1 SHIFT, VOICE2 SHIFT – Sets the shift amount of the voice in a range of +/- 12 half-steps.

VOICE1 LEVEL, VOICE2 LEVEL – Sets the volume level of the voice.

VOICE1 PAN, VOICE2 PAN – Sets the panning of the voice.

VOICE1 DELAY, VOICE2 DELAY – Sets the delay time of the voice. When a delay time is shown in parenthesis, it is being set automatically by a **TEMPO** parameter (see below). Set the corresponding **TEMPO** to "NONE" for manual control.

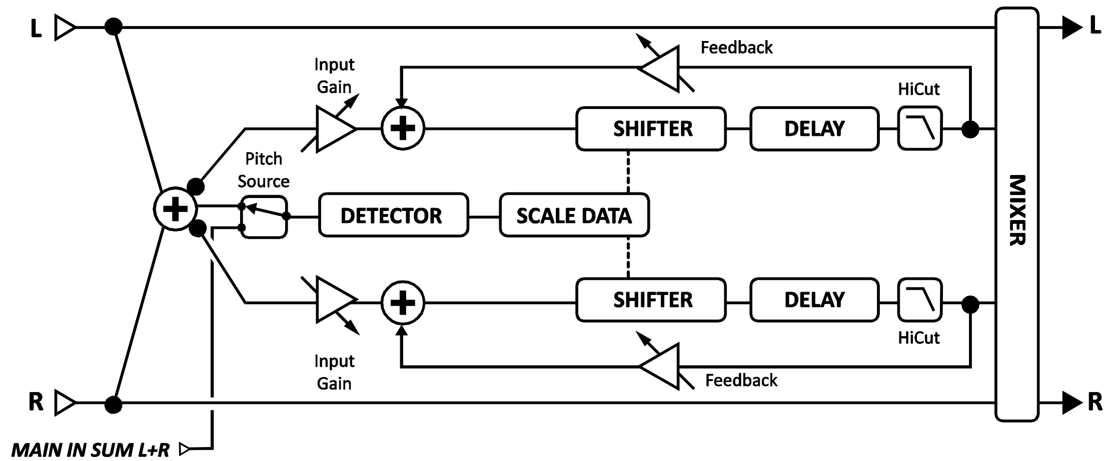
VOICE1 DLYTEMPO, VOICE2 DLYTEMPO – Locks the corresponding TIME parameter in rhythmic relation to the global tempo. For example, if the global tempo is 120 BPM, and **TEMPO** is set to "1/4" (one echo per beat), time will be 500 ms. To ignore the global tempo, set to "NONE."

VOICE1 FEEDBACK, VOICE 2 FEEDBACK – Sets how much of the output signal is returned to the input for "shift-upon-shift" effects. Using both feedback and delay (above) results in cool-sounding cascades.

5.24.3 Intelligent Harmony

The **Intelligent Harmony** Pitch Shifter block type creates harmonies within a certain musical key and scale. The **SCALES** parameter taps a small onboard database of music theory data to adjust shift amounts based on which note you play. This makes it possible, for instance, to harmonize a melody around a key center without the shifter playing any "wrong" notes. (With due respect, YOU still need to play the "right" notes for this to work.) Between the actual note played and the two shifted voices, three-note chords can be formed. Crank out some Maiden, Boston, or Thin Lizzy all by yourself!

Figure 5-29 – The Intelligent Harmony Pitch Shifter Type



VOICE1 DETUNE, VOICE 2 DETUNE – Sets the detune amount of each voice within +/- 50 cents.

KEY – Sets the key that the harmony will be in.

LEARN – While learn is ON, the **KEY** parameter will automatically change to whatever single note you play. Assign to a footswitch for modulations right in the middle of a phrase! Turn this OFF again to return to normal harmonizer function.

SCALE – Sets the scale or mode into which notes will be shifted.

TRACK MODE – Sets how the harmony will track the input. SMOOTH allows the shifted note to follow bends and vibrato in the input. STEPPED locks the harmony to the nearest chromatic tone.

GLIDE TIME – Sets the rate at which the harmony shifts from the one pitch to another as different notes are played.

TRACKING – Allows fine-tuning the pitch shifter "splice length." For large shifts, this control can help improve the quality of the shifted note. TRACKING is the same as "TRACK ADJ" in other Pitch Shifter types.

VOICE1 HARMONY, VOICE 2 HARMONY — Sets the scale degree that each voice will sound.

❗ It is important to recognize that this is NOT a "semitones" control 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. Compare IONIAN (MAJ) to AEOLIAN (MIN) in this way and you'll get the idea. 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.

VOICE1 DELAY, VOICE 2 DELAY – Sets the delay time of the voice in milliseconds.

VOICE1 DLYTEMPO, VOICE 2 DLYTEMPO – Locks the delay time to the global tempo. For example, if the global tempo is 120 BPM, and **TEMPO** is set to a quarter note "1/4," then the delay time will be 500 ms. To ignore the global tempo, set the tempo control to NONE.

Custom Scale

The Intelligent Harmonizer allows you to create a custom scale and save it with the preset. To use this feature, set the **SCALE** parameter (above) to CUSTOM, set the number of **CUSTOM NOTES**, and set up your scales notes in any key based on an arbitrary **TONIC**. As with other scales, the actual **KEY** setting (above) determines how your custom scale will be transposed and compared to the note played for creating custom harmonies.

In comparison to the Custom Shifter (**5.24.9**, below), in which any input note of the chromatic scale can be shifted +/- 24 semitones, the Intelligent Harmonizer requires that your creations contain from four to eight notes (including the tonic), and each scale degree must be at least ½ step higher than the previous.

CUSTOM NOTES – Sets the number of notes when using a custom scale. Custom scales can have between four and eight notes.

TONIC – This parameter has no effect on how the scale will sound in actual use but instead serves as an aid so you can see an example of your custom scale transposed to any arbitrary key. Change this value and other scale degrees will automatically update. It is the **KEY** parameter (above) which actually transposes your custom scale for use during actual performance.

NOTE 1,2,3…8 – These are the notes of your custom scale relative to the TONIC. Set these to define the scale degrees.

Scale Formulas

Here is a set of scale formulas for those scales used in the pitch shifter's Intelligent Harmonizer and Arpeggio modes. If scale names look slightly different from what you've learned, remember this joke: "Q: How many guys does it take to name a Jazz 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 TYPE	DEGREES/FORMULA							
	1	2	3	4	5	6	7	8
IONIAN (MAJOR)	1	2	3	4	5	6	7	
DORIAN	1	2	b3	4	5	6	b7	
PHRYGIAN	1	b2	b3	4	5	b6	b7	
LYDIAN	1	2	3	#4	5	6	7	
MIXOLYDIAN	1	2	3	4	5	6	b7	
AEOLIAN (MINOR)	1	2	b3	4	5	b6	b7	
LOCRIAN	1	b2	b3	4	b5	b6	b7	
MELODIC MINOR	1	2	b3	4	5	6	7	
HARMONIC MINOR	1	2	b3	4	5	b6	7	
DIMINISHED (whole-half)	1	2	b3	4	b5	b6	6	7
WHOLE TONE	1	2	3	b5	#5	b7		
DOMINANT 7 (aka dim half-whole)	1	b2	#2	3	#4	5	6	b7
DIMINISHED/WHOLETONE	1	b2	#2	3	#4	#5	b7	
PENTATONIC MAJOR	1	2	3	5	6			
PENTATONIC MINOR	1	b3	4	5	b7			
BLUES	1	b3	4	b5	5	b7		
CHROMATIC	All 12 tones							

5.24.4 Classic Whammy

The whammy, first introduced in 1991, is a relative newcomer to the field of guitar effects. The **Classic Whammy** brings all of the expected sounds to the Axe-Fx II. Its **CONTROL** parameter is designed to be operated remotely using a modifier (p. **136**), typically assigned to some source controlled by a pedal. In comparison to the Advanced Whammy (p. **108**) this type has only a few combinations of octave up/octave down.

Parameters

MODE – Selects the Whammy mode:

- UP 1 Octave
- DOWN 1 octave
- UP 2 Octaves
- DOWN 2 octaves
- UP/DOWN 1 Octaves
- UP/DOWN 2 octaves

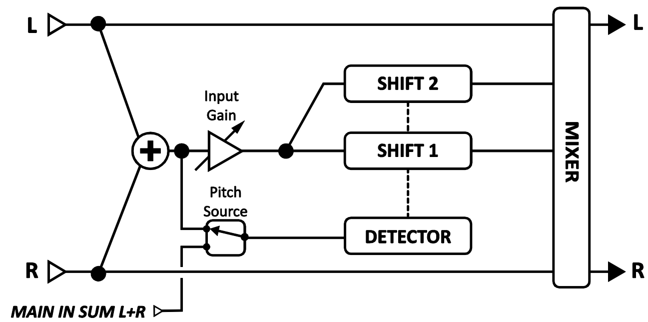


Figure 5-30 - The Classic Whammy Pitch Shifter Type

CONTROL – Adjusts the output pitch. Attach to a controller for dynamic control of the pitch. See the Wah pedal tutorial on p. **179** for more details on doing this.

The introduction to this section covers COMMON, MASTER, TRACKING, and MIX parameters.

5.24.5 Octave Divider

The **Octave Divider** simulates the classic analog effect and works by actually turning the input into a square wave and then dividing the signal by two using flip-flops. Like the classic effect, this effect only works on single notes and works best on notes above the fifth fret. Experiment with pickup selection and effect placement to achieve the best results.

Shift 1 and Shift 2 are fixed at one and two octaves down.

LVL1, LVL2 – Sets the volume levels of the octaves.

PAN1, PAN2 – Sets the panning of the octaves.

The introduction to this section covers COMMON, MASTER, TRACKING, and MIX parameters.

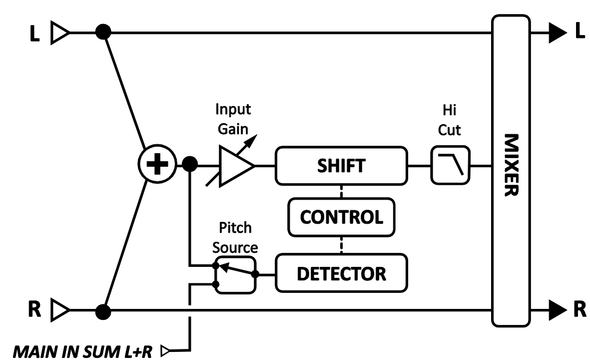
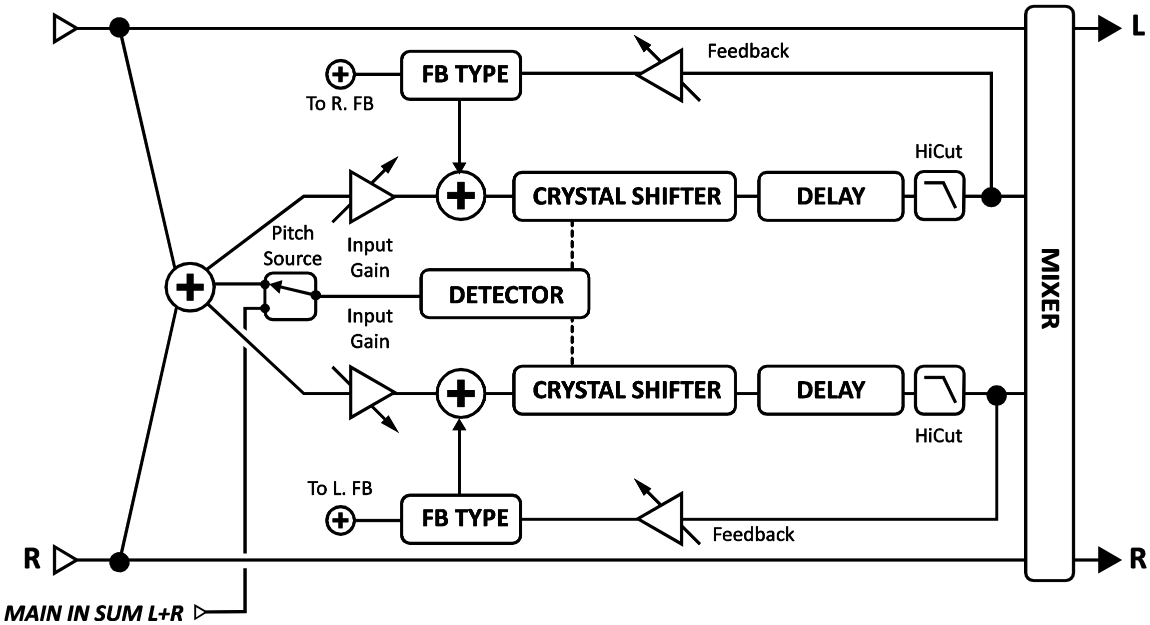


Figure 5-31 - The Octave Divider Pitch Shifter Type

5.24.6 Crystals

The **Crystals** Pitch Shifter is similar to the **Fixed Harmony** mode (p. **103**) but is designed for special effects. It features much longer possible “splice” times inside the shifter, reverse shifting, and a flexible feedback architecture.

Figure 5-32 – The Crystals Pitch Shifter Type



VOICE1 DETUNE, VOICE 2 DETUNE – Sets the detune amount of the voice in a range of +/- 50 cents.

VOICE1 SHIFT, VOICE2 SHIFT – Sets the shift amount of the voice in a range of +/- 12 half-steps.

VOICE1 LEVEL, VOICE2 LEVEL – Sets the volume level of the voice.

VOICE1 PAN, VOICE2 PAN – Sets the panning of the voice.

VOICE1 DELAY, VOICE2 DELAY – Sets the delay time of the voice. When a delay time is shown in parenthesis, it is being set automatically by a **TEMPO** parameter (see below). Set the corresponding **TEMPO** to “NONE” for manual control.

VOICE1 DLYTEMPO, VOICE2 DLYTEMPO – Locks the corresponding TIME parameter in rhythmic relation to the global tempo.

VOICE1 FEEDBACK, VOICE 2 FEEDBACK – Sets the feedback of the voice to the input. By delaying and feeding a voice back, strange pitch effects can be created as the note is shifted again and again.

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.

For sake of explanation, the Crystal algorithm “splicing” parameters detailed below are not listed in the order in which they appear on the display of the Axe-Fx II.

VOICE1 SPLICE, VOICE2 SPLICE – 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.

VOICE1 SPLTEMPO, VOICE1 SPLTEMPO – Sets splice time in rhythmic relation to the global tempo.

DIRECTION – Determines whether the processed granules are played back forwards or backwards. 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 (“Axɘ-ꟻx”). The length of the snippets depends on the **SPLICE** settings (above).

CROSSFADE – Sets the amount of overlap in the audio granules. A low setting will make the edges more discrete, while a high setting will tend to blend splices together.

Example: Imagine that the shaded rectangle below represents an audio signal, like a piece of reel-to-reel tape.

The signal is divided into **SPLICE**-parameter-sized “granules.”

Setting **DIRECTION** to “REVERSE” does this:

CROSSFADE applies overlaps with short fade-ins/fade-outs to blend spliced edges:

Figure 5-33 – Splicing

5.24.7 Advanced Whammy

The **Advanced Whammy** is identical to the **Classic Whammy** (p. **106**) except that its shift range may be set to any custom number of semitones within a range of +/- 24.

Parameters

The parameters for the Advanced Whammy are the same as the Classic with the following differences:

START – Sets the start pitch shift amount in semitones. This is the amount of pitch shift applied when the CTRL knob is in the minimum position.

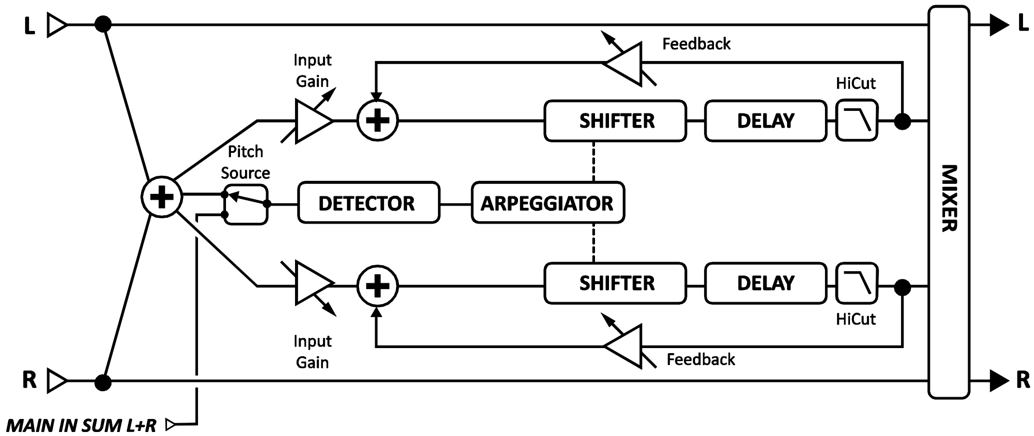
STOP – Sets the stop pitch shift amount in semitones. This is the amount of pitch shift applied when the CTRL knob is in the maximum position.

The introduction to this section covers COMMON, MASTER, TRACKING, and MIX parameters.

5.24.8 Arpeggiator

The **Arpeggiator** uses a 32-step sequencer to control the shift amount of a harmonizer so that complex arpeggio patterns can be created from a single note. Arpeggios "intelligently" transpose inside the 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).

Figure 5-34 – The Arpeggiator Pitch Shifter Type



Parameters

The Arpeggiator parameters are identical to those of the **Intelligent Harmony** shifter (p. **104**) except as noted below:

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. **143**) to re-trigger the sequence on each new note.

SCALE – Sets the type of scale, or scale mode, that the notes of the arpeggio will be drawn from. Examples are Ionian (major), Aeolian (minor), whole tone, etc. Set to CUSTOM to use custom scale tones (see Intelligent Harmony above for details).

KEY – Sets the key that the harmony will be in.

STAGES – Sets the number of stages in the pitch sequencer.

REPEATS – Sets the number of times the sequence will repeat once triggered. Set to INFINITE to loop forever.

TEMPO – Sets the duration of each sequencer step as a rhythmic value in relation to the global tempo.

GLIDE TIME – Sets the rate at which the harmony shifts from one pitch to another as the arpeggio shifts.

AMPLITUDE SHAPE, PAN SHAPE – Specifies how the volume or pan changes as the arpeggiator plays through one cycle. See Megatap Delay (p. **82**) for more detail on shapes and alpha.

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 – Tricky but ideally implemented for maximum flexibility, this parameter sets 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 place it in the key of C for comfort, with a scale type of IONIAN (MAJOR). When we play a C, the arpeggio will be heard as 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')

☞ Remember that the notes of the arpeggios and the steps required to arrive at them are drawn only from the current key/scale. Scales with more or fewer than seven notes in one octave (diminished, whole tone, custom, etc.) can require some mental math, and it's 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 the specified number of semitones.

The introduction to this section covers COMMON, MASTER, TRACKING, and MIX parameters.

5.24.9 Custom Shifter

The **Custom Shifter** is identical to the **Intelligent Harmony** Pitch Shifter (p. **104**) except that it uses the custom scales tables stored in the global memory. See section **8.3** on p. **147** for more information on setting up global scales.

VOICE 1 SCALE, VOICE 2 SCALE – Selects the custom scale to use for each voice.

KEY – This transposes both custom scales to the desired key. Internally the Axe-Fx II assumes all Custom Scales have a root of A so this shift is relative to that note. For example, if your custom scale was in A major and you wanted to perform in G major you'd set KEY to G. If your custom scale was in B minor and you wanted to play in E minor you'd set KEY to "D", since E is a 4th above B and D is a 4th above A.

5.25 **Quad Chorus [QCH]**

The **Quad Chorus** was designed to enable sounds beyond those of the legendary "Tri-Stereo" chorus, a fixture of 80s session player clean sound. It takes time to set up, but your efforts will be rewarded with incredibly lush and liquid chorus sounds. It is a four-voice chorus with a powerfully complex modulation generator.

Each Axe-Fx II preset can use two fully independent **Quad Chorus** blocks.

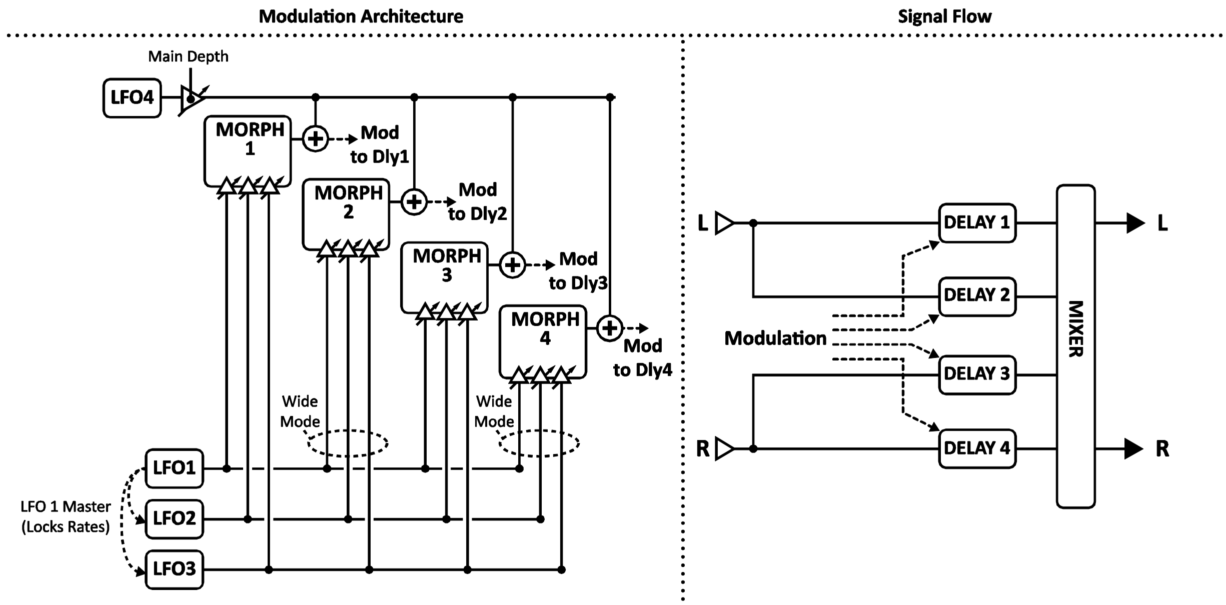


Figure 5-35 – The Quad Chorus Block

The parameters of the Quad Chorus are divided across pages for **Master**, **Chorus 1–4**, **Advanced**, and **Mix**.

Master Parameters

TIME – Scales the times set individually for each of the chorus voices.

RATE – Sets the master rate against which the rates of LFOs 1–4 are set as multiples.

DEPTH – Controls the overall depth of all chorus voices.

FDBK – Feedback for all four voices is controlled solely by this parameter. Classic chorus would typically require a setting of 0%, while flanging is possible with higher settings (though extreme settings can cause runaway oscillation).

INPUT MODE – The STEREO mode is shown above: left input feeds voices 1 and 2; right input feeds voices 3 and 4. In MONO mode, the two inputs are summed into all voices.

Chorus Unit Parameters

Each chorus unit has an identical set of parameters.

TIME – Sets the minimum time delay of the selected chorus voice. All modulation is positive/unipolar.

LEVEL – Sets the output level of the selected chorus voice.

PAN – Sets the panning of the selected chorus voice in the stereo field.

DEPTH – Sets the modulation depth for the selected chorus voice's **MORPH** blend of **LFOs 1/2/3**.
The depth of **LFO 4** is set for all voices simultaneously via the **MAIN DEPTH** parameter on the advanced page.

LFO MORPH – This parameter taps any one, or a blend of any two, of LFOs 1, 2, and 3 to use as a modulation source for the current voice. The scale below shows how the percentage setting selects between the LFOs. A value of 0% taps only LFO1, while 75% would be an even blend of LFO2 and LFO3.

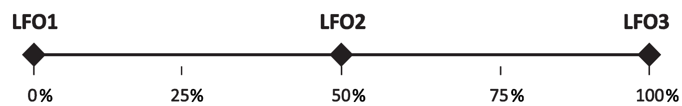


Figure 5-36 – Quad Chorus LFO Morph Scale

Advanced Parameters

The "MASTER" parameters on the Advanced Page are duplicates of items on the MASTER page (detailed above).

WIDE MODE – When this in set to "ON," the modulation to voices 2 and 4 from their LFO MORPH outputs is inverted to widen the stereo effect.

MAIN DEPTH – Controls the depth of LFO4, which modulates all four chorus voices.

MAIN PHASE – Sets the phase difference for the main LFO (LFO4) output to chorus voices 2 and 4.

LFO1 MASTER – If this is set to "ON," the controls for LFO 1 also control the rates.

LFO TYPE 1–4 – Selects the type of LFO for each LFO.

LFO RATE MULT 1–4 – Sets the rate of each LFO as a multiple of the **MASTER RATE**.
Note: **MASTER RATE** appears as "RATE" on the MASTER page and as "MASTER RATE" on the ADVANCED page.

Quad Chorus Mix Parameters

The **Quad Chorus** block has a **MIX** page with **MIX, LEVEL**, **BALANCE**, **BYPASS MODE,** and **GLOBAL MIX** parameters.

See **Common Mix Parameters** on p. **128** for more details on these.

5.26 Resonator [RES]

The **Resonator** consists of four resonant comb filters in parallel. By tuning the comb filters, a metallic or resonant timbre can be achieved from normally non-musical signals. The Resonator works best on transient signals like speech or percussion but can also be used to add unique character to musical inputs. In series with each comb filter is a bandpass filter tuned to the same frequency. These filters can be placed before or after the comb filters. They are shown in the "after" position in the diagram to the right.

Each Axe-Fx II preset can use two fully independent Resonator blocks.

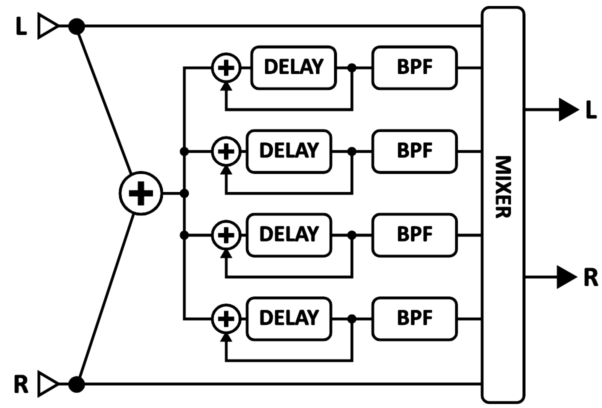


Figure 5-37 — The Resonator Block, Shown in its MONO INPUT MODE.

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.
- **CHORDAL**: Resonator/filter frequencies are set automatically based on a setting for **CHORD TYPE**. A base FREQUENCY scales all four voices to any pitch/key.

INGAIN – At high feedback levels, overload can easily occur. This scales the level into the effect for control.

MASTER FREQ/FREQUENCY – In MANUAL MODE, **MASTER FREQUENCY** scales the frequencies set manually for the four resonators/filters. In CHORDAL 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, 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 selected filter.

FEEDBACK 1–4 – Sets the resonance of the selected filter by varying the feedback.

FILTER LOC 1–4 – Selects the position of the bandpass filter in relation to the resonator.

FILTER Q 1–4 – Sets the Q of the selected bandpass filter.

LEVEL 1–4 – Sets the output level of the selected filter.

PAN 1–4 – Sets the panning of the selected filter.

Resonator Mix Parameters

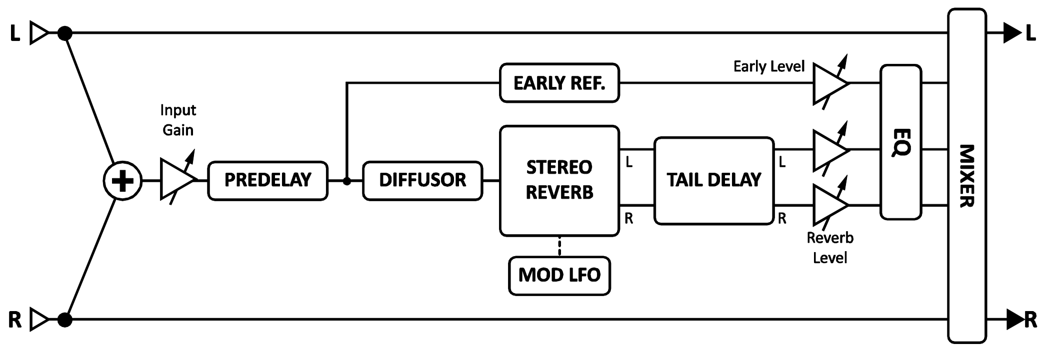
The **Resonator** block has a **MIX** page with **MIX, LEVEL**, **BALANCE**, **BYPASS MODE,** and **GLOBAL MIX** parameters. See **Common Mix Parameters** on p. **128** for more details on these.

5.27 **Reverb [REV]**

Aside from distortion, no effect has been more important to the electric guitar than reverb. Since the dawn of guitar amplification, guitarists in small spaces yearned for the sounds their amps made in a hall or large room. Early simulators incorporated metal springs and plates, but as with delay processing, reverberation effects were truly revolutionized by digital technology. The Axe-Fx II is one of the finest you are likely to have heard: realistic, lush, and dense, with the ability to emulate real spaces, vintage springs, classic digital effects, and more.

Each Axe-Fx II preset can use two fully independent **Reverb** blocks.

Figure 5-38 – The Reverb Block



The **Reverb** block supports X/Y Switching. See p. **36** for more information.

Basic Parameters

TYPE – Selects the reverb type. There 40+ variations of type based on a smaller number of core algorithms:

TYPE	NOTES
Room	Simulates an actual room. This is the type to use when you want the most natural, realistic reverb. Also great on vocals and percussion.
Hall	Similar to the Room a little less smooth with some response peaks and a unique character. Use this when you want your reverb to stand out a little.
Chamber	Simulates the sound of large, boxy chamber. Useful for a bright, resonant reverb.
Plate	Simulates the sound of a vintage plate characterized by smooth yet bright sound.
Cathedral	Surround your tone with heavenly reverb in this incredible simulation of a grand space.
Spring	Simulates the classic electro-mechanical reverb effect.
Cavern	Carl's cavern may be bad, but this one is wicked. A vast, cavernous space.
Studio	Models a classic digital studio reverb.
Quarry	Don't take this fantastic reverb for granite.

QUALITY – There are two "Quality" options in the Reverb block: Normal and High. High quality uses provides world-class reverberation algorithms and the utmost in sound quality. Normal quality reduces CPU usage considerably while still providing excellent sound quality suitable for most guitar reverb needs.

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).

PRE DELAY – 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.

MIX and **LEVEL** are duplicated from the MIX page.

Advanced Parameters

TYPE, TIME – Duplicated from the BASIC page (p.114).

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 pad sounds and drone notes/chords. You can assign this easily to a modifier for foot control!

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.

PRE DELAY – This is a duplicate of the Predelay parameter on the reverb's "Basic page.

EARLY LEVEL – Adjusts the relative level of the early reflections (has no effect for the "Spring "types.)

LATE LEVEL – Adjusts the relative volume level of the reverb tail.

HF TIME, LF TIME, LF XOVER – The Axe-Fx reverb algorithm is actually multi-band, allowing for very natural effects. These parameters control decay times for the bands, plus the crossover frequency.

EARLY DIFFUSION: This sets the amount of diffusion in the early reflections. Higher values result in fuzzier and less distinct echoes. Lower values result in sharp, distinct reflections.

EARLY DIFF TIME: This scales the delay time of the early reflections diffusers. Adjust this control to suit the size and character of the simulated environment.

EARLY DECAY: This parameter controls the decay rate of the early reflections. (Higher = faster decay.)

LATE DIFFUSION – Controls the amount of diffusion applied to the signal before it hits the main reverb generator. Greater diffusion reduces "distinctness" and increases the "density" of the tail.

LATE DIFF TIME – Controls the length of the input diffusor. Lower values simulate a small diffusion space. Higher values simulate a large space.

LATE INPUT MIX. In High Quality reverb mode, this parameter mixes the early reflections in with the (possibly diffused) input to be sent to the late reverb ("tail") generator. It uses a proprietary decorrelation technique which eliminates the metallic qualities associated with the typical diffusers used in other products.

WALL DIFFUSION – Controls how quickly the reverb tail's density builds. Lower values cause discrete echoes to be last longer. Higher values cause the echo density to build rapidly for a smoother effect.

ECHO DENSITY – Controls the initial 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 make the individual echoes more apparent, as will lower values of **LATE DIFFUSION**. For legato sounds, a low DENSITY value may be more suitable. For short, percussive sounds, a higher value may be more desirable as the reverb tail will be smoother. Echo Density has a pronounced effect on CPU Utilization and should be one of the parameters you check first when trying to lower the overall load.

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 re-set the mix as desired.

STEREO WIDTH – Determines the overall stereo separation of the reverb. Set to 0% for mono output.

MIC SPACING – Sets the stereo width of early reflections by simulating moving mics in the virtual space.

The Advanced page contains the spring controls, which are only active when using one of the Spring reverb types.

NUMBER SPRINGS – When the **TYPE** is set to "SPRING," this sets the number of springs in the simulation.

SPRING TONE – When the **TYPE** is set to "SPRING", this changes the character, emphasizing different tonal aspects of the simulation. Lower values create a darker tone.

SPRING DRIVE – This simulates overdriving the circuit of the reverb when the **TYPE** is set to "SPRING."

The Advanced page also contains the ducking controls. Ducking causes the "wet" level to be lowered automatically when you play (or play louder—depending on the threshold.)

DUCKER ATTEN – Attenuation sets the amount that the wet signal will duck (decrease). A setting of 12 dB, for example, will decrease the reverb level by 12 dB when the signal ducks.

DUCKER THRSHLD – Sets the trigger level of the ducker (lower = ducking occurs more easily). If the signal at the input of the reverb block is greater than the threshold, ducking will occur.

DUCKER RELEASE TIME – Sets how long it takes for the wet signal to return to normal after the ducker stops being triggered. A short (low) value causes a faster return to normal level. A slow (high) value causes a gradual return.

Reverb EQ Parameters

The reverb features a powerful equalizer with high and low-pass, plus two peaking-type filters.

LOW CUT – Sets the frequency of the low-cut filter. Increase for thinner sounds.

HIGH CUT– Sets the frequency of the high-cut filter. Decrease for darker sounds.

FREQ 1, GAIN 1, Q 1 – Controls the first peaking filter. Select the frequency and amount to boost or cut. Set Q to determine the width of the effect.

FREQ 2, GAIN 2, Q 2 – Controls for the second peaking filter.

The **Reverb** block **MIX** page has **INPUT GAIN, MIX, LEVEL**, **BALANCE**, **BYPASS MODE,** and **GLOBAL MIX** parameters.

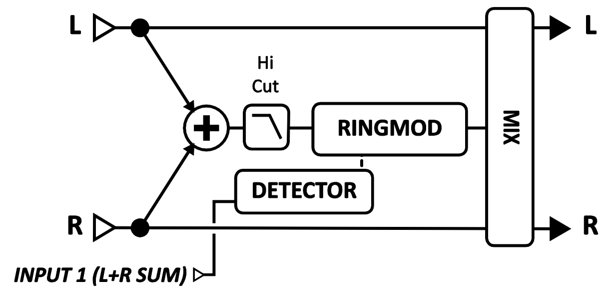
See **Common Mix Parameters** on p. **128** for more details on these.

A Word on "Spillover"

The Axe-Fx II Reverb is capable of "spillover," which means that effect tails ring out when the effect is bypassed or when you change presets. For more on this subject, please see **Setting Up Spillover** on p. **181**.

5.28 **Ring Modulator [RNG]**

Figure 5-38 – The Ring Modulator Block



Commonly used on synth and fusion electric piano sounds (or to create Dalek/X-wing pilot voices), the **Ring Modulator** uses fast changes in amplitude to create subtones (or supertones?) harmonically unrelated to the input. The Axe-Fx II Ring Modulator 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.

Each Axe-Fx II preset can use two fully independent **Ring Modulator** blocks.

Parameters

FREQ – Sets the frequency of the oscillator.

FMULT – Sets the frequency multiplier for the oscillator. The actual oscillator frequency is the **FREQ** value multiplied by the **FMULT** value.

TRACK – When this is set to "ON," the frequency of the oscillator tracks the input pitch. The actual frequency is then the pitch multiplied by the **FMULT** value.

HICUT – Sets the cutoff frequency of a low-pass filter on the output.

Ring Modulator Mix Parameters

The **Ring Modulator** block has a **MIX** page with **MIX, LEVEL**, **BALANCE**, **BYPASS MODE,** and **GLOBAL MIX** parameters.

See **Common Mix Parameters** on p. **128** for more details on these.

5.29 Rotary Speaker [ROT]

A Hammond B3 Organ without a Leslie cabinet is like a BLT without lettuce and tomato. Guitar players also revel in 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 Axe-Fx II **Rotary Speaker** simulator reproduces all these classic sounds and offers more control. Also, it doesn't weigh 300 pounds or require four guys to move it up a flight of stairs. Isn't technology great?

XY The Rotary block supports X/Y Switching. See p. **36** for more information.

Each Axe-Fx II preset can use two fully independent **Rotary Speaker** blocks.

Parameters

RATE – Controls the rate at which the "drum and rotor" spin. Connect this to a controller for real-time control. When **RATE** is shown in parenthesis, it is being controlled by the tempo parameter (below). Set **TEMPO** to "NONE" for manual rate control.

TEMPO – Locks the rate to the global tempo. For example, if the global tempo is 120 BPM, and **TEMPO** is set to a "1/4," the LFO rate will be 2 Hz (120 BPM / 60 seconds = 2). To ignore the global tempo set the tempo control to NONE.

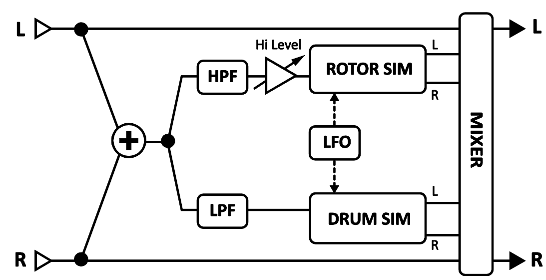


Figure 5-39 – The Rotary Speaker Block

LOW DEPTH – Sets the modulation depth of the drum. Higher settings provide a more pronounced throb.

HI DEPTH – Sets the modulation depth of the rotor. To simulate a rotating drum-only cabinet, reduce this.

HI LEVEL – Sets the output level of the rotor. Use this to balance the level between the drum and rotor.

ROTOR LENGTH – This parameter 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 rotation compared to the rotor (which always spins at the value set for **TEMPO**, above).

LOW TIME CONSTANT, HI TIME CONSTANT – Sets acceleration/deceleration rates of the drum/rotor.

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.

DRIVE – Give your virtual rotary speaker the grind of the classic power amp with this overdrive control.

The **Rotary Speaker** has **MIX, LEVEL**, **BALANCE,** and **BYPASS MODE** controls. See **Common Mix Parameters** on p. **128**.

5.30 Synth [SYN]

The Axe-Fx II **Synth** block contains a 3-voice monophonic synthesizer that can be used as a tone generator or to track the pitch of your playing for synth leads with a guitar or other instrument. Each voice has its own resonant filter and may be set to produce any of seven different oscillator waveforms. The global ADSRs or LFOs may be used to modulate a variety of functions.

Each Axe-Fx II preset can use two fully independent **Synth** blocks.

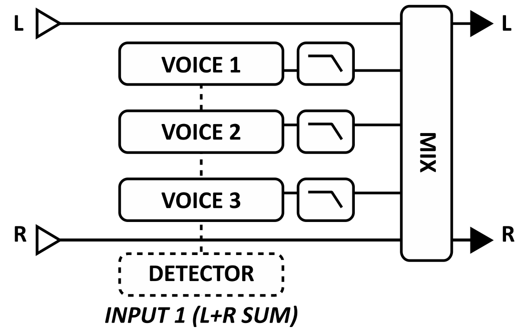


Figure 5-40 — The Synth Block

Parameters

Each synth has two voices with the following parameters:

TYPE – Sets the waveform to Sine, Triangle, Square, Sawtooth, Random, White Noise, Pink Noise, or OFF. (Setting a synth voice to OFF helps conserve CPU usage).

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.

FREQ – 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 – When using the TRIANGLE or SQUARE waveforms, this parameter controls the symmetry or pulse width of the waveform.

LEVEL – Controls the output level of the oscillator.

PAN – Controls the panning of the oscillator.

FILTER – Sets the cutoff frequency of a low-pass filter after the oscillator.

Q – Sets the Q or resonance of the post-oscillator filter.

ATTACK – Sets the attack time of the envelope follower on the input.

Synth Mix Parameters

The **Synth** block has a **MIX** page with **MIX, LEVEL**, **BALANCE**, **BYPASS MODE,** and **GLOBAL MIX** parameters.

See **Common Mix Parameters** on p. **128** for more details on these.

5.31 Tone Matching [TMA]

The **Tone Matching** block matches the sound of the Axe-Fx II to the sound of another amp, recording, or device. It does this by analyzing the difference between a "reference signal" and the sound of a starting point preset (the "local" signal). This process is covered by a separate mini-manual available on our web site at:

http://www.fractalaudio.com/downloads/manuals/axe-fx-2/Axe-Fx-II-Tone-Match-Manual.pdf

CAPTURE PAGE

This page allows you to Start and Stop the local and reference signals, and create the Match.

EXPORT PAGE

Tone Matching Data can be "Exported" directly to an onboard USER CAB memory.

PROCESS PAGE

REF SOURCE – This specifies the signal to be used for reference. For example, to reference the feed from a mic on a live amp, you'll typically use INPUT 2. For a computer audio source, use USB.

REF CHAN, LOCAL CHAN –Tone Matching operates in mono only. If the reference or local signal is stereo, use these parameters to determine whether left, right, left/right summed channel should be used.

REF SOLO – When set to "ON" the reference signal is sent directly to the block output and other block inputs are muted. If you assign a footswitch to this function, you can easily compare the reference and the result.

AVG TIME – Determines how long of a time window is used when analyzing the reference signal.

AMOUNT – Scales the Tone Match effect from full to flat, determining the magnitude of its effect.

MODE – Selects between OFFLINE and LIVE modes. Use LIVE mode when you use one guitar with a Y cable to generate both reference and local signals at the same time.

SMOOTHING – This reduces the prevalence of peaks and valleys in the IR to ease a "notchy" sound.

RESOLUTION – Normally, you'll run this in HIGH mode. To save CPU, change to LOW mode, which sacrifices 50% of the match resolution but can still sound great.

MONITOR PAGE

The monitor page allows you to monitor the difference between local and reference signals in real time. This can be used when matching real amps to identify differences due to manufacturing tolerances, etc.

Tone Matching Mix Parameters

The **PROCESS** page also has **LEVEL, BALANCE,** and **BYPASS MODE** parameters.
See **Common Mix Parameters** on p. **128** for more information.

5.32 Vocoder [VOC]

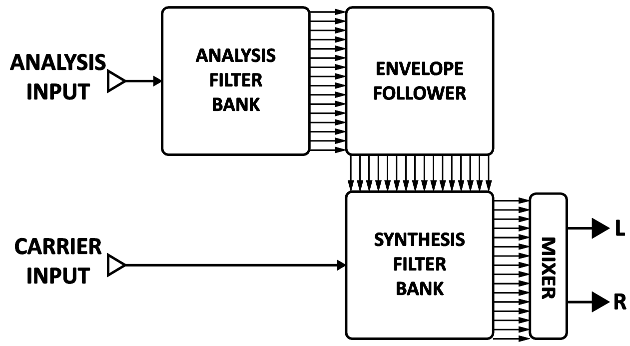


Figure 5-41 — The Vocoder Block

The Axe-Fx II has a digital recreation of the classic analog vocoder. 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.

The Axe-Fx II **Vocoder** block pays faithful homage to those early analog vocoders. Using a true constant-Q approach, it can be used to make your guitar "talk" or to make your voice sound like a robot [chicken]. When using the vocoder with your guitar or other instrument as the carrier (synthesis bank), resist the urge to sing along. Speaking in a monotone voice yields the best results.

Each Axe-Fx II preset can use one **Vocoder** block.

Parameters

INSEL – Selects the input to use for the synthesis (carrier) channel. This is the input to use for your guitar or other instrument. The other input is then the analysis channel and is typically used for vocal input.

BANDS – Selects the number of bands to use in the analysis and synthesis filter banks.

MIN FREQ – Sets the frequency of the lowest filter band.

MAX FREQ – Sets the frequency of the highest filter band.

RES – Sets the Q, or bandwidth, of the filters. Higher values yield narrower filters.

SHIFT – Shifts the frequency of the 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.

HPMIX – Sets the amount of high-pass filtered signal to mix in with the synthesis output. This can be used to increase the intelligibility of the vocoding by allowing certain consonants and air sounds to pass right through.

ATT – Sets attack filter time for the envelope followers.

REL – Sets release filter time for the envelope followers.

FREEZE – Turning this to ON freezes the output of the envelope followers. This can be used to hold the vocal formant.

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 1–16 – These parameters appear as sliders across two pages. They set the out level for each of the 16 bands.

PAN 1–16 – These parameters appear as sliders across two pages. They set the out pan for each of the 16 bands.

Vocoder Mix Parameters

The **Vocoder** block has a **MIX** page with **MIX, LEVEL**, **BALANCE**, **BYPASS MODE,** and **GLOBAL MIX** parameters.

See **Common Mix Parameters** on p. **128** for more details on these.

5.33 Volume/Pan [VOL]

The **Volume/Pan** block can be used either as a "spot fix," to adjust volumes or pans within a preset, or as a dynamic control with an external expression pedal.

Each Axe-Fx II preset can use four fully independent **Volume** blocks.

VOLUME – Scales the output level out of the block. Assign a modifier to create a volume pedal. See the Wah pedal tutorial for more details on doing this (p. **179**).

BALANCE – Sets the balance (L/R) out of the block.

VOLUME TAPER – Sets the "taper" of the volume control. "LINEAR" selects a linear taper. Log 30A, 20A, 15A, 10A and 5A select various tapers 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 L, PAN R – Controls the panning of the left and right output signals.

BYP MODE – Sets the bypass mode of the block. See **Common Mix Parameters** on p. **128** for more information.

LEVEL – Sets the output level of the block independent of the setting for the **VOLUME** control.

5.34 **Wahwah [WAH]**

From Jimi Hendrix's "Voodoo Chile" to, well, Stevie Ray Vaughan's "Voodoo Chile," the wah pedal holds a unique place in the annals of rock history. The Axe-Fx II **Wahwah** is the embodiment of this legacy but with modern reliability and control and a *smooooth* feel. In fact, a wah is actually a very straightforward device. The signal is passed through a resonant filter whose frequency is controlled by a pedal. Place it before distortion for a classic sound or after distortion for a more prominent and "synthy" sound.

The wah is stereo-in/stereo-out. Each Axe-Fx II preset can use two fully independent Wahwah blocks.

XY The **Wah** block supports X/Y Switching. See p. **36** for more information.

Parameters

TYPE – Selects between different wah types based on classic vintage and cutting edge modern designs.

Type	Based On…
FAS Standard	Equivalent to the "Bandpass" setting in earlier firmware.
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	Modeled after the "Shaft" sound.
Mortal	Based on a Morley wah/volume pedal.
VX845	Based on a Vox V845.

FMIN – 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.

FMAX – Sets the frequency of the filter when the frequency control is at its highest value.

RES – Sets the resonance ("Q") of the filter. Higher values give a more pronounced response.

TAPER – Defines the "Sweep Curve" by allow selection from various popular potentiometer tapers.

DRIVE – This simulates overdriving the circuit of the wah pedal.

TRACK – Causes "Q" to inversely track filter frequency. Due to design limitations, classic wah pedals usually become less resonant (less "peaky") as you push the pedal down. This control can be used to mimic those pedals. As frequency is increased, the resonance decreases. If this is zero, the resonance is the same at all frequencies.

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 manually for a "parked" wah (so you can play the guitar on the MTV.

COIL BIAS – This adjusts the DC offset of the virtual inductor which interacts with the wah's **DRIVE** parameter to replicate the subtle (and awesome) sound of some of the most coveted wah pedals.

LOW CUT FREQUENCY – Applies a highpass filter, as created by the internal coupling capacitor of a real wah pedal.

5.35 Input Noise Gate

Every Axe-Fx II preset includes a "built-in" **Noise Gate** connected directly to the main inputs. To edit noise gate parameters, press LAYOUT and navigate all the way to the left to select the IN/GATE column at the left of the grid, then press EDIT. The Noise Gate is always active but can be defeated by turning the **THRESH** control fully counterclockwise.

The Noise Gate can be set up as a Global Block. See **Global Blocks** on p. 131.

The Noise Gate is a downward expander with dynamic filtering. Any signal below the threshold is reduced by the expansion ratio. This can provide smooth transitions as well as abrupt open/close-style gating.

Parameters

TYPE – The noise gate features two types: "Classic" and "Intelligent". The Classic type is a basic downward expander. The Intelligent type is faster and more stable and features a proprietary noise reduction algorithm.

THRESH – Threshold control. Sets the level at which the Noise Gate will start its downward expansion. If the input signal drops below this level, it will be attenuated by an amount controlled by the ratio.

NOTE: As of firmware version 14, the threshold can be modified *globally* for all presets at once using the **NOISEGATE OFFSET** parameter found on the Global: Config page. See Configuration Parameters on p. 145.

RATIO – Sets the downward expansion ratio of the noise gate, thereby determining how much quieter the signal will sound when the gate is *closed*. The ratio acts as a *multiplier* to further reduce signals below the threshold by a factor of "x."

For example, with threshold is set to "-50" and a ratio of "2.0," an input signal at -60 dB (**10 dB** below the threshold) will actually sound **20 dB** below the threshold, (so, -80 dB).

ATTACK – Attack time control. Sets the rate at which the Noise Gate opens the gate.

RELEASE – Release time control. Sets the rate at which the Noise Gate attenuates the signal once the threshold has been crossed. Higher values will make the signal gradually fade once it drops below the threshold.

LEVEL – Controls level at the output of the noise gate. Can be used to boost the overall preset input level.

5.35.1 Input Impedance

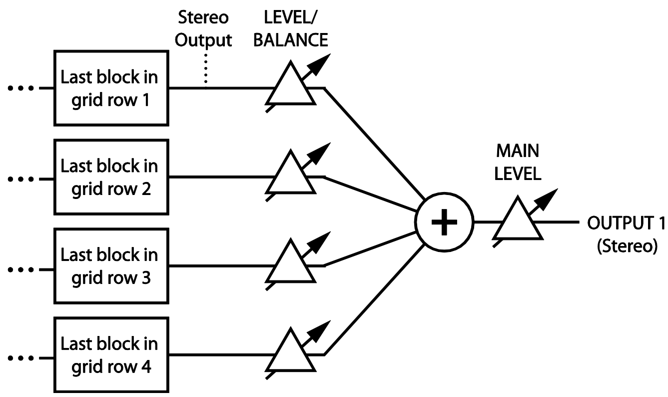
The **INPUT IMP** parameter appears on the page with Noise Gate parameters, but is not part of the Noise Gate. Instead, it changes the actual analog circuitry of the INSTR input jack to alter the way the Axe-Fx II interacts with your guitar. This recreates the way that some classic effects (e.g. Vibe) “load down” the pickups, causing a change in frequency response. The Axe-Fx II recreates this effect by switching various (real) resistors and a capacitor in and out of the signal path. In **Auto** mode, the impedance is automatically set based on the first active effect the input "sees." Normally you will want to leave this on AUTO, but you may also select any of the following values manually. This setting is saved with the preset.

- 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

5.36 **Output Mixer**

Every preset includes a fully programmable output mixer. This provides four pairs of controls for setting the output level, balance for each of the four grid rows, and a master level adjustment control. To edit Output Mixer parameters, press LAYOUT and navigate all the way to the right to select the OUTPUT column at the right of the grid, then press EDIT.

Figure 5-42 – The Output Mixer



LEVEL 1–4 – Think of these as input faders. Each is connected to one row of the grid as shown above.

BAL 1–4 – Each of these is connected to one row of the grid as shown above and determines the left-right balance for the incoming signals. See **BALANCE** under **Common Mix Parameters** (below) for more information.

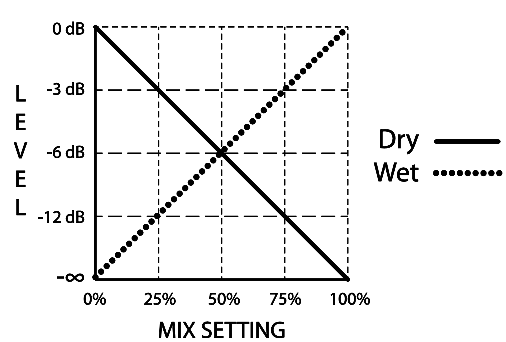
MAIN – Sets the overall level of the main mix for the selected preset. Use this control to adjust the relative levels of different presets, but be careful not to clip the main outputs.

This parameter can be adjusted via MIDI using the **VOL INCR/VOL DECR** feature. See section **9.4** on p. **152**.
The setting for **MAIN** is stored per scene (see SCENES on p.184)

5.37 **Common Mix Parameters**

Almost every block in the Axe-Fx II has a **MIX** page with parameters to determine how the output of that block contributes to the overall preset signal. On some blocks, these controls appear on other pages. Take a moment to familiarize yourself with these important controls and the differences between their settings.

MIX – Determines the balance of wet and dry signals produced at the block output. For basic purposes, setting the mix by ear is generally the best way to achieve a desired result. 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 the dry and the wet being at equal levels of attenuation (-6 db) in comparison to their maximum output levels, as shown at right. Note that the Delay block mix control uses a different mix law.

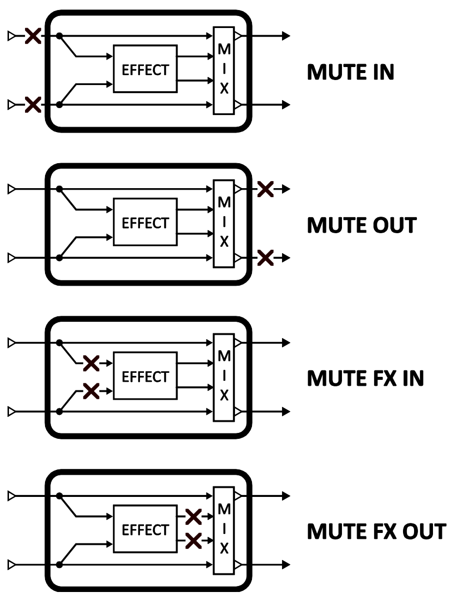


See **DELAY** on p. **60** for details.

LEVEL – As you would expect, the **LEVEL** controls set the output level of a block. Almost all **LEVEL** controls have a range from -80.00 to +20.00 dB. Exceptions include the Compressor and Filter (+/-20.00 dB) and the Drive, which ranges from 0-10 dB.

BALANCE – Determines how the mixed signal of a block will appear at its two outputs. A centered **BALANCE** setting of 0.0 results in both left and right signals being at full volume. As the control is turned either way from the center position, the opposite channel gets quieter. Both the wet and the dry are affected.

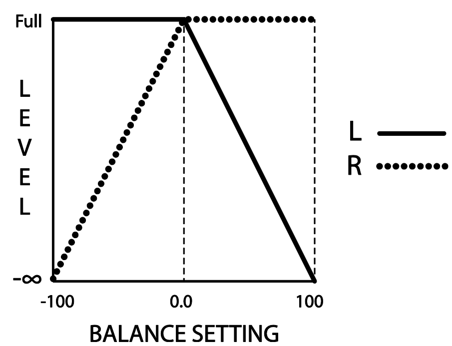
BYPASS MODE – Determines exactly what happens when a block is bypassed. The different options, not all of which are available for every block, are detailed below.



MUTE IN –When the block is bypassed, its inputs are disconnected, silencing the dry immediately but allowing existing effect "tails" to ring. New signals are prevented from entering the effect until it is re-engaged.

MUTE OUT – When the block is bypassed, its inputs remain connected, but its outputs are muted. With this setting, effect tails are silenced when the block is bypassed, but signals can still enter before it is switched on.

MUTE FX IN – When the block is bypassed, the inputs to its internal Effect Processor are disconnected. This allows effect "tails" to ring and leaves the dry unaffected when the block is bypassed. The Dry is completely unchanged—**LEVEL** and **BALANCE** settings remain in effect.



MUTE FX OUT – When the block is bypassed, the outputs of its internal processor are pulled, but dry signal is totally unaffected. With this setting, signals can enter a reverb or delay before it is engaged.

MUTE – When the block is bypassed, 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 a shunt would in its place.

With **BYPASS MODE** settings of "MUTE FX IN" or "MUTE FX OUT," the **LEVEL** and **BALANCE** controls will still affect the dry signal when a block is bypassed.

The MODIFIER slot of the **BYPASS MODE** parameter does not actually connect to the bypass mode parameter, but to the block's BYPASS SWITCH (the same one that the FX BYP button controls).

IMPORTANT: When a modifier is attached to this switch, it becomes the ONLY way that you can bypass or engage the effect. If you find an effect that won't bypass/un-bypass, check this setting.

INPUT GAIN – This parameter, available on reverb, pitch shift, and all three types of delay blocks, determines the amount of signal fed to the effect portion of the block. It has no effect on the dry signal. Within the block, this simulates the way an "Aux Send" would normally feed an effect routed in parallel.

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 (p. **145**).

This feature is provided so you can design presets with the built-in ability for one-touch mix compensation in playing environments that require more or less wet mix. It is offered on the following effect block types:

Delay	**Formant**	**Pitch**	**Ring Mod**
Chorus	**Megatap Delay**	**Quad Chorus**	**Rotary**
Feedback Return	**Multi-Delay**	**Resonator**	**Synth**
Flanger	**Phaser**	**Reverb**	**Vocoder**

6 Global Blocks

6.1 Introduction

The **Global Blocks** feature is completely new and exclusive to the Axe-Fx II. Those familiar with **Global Amps** from previous Axe-Fx products will find this system greatly expanded and improved. Those new to the Axe-Fx will appreciate how Global Blocks enable central control of blocks shared across multiple presets.

With this feature, "links" keep designated blocks in your presets synchronized to their global "masters," which are stored in a separate and independent memory area of the Axe-Fx II. Changes saved to a Global Block cause linked blocks to update when the presets that contain them are recalled. Links to Global Blocks remain in place until you manually remove them—even if you make and save other changes to the block or the preset!

This enables you to create a favorite sound setting and use it to create one or more global blocks. When you load these into multiple presets, each with other different effects, mix levels, routing—whatever suits your needs—they are automatically "linked" back to the original global entries. Now, as favorite (global) sound settings evolve (as we all know these things tend to do…) you no longer need to update all of the individual presets that rely on them. You just save your tweaks into the Global Blocks, and the latest and greatest settings are automatically applied to linked blocks as normal presets are recalled.

Any block instance (except Tone Match, but including the Input/Noise Gate) can link to a global block, with 10 global memories for each of them. Should you choose to REMOVE a link between a block and its global counterpart, this leaves both the normal and the global blocks fully intact and able to be edited independently of one another again.

6.2 Using Global Blocks

The **Global Blocks** feature includes 10 global memories for each and every instance of every type of block (except Tone Matching). There are 10 global "Amp 1" memories, 10 global "Amp 2" memories, 10 for Cab 1, Cab 2, Cho 1, Cho 2 …right down the line to "Wahwah 2."

One important thing to note is that you can only **save to**, **load from**, or **link with** the Global Block that precisely corresponds to the numbered block *instance* you are using in a preset. So for example, global "Cabinet 1" blocks may only be used with the "Cabinet 1" blocks in your presets and not with "Cabinet 2" blocks.

AMP 1	AMP 2	CAB 1	CAB 2	CHORUS 1	CHORUS 2
Global AMP 1 #1	Global AMP 2 #1	Global CAB 1 #1	Global CAB 2 #1	Global CHORUS 1 #1	Global CHORUS 2 #1
Global AMP 1 #2	Global AMP 2 #2	Global CAB 1 #2	Global CAB 2 #2	Global CHORUS 1 #2	Global CHORUS 2 #2
Global AMP 1 #3	Global AMP 2 #3	Global CAB 1 #3	Global CAB 2 #3	Global CHORUS 1 #3	Global CHORUS 2
Global AMP 1 #4	Global AMP 2 #4	Global CAB 1 #4	Global CAB 2 #4	Global CHORUS 1 #4	Global CHORUS 2
Global AMP 1 #5	Global AMP 2 #5	Global CAB 1 #5	Global CAB 2 #5	Global CHORUS 1 #5	
Global AMP 1 #6	Global AMP 2 #6	Global CAB 1 #6	Global CAB 2 #6	Global CHORUS 1 #6	**AND SO ON...**
Global AMP 1 #7	Global AMP 2 #7	Global CAB 1 #7	Global CAB 2 #7	Global CHORUS 1 #7	
Global AMP 1 #8	Global AMP 2 #8	Global CAB 1 #8	Global CAB 2 #8	Global CHORUS 1 #8	
Global AMP 1 #9	Global AMP 2 #9	Global CAB 1 #9	Global CAB 2 #9	Global CHORUS 1 #9	
Global AMP 1 #10	Global AMP 2 #10	Global CAB 1 #10	Global CAB 2 #10	Global CHORUS 1 #10	

To review, the Global Blocks feature allows you to:

- SAVE the settings for any "normal" block into one of the 10 global memories for that block type/instance. This also creates a "link" between the original block and the fully independent Global Block.
- LOAD the settings from any Global Block into a corresponding normal block, with or without creating a link. (Loading without linking simply applies the settings from a global onto a normal block).
- Use LINKS to keep normal blocks in sync with Global Blocks. As a preset is loaded, any linked blocks are seamlessly and instantly updated from their global masters, ensuring up-to-the-moment settings. You can also UNLINK at any time, leaving both global and normal block settings intact.

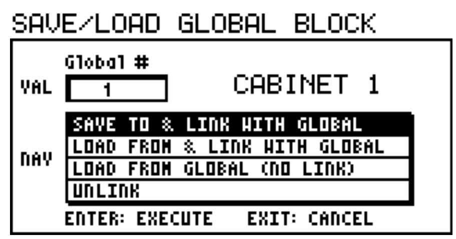
Modifier settings are NOT saved with Global Blocks, but blocks that have an X/Y switch (p. **36**) will have all parameter settings for BOTH states saved in the Global Block.

Global Blocks are included in an Axe-Fx II SYSTEM backup or dump.

Without further ado, let's look at a "how-to."

6.2.1 Saving to a Global Block

Let's start with how to SAVE the settings of a block to a Global Preset. This assumes that you've already inserted a block on the grid and adjusted its parameters to create a setting you want to save as a Global Block.



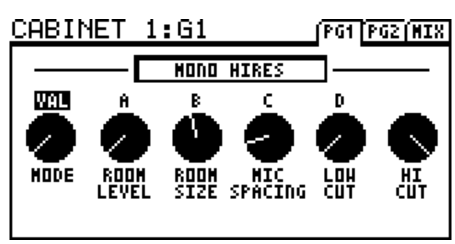
- Select a block on the grid and press **EDIT** to open its EDIT menu.
- Double-click (or press and hold) the **FX BYP** button to open the SAVE/LOAD GLOBAL BLOCK screen.

☞ *Make a mental note of the name (and if there is one, the number) of the block type you are saving, shown in the upper right of the SAVE/LOAD screen ("CABINET 1" in the example above).*

- Turn the **VALUE** wheel to select the number of the Global Block you want to save to.

☞ *It might be good to keep a written log of your Global Blocks, e.g. "Global Amp1/#1: My Hot Plexi," "Global Delay1/#3: Super Spacey Echoes," "Global Cab2/#4: Hi Res, 4×12 Recto (OH), panned left."*

- Ensure that "SAVE TO & LINK WITH GLOBAL" is selected on the display and press **ENTER** to save. A confirmation message will be displayed: "OPERATION COMPLETE! YOU MUST SAVE PRESET TO COMMIT CHANGES." Almost there…



- You will be returned to the EDIT menu of the block you began with. This will now show a "G" and the number of the currently linked global preset in its title area. In the example at right, you'll notice "**CABINET 1:G1**" (G1 = "Global Block #1").
- **IMPORTANT!** Once you have *created* (or *updated*) and *linked to* a Global Block, you must ALSO store the preset for the Global Block changes and the link to be made permanent.

"You must save the preset for any changes made to global blocks to actually be committed permanently to memory."

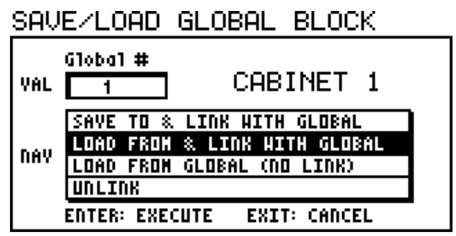
To review:
1) Open the Global Blocks Screen. 2) Save the Global Block with a Link. 3) Save the Preset. If you miss that last step, your changes to the Global Block will be lost the moment you recall a new preset.

To edit or update a Global Block: 1) open any linked instance; 2) make desired changes; and 3) save using the same process outlined above, remembering to also save the preset to commit changes.

Note that once a preset is stored with one or more links to Global Blocks, the RECALL screen will indicate this with the text "USES GLOBAL BLOCKS".

6.2.2 Loading and Linking a Global Block

Once you've saved a Global Block, it is a simple process to load it into other presets and create the links that ensure different instances will stay in sync. It is up to you to remember or document your Global Blocks, but you can always RECALL one and inspect it if you lose track.



To LOAD a Global Block:

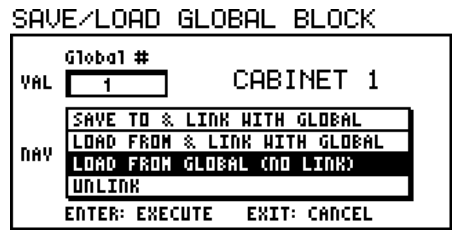
- First insert or select a block of the appropriate type in the current preset. Remember that Global Blocks are limited to use with normal blocks of the same "type and instance," so a Global FILTER 4 Block, for instance, cannot be loaded into FILTER 1, FILTER 2 or FILTER 3.
- With the desired block selected on the grid, press **EDIT** to open its EDIT menu.
- Double-click (or press and hold) the **FX BYP** button to open the SAVE/LOAD GLOBAL BLOCK screen.
- Turn the **VALUE** wheel to select the number of the Global Block you want to load from.
- Ensure that "LOAD FROM & LINK TO" is selected on the display and press **ENTER**.
  A confirmation message will be displayed: "OPERATION COMPLETE! YOU MUST SAVE PRESET TO COMMIT CHANGES."
- You will be returned to the EDIT menu of the current block, which will now be linked to the Global Block as indicated by its title area (see the "G1" atop the second illustration of section 6.2.1).
- In order for the loaded settings and the link to be retained, you need to STORE the current preset.

Once a link has been created, it will cause the block in the current preset to update from the current settings of the linked Global Block—seamlessly and instantly—as its preset is recalled.

6.2.3 Loading Global Blocks without Linking

It is also possible to load normal blocks from Global Blocks without creating a link. This offers a way to "stamp" settings into a preset without enabling the automatic synchronization that normally accompanies the use of Global Blocks. This is useful, for instance, if you want to use a favorite setting as a starting point for a "disconnected" variant or if you want to share a preset with another Axe-Fx II owner who may not have the same Global Block settings as you.

To load a Global Block without linking:



- First select a block of the appropriate type to insert into the current preset. As always, remember that Global Blocks pair with normal blocks of the same "type and instance," so a Global "Wahwah 1" block can be loaded into a "Wahwah 1" block but not into a "Wahwah 2" block.
- With the desired block selected on the grid, press **EDIT** to open its EDIT menu.
- Double-click (or press and hold) the **FX BYP** button to open the SAVE/LOAD GLOBAL BLOCK screen.
- Turn the **VALUE** wheel to select the number of the Global Block you want to load from.
- Ensure that "LOAD FROM GLOBAL (NO LINK)" is selected on the display and press **ENTER**.
  A confirmation message will be displayed: "OPERATION COMPLETE! YOU MUST SAVE PRESET TO COMMIT CHANGES."
- You will be returned to the EDIT menu of the current block.
- In order for the loaded settings to be retained, you must STORE the current preset.

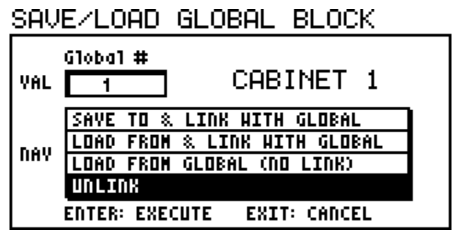
"Global Blocks are limited to use with preset blocks of the same "type and instance," so Global *Wahwah 1* may be used with *Wahwah 1*, but not the *Wahwah 2* block.

To review:

1) Open the Global Blocks screen. 2) Load the Global Block without linking. 3) Save the Preset. Your local block settings will now have all the settings from the Global Block, but there will be no link between the two entries.

6.2.4 Unlinking Preset and Global Blocks

You will sometimes want to remove the link between a normal block and its global master. Removing a link does not change the settings of the local block or its global master.



- With the desired block selected on the grid, press **EDIT** to open its EDIT menu.
- Double-click (or press and hold) the **FX BYP** button to open the SAVE/LOAD GLOBAL BLOCK screen.
- Ensure that "UNLINK" is selected on the display and press **ENTER**. A confirmation message will be displayed: "OPERATION COMPLETE! YOU MUST SAVE PRESET TO COMMIT CHANGES."
- You will be returned to the EDIT menu of the current block.
- In order for the link to be removed permanently, you must STORE the current preset.

6.2.5 Backing Up/Sharing Presets Containing Global Blocks

As you learned above, presets with links to Global Blocks refer to the "System" area of the Axe-Fx II to obtain their parameter settings. When such presets are backed up or dumped for sharing—whether individually or in a bank—their Global Blocks links point to data which is not present in the preset or bank dump. Instead, the Axe-Fx looks to its own local Global Block data, which could be totally blank or unrelated.

There are two approaches to handling this issue.

1. Make sure you also back up the entire SYSTEM area of the Axe-Fx II. This contains the Global Block data will be needed to reconstitute the presets.

2. Or, remove links to global blocks from each preset before dumping it. You can use the method described above in section **0**, or use one of two options described below:

    i. Once a preset is stored with one or more links to Global Blocks, the RECALL:PRESET screen displays an option to "PRESS ENTER TO UNLINK". Doing so will unlink ALL global blocks. You can then STORE or DUMP the preset with no risk missing data.

    ii. At any time, you can use the STRIP GLOBAL DATA utility to unlink all global blocks in the current preset. This is located on the PRESET page of the UTILITY menu.

Tip: You don't need to STORE after removing global blocks to dump a preset. If you want to keep the Global Blocks version in your system, remove the links, dump the preset, and then discard the changes by NOT saving.

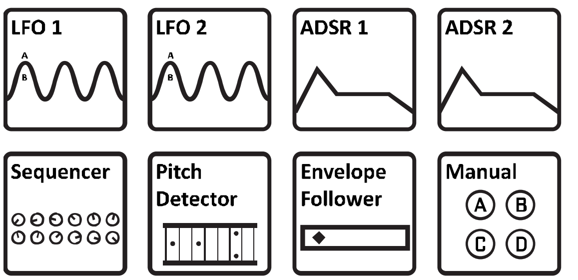
7 Modifiers & Controllers

7.1 Introduction

Modifiers link parameters to **controllers**, allowing sound features to be automated or remotely controlled in real-time. For example, the sweep of a WAH block might be "attached" to a pedal as it ordinarily is, but you could just as easily assign an LFO for a little Auto-Wah, or an "Envelope Follower" for some funky Mutron™-style action.

Besides Wahwah control, there are hundreds of other parameters you can "modify" on the Axe-Fx II, with over 20 different controller **sources** to connect them to. Some, like the built-in LFOs and Envelope Follower, are "internal" to the Axe-Fx II, while others, like a connected expression pedal or a footswitch which sends a MIDI message, are referred to as "external" controllers.

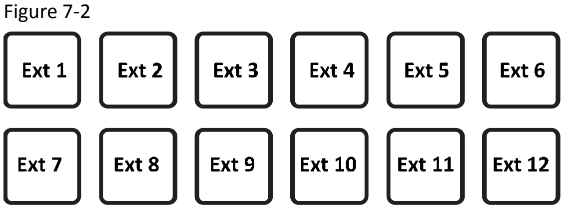
Figure 7-1



INTERNAL CONTROLLERS

These are built in to the Axe-Fx II. Every preset can have its own settings for 2 **LFOs**, 2 **ADSRs**, **Envelope Follower**, **Sequencer**, 4 **Manual** knobs and 2 **Scene Controllers**. These are accessed via the CONTROL button on the front panel. See section **7.3** below for details on each of these controllers.

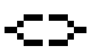
Figure 7-2

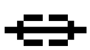


EXTERNAL CONTROLLERS

Each one of the 12 External Controllers must be assigned either to the onboard PEDAL jack(s), *or* a **MIDI CC#** (0–127) in the CTRL page of the I/O menu (p. **152**). These assignments are global, but you can use external controllers for different things in different presets.

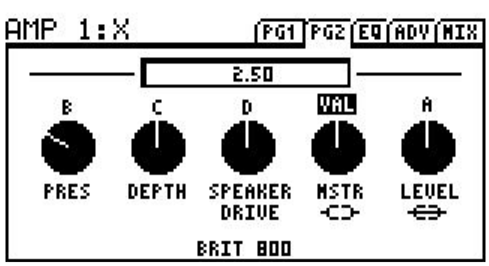
7.2 Creating a Modifier

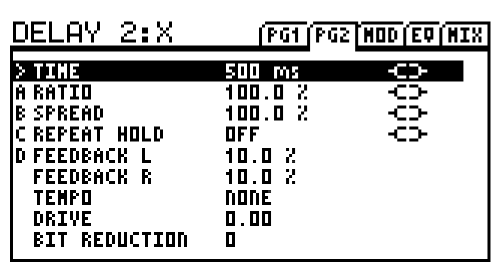

OPEN


FILLED

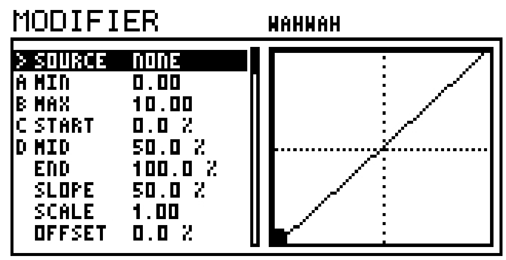
The process of creating a modifier begins at the parameter you want to control. Parameters that can be controlled are marked with a special symbol (left top). Look for it beneath a "soft" knob or to the right of a text parameter. If a modifier is already present, the symbol will have a line through it (left bottom).

In the examples below, the **MASTER** and **LEVEL** (left), and **TIME**, **RATIO**, **SPREAD,** and **REPEAT HOLD** (right) may be controlled with modifiers. **LEVEL** has one assigned already. Parameters without the symbol cannot be controlled.





To Create a Modifier…



- Select any controllable parameter ( ) and press the **ENTER** button to show the MODIFIER screen.
- Select a **SOURCE** to assign to the current parameter or choose NONE to remove an existing modifier.
  - The graph shows the relationship between the control source (x-axis) and the sound parameter (y-axis). The "dot" in the graph tracks the SOURCE as its value changes.
- You may leave the MODIFIER screen to return to the parameter's main menu at any time by pressing **EDIT** or **EXIT.**
  - Back in the EDIT menu, a modified **Knob**, **Slider,** or **Graph** parameter will be animated as the source changes. Modified text parameters do not update in this fashion.
  - The text box above a knob shows the value that was set before the modifier was applied.
- You must **STORE** the current preset to make the modifier setting permanent.

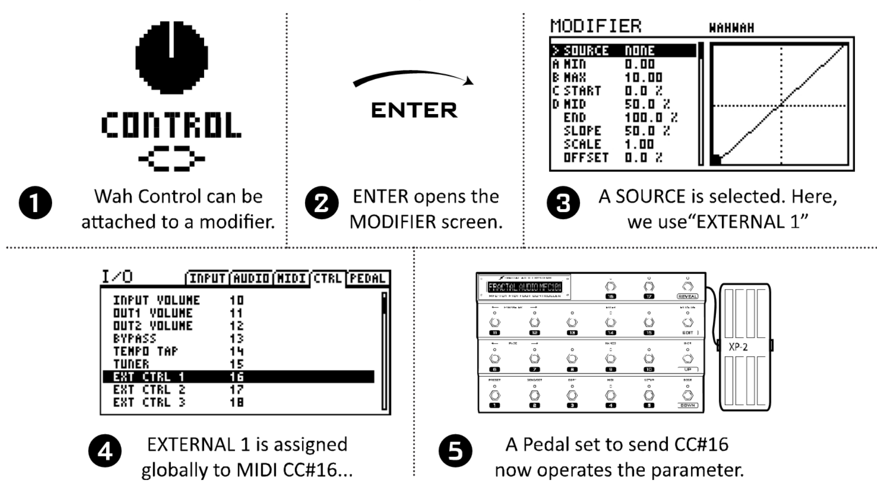
To Remove a Modifier…

To remove a modifier, just change its **SOURCE** parameter to NONE.

Modifier Example: Wahwah Control

Before we get into further detail about the other parameters on the MODIFIER screen, let's look at the basic Wahwah example from the introduction above. Z

Figure 7-3



Let's assume that the controller called "EXT 1" has already been set up (in the CTRL page of the I/O menu) for MIDI CC# 16 (its default setting), and that we have connected a MIDI foot controller with an expression pedal that is set up to send this same message on the correct MIDI channel. (In fact, this is the default setting for the EXPRESSION PEDAL 2 jack on the MFC-101).

To start, a **WAH** block is inserted on the grid. Pressing EDIT opens its EDIT menu. The modifier symbol beneath the **CONTROL** knob indicates that a modifier can be added here. Selecting this parameter and then pressing ENTER opens the MODIFIER screen. Selecting EXT1 for the **SOURCE** attaches this controller to our parameter, and the wah pedal starts working! "Follow the bouncing ball" as the dot on the graph follows the motion of your foot on the pedal.

If we change the modifier **SOURCE** to "ENVELOPE," our wah is disconnected from the pedal and becomes controlled instead by the level of the input signal to create a "touch wah". Changing the source to one of the LFOs, creates an oscillating "auto-wah." Externals, Envelopes, LFOs, and other sources are detailed below in section **7.3**.

7.2.1 Transformations

The MODIFIER screen also contains several parameters that enable you to set up a custom relationship between changes at the source and changes in the destination parameter. This makes it possible to transform or "tune" the feel and sound of a dynamic effect. This can be especially important when one control source is attached to multiple different parameters.

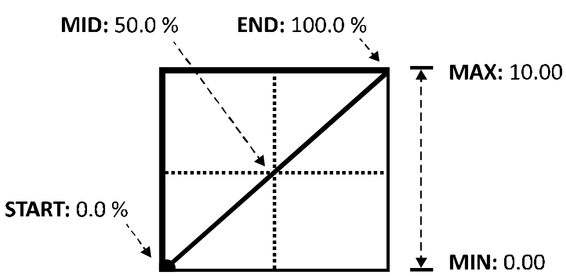
The **MIN** and **MAX** parameters determine the range over which the controlled parameter will respond, using the same units. For example, the **MIN** to **MAX** range for a **LEVEL** parameter might be set from -9 to +4 dB, while a delay **TIME** might be set for 200–400 ms.

The **START**, **MID**, **END,** and **SLOPE** settings are used to re-map the ways in which parameters respond to changes in the source.

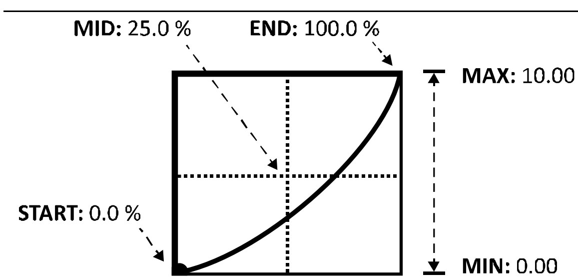
The **SCALE** and **OFFSET** parameters allow you to vertically resize or shift the modifier curve.

<u>Example 1: Creating a Custom Curve</u>

For the first example, let's imagine a **VOLUME** parameter being controlled by a pedal (via EXT1/MIDI CC#16).

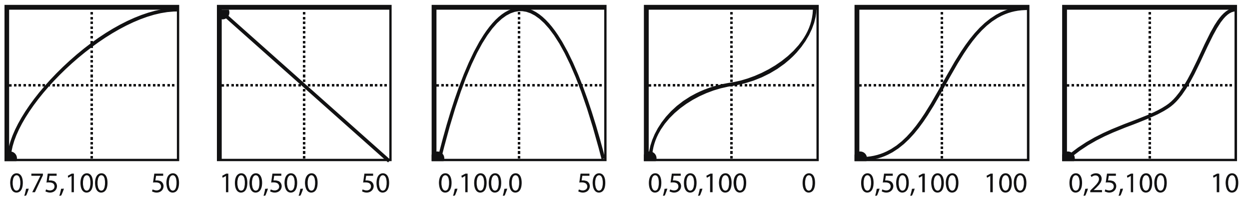


The default settings for **START**, **MID,** and **END** create a perfectly linear relationship between the source and the target. As the pedal is depressed, the volume increases in direct proportion. **MIN** and **MAX** are set to their extreme limits, so volume goes from 0.00-10.00 (OFF to FULL).
But linear response is generally unsatisfactory for volume control because of the non-linear characteristics of our hearing. Ears aren't very good at math…



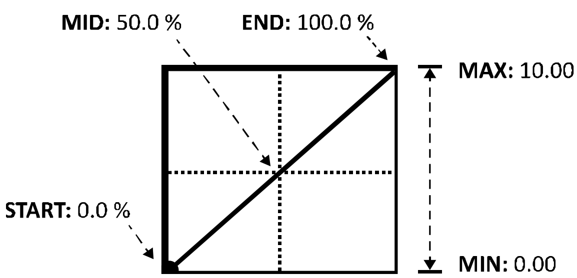
As we lower the value of **MID,** the response starts to take on a more comfortable curve, closer to the classic log or audio taper typically used for volume control. **MIN** and **MAX** are still set to "0.0" and "10.0" so that volume goes from off to full, but the way it *swells* has changed.

Here are some more examples of the kinds of curves you can create by changing **START, MID, END,** and **SLOPE**. With a bit of practice, you will learn to achieve desired modifier effects quickly.

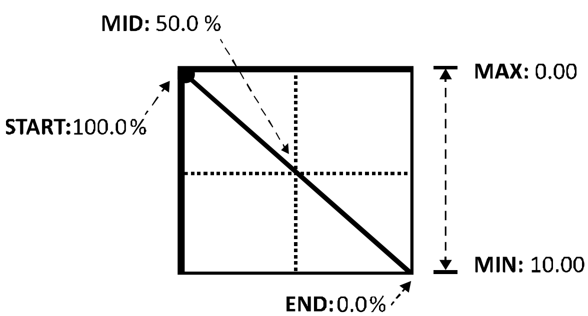


Example 2: Setting MIN and MAX

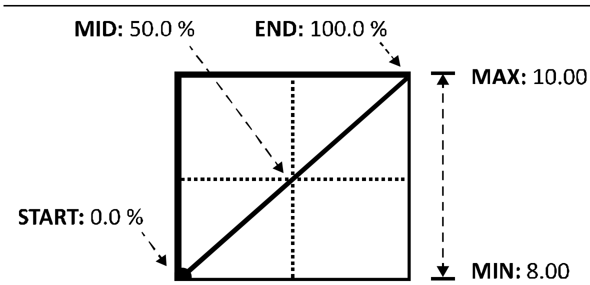
MIN and **MAX** allow a modifier's range to be "pegged" to precise values by setting the y-axis scale of the modifier graph using the same units as the parameter being modified. This ultimately determines the relationship between the source and the target parameter. The following examples illustrate **MIN** and **MAX** in action.



A now-familiar example: the default modifier setting for a **VOLUME** parameter, showing a linear sweep, with **MIN** and **MAX** set to their extremes.



At first glance this seems to be a reversal of the example above, where pushing the pedal forward decreases the volume. But look closer and you'll find that **MIN** is set to "10.0" and **MAX** to "0.0" With both sets of values inverted, this setup sounds and works *exactly* like the one above!



Unlike the preceding examples, this one does not show **MIN** and **MAX** set to their extremes. Instead, the range goes from 8.0 to 10.0, so our volume pedal will create only a small swell as it is rocked from heel to toe. You might use this to create a variable solo boost pedal.

7.2.2 Scale and Offset

SCALE and **OFFSET** change the vertical scale and positioning of a modifier curve. **SCALE** exaggerates (or compresses) the shape of the curve, while **SHIFT** moves up or down with reference to the axes. Curves pushed or pulled outside the modifier graph boundaries will be clipped and replaced by line segments.

7.2.3 Damping

If MODIFIERS add automation or remote-control in "real time," **damping** allows these changes to happen in "stretched time." Normally, parameters change at the same rate as modifier sources. Their values may be re-mapped as we saw above, but changes happen simultaneously. Every little tremble of your foot is reflected by a little tremble in the sound. **DAMPING** allows you to add hysteresis—a little "viscosity" or "elasticity"—so changes in the parameter don't accelerate as quickly as those in the source.

At low settings, **DAMPING** adds a little smoothing. Try a setting of 4–12 ms to "grease" a wah or to ease the edges of a step-sequence to eliminate clicks and pops. Medium settings can "relax" a whammy or cover up for a "scratchy" expression pedal. With high damping, sound changes glide like honey. Quick motions are entirely swallowed up, while a simple footswitch can be used to create a 1000-ms-long ramp of sound change. Used appropriately, damping adds tremendous power to the MODIFIER system.

7.2.4 Auto Engage

While it may sound like the precursor to an arranged marriage, **AUTO ENGAGE** is actually a powerful feature that brings a block out of bypass when the **SOURCE** for a modifier on one of its parameters changes in a predefined way. The classic example is a Wah that turns on automatically when you rock the pedal and then turns off when you return to the heel-down position. You might also use Auto Engage to create a pedal that controls the speed of a Rotary Speaker and then bypasses the effect when pulled all the way back.

Try it! You'll quickly find that Auto Engage comfortably eliminates the need for expression pedal "toe" switches.

How the effect turns ON or OFF is determined by the values of two parameters:

AUTOENG – Determines whether or not the block containing a modifier will automatically engage or bypass based on the level of a modifier source. The SPEED ("SPD") options engage the effect when the controller changes more than 5% in any 20 ms interval. The POSITION ("POS") options engage the effect when the controller value is 5% greater than or less than the **OFF VALUE** (depending upon whether the **OFF VALUE** is less than or greater than 50% respectively).

For example, setting the Auto-Engage to "SLOW SPD" will bypass the effect when the controller value is less than, say, 5% (default). To engage the effect the controller (foot pedal) must be moved more than 5% in a 20 ms interval. This requires that the pedal be moved somewhat rapidly and prevents the effect from engaging erroneously if the pedal droops. Setting the Auto-Engage to SPD POS will bypass the effect in the same manner but the effect will engage when the controller exceeds 10% (5% + 5%) regardless of the rate of change.

OFF VAL – Sets the threshold that the value of the current **SOURCE** must cross for auto-engage to occur. When **OFF VALUE** is set below 50%, the effect is bypassed when the controller goes BELOW the **OFF VALUE.** If **OFF VALUE** is set to 50% or higher, the effect is bypassed when the controller goes ABOVE than the **OFF VALUE.** For "HEEL down = bypassed, set to 5%. For "TOE down = bypassed, set to 95%.

Use SLOW settings to "loosen" Auto Engage so your effect doesn't snap on or off unexpectedly. Set to OFF to disable Auto Engage.

7.2.5 Program Change Reset

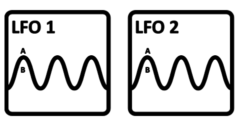
Normally, the last value of an external control source is retained—even across preset changes—until a new value is received. So if you "park" a pedal-controlled Wah, for example, then change to a new preset with the same Wah settings, the newly loaded preset will load with the Wah in the same parked position.

Program Change Reset (**PC RST**) allows you to *override* this behavior, causing a parameter to use a previously stored setting when it loads, rather than referring to the retained source value. As soon as the external controller is updated—the pedal moved, a new MIDI message received—the attached parameter snaps back to track it again.

Here's how it works: the parameter box above a modified knob always shows the "reset" value so you can freely set and save it. Note: Knob and filter "graph" parameters with modifiers are animated as their sources change, so you won't see graphical elements update in the display as you turn the value wheel. You can either set the stored value before turning **PC RST** to ON or change it afterwards. Note: Internal controllers (q.v.) are not subject to **PC RST** because their values are updated immediately when a preset loads.

7.3 Control Sources

7.3.1 LFO1 & 2



An **LFO**, or Low-Frequency Oscillator, generates control signals in the form of a variety of familiar wave shapes or random signals. Familiar examples of LFOs in action include the pulsing of a tremolo, the steady back and forth sweep of a phaser, or the modulation of a chorus. The Axe-Fx II contains two global LFOs that can be individually programmed per-preset for use as modifier sources to control various other parameters. Press the front panel CONTROL button to find the LFO1 and LFO2 menu pages for the current preset.

Each of the two LFOs outputs two signals (A and B), so the list of modifier sources contains four entries: LFO 1A, LFO 1B, LFO 2A, and LFO 2B. By default, A/B pairs are complementary, meaning that as A swings from 0–100, B swings from 100–0, but the phase of output B is fully adjustable for in-phase or in-between settings.

Besides being available as a modifier source, LFO1 may also be used to SYNC the rates of the Chorus, Flanger, Phaser, Tremolo, and Multi-Delay blocks. This not only allows the sweeps of these effects to be aligned to each other, but to other modifiers as well. Set the "native" rate fully counterclockwise to enable LFO1 SYNC.

TYPE – This sets the waveform or shape of the selected LFO.

RUN – Tucked into the MODIFIER slot beneath the TYPE control, this parameter starts and stops the LFO. When the LFO is stopped, its output drops to "zero," and the wave cycle resets. When using a MIDI CC via an external controller to toggle RUN, a value of 70 or higher will START the LFO, and a value of 57 or lower will STOP the LFO (assuming the modifier has default settings for **MIN/MAX/START/MID/END/SLOPE**).

> ***Tip:*** *An LFO that is locked to tempo can still "drift" from another system or device. To stay locked to "song position," just stop and restart periodically from your DAW/sequencer with a quick pair of OFF/ON messages sent at regular intervals.*

RATE – Sets the frequency of the LFO from 0.05–30.0 Hz. When **RATE** is shown in parenthesis, it is being controlled by the tempo parameter (below). Set **TEMPO** to "NONE" for manual control.

DEPTH – Sets the amplitude or "intensity" of the LFO from 0-100%.

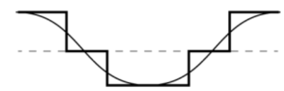
DUTY – Varies the duty cycle, or "symmetry," of the Triangle, Square, and Trapezoid waveforms.

OUTB PHASE – Adjusts the phase angle of the LFO's output B with respect to A. At 180°, the outputs are phase-opposite, so while A swings from 0–100%, B swings from 100–0%. At 0°, A and B are in phase.

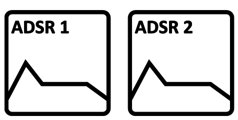
TEMPO – Sets the LFO rate in rhythmic relation to the global tempo. For example, if the tempo is set to "1/8," the LFO will cycle twice per tempo beat (8x/measure). Tempo changes are reflected in real-time. To ignore the global tempo, set the tempo control to NONE.

LFO1 and LFO2 are also subject to interesting interactive variations made possible by the modifier slots available on their own parameters. See p. **174** of the Appendix for a guide to LFO Waveforms, Duty, and Phase.

QUANTIZE – Divides continuous LFO waveforms into "sample and hold" segments. In the diagram at right, a "SIN" wav is quantized as "3" values. (Use the "TRI" wav for rhythmically equal segments.)



7.3.2 ADSR 1 & 2

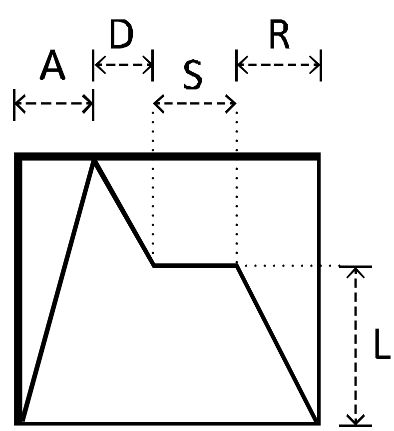


The Axe-Fx II contains two **ADSR** or "envelope" generators that can be used as control sources. ADSR stands for "attack, decay, sustain, release"—the four time segments that determine how long it takes for the entire envelope to run its course. The graph below illustrates this concept. Press the front panel **CONTROL** button to find the ADSR1 and ADSR2 menu pages for the current preset.

MODE – The mode determines how the ADSR generator operates in response to signals above the threshold.

- **ONCE**: The ADSR plays through when the threshold is exceeded.
- **LOOP**: The ADSR loops as long as the signal is above the threshold.
- **SUSTAIN**: The ADSR begins when the threshold is exceeded, but HOLDS at the sustain level until the signal drops below the threshold, at which point the release phase occurs.

RETRIG – When Retrigger is ON, the ADSR will reset to the beginning whenever the threshold is crossed from below to above. If Retrigger is OFF, the ADSR must reach the end of its release phase before it can be retriggered again.



ATTACK – The envelope begins at zero and increases to 100% over the duration of the attack time (A).

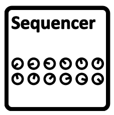
DECAY – When the attack phase completes, the envelope descends for the duration of the decay time (D) until it reaches the sustain level (L).

SUSTAIN, LEVEL – After the decay, the envelope remains at the sustain level (L) for the duration of the sustain time (S).

RELEASE – At the end of the sustain phase, the envelope proceeds to zero over the duration of the Release time (R).

THRESHOLD – Sets the level at which the LFO will be triggered (or re-triggered; see MODE above).

7.3.3 Sequencer



Like any step-sequencer, that of the Axe-Fx II generates repetitive control patterns by looping through a series of steps or “stages,” each of which outputs a specified value. The sequence can be run at a specified rate or be synchronized to the global tempo. Press the front panel **CONTROL** button to find the SEQUENCER page for the current preset.

RATE – Sets the rate at which the sequence is stepped through. At 1 Hz, each step will last for 1 second. When **RATE** is shown in parenthesis, it is being controlled by the tempo parameter (below). Set **TEMPO** to “NONE” for manual control.

TEMPO – Sets the sequencer rate in rhythmic relation to the global tempo. For example, if the tempo is set to “1/16” the sequencer will play 16 steps per measure (4 beats). To ignore global tempo, set this to “NONE.”

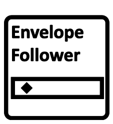
RUN – This parameter starts and stops the sequencer. When the sequencer is stopped, it remains at the value set for **STAGE 1** (see below). When using a MIDI CC via an external controller to toggle RUN, a value of 70 or higher will START the LFO, and a value of 57 or lower will STOP the LFO (assuming default modifier settings for IN/MAX/START/MID/END/SLOPE).

> ***Tip:*** *A Sequencer that is locked to tempo can still “drift” from another system or device. To stay locked to “song position,” just stop and restart periodically from your DAW/sequencer with a quick pair of OFF/ON messages at regular intervals. Assign the CC# to an External controller (p. 144) and assign this as the source of the modifier on the RUN parameter.*

STAGES –Sets the number of steps in the looped sequence from 1–32. For example, if STAGES is set to “3”, the sequencer will play steps 1, 2, and 3 in constant rotation: 1,2,3,1,2,3,1,2,3,1,2,3…

STAGE # – Each of the **STAGE** parameters sets the value for one step of the sequence. You can randomize the values of ALL steps in the sequence by pressing ENTER with any STAGE (or the STAGES parameter) selected.

7.3.4 Envelope Follower



The **Envelope Follower** “tracks” the level of the main input signal. The harder you play, the greater a value it produces. The Envelope Follower is designed to enable “touch-control” of wah, filter, and other effects, plus “ducking” and other types of dynamics control. Press the front panel **CONTROL** button to find the ENVELOPE page for the current preset.

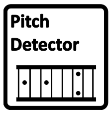
THRESH – The threshold controls the sensitivity of the Envelope Follower by setting the level at which tracking kicks in or out. When the input level is greater than the threshold, the follower tracks it at the attack rate. When the signal drops below this level, the output of the follower will decay to zero at the release rate.

ATTACK – The rate at which the follower output follows signals increasing in power.

RELEASE – The rate at which the follower output follows signals decreasing in power.

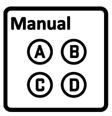
GAIN – The gain control works like a classic “sensitivity” control to set the relationship between incoming levels and outgoing control signals. By boosting the input of the envelope follower, GAIN allows a weaker input signal to exert a greater effect.

7.3.5 Pitch Detector



The **Pitch Detector** has no menu page or parameters, but appears in all modifier screens as a source. This module monitors the main input signal and analyzes its pitch, outputting a low value for low notes and a high value for high notes.

7.3.6 Manual Knobs



New to the Axe-Fx II are four front panel Quick Control knobs, detailed in section **4.3.1** on p. **36**. When you select the MANUAL page of the CONTROL menu, these knobs work as MODIFIER control sources that can be used for making sound adjustments without menu diving. The current value of each knob is saved with each preset.

Remember that while you are setting up a MODIFIER, the Manual knobs will be assigned to edit parameters on the screen instead of operating as SOURCE. To test your modifier, you'll need to leave the EDIT menu and return to CONTROL MANUAL to test your work.

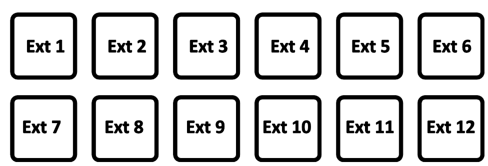
7.3.7 Scene Controllers

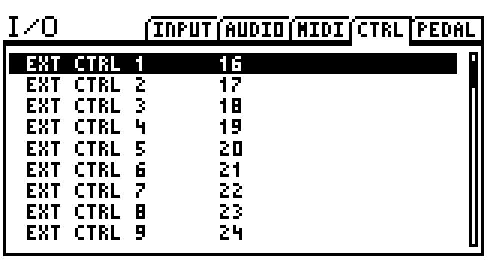
Scene Controllers are similar to Manual Knobs above in that you manually set the value for each of two virtual knobs that can be assigned as SOURCES for modifiers. Unlike Manual Knobs however, the value of each SCENE controller can be pre=programmed individually for each SCENE in each preset. See p.184 for more on SCENES.

For example, you might assign SCENE CONTROLLER 1 to delay feedback, with a value of 10% in SCENE 1, 30% in SCENE 2, 0% in SCENE 3, and so on. SCENE CTRL 2 could meanwhile be used for something completely different.

7.3.8 External Controllers

The onboard PEDAL jack or any MIDI Control Change message can be assigned as an **External Controller** to be used as modifier source, as shown in the example in Figure 7-3. The Axe-Fx II allows you to set up 12 global External Controller assignments, but remember: you can use each of these to modify multiple parameters per preset.





To set these EXTERNAL CONTROL assignments, use the CTRL page of the I/O menu. Select the desired EXT CTRL entry and turn the value wheel until PEDAL or the desired MIDI CC number is shown. Set an External Control to NONE to disable remote control.

The Axe-Fx II also has a "learn" feature that allows it to detect control sources automatically. Select the desired item, press ENTER, and use the remote controller to send some data to the Axe-Fx II; the source will be set automatically. (This is also a good way to ensure that remote devices are in fact set up and transmitting correctly.)

Remember that the ***channel*** of incoming CC# messages must match that of the Axe-Fx II under I/O:MIDI.

The Axe-Fx II uses a set of system parameters—**EXTERNAL CONTROLLER INITIAL VALUE 1–12**, located on the MIDI page of the I/O menu (p. **150**)—to determine what value should be used for each External Controller between the time when the Axe-Fx II is booted up and when external data is first received. Options are 0% or 100%.

8 Global Parameters

The Global Parameters section, accessed by pressing the front panel GLOBAL button, contains four pages of menus that control sound settings across all presets and modes. Changes made in this area take effect immediately without needing to be STORED. The settings for all global parameters are included in a backup of the Axe-Fx II "SYSTEM" (See p. **160**). Default assignments for all system parameters are listed beginning on p. **192**.

8.1 Configuration Parameters

The Configuration page ("CONFIG") of the GLOBAL menu contains parameters that affect all presets at once.

PARAMETER	Description
DEFAULT MODELING VERSION (LATEST/etc.)	**(XL/XL+ ONLY!)** Selects which version of modeling to use when choosing a new amp model or when loading presets made with previous firmware versions. You can override this setting in any preset with the **MODELING VERSION** parameter in the Amp block.
FORCE DEFAULT VERSION (ON/OFF)	When set to "ON", this forces all presets to load the selected **DEFAULT MODELING VERSION** (above), in effect overriding any per-preset overrides.
POWER AMP MODELLING (OFF/ON)	Enables or disables power amp simulation for all AMP blocks in all presets. This capability is provided for cases when the Axe-Fx II 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 Axe-Fx II and once in the real power amp—would result in an over-processed tone. Preamp (gain, tone, etc.), Graphic EQ, and Mixer (Level, Balance, etc.) sections of the AMP block continue to affect the sound.
CABINET MODELLING (ACTIVE/BYPASSED)	This parameter enables or disables all CAB block processing in all presets. The CAB blocks will not physically appear to be bypassed, but they will act exactly as if they have been replaced by shunts. Use this when you are using the Axe-Fx II with an amp powering real guitar speaker cabinets or the sound will be muffled, muddy, and boomy.
SPILLOVER (OFF/DELAY/ REVERB/BOTH)	Allows delays and reverbs tails to ring out or "spill over" across preset changes. You can select whether DELAY, REVERB, or BOTH effects will spill over. Setting this to OFF causes effect tails to be "cleared" upon preset change. See section **16.10** on p. **177** for more.
REVERB MIX (+/- 50%)	Allows you to boost or cut the MIX setting across all REVERB blocks in all presets at once. Note that an offset applied here will NOT be reflected in the value shown in the EDIT menu of the actual Reverb blocks. This capability is provided to compensate for the fact that certain performance spaces may require more or less reverb "across the board."
EFFECTS MIX (+/- 50%)	This parameter allows you to boost or cut the setting of the MIX parameter 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 following block types: **Chorus**, **Delay**, **Flanger**, **Formant**, **Megatap Delay**, **Multi-Delay**, **Phaser**, **Pitch**, **Quad Chorus**, **Resonator**, **Reverb**, **Ring Mod**, **Rotary**, **Synth**, **Vocoder** This feature is provided so you can design presets with the built-in ability to compensate for how different performance spaces may require more or less "wet" mix.

NOISEGATE OFFSET (+/- 40.00 dB)	Different environments (e.g. studios, stages) and different guitars create different levels of noise interference. This parameter allows you to *globally* raise or lower the **THRESHOLD** of the **Noise Gate** (p. 126). Note that if the **THRESHOLD** for a given preset is set to "OFF" the Global Offset will have no effect.
AMP GAIN (+/- 12.00 dB)	This control provides +/- 12 dB of relative gain for all amp blocks and can be used to adjust the gain of all presets to compensate for the differences between guitars.
IR CAPTURE MODE	Determines whether the Impulse Response Capture utility operates in UltraRes™ or Standard Resolution (2048).

8.2 Output Parameters

The OUT1 and OUT2 pages of the GLOBAL menu provide tone and level control tools for the two main outputs.

PARAMETER	Description
BANDS 1–10 (+/- 12 dB)	Output 1 and Output 2 are individually equipped with 10-band graphic equalizers for fine-tuning of the tone across all presets. The outermost bands are shelf type filters. These come in handy when using the Axe-Fx II in different acoustical environments, or when you change amps/speakers with no opportunity to adjust your presets. Output 1 settings apply to *all instances* of the Output 1 mix signal including the Headphone jack, the OUT1 balanced jacks, the OUT1 unbalanced jacks, and signal which may be routed to USB or DIGITAL OUTS. One exception to note: when **OUTPUT 2 ECHO** is set to "OUTPUT 1" in the I/O:AUDIO menu, the global Graphic EQ setting for OUT1 is NOT copied to Output 2; Output 2 is processed by its own global graphic equalizer in the usual way. Output 2 settings apply to *all instances* of the Output 2 mix signal including the Output 2 unbalanced jacks (FX send) and the Output 2 signal, which can be routed to the DIGITAL OUTS. (See I/O:AUDIO on p. **148** for more on routing).
GAIN (+/- 12 dB)	The GAIN slider to the right of the graphic equalizer can adjust the output signal at the selected output by +/- 12 dB. This level adjustment happens in the digital domain, and a boost can cause output clipping if not used judiciously. Conversely, a reduction made here can provide a good temporary remedy until you have time to address a multi-preset clipping problem.

8.3

8.4 Custom Scales

The Custom Scales ("SCALES") page of the GLOBAL menu is used to configure custom scales for the Custom Shifter type found in the Pitch Shifter block.

PARAMETER	Description
Custom Scale Number (1–32)	This selects from among the 32 global custom scales available to edit using the 12 parameters below.
Input Pitch/Output Shift	These 12 parameters are used to determine the relationship between notes played and pitch shift applied. Each of the 12 steps of the chromatic scale can be individually shifted within a range of +/- 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. Settings are applied immediately with no need to STORE.

9 Input/Output Parameters

The Input/Output ("I/O") Parameters section, accessed by pressing the front panel **I/O** button, contains six pages of menus used to configure audio, MIDI, and control settings for the Axe-Fx II. I/O settings are global, and changes made in this area take effect immediately without needing to be STORED. The settings for all I/O parameters are included in a backup of the Axe-Fx II "SYSTEM" (See p. **160**). Default assignments for all system parameters are listed beginning on p. **192**. See also the diagram "Using Send and Return" on p. **182** of the Appendix.

9.1 Input Parameters

The INPUT page of the I/O menu contains parameters to manage input levels. For more on setting levels, see section **3.1** on p. **15.**

PARAMETER	Description
INSTR IN (0-100%)	This scales the level of signal at the front panel INSTR input jack, determining its level at the input of the A/D converter.
INPUT 1 (0-100%)	This scales the level of signal at the rear balanced 1/4 " INPUT 1 jacks, determining their level at the input of the A/D converter.
INPUT 2 (0-100%)	This scales the level of signal at the rear balanced 1/4 " INPUT 2 jacks ("FX RETURN"), determining their level at the input of the A/D converter.

9.2 Audio Parameters

Routing and format parameters appear on the AUDIO page of the I/O menu.

PARAMETER	Description
MAIN INPUT SOURCE (ANALOG (IN 1)/ USB/ SPDIF/AES)	This selects which signal source should be routed to the Input of the grid. "ANALOG (IN 1)" selects the front INSTR input jack or the rear INPUT 1 jacks, based on the setting of **INPUT 1 LEFT SELECT** (below). When "USB" is selected, the Axe-Fx II will process signals sent from the **AXE-FX II OUT 0** and **AXE-FX II OUT 1** audio outputs of the connected computer. When "SPDIF/AES" is selected, digital inputs will be used.
WORD CLOCK (AUTO/SPDIF-AES IN)	This selects the clock source for the A/D and D/A converters as follows: **Auto:** uses the internal clock if the input source is Analog or USB. Uses the recovered SPDIF/AES clock if the input is SPDIF/AES. **SPDIF/AES IN:** uses the recovered clock for all input sources. A valid 48 kHz data stream must be present at the AES or SPDIF input or the unit will fall back to the internal clock and display "NO INPUT CLOCK!"
INPUT 1 LEFT SELECT (REAR/FRONT)	When "ANALOG (IN1)" is selected as the Main Input Source (above), this determines whether the front INSTR input jack or the rear IN1 LEFT/MONO jack should be used as the LEFT channel of the input signal. This option defeats the jack that is NOT selected.
INPUT 1 MODE (LEFT ONLY/ L+R SUM/ STEREO)	This determines whether the signal received at INPUT 1 will be processed in stereo or mono, and, if in mono, whether the "LEFT ONLY" or a sum of both channels should be used. For typical "Guitar In/Processed Stereo Audio Out" applications, use default settings: **MAIN INPUT SOURCE** to ANALOG (IN1), **INPUT 1 LEFT SELECT** to FRONT, and **INPUT 1 MODE** to LEFT ONLY.

INPUT 2 MODE (LEFT ONLY/ L+R SUM/ STEREO)	This determines how the Axe-Fx II handles signals received at the balanced INPUT 2 ("FX RETURN") jacks, setting whether they should be processed in stereo or mono, and, if in mono, whether the "LEFT ONLY" or a sum of L+R channels should be used. The outputs of the connected device and the nature of the source material will determine which setting is best.
OUTPUT 1 MODE (STEREO/ SUM L+R/ COPY L>R)	This determines how OUTPUT 1 signals will be processed after the output mixer of the grid. This control makes it easy to use the same Axe-Fx II presets in a variety of stereo and mono performance or recording environments. The decision to use SUM L+R or COPY L>R should be based on the source material. See Mono and Stereo on p. **176** of the Appendix for more detail.
OUTPUT 1 BOOST/PAD (0-18 dB)	This is NOT a boost intended for use during musical performance, as you might find on an amp or pedal. Rather, this parameter is designed to boost signals to the OUT1 D/A converters so they can operate as close to full-scale as possible, while simultaneously padding converter outputs to lower the noise floor. To set, increase boost/pad until hard playing on a loud preset causes the front panel OUT1 CLIP LED to light. Then reduce the Boost/Pad setting a few dBs to prevent further clipping. Note that the BOOST amount is also applied to outgoing DIGITAL or USB signals.
OUTPUT 1 PHASE (NORMAL/INVERT)	This determines whether signal at the OUTPUT 1 jacks will be normal or phase-inverted relative to its state at the output of the grid. This lets you compensate for inversions elsewhere in the signal chain. (inverts BOTH channels of the stereo pair.
OUTPUT 2 MODE (LEFT ONLY/ L+R SUM/ STEREO)	This determines how signals will be processed after leaving the output mixer of the FX LOOP block, or after being created as a COPY of OUT 1 (see below). This control makes it possible to use the same FX-LOOP-block-based presets in either stereo or mono conditions, or to use OUT 2 to create a mono copy of the stereo signal appearing at OUT 1. See Mono and Stereo on p. **176** of the Appendix for more on using the Axe-Fx II in mono.
OUTPUT 2 BOOST/PAD	See OUTPUT 1 Boost/Pad, above.
OUTPUT 2 PHASE	See OUTPUT 1 Phase, above.
OUTPUT 2 ECHO NONE OUTPUT 1 INPUT 1	No, not as in repeats… repeats… repeats. Echo in this usage refers to an exact simultaneous *copy* of a signal. The default value "NONE" connects Output 2 to the inputs of the FX LOOP block (p. 70). Selecting "OUTPUT 1" creates an exact copy of the OUTPUT 1 signal at OUTPUT 2. Use this when you need to feed both front of house and full-range monitors, for example, and want independent level control over each using the front panel OUTPUT knobs. Selecting "INPUT 1" copies the raw unprocessed INPUT 1 signal to OUTPUT 2, and is ideal for capturing a dry track for "reamping" without using the FX Loop block or USB. Note that the echo feature will not function when an FX Loop block is present in a preset as the FX Loop has priority.
SPDIF/AES SELECT SPDIF/AES	This switch selects whether the S/PDIF or AES digital inputs and outputs are active. Only one may be active at any time.
SPDIF/AES SOURCE	This selects the signal to feed *both* the digital outs: OUTPUT 1, OUTPUT 2, INPUT
USB BUFFER SIZE	Set this to lower values for less latency, set to higher values if experiencing distorted audio. Low values generally work fine with Windows machines. OS X computers may need higher values. 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 or by disconnecting the USB cable.
USB RETURN LEVEL	This parameter sets the level of the USB input sent to the main outputs. You can lower this level to prevent excessively high level computer audio signals (approaching 0 db) from clipping the outputs of the Axe-Fx II.

9.3 MIDI Parameters

The MIDI page of the I/O menu contains parameters related to MIDI channel, thru, and program changes.

PARAMETER	Description
MIDI CHANNEL 1–16, OMNI	Sets the channel on which the Axe-Fx II will receive MIDI messages. OMNI causes the unit to respond to incoming messages on ANY channel.
MFC PORT (XL Only) Disabled/FASLINK/EtherCON	Set this to the appropriate port when you have an MFC-101 MIDI Foot Controller connected at either the "FASLINK" or "ETHERCON" port.
MFC ECHO TO MIDI OUT (XL Only) ON/OFF	MIDI signals received from an MFC-101 at the FASLINK or Ethernet/EtherCON jacks are never passed to the MIDI THRU. When you want to pass MIDI from an MFC-101 to a downstream device, enable this option to create a "soft" MIDI Thru at the MIDI OUT port of the Axe-Fx II XL/XL+.
MIDI THRU **(NON-XL Only)** OFF/ON	Turning this ON will cause MIDI data received at the MIDI IN port to be forwarded to the MIDI OUT port, where it is **merged** with regular outbound MIDI—SysEx dumps or Realtime Tuner/Tempo Sysex for example. Thru is disabled during firmware update.
PROG CHANGE ON/OFF	This parameter determines whether the Axe-Fx II will process or ignore incoming MIDI program change messages.
MAPPING MODE NONE/CUSTOM	This determines whether the Axe-Fx II will respond explicitly to MIDI program change messages received, or whether it will re-map incoming program change messages to load user-specified presets. For example, an incoming program change message of 15 would normally recall preset 15 (or preset 16 if DISPLAY OFFSET is on; see below). With Custom Mapping set up, program change message 15 might be set to recall preset 100, or any other preset you choose.
MAP FROM/TO PRESET/SCENE 0–127, 0–383, 1–8	These three parameters work to specify which preset and scene will be loaded for each incoming program change message when **MAPPING MODE** is set to "CUSTOM." For example, if you select "15" for **MAP FROM** and **MAP TO** values of "100," and "2" incoming Program Change #15 will load Preset 100, scene 2 (or 101:2 if display offset is on; see below).
SCENE REVERT	Selects between one of two behaviors for scene recall via MIDI. **OFF** (Default): Scene edits are **retained** across scene changes until you load an entirely new preset or reload the current one. If you tweak Scene 1, switch to Scene 2, then back to scene 1, your tweaks will remain intact. **ON:** Scene edits are **lost** when you change the scene without saving first. 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.
SYSEX ID (INFO ONLY)	This is **00 01 74** for the Axe-Fx II and cannot be modified.
DISPLAY OFFSET 0/1	Display Offset causes Axe-Fx II presets to appear to be numbered from 001 instead of from 000. The offset does not change which preset is recalled by a given Program Change message. *Note: Display Offset requires a corresponding setting in a connected MFC-101 MIDI Foot Controller.*

IGNORE REDUNDANT PC OFF/ON	This determines whether the Axe-Fx II should re-process or ignore a MIDI program change command for the currently loaded preset. With this setting OFF, the current preset will be reloaded (and changes discarded) if it is selected again via MIDI PC. Turning this OFF allows you, for instance, to load a preset, use various MIDI “instant Access” switches or pedals to bypass effects or otherwise modify the sound, and then stomp on the footswitch that originally selected the preset to have it revert to its saved state. With this setting ON, redundant PC messages are ignored.
SEND REALTIME SYSEX NONE/ALL/TUNER/TEMPO	Selectively determines whether real-time SysEx messages for TUNER and TEMPO will appear at the MIDI OUT port. The default setting of ALL ensures that a connected MFC-101 Midi Foot Controller will be able to show the Axe-Fx II Tuner on the floor, and that its TEMPO LED will flash in time with the current Axe-Fx II system tempo.
USB ADAPTER MODE ON/OFF	This setting changes the way MIDI data is handled between the computer and the physical MIDI IN/OUT and MFC ports of the Axe-Fx II. When Set to OFF: 1. Inbound **MIDI** data at the MIDI IN or MFC ports is processed by the Axe-Fx II. 2. Inbound **MIDI-over-USB** data sent via the *AXE-FX II MIDI OUT* port of a connected computer is processed by the Axe-Fx II. When Set to ON: 1. Inbound **MIDI** data at the MIDI IN or MFC ports is processed by the Axe-Fx II and also forwarded to the *AXE-FX II MIDI OUT* port of a connected computer. a. Use this so your host sequencer/DAW can record the MIDI data generated by an expression pedal connected to the MFC-101. b. Use this to connect a MIDI keyboard or other device to the MIDI IN port of the Axe-Fx II and play plugins or record performances in your host sequencer/DAW. 2. Inbound **MIDI-over-USB** data sent via the *AXE-FX II MIDI OUT* port of a connected computer is processed by the Axe-Fx II and also forwarded to the MIDI OUT and MFC ports. a. Use this so your host sequencer/DAW can also control devices chained at the MIDI OUT/THRU port of the Axe-Fx II: synth modules, 3rd-party processors, etc. b. Use this to send SysEx backups or firmware updates to an MFC-101 connected at the MFC port. Note that the USB ADAPTER MODE function is completely independent of the setting for MIDI THRU (above). It is therefore possible, with both settings turned ON, for the physical MIDI OUT/THRU port to sum/output **four different sources** simultaneously: 1) The Axe-Fx II regular MIDI out functions; 2) events received at the MIDI IN port; 3) events received at the MFC port; and 4) events received via *AXE-FX II MIDI OUT* from a connected computer. Adapter mode is disabled during firmware updates.

MIDI PC OFFSET 0–255	Adds the value specified to all incoming MIDI Program Change requests before they are processed. This makes it possible, for instance, for the same foot controller programs to access presets 0-127 for one gig, presets 128-255 for another, and presets 256-383 for a third.
EXT CTRL # INIT VAL 0% OR 100%	This specifies an initial value to be used for each of the 12 External Controllers (p. **144**) until remote data is received. This also applies when the controller is absent. For example, if you normally use an expression pedal to control the **VOLUME** block in a particular preset, the absence of that pedal might mean that the preset gets "stuck" in a silent 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 be all the way up instead of all the way off.

9.4 Control Parameters

The Control page ("CTRL") of the I/O menu allows external controllers to be assigned to onboard functions. Besides extensive MODIFIER capabilities (p. **136**), the Axe-Fx II also has 100+ dedicated functions that can be assigned for remote control via MIDI CC messages or the onboard PEDAL jack. These include global input and output volumes, tap tempo, tuner, the bypass functions for every individual block in the inventory, the block X/Y switches, the Looper controls, and more.

To assign a controller to a desired item, select it and turn the VALUE dial to select a MIDI CC number, or choose "PEDAL 1" or "PEDAL 2" for the onboard jacks. (An Axe-Fx II Mark II or older has only one pedal jack.) Set to NONE to disable remote control. The Axe-Fx II also has a "learn" feature that allows it to detect control sources automatically. Select the desired item, press ENTER, use the controller to send some data, and the source will be set automatically. This is also a good way to ensure that remote devices are in fact transmitting correctly.

Unless otherwise noted, all functions interpret MIDI CC values of 0–63 as "OFF" and 64–127 as "ON."

PARAMETER	Description
INPUT VOLUME NONE/PEDAL…/0-127	This controls the global input volume at the start of the preset chain and is useful to assign when you want to emulate the behavior of a volume pedal between the guitar and the amp. Changes in level equate to changes in distortion amount, and effect tails ring out when volume is lowered.
OUT 1 VOLUME NONE/PEDAL…/0-127	This controls global OUTPUT 1 volume (after the preset's output mixer) and is useful to assign when you want to control playback levels without changing any other aspect of the sound. Amplifier input levels are unaffected for consistent distortion at any level, and effect tail levels will be adjusted as well.
OUT 2 VOLUME NONE/PEDAL…/0-127	This controls global OUTPUT 2 volume (after the output the FX Block, or after it is copied from OUT1, see section **9.2**). This is useful when you want to control playback levels for OUT1 and OUT2 independently, without changing any other aspect of the sound, or if you want a pedal to control the SEND level to an outboard device in the loop.

BYPASS NONE/PEDAL…/0-127	Operates the front panel **BYPASS** button (p. **8**) remotely.
TEMPO TAP NONE/PEDAL…/0-127	Provides the ability to set the Global Tempo (see p. **158**) through a remote switch. **IMPORTANT:** Any data value (0-127) for the assigned CC# counts as a tap, so do not use a momentary switch set up to send 127 for ON and 0 for OFF, or you will end up with double-time!
TUNER NONE/PEDAL…/0-127	Provides a way to enter or exit the TUNER function remotely.
EXT CTRL 1–12 NONE/PEDAL…/0-127	This is where you specify which incoming MIDI CC# should be assigned to each of the 12 External Controllers available as Modifier sources (p. **144**).
LOOPER REC, PLAY, ONCE, DUB, REV, BYPASS, HALF, UNDO, METRONOME	Each of the controls of the Looper block (p. **80**) may be remote controlled.
SCENE SELECT, **SCENE INCR,** **SCENE DECR**	These options allow selecting SCENES (p.184) via MIDI CC#. For SCENE SELECT, the value of the Control Change message determines which scene is loaded. The easy rule is CC# Value + 1 = Scene number. (Ex: Value 0 = Scene 1). However, values greater than 7 continue to select scenes, in order. To determine the scene for values >7, divide by 8. The REMAINDER is the scene that will be selected. SCENE INCR and SCENE DECR are triggered only by CC# data values greater than 63.
VOLUME INCR, **VOLUME DECR** NONE/PEDAL…/0-127	These two options provide a convenient way to **permanently** increase or decrease the MAIN volume (OUTPUT 1) of the currently loaded preset. Each time VOL INCR is triggered by a CC# value greater than 63, the MAIN is increased by 1.0 dB and the preset is saved. VOLUME DECR, works the same way, decreasing volume. NOTE: The MAIN fader is located on the OUT page of the Layout menu. **IMPORTANT!** Any other unsaved changes such as altered effect parameters or bypass states will also be stored if either **VOLUME INCR** or **VOLUME DECR** is triggered. **WARNING!** 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!
AMP1 BYP **AMP 2 BYP** **CAB 1 BYP** … THROUGH… **WAH 2 BYP**	Every block instance in the inventory of the Axe-Fx II except FB SEND, FB RETURN, and SHUNT can be set up with a global, dedicated MIDI CC assignment to control its bypass state. The complete list of default effect bypass assignments can be found in the **Factory Defaults** section of the Appendix.
AMP1 XY **AMP 2 XY** **CHO1 XY** … THROUGH **WAH 2 XY**	The X/Y switching feature equips various Axe-Fx II blocks with dual independent sets of parameters, making it possible to have two completely different switchable settings for a single block. Data values from 0–63 will select the Y state, while values from 64–127 will select X. See section **4.4** on p. **36** for more on X/Y switching.

9.5 Pedal Parameters

The PEDAL page of the I/O menu contains parameters to set up and use an expression pedal or switch through the PEDAL jack(s) on the rear panel of the Axe-Fx II. (The Axe-Fx II original and Mark II models have only one pedal jack; XL and XL+ models have two). Simply connect the pedal, set its type, and calibrate. Any PEDAL may be assigned to any of the items listed in the CONTROL page of the I/O menu (p. **152**) Expression pedals should have a linear resistance taper and max resistance of 10kΩ to 100kΩ, and 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 1,2 TYPE CONTINUOUS/ MOMENTARY/ LATCHING	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.
PRESET INCR OFF/ON	When this is turned ON, a connected at the PEDAL jack may be used to increment presets.
PRESET START, **PRESET END**	These set the start and endpoints for a cycle of presets that may be stepped through by a connected footswitch or pedal when PRESET INCR is on. When the series reaches the END, it starts from the beginning again. Range is limited to the first 128 presets.
PEDAL 1,2 CAL	To calibrate an expression pedal connected to an onboard 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.

9.6 X/Y Quick-Jump Assign

The QUICK-JUMP page of the I/O menu contains two parameters: **X QUICK-JUMP ASSIGN** and **Y QUICK-JUMP ASSIGN.**

In addition to their functions for X/Y Parameter switching (p. **36**) the X and Y buttons are used for one-touch access to two edit menus of your choice. The EDIT menus of the blocks specified here will be opened instantly when you press X or Y in any screen of any menu except **EDIT** (including MODIFIER and SAVE/LOAD GLOBAL BLOCK) or **STORE**.

10 Utilities

The Utilities section, accessed by pressing the front panel UTILITY button, contains five pages of menus containing settings and tools that do not affect the sound or routing of the Axe-Fx II.

10.1 LCD Contrast

The **LCD** page of the UTILITY menu contains a single adjustment slider, used to set the contrast of the built-in display to ensure good readability in different viewing environments.

10.2 Preset Utilities

The **PRESET** page of the UTILITY menu contains dump and backup utilities. Each of the options listed is executed by selecting it and pressing ENTER to transmit or "dump" the selected memory area to an external device for backup, editing, or other purposes. Be sure to note that dumps to MIDI and USB are initiated with separate menu choices.

Backup and restore operations are also detailed in **Section 13** beginning on p. **160.**

- **STRIP ALL GLOBAL DATA** removes all Global Block links from the *current* preset. Only the links are removed; parameter values are not affected.
- **SET AMP(S) TO DEFAULTS** - This resets all advanced parameters in each amp block in the current preset to default values. Tone controls and Drive levels are not altered.
- **UPDATE AMPS ALL PRSTS** – Same as above, but acts on all amps in all presets.
- **ERASE ALL PRESETS** – (XL Only) – Completely erases all presets in the unit.
- **ERASE ALL CABS** – (XL Only) – Completely erases all user cabs in the unit.

10.3 Status Meters

The Status meter page of the UTILITY menu contains audio meters for Input 1 Left and Right (L1, R1), Input 2 Left and Right (L2, R2), Output 1 Left and Right (L1, R1), and Output 2 Left and Right (L2, R2). The scale is -80 to 0 db.

The "USB" bar graph displays the amount of data in the USB FIFO buffer. Ideally this should be around 50%. If the buffer overflows or underflows, the **USB BUFFER SIZE** (p.**148**) should be increased. The number of buffer errors that have occurred since the last reset is also indicated above the graph.

The front panel also shows the levels for Input 1 L+R and Input 2 L+R on stereo LED meters.

To the right of the I/O section, CPU Utilization (CPU%) is shown on its own meter. The total CPU usage must not exceed 98% or the entire system could destabilize. The Axe-Fx II works to prevent this condition from occurring. See **Understanding Preset Size Limits** on p. **173** for more information on this subject.

10.4 **VU Meters**

The VU Meter page of the main menu shows preset sound levels in a horizontal meter, scaled from -20 to +10 dB with a line to mark 0 dB. The levels of one or both amp blocks in your preset can be adjusted using the A and B knobs. The value of the Output Level knobs is displayed for reference only and does not affect the VU meters as the measurement is prior to the Output Level potentiometers.

Note: It is generally recommend to use the Amp block to set levels for consistency, but if you are using a complex grid routing, however, or if there are *level-dependent* blocks AFTER your amp, you'll need to adjust levels elsewhere. (Level-dependent blocks include Compressor, Drive, Gate, and possibly the FX LOOP if the loop contains outboard gear) Adjust the last of these blocks in your signal chain for best results.

If you need to set levels PER SCENE, see “To Set the Main Output Level for a Scene:” on p. 43.

10.5 **Reset System**

This menu page includes a single parameter that is used to restore the factory default settings to **System** Parameters, specifically:

- GLOBAL Configuration parameters, Global OUT1 and OUT2 settings.
- IO Input, Audio, MIDI, Control and Pedal Parameters.
- LCD Display Contrast.

The following areas are NOT affected by **RESET SYSTEM PARAMS:**

- User scale settings are NOT modified.
- User Cabs are NOT affected.
- Global Blocks are NOT affected.
- Preset memories are NOT affected.
- Firmware is NOT affected.

To Reset System Parameters, select this menu page, then press **ENTER.** A dialog will prompt you to “RESET ALL?” Press **ENTER** to execute the reset.

10.6 **IR Capture**

The Axe-Fx II includes an onboard capture utility. This is now covered by a separate manual, the **IR CAPTURE GUIDE**, available at http://www.fractalaudio.com/support

10.7 **Firmware**

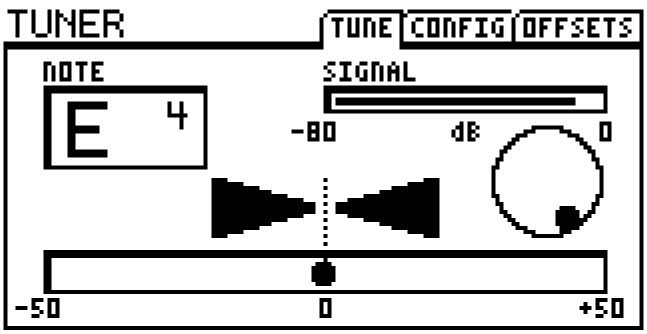
The firmware page in the UTILITY menu utility shows you the **VERSION** of the currently loaded firmware, and includes a function to place the unit into an UPDATE mode. Note that Fractal-Bot automatically enters UPDATE mode, meaning you don’t need to press ENTER.

See Section **14** on p. **162** for details on how to update firmware.

11 Tuner

The Axe-Fx II includes a state-of-the art onboard TUNER, an essential tool for the performing or recording musician. It is easy to operate and features high-resolution automatic pitch detection, a calibration control, offsets for modified tuning schemes, and the option to mute audio while tuning. The TUNER button shows the tuner. The PAGE buttons allow access to the tuner **CONFIG** and **OFFSET** pages. To exit the tuner screen, press EXIT or RECALL. The tuner may be set up via the CONTROL page of the I/O menu (p. **152**) to be remotely engaged/disengaged using MIDI or the onboard Pedal Jack.

Firmware 6.0 increased the range of the Axe-Fx tuner so it is equally well suited for bass or drop-tune guitars, and also added a spinner that rotates clockwise when the note being tuned is sharp and counterclockwise when flat.



Configuration Parameters

PARAMETER	Description
CAL 430.0 – 450.0 Hz	This calibrates the tuner by setting the frequency of A4 (in the 8va above middle C).
MUTE OFF/INPUT/OUTPUT	This setting determines whether the main inputs or outputs will be MUTED when the tuner is engaged.
USE OFFSETS OFF/ON	Determines whether the settings on the OFFSET page (see below) are applied or ignored.

Offset Parameters

PARAMETER	Description
E1, B2, G3, **D4, A5, E6** +/-12.7 Hz	Offsets allow the tuner to be calibrated so individual notes diverge from standard concert tuning by a defined amount.

12 Tempo

Tempo is used in electronic music for synchronizing different rates and times, whether inside one machine or across different devices. The **Global Tempo** of the Axe-Fx II allows for both internal and external synchronization, providing effects with a central BPM clock that can run autonomously or march to the rhythm of an upstream device outputting MIDI Beat Clock. The tempo is available to control a variety of rates and times in the Axe-Fx II.

12.1 Setting the Tempo

The **Global Tempo** may be set to any whole number value in the range 30 BPM (*grave*) to 250 BPM (*prestissimo*). The current tempo is shown by a flashing LED inside the TEMPO button of the Axe-Fx II front panel, and it will also flash on the tempo footswitch of a connected MFC-101 MIDI Foot Controller.[1]

To set the tempo, tap two or more times on the front panel TEMPO button, or press the button once and adjust the **TEMPO** knob that appears in the onscreen display. By default, the tempo averages across ten taps, but you can also set this to two using an option found under Global: Config: Tap Tempo (p.**145**)

The tap function can also be controlled remotely by assigning a **MIDI CC#** or a footswitch connected to the onboard PEDAL jack to **TAP TEMPO** in the Control page of the I/O menu (p. **152**).

The **Global Tempo** will automatically synchronize to MIDI Beat Clock if it is detected at the MIDI IN port or in the incoming MIDI stream of the USB interface. The Axe-Fx II does not recognize MIDI Time Code or SMPTE, nor does it transmit MIDI Beat Clock.

12.2 Synchronizing Sound Parameters

Rates and times in a preset can be made to sync rhythmically to the **Global 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” to follow the quarter note pulse of the tempo, find the **TEMPO** parameter in the delay’s EDIT menu (it is on **PG2**) 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.

The following parameters may be synchronized to the **Global Tempo**:

- **Chorus**, **Flanger**, **Phaser** and **Tremolo** Modulation Rates.
- **Delay**: All Delay Times and Modulation Rates.
- **Multi-Delay**: All delay times, All Modulation rates, Rhythm Tap Quantization value.
- **Pitch**: All delay times, Crystal Splice time, Arpeggiator tempo.
- **Controllers**: Global LFO 1 and LFO 2 rate, Sequencer Rate.

[1] This requires that the SEND REALTIME SYSEX option in the MIDI page of the I/O menu be set to “ALL” or “TEMPO.”

Tempo-synchronized sound functions track changes to Global Tempo in real time, increasing or decreasing as it speeds up or slows down.

12.3 Tempo to Use

Every preset contains two saved **TEMPO** settings: an actual **TEMPO** (BPM) and a second setting called **TEMPO TO USE.**

If a preset's **TEMPO TO USE** parameter is set to "PRESET," the **Global Tempo** will change to the saved BPM value whenever that preset is loaded. Factory presets in the Axe-Fx II are saved with a tempo of 120 BPM and a **TEMPO TO USE** setting of "PRESET."

If a preset's **TEMPO TO USE** parameter is instead set to "GLOBAL," its saved BPM value will be *ignored*, and the current Global Tempo will be used.

12.4 Auto Delay

When set to "ON," any delay blocks that are bypassed will become active whenever a tempo is tapped in. This allows you to set the tempo and un-bypass your delay block(s) from a single footswitch.

12.5 Metronome

The Axe-Fx II features a built in metronome whose signal is mixed in at Output 1.

METRONOME – Turns the metronome on or off.

METRO LEVEL – Sets the level of the metronome from +/- 20db

13 Backing Up and Restoring

The best way to backup your Axe-Fx II is to a computer using Fractal-Bot, which automates the entire process. Previous firmware versions provided the capability to manually dump and receive presets, banks and system files via MIDI or USB, but this was removed to encourage the use of Fractal-Bot.

13.1 Onboard ROM Backup and Restore

The Axe-Fx II original and Mark II versions (but not the XL or XL+) contain an internal FLASH ROM where PRESET BANKS can be stored for speedy recovery without a computer. ALL models provide this capability for backing up or restoring SYSTEM settings. The contents of this FLASH ROM are not overwritten or modified during firmware updates or when you load presets, banks, or the system from computer backup files.

Select the appropriate entry on the **UTILITY:PRESET** page and press ENTER to begin the backup or restore.

Warning! Both BACKUP and RESTORE operations are permanent and cannot be UNDONE. When you BACKUP to the onboard FLASH ROM, the old contents of the selected area will be irrevocably overwritten by the new backup. When you RESTORE from the FLASH ROM, the selected Preset Memories or System Settings of the Axe-Fx II will be overwritten by the restored data and cannot be recovered.

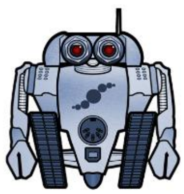
Warning! Never interrupt a FLASH ROM Backup or Restore operation in process, or data loss/corruption could occur. Do not unplug or power off the unit during BACKUP or RESTORE.

- **BACKUP BANK A, B, or C (Axe-Fx II Mark II Only!)** copies a group of 128 presets to the onboard backup ROM. Bank A contains presets 0-127, B contains 128-255, and C contains 256-383 (or 1-128, 129-256, and 257-384 when **DISPLAY OFFSET** [p. **150**] is turned ON). A meter indicates progress. It takes less than 10 seconds for a backup to complete.
- **BACKUP SYSTEM (every Axe-Fx II)** copies all of the system settings of the Axe-Fx II to the onboard Backup. Specifically, this includes:
  - All **GLOBAL** and **I/O** parameter settings
  - All **TUNER** settings
  - **USER CAB** IRs 1-50 (*Axe-Fx II Mark II ONLY – User Cabs are not part of an XL System dump!*)
  - All **GLOBAL BLOCKS**
- **RESTORE USER BANK A, B, or C (Axe-Fx II Mark II Only)** reads a group of 128 presets from the onboard backup ROM and writes these into the onboard preset memories of the Axe-Fx II when you press ENTER to execute the selected function. Bank A contains presets 0-127, B contains 128-255, and C contains 256-383 (or 1-128, 129-256, and 257-384 when DISPLAY OFFSET (p. **150**) is turned ON.
- **RESTORE SYSTEM** reads all of the non-preset settings of the Axe-Fx II from the onboard Backup ROM and writes these into their respective areas when you press ENTER. This specifically includes
  - All **GLOBAL** and **I/O** parameter settings
  - All **TUNER** settings
  - **USER CAB** IRs 1-50 (Cabs 51-100 are not included and must be restored separately.)
  - All saved **GLOBAL PRESET** settings

- **RESTORE FACTORY BANK A, B, or C (Axe-Fx II Mark II Only)** reads from a separate different ROM memory which always contains a fresh copy of the factory preset banks regardless of what changes you save to the preset memories or backup ROM. These otherwise operate the same as **RESTORE USER BANKS,** above.
- **FETCH BACKUP PRESET (Axe-Fx II Mark II Only)** loads an individual preset from the USER BANKS in ROM into the temporary edit buffer of the Axe-Fx II. You need to STORE this preset manually.
- **FETCH FACTORY PRESET (Axe-Fx II Mark II Only)** is the same as the above, except the preset is recalled from the FACTORY ROM.

14 Firmware Updates

Firmware is the built-in software that gives the Axe-Fx all of its features, functions and capabilities. New firmware updates are made available through our website http://www.fractalaudio.com/support. The firmware page in the UTILITY menu utility shows you the **Version** of the currently installed.



Updating using Fractal-Bot

Fractal-Bot is hands-down the best way to upgrade firmware. This application for Mac or Windows enables you to easily transmit updates quickly and easily. It’s small, lightweight, and has built-in step-by-step instructions.
Visit http://www.fractalaudio.com/fractal-bot.php to download Fractal-Bot.

Updating with a 3rd Party MIDI Utility or Sequencer

It is also possible to update using a legacy MIDI application, for instance if you can’t install the Axe-Fx II driver.

- Connect your computer MIDI interface OUT to the MIDI IN of the Axe-Fx II.
- Launch your MIDI Utility and prepare it to send the firmware SysEx file.
- On the Axe-Fx II, press **UTILITY**. Select the FIRMWARE page. Press **ENTER**.
- Transmit the file from the computer to the Axe-Fx II. A progress bar will appear on the Axe-Fx II while it receives the firmware. It will take several minutes to transfer the file.
- If all goes well, the Axe-Fx will display "GOOD CHECKSUM” and will then erase, flash and reboot.
  - If a firmware update fails, you may need to reboot the Axe-Fx II and try again after slowing the MIDI transmit rate of your application.

14.1.1 Axe-Fx II XL/XL+ Failsafe Firmware

The Axe-Fx II XL/XL+ has a built-in recovery mechanism to protect against firmware update issues. (The original and Mark II models do not include this feature. Contact support if you require assistance.) In the rare event that an error occurs during firmware update and your Axe-Fx II XL/XL+ will not boot, please perform the following steps:

- While holding both Page buttons, power your Axe-Fx II OFF and then ON.
- Once the unit boots, release both buttons.
  The unit will boot into failsafe update mode and a special UTILITY menu will be shown.
  WARNING: Do NOT execute any of the <ENTER> functions on this page or data loss may occur!
- Press PAGE LEFT to select the FIRMWARE page. It will show “Version 0.00”.
- Start Fractal-Bot and update the firmware as you normally would.
- If you require assistance, please visit http://support.fractalaudio.com

14.1.2 Firmware Compatibility

Presets created under different firmware versions are generally forward compatible. You can load older presets in newer firmware version (though they may not sound as intended; see firmware Release Notes for details). Updates are otherwise compatible with each other with one exception: due to space limitations in the Mark I/II boot ROM, firmware versions *newer* than Quantum 3.03 are NOT compatible with **presets** that were last saved prior to firmware version 15.0. You must first load and save such presets in an intermediate version such as Quantum 3.0

15 Troubleshooting

Fractal Audio Systems offers support through its web site at http://www.fractalaudio.com
You can also get answers to most questions in our online forum at http://forum.fractalaudio.com
The Axe-Fx Wiki at http://wiki.fractalaudio.com/axefx2 is also an excellent source of information.

Here are some frequently asked questions that might help you with very basic issues.

Q: How do I connect the Axe-Fx II to my computer?
A: Install the driver, available in the support section of our web site, and then connect the two devices together with a standard USB cable. The Audio and MIDI ports will appear in your applications that support these functions.

Q: Can I use the Axe-Fx II with MIDI controller "Brand X"?
A: MIDI is MIDI. You won't get all of the deep integration features provided by "Axe-Fx Mode" on the MFC-101, nor will you be able to connect via FASLINK™ or Ethernet/CON, but any device that can send MIDI program and control change messages can interface with the Axe-Fx II.

Q: How do I set up a pedal to control a Wah?
A: See the tutorial on page **179**.

Q: My Axe-Fx II is behaving erratically or "froze up" while I was using it. What should I do?
A: First try a simple reboot. If that doesn't work, Disconnect MIDI, USB and MFC and hold RECALL while you power on the unit. This will load an EMPTY preset, but won't affect any of your saved settings.

Q: One or more of my presets produces no sound.
A: This might be any one (or several) of a number of things:

- Is everything still wired up correctly? Most of the time, the problem is a faulty or disconnected cable!
- Have you double-checked to ensure that you have a complete path from the input to the output? See p. **30**.
- Is your guitar plugged into the front "INSTR" jack while **INPUT 1 LEFT SELECT** is set to "REAR" or vice versa? (See I/O : AUDIO on p. **148**.)
- Is there a MODIFIER assigned to a volume or level control while the pedal or external switch is not present? Find and remove that modifier (p. **137**) or change its **EXT CTRL INIT VALUE** from 0% to 100% (p. **150**).
- Does the preset require a USER CAB which is not loaded? Try changing the CAB block to a Factory cab.

Q: The amps I've dialed in sound "wrong."
A: The amp simulations of the Axe-Fx II are extremely accurate and should sound instantly familiar to those who know the original amplifiers that inspired them. If things don't sound correct, a few quick checks can confirm that basic settings are correct. First, check the speaker cabinet type to make sure it is an appropriate match. You won't get typical "DAS METALL" sounds from a 1x10 speaker! Next, consider the setting of the **MASTER**. If the original amp had no master, try a high setting (8+) and then dial in the amount of distortion you want using the **DRIVE**

parameter. Check the CONFIG page of the Global menu to make sure that power amp and cabinet simulations are set correctly (see Chapter 3). Finally, consider trying default settings and the headphone jack for reference.

Q: All my presets suddenly sound harsh and bright. What's going on here?
A: Check the CONFIG page of the Global menu (p. **145**) to ensure that the Global Power Amp and Cab Simulations have not been turned off. If a single preset is affected, make sure you haven't set the SAG parameter of its AMP block(s) fully counterclockwise. The Global Graphic EQ (p. **146**) may also have an unusual setting.

Q: Many of my presets suddenly sound hollow and phasey. What could cause this?
A: These terms are sometimes used to describe the sound of problems caused when stereo sounds have been summed to mono. Check the AUDIO page of the I/O menu to ensure that the **OUTPUT1 MODE** and **OUTPUT 2 MODE** are set correctly according to how you are listening to your Axe-Fx II. Listening with headphones while making this change can also confirm or rule out this possible problem.

Q: The sound is distorted even on what should be "clean" presets.
A: Check the input and output levels of the Axe-Fx II and check the input levels of any downstream amps/monitors. See p. **15** for more on how to set levels. Also, try the obvious and turn **DRIVE** even lower on the amp, or consider adjusting global **AMP GAIN** on the CONFIG page of the GLOBAL menu.

Q: The front panel CLIP LED is coming on. What does this mean?
A: The two CLIP LEDs indicate whether the D/A converters are being overdriven. See p. **15** for more on how to set levels. Adjust the output levels of your presets, or reduce the global BOOST/PAD setting for the offending output. In a pinch, you can also lower the GAIN setting on the Global Graphic EQ.

Q: My screen is blinking EXCESS CPU UTILIZATION! REDUCE LOAD.
A: First, try undoing the last change you made by turning the value wheel in the opposite direction. You can also return to the grid, navigate around the message, and remove one or more blocks. You can also load a different preset or reboot the unit. Changing CAB mode to "MONO LO-RES" is also a good solution in many cases.

Q: I am pressing FX BYP, but the state of the selected block will not toggle. Why is this not working?
A: The BYPASS MODE parameter, available on almost all blocks, has a **modifier** slot to control bypass state. Once attached to this slot, the modifier assumes total control over the block's bypass switch. Either utilize the MODIFIER to change the bypass state, or remove it.

Q: I loaded a preset from a backup and it sounds nothing like it did when I first saved it. What's wrong?
A: The most likely scenario is that something else changed: the firmware version, guitar, amp, or perhaps the settings of external controllers. It is also possible that the preset uses Global Blocks that have changed.

Q: When I play audio from the computer through the Axe-Fx II, the sound is processed by the effects, and I can't hear the guitar. What's wrong?
A: Change the MAIN INPUT SELECT from USB to ANALOG (IN 1). This will allow you to play along with audio from the computer.

Q: Can I use the onboard MIDI ports to connect a keyboard/synthesizer/etc. to my computer?
A: Yes. Just remember to turn on **USB ADAPTER MODE** in the MIDI page of the I/O menu. See p. **150**.

Q: My Axe-Fx is warning me about presets not being installed correctly.
A: This rare error can be corrected through a series of simple steps. Visit http://support.fractalaudio.com

16 Appendix

The following material is designed for reference and aims to maximize your enjoyment of the Axe-Fx II. Please also visit our online forum at http://forum.fractalaudio.com for discussions on these and many other subjects.

16.1 Table of Amp Types

Amp Types in the Axe-Fx II are exacting models of the actual amps they are based on. Our approach is commonly referred to as “physical modeling,” where the individual components are modeled and assembled so the virtual model reproduces every aspect of how the real amp sounds and works as a system. Amplifier Types are subject to change without notice. Please check firmware release notes.

If you’re unfamiliar with the models, the Fractal Audio wiki, a publicly managed document, contains tons of helpful information, especially this guide by *Yek*, an illustrious member of the Fractal Audio community:
http://wiki.fractalaudio.com/axefx2/index.php?title=Yeks_Guide_to_the_Fractal_Audio_Amp_Models

Manufacturer names and product names mentioned below are trademarks or registered trademarks of their respective owners, which are in no way associated with or affiliated with Fractal Audio Systems. The names are used only to illustrate sonic and performance characteristics of the Fractal Amplifier TYPES, which have been created by incredibly detailed analysis of the actual amps that inspired them.

MODEL NAME	BASED ON	NOTES
5F1 Tweed	Fender® Champ®	This "practice amp" from the Tweed era exhibits a unique breakup characteristic.
5F8 Tweed	Fender® Twin®	5F8 Tweed: based on Keith Urban's own #1 high power Fender Twin-Amp
6G4 Super	Fender® Super 6G4	Based on a pre-CBS 1964 blackface Fender Super Reverb, AB763 circuit
6G12 Concert	Fender® Concert 6G12	Based on a brown-era '60 Fender Concert with a 6G12 circuit.
59 Bassguy	1959 Fender® Bassman®	A low-to-medium gain amp designed for bass but more widely adopted by guitarists.
65 Bassguy Bass	1965 Fender® Bassman®	The bass channel of an AB165 Bassman.
65 Bassguy Nrml	1965 Fender® Bassman®	The normal channel of an AB165 Bassman.
1959SLP Jump	Marshall® 1959 Super Lead Plexi	Marshall 100W “Super Lead Plexi” model no. 1959, both Treble and Normal Channels "jumpered"
1959SLP Normal	Marshall® 1959 Super Lead Plexi	Normal channel of a Marshall 100W “Super Lead Plexi” model no. 1959
1959SLP Treble	Marshall® 1959 Super Lead Plexi	The treble channel of a Marshall 100W “Super Lead Plexi” model no. 1959
1987X Jump	Marshall® 1987x Vintage Series	Both Treble and Normal channels as when “jumpering the inputs” of this amp.
1987X Normal	Marshall® 1987x Vintage Series	Features what many consider to be an “essential” mod to the tonestack of this Plexi.
1987X Treble	Marshall® 1987x Vintage Series	The treble channel of the 1987x Vintage Series Plexi.
5153 50W Blue	Fender® EVH® 5150III®	Red is the “High Gain” channel of this amp.
5153 100W Blue	Fender® EVH® 5150III®	Blue is the “Medium Gain” channel of this amp.
5153 100W Green	Fender® EVH® 5150III®	The 50W Model.
5153 100W Red	Fender® EVH® 5150III®	Based on the clean channel of a “holy grail of modern tone.”
AC-20 12AX7 B	Morgan AC20 Deluxe	Rear switch set to 12AX7 and the Bass/Treble switch set to Bass.
AC-20 12AX7 T	Morgan AC20 Deluxe	Based on the Treble channel with the EF86/12AX7 switch in the 12AX7 position.
AC-20 EF86 B	Morgan AC20 Deluxe	Input tube in EF86 position, Normal mode.
AC-20 EF86 T	Morgan AC20 Deluxe	Treble channel of the same.
Angle Severe 1	Engl Savage	Modeled with the contour switch off.
Angle Severe 2	Engl Savage	Modeled with the contour switch depressed.
Atomica High	Cameron Atomica	Based on the Atomica from Cameron Amps
Atomica Low	Cameron Atomica	The low gain channel of the same.
Band-Commander	Fender® Bandmaster®	Based on an Silverface with the AB763 circuit
Big Hair	Created by Fractal Audio	Mids without mud. Revive the 80s metal scene. (Spandex not included.)
Blanknshp Leeds	Blankenship Leeds	Based on a boutique version of an 18W Marshall with a big sound at low power.
Bludojai Clean	Bludotone Ojai	The Ojai is reported to be an exact clone of Robben Ford's Tan Dumble.® Clean channel.
Bludojai LD 1	Bludotone Ojai	Lead channel of the Ojai with PAB on.
Bludojai LD 2	Bludotone Ojai	Lead channel of the Ojai with PAB off.
Bogfish Brown	Bogner Fish preamp	Based on a Bogner Fish preamp, Brown Channel.
Bogfish Strato	Bogner Fish preamp	Based on a Bogner Fish preamp, Strato Channel.
Boutique 1	Matchless Chieftain®	A medium-gain amp: thick, yet crisp, with a fair amount of power amp breakup.
Boutique 2	Matchless Chieftain® + Boost	Based on the same amp but with a boost for more gain and high-frequency emphasis.
Brit 800 Mod	Marshall® JCM 800®	Based on an 800 modded to be "heavier" and "less strident"
Brit 800	Marshall® JCM 800®	Based on the vaunted model 2204. Bring the Master up for true 80s tone.
Brit AFS100 1	Marshall® AFD100SCE®	Based on a Marshall AFD100SCE with #34/AFD switch in the #34 mode (LED off)
Brit AFS100 2	Marshall® AFD100SCE®	Based on a Marshall AFD100SCE in AFD mode (LED on)
Brit Brown	“The Brown Sound”	A faithful recreation of the legendary “Brown Sound” —The modded “#1” Marshall®.

Brit JM45 Jump	Marshall® JTM45® Jumpered	Based on this popular high gain 4 channel 100W head.
Brit JM45	Marshall® JTM45®	Made famous by Clapton and others; actually a modified Bassman® design.
Brit JVM OD1 Gn	Marshall® JVM 410®	The OD1 green channel of this popular amp.
Brit JVM OD1 Or	Marshall® JVM 410®	The OD1 channel in Orange Mode (more gain).
Brit JVM OD1 Rd	Marshall® JVM 410®	The OD1 channel in Red Mode (even more gain).
Brit JVM OD2 Gn	Marshall® JVM 410®	The OD2 channel in Green Mode, with lower mids than OD1.
Brit JVM OD2 Or	Marshall® JVM 410®	The OD2 channel in Red Mode, with even more gain and lower mids than OD1.
Brit JVM OD2 Rd	Marshall® JVM 410®	The OD2 channel in Orange Mode, with more gain and lower mids than OD1.
Brit Pre	Marshall® JMP-1 Preamp®	Based on a rack-mount preamplifier version of the Brit 900. Crunchy “ZZ” tone.
Brit Silver	Marshall® Silver Jubilee	Based on a distinctive commemorative “25/50” anniversary 100W British amp model
Brit Super	Marshall® AFD100®	Based on a Marshall® AFD100 (AFD = “Appetite For Destruction”)
Buttery	Budda® Twinmaster	Based loosely on a late 90s specimen. Relies mostly on power amp distortion.
CA OD-2	CarolAnn OD2r®	The celebrated OD2r. Model fine-tuned by the highly respected Alan Phillips himself!
CA Triptik Cln	Carol-Ann TripTik	Based on the Carol-Ann TripTik, a "true rock amp" capable of great versatility.
CA Triptik Clsc	Carol-Ann TripTik	Triptik "Classic" overdrive channel, for 70s and 80s British tones
CA Triptik Mdrn	Carol-Ann TripTik	Triptik "Modern" overdrive channel with more gain and low end. Also a great lead channel.
CA Tucana Cln	Carol-Ann Tucana® 3	Based on the Carol-Ann Tucana 3 clean channel
CA Tucana Lead	Carol-Ann Tucana® 3	Based on the Carol-Ann Tucana 3 lead channel
CA3+ Clean	CAE 3+ SE®(Ch 3)	Based on a preamp designed by Custom Audio Electronics®.
CA3+ Lead	CAE 3+ SE®(Ch 3)	Lead channel of the same.
CA3+ Rhy	CAE 3+ SE® (Ch 2)	Rhythm channel of the same.
Cali Leggy	Carvin® Legacy I	A model based on Steve Vai’s original signature Legacy amplifier.
Cameron CCV 1A	Cameron CCV100	An amp its creator Mark Cameron calls “one pissed off amp.”
Cameron CCV 1B	Cameron CCV100	The higher gain tone of the Cameron.
Cameron CCV 2A	Cameron CCV100	Bright1 switch left, Bright2 switch left, Gain Style switch left.
Cameron CCV 2B	Cameron CCV100	Bright1 switch left, Bright2 switch right, Gain Style switch left.
Cameron CCV 2C	Cameron CCV100	Bright1 switch left, Bright2 switch left, Gain Style switch right .
Cameron CCV 2D	Cameron CCV100	Bright1 switch left, Bright2 switch right, Gain Style switch right.
Capt Hook 1A	Hook Captain 34	Based on a Hook Captain Classic 34, Channel 1 with EQ and Boost switches off.
Capt Hook 1B	Hook Captain 34	Based on Channel 1 of this amp with EQ and Boost switches on.
Capt Hook 2A	Hook Captain 34	Based on Channel 2 of this amp with with Edge switch off.
Capt Hook 2B	Hook Captain 34	Based on Channel 2 of this amp with with Edge switch on.
Capt Hook 3A	Hook Captain 34	Based on Channel 3 of the same amp with with Edge switch off.
Capt Hook 3B	Hook Captain 34	Based on Channel 3 of the same amp with with Edge switch on.
Car Roamer	Carr Rambler	Based on a clean dream with large headroom.
Citrus A30 Cln	Orange® AD30HTC	Based on the Orange AD30 Twin Channel.
Citrus A30 Drty	Orange® AD30HTC	Based on the Orange AD30 Twin Channel.
Citrus Bass 200	Orange AD200B	Based on an Orange AD200B
Citrus RV50	Orange® Rockerverb®	Based on the dirty channel of the 50W head known for warmth and rich harmonics.
Citrus Terrier	Orange® Tiny Terror®	Based on the Orange Tiny Terror.
Class-A 15W TB	Vox® AC-15®	The heart of this amp’s tone comes from its power section and no negative feedback.
Class-A 30W Brt	Vox® AC-30®	Based on the Bright channel of a non-Top Boost Vox AC30.
Class-A 30W Hot	Vox® AC-30HW	Based on a Vox AC30HW with the Hot/Cool switch in the Hot position
Class-A 30W TB	Vox® AC-30TBX®	Created in response to demand for “More Treble.” Great highs + slightly reduced bass.
Class-A 30W	Vox® AC-30®	A combo that dominated the British Invasion. Gritty character, warm tone, great feel.
Comet 60	Komet™ 60	Capable of delivering a full range of tones from clean to singing leads.
Comet Concourse	Komet™ Concorde	Amp was modelled with the Response switch in the FAST position.
Corncob M50	Cornford MK50II®	Based on a boutique British amp. Plexi-Meets-Modern tone with big cojones.
Das Metall	Diezel™ VH4®	Based on a high-gain, boutique amp famous for its powerful, heavy, aggressive sound.
Deluxe Tweed	Fender® 5E3 Deluxe	The earliest and most popular of the so-called Tweed amplifiers.
Deluxe Verb Nrm	Fender® Deluxe Reverb®	Great, chimey tone with nice power amp breakup when you push the MASTER. Normal Channel.
Deluxe Verb Vib	Fender® Deluxe Reverb®	Vibro Channel of this amp.
Dirty Shirley	Friedman Dirty Shirley	“Designed to be an Ultra Fat Sweet sounding Classic Rock Amp”
Div/13 CJ Boost	Divided by 13® CJ11	Modeled with the Volume knob pulled out (boost switch).
Div/13 CJ	Divided by 13® CJ11	Based on a high performing “Tweed” meets “EL34” meets “Master Vol” 1x12 combo
Div/13 FT37 Hi	Divided by 13® FTR 37	Based on the FTR 37 with Gain Boost ON.
Div/13 FT37 Lo	Divided by 13® FTR 37	Based on the FTR 37 with Gain Boost OFF.
Dizzy V4 Blue 2	Diezel® VH4	Based on channel 2 of this 100w amp, said to be great for “gritty funk, dynamic clean.”
Dizzy V4 Blue 3	Diezel® VH4	Based on channel 3 (Diezel Heavy), with higher gain but still big dynamic range.
Dizzy V4 Blue 4	Diezel® VH4	Based on channel 4, a monster of gain which still has great definition and authority.
Dizzy V4 Slvr 2	Diezel® VH4	Based on channel 2 of the silver-faced version of a Diezel VH4.
Dizzy V4 Slvr 3	Diezel® VH4	Based on channel 3 (Diezel Heavy) of the silver-faced version of a Diezel VH4.
Dizzy V4 Slvr 4	Diezel® VH4	Based on channel 4 of the silver-faced version of a Diezel VH4.
Double Verb Nrm	Fender® Twin Reverb®	An indispensable icon, known for amazing cleans. Based on the “Normal” channel.
Double Verb SF	Fender® Twin Reverb®	Based on the Vibrato channel of a 1971 “Silverface” Fender Twin Reverb.
Double Verb Vib	Fender® Twin Reverb®	Based on the “Vibrato” channel of this amp.
Dweezil's B-Man	1965 Fender® Bassman®	The blackface version with a different circuit design. Dweezil Zappa’s personal amp.
Energyball	ENGL Powerball®	Very high-gain German model. Lots of bass. Great for aggressive, drop-tuned riff work.
Euro Blue Mdrn	Bogner® Ecstasy Blue Channel®	Same as above, with structure switch is set to ‘M’ (Modern).
Euro Blue	Bogner® Ecstasy Blue Channel®	Based on the 20th Anniversary model. OD channel w/ BOOST + STRUCTURE OFF.
Euro Red Mdrn	Bogner® Ecstasy Red Channel®	Same as above, with structure switch is set to ‘M’ (Modern).
Euro Red	Bogner® Ecstasy Red Channel®	Same as above but with OD channel w/ BOOST + STRUCTURE ON.
Euro Uber	Bogner® Überschall	Based on “High Gain” channel of this 120W head. Heavy grinding lows and insane gain.
FAS 6160	PVH® 6160	A modified version of the PVH 6160 – less fizzy than the original, with a bouncier feel.
FAS Bass	Created by Fractal Audio	Our take on how a bass amp should sound

FAS Brootalz	Created by Fractal Audio	This model brings teh brootalz.
FAS Brown	“The Brown Sound”	The original BROWN model from the Axe-Fx Ultra.
FAS Class-A	Created by Fractal Audio	A “Blackface” preamp into a cathodebiased 6L6 power amp with no negative feedback.
FAS Crunch	Created by Fractal Audio	Our take on the ultimate British-sounding amp. More dynamic/open plus more gain.
FAS Hot Rod	Created by Fractal Audio	Based on nothing that we know by now.
FAS Lead 1	Created by Fractal Audio	Neutral high-gain lead with a tight midrange.
FAS Lead 2	Created by Fractal Audio	Hot-rodded British lead sound with a tonestack by Custom Audio Electronics.
FAS Modern II	Created by Fractal Audio	Tighter version with a 5153-style bass boost in the tone stack.
FAS Modern III	Created by Fractal Audio	Similar to a Recto but with tighter bass and a cathode-biased power amp.
FAS Modern	Created by Fractal Audio	A high-gain hybrid. Equally well-suited to modern rhythm and lead work.
FAS Rhythm	Created by Fractal Audio	Combines the best features of the British and USA crunch models.
FAS Wreck	Trainwreck™ Express	The original WRECKER 1 model from the Axe-Fx Ultra.
Fox ODS Deep	Fuchs® Overdrive Supreme 50	Same model with the "MID" switch OFF.
Fox ODS	Fuchs® Overdrive Supreme 50	Based on an amp that's based on an amp ;-)
Friedman BE V1	Friedman Brown Eye "Marsha"	New Friedman BE model based on a newer Friedman BE. Voice switch toggled right.
Friedman BE V2	Friedman Brown Eye "Marsha"	New Friedman BE model based on a newer Friedman BE. Voice switch toggled left.
Friedman BE	Friedman Brown Eye "Marsha"	What many call “the ultimate modded Plexi” by Dave Friedman of Rack Systems.
Friedman HBE V1	Hairy Brown Eye “Marsha”	New Friedman HBE model based on a newer Friedman HBE. Voice switch toggled right.
Friedman HBE V2	Hairy Brown Eye “Marsha”	New Friedman HBE model based on a newer Friedman HBE. Voice switch toggled left.
Friedman HBE	Hairy Brown Eye “Marsha”	The BE amp’s alternate voicing with a gain boost. A killer hi-gain tone in your arsenal.
Friedman SM Box	Friedman Smallbox	Based on the amp by Dave Friedman of Rack Systems. 50W, EL34.
Fryette D60 L	Fryette D60®	Based on the Fryette Amplification D60 in the “Less” mode.
Fryette D60 M	Fryette D60®	Based on the Fryette Amplification D60 in the “More” mode.
Gibtone Scout	Gibson® Scout	If you love vintage clean tones, this 17-watter has got ‘em!
Herbie Ch2+	Diezel® Herbert	Based on an amp called “looser” and “more “familiar” than the VH4®
Herbie Ch2-	Diezel® Herbert	Channel 2+ of the same.
Herbie Ch3	Diezel® Herbert	Channel three of the same.
Hipower Brillnt	Hiwatt® DR103 (Brilliant)	Amp with a unique tone-stack and a brilliant “chimey” tone.
Hipower Jumped	Hiwatt® DR103 (Both)	Both Brilliant and Normal channels as when “jumpering the inputs” of this amp.
Hipower Normal	Hiwatt® DR103 (Normal)	Medium-gain, full sound normal channel of this amp.
Hot Kitty	Bad Cat® Hot Cat	Based on an amp voted by Guitar Player as “the second best combo of all time.”
Jazz 120	Roland® JC-120®	The only solid-state-based model in our collection; a quintessential clean tone.
JMPre-1 OD1 BS	Marshall® JMP-1	Based on a Marshall JMP-1 rack preamp OD1 channel with Bass Shift engaged.
JMPre-1 OD1	Marshall® JMP-1	Based on a Marshall JMP-1 rack preamp OD1 channel with Bass Shift off.
JMPre-1 OD2 BS	Marshall® JMP-1	Based on a Marshall JMP-1 rack preamp OD2 channel with Bass Shift engaged.
JMPre-1 OD2	Marshall® JMP-1	Based on a Marshall JMP-1 rack preamp OD2 channel with Bass Shift off.
Jr Blues Fat	Fender® Blues Jr.®	The same as the Jr Blues model but with the “Fat” switch engaged
Jr Blues	Fender® Blues Jr.®	A gutsy little classic with dual EL84s.
JS410 Crunch Or	Marshall® JVM410HJS	Based on a Marshall JVM410HJS, Joe Satriani's 4-channel 100w signature amp. Crunch OR mode.
JS410 Crunch Rd	Marshall® JVM410HJS	Same, Crunch RD mode, based on a modded JCM 2203.
JS410 Lead Or	Marshall® JVM410HJS	Same, more gain.
JS410 Lead Rd	Marshall® JVM410HJS	Same, even more gain.
Legato 100	Carvin Legacy VL100	Based on a certain famous Carvin Legacy 100. 100 watts, EL34 tubes.
Matchbox D-30	Matchless DC-30	Based on this first of all Matchless Amps.
Mr Z Hwy 66	Dr. Z Route 66	Based on this new approach of a classical design.
Mr Z Mz-8	Dr. Z Maz-8	Based on a versatile single-ended Class-A amp.
Mr Z Mz-38	Dr. Z Maz 38 SR®	Based on an amp popular with country and roots players.
Nuclear-Tone	Swart Atomic Spacetone	Based on a cool little, gritty, stylishly retro all-tube combo. Use with Bias Trem!
ODS-100 Clean	Dumble™ OD Special®	Based on the Clean channel of a ODS-100 serial number 213, an “HRM” version.
ODS-100 Ford 1	Dumble™ OD Special®	This is a "non-HRM" version of the ODS-100 Lead.
ODS-100 Ford 2	Dumble™ OD Special®	A "non-HRM" version of the ODS-100 Lead with the Preamp Bypass switch off.
ODS-100 Ford Md	Dumble™ OD Special®	Same as ODS-100 Ford 1 but with the Mid switch engaged
ODS-100 HRM Mid	Dumble™ OD Special®	Same with the “Mid” (sometimes called "Deep") switch engaged.
ODS-100 HRM	Dumble™ OD Special®	The OD channel of the same.
Plexi 50W 6550	Marshall® Super Lead®	Based on the High input of a 1972 50W Marshall “Plexi” with 6550 power tubes.
Plexi 50W Hi 1	Marshall® Super Lead®	Based on the “High Treble” channel of the 50W Plexi.
Plexi 50W Hi 2	Marshall® Super Lead®	Same with a 0.68uF cathode bypass capacitor in the second triode stage. Slightly brighter tone.
Plexi 50W Jump	Marshall® Super Lead®	Both Treble and Normal channels as when “jumpering the inputs” on a 4-hole amp.
Plexi 50W Nrml	Marshall® Super Lead®	Based on the “Normal” channel of the 50W Plexi.
Plexi 100W 1970	Marshall® Super Lead® 1970	Based on a 1970 Marshall 1959SLP 100 which has a darker, smoother sound than earlier Plexis.
Plexi 100W High	Marshall® Super Lead® 1959	Based on the 100W version of the legendary "original Plexi"
Plexi 100W Jump	Marshall® Super Lead® 1960	Both High and Normal channels as when “jumpering the inputs” of the amp.
Plexi 100W Nrml	Marshall® Super Lead® 1961	Normal channel of the same.
Prince Tone NR	Fender® AA964 Princeton®	Based on an early CBS “Silverface” but still using pre-CBS design and components.
Prince Tone Rev	Fender® Princeton® Reverb	Based on a 1966 Fender® Princeton® Reverb.
Prince Tone	Fender® Princeton® Tweed	Based on a compact amp with a single-ended power section.
PVH 6160 Block	Peavey® EVH® 5150™	Based on the high- input lead channel of an amp named after the criminally insane.
PVH 6160+ LD	Peavey® 6505+®	Based on an amp that's called the new standard for "brutality and aggression."
PVH 6160+ Rhy B	Peavey® 6505+®	Based on channel 1 with the Crunch and Bright switches depressed.
PVH 6160+ Rhy	Peavey® 6505+®	Based on Channel 1 of a Peavey 6505+ with the Crunch switch depressed and Bright switch out.
Recto1 Org Mdrn	Boogie™ 2 Ch. Dual Rectifier®	Based on an original Mesa Boogie 2-channel Dual Rectifier. Orange channel, "Modern" Mode.
Recto1 Org Norm	Boogie™ 2 Ch. Dual Rectifier®	Based on an original Mesa Boogie 2-channel Dual Rectifier. Orange channel, "Normal" Mode.
Recto1 Red Mdrn	Boogie™ 2 Ch. Dual Rectifier®	Based on an original Mesa Boogie 2-channel Dual Rectifier. Red channel, "Modern" Mode.
Recto2 Org Mdrn	Boogie™ 3-Ch. Dual Rectifier®	Based on the modern, more aggressive version of the Dual Rectifier amp.
Recto2 Org Vntg	Boogie™ 2 Ch. Dual Rectifier®	Based on the original Mesa Boogie 2-channel Dual Rectifier. Orange channel.

Recto2 Red Mdrn	Boogie™ 3-Ch. Dual Rectifier®	Based on the modern version of the Dual Rectifier amp. Red channel.
Recto2 Red Vntg	Boogie™ 2 Ch. Dual Rectifier®	Based on the original Mesa Boogie 2-channel Dual Rectifier Red Channel.
Ruby Rocket Brt	Paul Ruby Rocket	Same with the Bright switch in the up position.
Ruby Rocket	Paul Ruby Rocket	Based on Paul Ruby's amp, in turn based on a Trainwreck® Rocket. Bright switch down.
Shiver Clean	Bogner® Shiva Clean Channel	Based on the 90W anniversary model. Powerful shimmering cleans.
Shiver Lead	Bogner® Shiva Lead Channel	Lead channel: sweet, rich- sounding with aggressive, English- style midrange punch.
Solo 88 Clean	Soldano® X-88	Based on the clean channel of a Soldano X-88.
Solo 88 Lead	Soldano® X-88	Based on the lead channel of a Soldano X-88.
Solo 88 Rhythm	Soldano® X-88	Chosen because the Rhythm channel of the 99 is identical to the that of the 100
Solo 99 Clean	Soldano® X99® Preamp	Based on the clean channel of this midi-equipped motorized preamp.
Solo 99 Lead	Soldano® X99® Preamp	Based on the lead channel of the X99.
Solo 100 Clean	Soldano™ SLO-100®	Based on the clean channel of the SLO-100
Solo 100 Lead	Soldano™ SLO-100®	Based on the snarling lead channel of the above amp.
Solo 100 Rhy	Soldano™ SLO-100®	Based on the rhythm channel of the same.
Spawn Nitrous 1	Splawn® Nitro	Based on the OD-1 mode of a Splawn Nitro with KT-88 power tubes.
Spawn Nitrous 2	Splawn® Nitro	Same, with OD-2 mode. Lots of saturation and big lows.
Spawn Rod OD1-1	Splawn® Quickrod	Based on an amp with bold body and a bit of bite. This is the 'Hot Rod Plexi' channel.
Spawn Rod OD1-2	Splawn® Quickrod	This is the 'Hot Rod 800' channel of the same.
Spawn Rod OD1-3	Splawn® Quickrod	This is the 'Super Hot Rod 800' channel of the same.
Spawn Rod OD2-1	Splawn® Quickrod	OD2 switches in a cathode bypass cap which increases the gain of that stage.
Spawn Rod OD2-2	Splawn® Quickrod	Hot Rod 800 OD2
Spawn Rod OD2-3	Splawn® Quickrod	Super Hot Rod OD2
Suhr Badger 18	Suhr® Badger 18 W	Based on the 18w version of this EL-84 powered tube rectifier classic from Suhr®
Suhr Badger 30	Suhr® Badger 30W	In comparison to the 18w, the 30w features a solid state rectifier.
Super Verb Nrm	Fender® Super Reverb	Based on a pre-CBS 1964 blackface version of this amp. Normal Channel.
Super Verb Vib	Fender® Super Reverb	The Vibro Channel of this amp.
Supertweed	FAS Supertweed	Like a favorite vintage tweed on steroids…
Supremo Trem	Supro® 1964T	A cool classic, built in Chicago by Valco. Originally intended for bass!
SV Bass	Ampeg SVT®	Based on a head used for decades by famous bassists the world over.
Thordendal Mdrn	Custom Amp	Built to the specifications of Meshuggah's Fredrik Thorendal.
Thordendal Vint	Custom Amp	Built to the specifications of Fredrik Thorendal. A less aggressive channel.
Tremolo Lux	Fender AA763 Tremolux	Based on a Fender AA763 Tremolux
Tube Pre	Studio Tube Preamp	A completely neutral, low-gain tube pre useful for "warming up" various sources.
Two Stone J35 1	Two Rock® Jet 35	The amp was modeled in the LEAD mode with the input tone stack bypassed.
Two Stone J35 2	Two Rock® Jet 35	Modeled with the Preamp Bypass switch off.
TX Star Clean	Mesa Boogie® Lonestar™	This model is based on the clean channel of a Mesa Lonestar.
TX Star Lead	Mesa Boogie® Lonestar™	This model is based on the lead channel of a Mesa Lonestar.
USA Bass 400 1	Mesa Boogie® Bass 400	Based on a Mesa Boogie® Bass 400
USA Bass 400 2	Mesa Boogie® Bass 400	Modeled with the Bass Shift on.
USA Clean	Mesa Boogie™ MKIV™	A beautiful clean that can pushed into warm clipping.
USA IIC+ Bright	Mesa Boogie® Mark II™	Based on a US-made amp famous for its smooth overdrive sound. Pull Bright ON, Pull Deep OFF.
USA IIC+ Brt/Dp	Mesa Boogie® Mark II™	Based on a US-made amp famous for its smooth overdrive sound. Pull Bright ON, Pull Deep ON.
USA IIC+ Deep	Mesa Boogie® Mark II™	Based on a US-made amp famous for its smooth overdrive sound. Pull Bright OFF, Pull Deep ON.
USA IIC+	Mesa Boogie® Mark II™	Based on a US-made amp famous for its smooth overdrive sound. Pull Bright OFF, Pull Deep OFF.
USA IIC++	Mesa Boogie® Mark II™ C+	Based on a Mesa/Boogie Mark IIC+. This model is used by Metallica for their live sound.
USA Lead +	Mesa Boogie™ MKIV™	Same amp with the Mid Gain switched on.
USA Lead Brt +	Mesa Boogie™ MKIV™ (Lead)	The same, with both the Treble Shift and Mid Gain on.
USA Lead Brt	Mesa Boogie™ MKIV™	Treble Shift gives this amp a slightly different character with a little more cut.
USA Lead	Mesa Boogie™ MKIV™	This model has a tight, focused, hi-gain sound. Great for fusion and rock leads.
USA Pre Clean	Mesa Boogie™ Triaxis™	Based on the clean channel of a Mesa Triaxis™ preamp
USA Pre LD1 Red	Mesa Boogie™ Triaxis™	Based on the LD1 Red Mode of a Mesa Triaxis™ preamp with the TX-4 board.
USA Pre LD2 Grn	Mesa Boogie™ Triaxis™	Based on the LD2 Green "Mid Gain Mark IV Lead channel".
USA Pre LD2 Red	Mesa Boogie™ Triaxis™	Based on the LD2 Red Mode of a Mesa Triaxis preamp.
USA Pre LD2 Ylw	Mesa Boogie™ Triaxis™	Based on the LD2 Yellow "Classic MKII Lead channel".
USA Rhythm	Mesa™ Boogie MKIV™	Based on "the" California crunch rhythm sound. Rhythm Ch. 2 with "Fat" switch OFF.
USA Sub Blues	Mesa™ Subway Blues	Based on the 20W Subway Blues.
Vibra-King Fat	Fender® Vibro-King®	Modeled with the Fat switch on
Vibra-King	Fender® Vibro-King®	Based on the venerable Vibro-King®, famous for crystal cleans and powerful overdrive
Vibrato Lux	Fender® Vibro-Lux®	Based on the "legendary" amp that connoisseurs call "the little Vibro-King™"
Vibrato Verb AA	Fender® Vibroverb®	Based on a AA763 Fender VibroVerb.
Vibrato Verb AB	Fender® Vibroverb®	Based on a AB763 Fender VibroVerb.
Vibrato Verb CS	Fender® Vibroverb®	Based on the "64 Vibroverb Custom" a modified version of SRV's blackface Vibroverb. Mod switch ON.
Vibrato Verb	Fender® Vibroverb®	Based on a 40W combo that's great for clear or grinding cleans and gutsy blues.
Wrecker Express	Trainwreck™ Express	Based on the Trainwreck Express—designed and built by the late, great Ken Fischer.
Wrecker Lvrpool	Trainwreck™ Liverpool	Based on a Trainwreck Liverpool.
Wrecker Rocket	Trainwreck™ Rocket	Rounding out the collection is the Rocket, with four EL84s and a tube rectifier.

16.2 **Table of Cab Types**

The following lists the options available when selecting TYPE in the CAB block (p.50). Each Impulse Response (IR) was created by measuring the actual cabs named below. Selections which include the word "Mix" were created using a blend of microphone "colors" and may sound best with **MIC** set to "NONE". Factory cabs include custom creations by Fractal Audio Systems, selections from 3rd-party libraries by ML Sound Lab, RedWirez, OwnHammer, and more.

Factory Cabs are subject to change without notice. Please check firmware release notes.

Manufacturer names and product names mentioned above are trademarks or registered trademarks of their respective owners, which are in no way associated with or affiliated with Fractal Audio Systems. The names are used only to illustrate sonic and performance characteristics of the Fractal Amplifier TYPES, which have been created by incredibly detailed analysis of the actual amps that inspired them

001. 1x6 Oval
002. 1x8 Tweed
003. 1x10 Prince Tone AT4047
004. 1x10 Prince Tone M160
005. 1x12 Brown M160
006. 1x12 Black SM57
007. 1x12 G12T R121
008. 1x12 E12L (RW)
009. 1x12 Studio
010. 1x12 Emi Open Back (JM)
011. 1x12 Bludo Mix
012. 1x12 Shiver R121 (BG)
013. 1x12 Tweed Blue (RW)
014. 1x12 Tweed Deluxe (RW)
015. 1x12 Brit Blue (RW)
016. 1x12 Brit G12H30 (RW)
017. 1x15 Blues
018. 1x15 Thunderbolt (RW)
019. 2x12 TX Star M160
020. 2x12 Double Amp KSM313
021. 2x12 Double Verb R121
022. 2x12 Brown Super M160
023. 2x12 Blue
024. 2x12 Top Boost Blue (RW)
025. 2x12 Top Boost Silver (RW)
026. 2x12 Boutique (RW)
027. 2x12 Fuzzbomb M160
028. 2x12 Gold 30 Far-Field (JM)
029. 2x12 G12-65 Far-Field (JM)
030. 2x12 Boutique R121
031. 2x15 Doubleshow (RW)
032. 4x10 Bassguy M160
033. 4x10 Bassguy P10 (RW)
034. 4x12 Basketweave G12H30 (RW)
035. 4x12 Basketweave G12L (RW)
036. 4x12 Basketweave G12M20 (RW)
037. 4x12 Basketweave G12M25 (RW)
038. 4x12 1960A G12M (RW)
039. 4x12 1960B T75 (RW)
040. 4x12 1960B K120 (RW)
041. 4x12 1960B V30 (RW)
042. 4x12 Hi-Power (RW)
043. 4x12 Recto SM57
044. 4x12 Recto M160
045. 4x12 Solo V12 (RW)
046. 4x12 Solo S12X (RW)
047. 4x12 German V30 (RW)
048. 4x12 German Boutique
049. 4x12 PVH6160 (RW)
050. 4x12 Uber T75 (RW)
051. 4x12 Uber V30 (RW)
052. 4x12 Uber T75+V30 (RW)
053. 4x12 Citrus V30 (RW)
054. 4x12 Pre-Rola 55 M160 (ML)
055. 4x12 Pre-Rola 75 M160 (ML)
056. 4x12 Brit 80S R121 (ML)
057. 4x12 Slm H75 (OH)
058. 4x12 TV Mix C1 (ML)
059. 4x12 TV Mix C4 (ML)
060. 4x12 Fractal GB M160
061. 4x12 Fractal V30 AT4047
062. 4x12 V30
063. 4x12 German
064. 4x12 30W (Ultra)
065. 4x12 Cali
066. 1x15 L.A. Bass
067. 4x10 Aluminum Bass (RW)
068. 8x10 SV Bass (RW)
069. 4x12 Pre-Rola GB C414
070. 4x12 Beatle GB
071. 4x12 D120
072. 4x12 Sorcerer
073. 4x12 USA Trad 57-121 (ML)
074. 4x12 USA Trad 906-421 (ML)
075. 1x8 Champlifier Mix
076. 1x8 Vibrato Champlifier Mix
077. 1x10 Prince Tone Black Mix
078. 1x10 Prince Tone Silver Mix
079. 1x12 Junior Blues M160
080. 1x12 Deluxe Verb Mix
081. 1x12 Deluxe Tweed Mix
082. 1x12 Vibrato Lux Mix
083. 1x12 Class-A 15W Blue Mix
084. 1x12 Division 13 Mix
085. 1x12 Hot Kitty Mix
086. 1x12 Hawaii Mix
087. 1x15 Tweed Pro Mix
088. 1x15 Empire Mix
089. 2x10 Super Tweed Mix
090. 2x10 Vibrato Lux Mix
091. 2x12 Double Verb Mix
092. 2x12 Pro Verb Mix
093. 2x12 Class-A 30W Blue Mix
094. 2x12 Class-A 30W Silver Mix
095. 2x12 Supremo Mix
096. 2x12 Santiago EJ1250
097. 2x12 Santiago Altec
098. 3x10 Vibrato King Mix
099. 4x10 Bassguy Mix
100. 4x10 Super Verb Mix
101. 4x12 Basketweave Green Mix
102. 4x12 Basketweave AX Mix
103. 4x12 Basketweave TV Mix
104. 4x12 Cali Lead 80S Mix
105. 4x12 Rumble EV12L RNR1
106. 4x12 Rumble EV12S M160
107. 4x12 PVH6160 Mix
108. 4x12 Petrucci V30 Mix
109. 1x15 SV Bass M88
110. 1x15 SV Bass Subkick
111. 4x10 SV Bass M88
112. 4x10 SV Bass Subkick
113. 4x10+Tweeter SV Bass M88
114. 1x12 AC-20 Dlx Mix
115. 1x12 Nuclear Tone Mix
116. 1x12 Scumtone 25W Mix
117. 2x12 Boutique Mix
118. 2x12 SV Legend Mix
119. 1x12 AC-20 Dlx Mix
120. 1x12 Roamer Mix
121. 1x12 Triptik Mix
122. 2x12 Class-A Mix
123. 2x12 Double Verb Mix
124. 4x12 5153 Mix #1
125. 4x12 5153 Mix #2
126. 4x12 Citrus Mix
127. 4x12 Lerxst R121
128. 4x12 Cali Mix
129. 4x12 Recto Mix
130. 4x12 Recto New Mix
131. 4x12 TV Mix #1
132. 4x12 TV Mix #2
133. 1x8 EC Champlifier I5
134. 1x12 Tweed-Verb R121
135. 1x12 AC-20 Dlx M160
136. 1x12 Roamer R121 Reverse
137. 2x12 Double Verb M160
138. 2x12 Class-A Blues Mix
139. 4x12 USA Lead 80S R121
140. 1x12 Dlx Aln-Slv Mix (OH)
141. 1x12 Dlx Fn-42 Mix (OH)
142. 1x12 Dlx J12-Pr Mix (OH)
143. 2x12 Bog-Sh Fn-42 Mix (OH)
144. 4x12 Mar-Cb EV-S Mix (OH)
145. 4x12 Mar-Cb Fn-42 Mix (OH)
146. 4x12 Mar-Cb H-Pr-55 Mix (OH)
147. 4x12 Mar-Cb M-BB-55 Mix (OH)
148. 4x12 Mar-Cb Sb-75 Mix (OH)
149. 4x12 Mar-Cb V30-Ch Mix (OH)
150. 1x12 Shadow Mix (TAF)
151. 1x12 Vintage Mars Mix (TAF)
152. 2x10 Fen Room Mix (TAF)
153. 2x12 Art+Tango Jr Mix (TAF)
154. 2x12 Acrox Mix (TAF)
155. 4x12 Wat Mix (TAF)
156. 4x12 Starfound Mix (TAF)
157. 4x12 Mars G12T Room Mix (TAF)
158. 4x12 Mars Bw G12 Room Mix (TAF)
159. 4x12 Vintmars+Bw Room Mix (TAF)
160. 4x12 5153 121 G
161. 4x12 5153 4047 G
162. 4x12 5153 57 C
163. 4x12 Citrus 121 B
164. 4x12 Citrus 160 C
165. 4x12 Citrus 57 C
166. 4x12 Rumble L 121 A
167. 4x12 Rumble L 4047 A
168. 4x12 Rumble L G44 A
169. 4x12 Rumble S 121 C
170. 4x12 Rumble S 4047 B
171. 4x12 Rumble S R1 D
172. 4x12 Recto 121 C
173. 4x12 Recto 4047 E
174. 4x12 Recto 57 B
175. 4x12 TV 160 B
176. 4x12 TV 57 D
177. 4x12 USA 121 B
178. 4x12 USA 4047 B
179. 4x12 USA 57 A

16.3 **Loading User Cab IRs**

In addition to the onboard Factory Cabs, the Axe-Fx II XL/XL+ allows you to store up to 1024 “User Cabs” onboard (100 on the Mark II). User Cabs are a great way to explore tones and define your signature sound. Fractal Audio Systems offers professionally produced user “Cab Packs” at http://shop.fractalaudio.com. Axe-Change, our file sharing site, is a great resource for FREE cabs. Other commercial libraries are also worth a look. You can even create your own cabs if you have some basic equipment. (See **IR Capture** on p. **156** for details.)

Axe-Fx II Impulse Responses are transferred into the unit as MIDI System Exclusive data. The recommended way to do this is using one of our software applications:

Axe-Edit – Our basic backup/restore utility allows you to drag and drop cab files into memory slots using Axe-Manage Cabs. You can also manage entries already in the memory of the Axe-Fx II.

Cab-Lab – Cab-Lab is a powerful Cab IR Mixer. It also includes a guided utility to capture your own cabs.

Fractal-Bot – The most basic (and essential) Axe-Fx utility can send individual cab files to any memory location.

You can also use a 3rd-party MIDI utility such as MIIDI OX or Snoize Sysex Librarian. The process for sending User Cab IR Files to the Axe-Fx II is as follows:

1. Ready the Axe-Fx II to receive the file:
    a. Open any preset with the **CABINET 1** block in it.
    b. Select this block and press EDIT to open its EDIT menu.
    c. Ensure that the **MODE** is not set to “STEREO”
    d. Change the **CAB** parameter to the numbered USER slot you wish to load into.
2. Connect the Axe-Fx II and the computer. This will typically be done using USB, but legacy MIDI also works.
3. Launch your MIDI Utility, ensure it is configured correctly to send SysEx to the Axe-Fx II, and ready it to send the desired cab file. (Remember to set the Axe-Fx II as your application’s MIDI device!)
4. Transmit the file from the computer to the Axe-Fx II. The Axe-Fx II will not show a progress bar, but its MIDI IN LED will flash. There is no need to STORE or otherwise commit the change; it is permanent once complete. If the IR File you transmitted has an embedded name, this will be displayed on the Axe-Fx II in the bottom line of the CAB PG2 menu once the file has been received successfully.

The XL never includes Cabs in a SYSTEM dump/restore.
User Cab IRs 1-50 are included when you backup or restore the **System** area of an Axe-Fx II Mark II.
User Cabs 51-100 on the Mark II are not included in the SYSTEM and must be backed up or loaded individually.

16.4 Shortcuts Overview

The Axe-Fx II has several shortcuts and hidden features. These are summarized below.

IN THE AMP BLOCK

- With the BASS knob selected, press ENTER to engage or disengage the CUT switch.
- With the MID knob selected, press ENTER to engage or disengage the FAT switch.
- With the TREBLE knob selected, press ENTER to engage or disengage the BRIGHT switch.
- To reset the amp's GRAPHIC EQ, press ENTER

IN PRESET RECALL MODE

- Press NAV UP or DOWN to load the next or previous preset.
- Press NAV LEFT or RIGHT to load the 10th preset in either direction.
- Turn the 'A' knob to change SCENES.

ON THE GRID:

- With a non-shunt grid block selected, press EXIT…ENTER to convert to a SHUNT.
- With a shunt selected, press EXIT…ENTER to convert to an EMPTY SPACE.
- With any block selected, PRESS AND HOLD ENTER to create a series of connectors and shunts 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.
- Double-tap BYPASS to return the current block to defaults (also works for INPUT gate, OUTPUT mixer, all internal CONTROLLERS, and both Global Graphic EQs).
- Double-tap FX BYP to access the SAVE/LOAD GLOBAL BLOCKS screen.
- Double-tap X or Y for XY copy options.

IN THE I/O CONTROL MENU:

- Press ENTER to start LEARNING MODE for any function on the page. Move the external controller or send a MIDI CC# to the Axe-Fx II, and the selected function will learn the assignment.

IN THE SEQUENCER MENU

- With any STAGE selected, press ENTER to randomize the values of all stages.

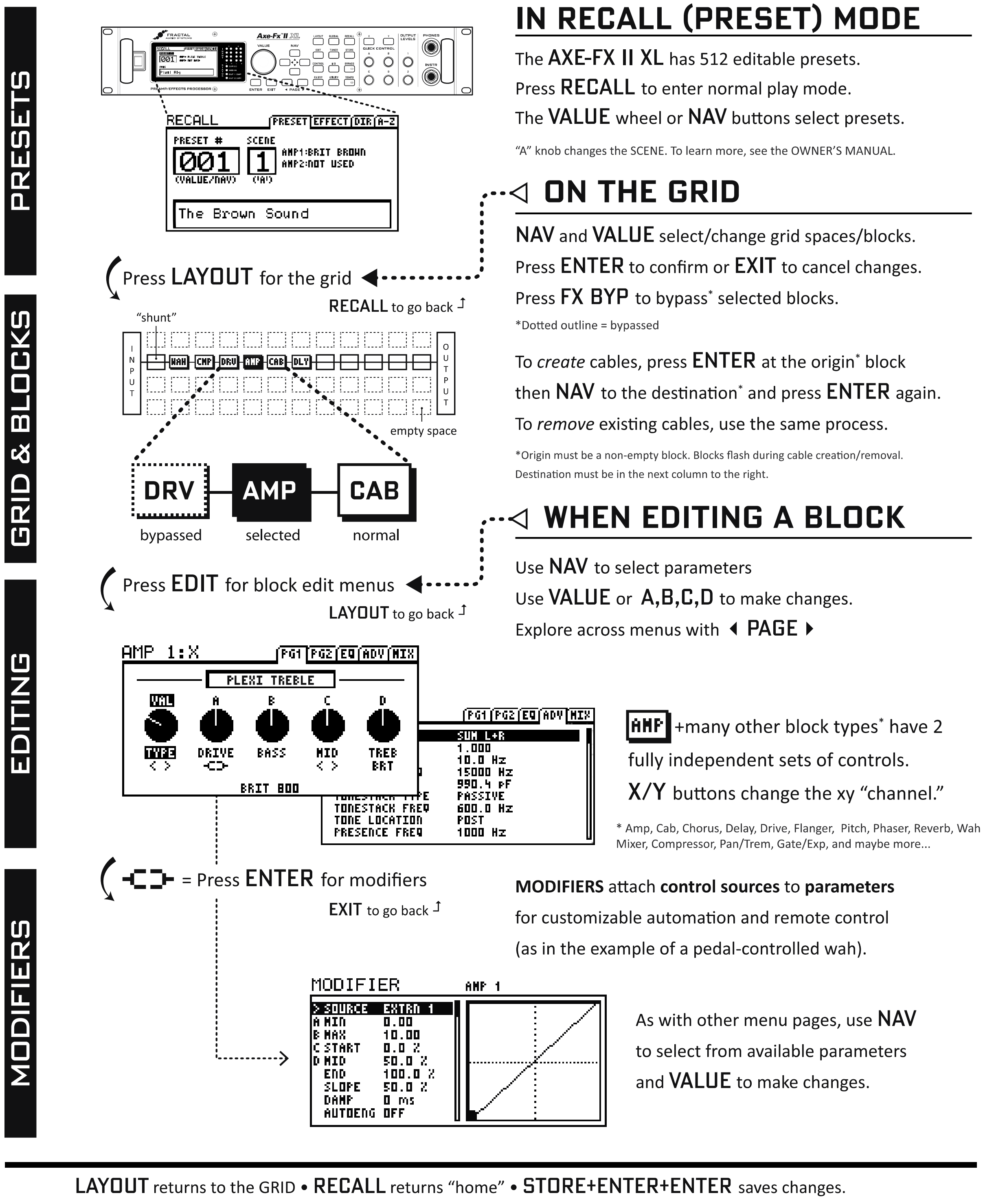
ANYWHERE EXCEPT IN AN "EDIT" or "STORE" MENU or SUBMENU:

- Press X or Y to jump to the EDIT menu of either **Quick Jump** block (p. 154).

16.5 60-Second Edit Guide

The following is provided as a quick start-up or reminder about editing on the Axe-Fx II XL/XL+.

Figure 16-1 – 60-Second Edit Guide



(Note: The Axe-Fx II Mark II has 384 presets and supports fewer block types with XY.)

16.6 **Understanding Preset Size Limits**

Each block you add to the grid contributes to a preset's total CPU load. So do connector cables, modifiers, and "general overhead," albeit to a far lesser extent.

As a preset grows in size and complexity, load on the CPU increases. You can check the current load at any time by pressing UTILITY and switching to the STATUS page. A thermometer-like meter on the right side of the screen shows the CPU usage (and provides a specific numerical readout above).

If the total CPU load were to exceed 98%, the Axe-Fx II would become unable to do much of anything, so there are safeguards that prevent this condition from occurring. First, you are stopped from inserting any block whose potential CPU usage might cause an overload by the message "INSUFFICIENT CPU"; (the Axe-Fx II assumes that a block will be used to its limits when making this determination).

If you are prevented from inserting an effect, you can make changes to reduce the current CPU utilization and try again. One strategy might be removing redundant blocks. Bypassed effects are in fact always "running" at full CPU utilization, so these are prime candidates. Adjusting certain parameters can also help. Lowering the number of voices in a chorus, for example, or switching the **CAB TYPE** from high to low-resolution might make the difference of fitting that last effect block…or not.

As a second safeguard, the Axe-Fx II will alert you if a parameter change pushes the CPU too far. In this case, the preset will be muted, the message "EXCESS CPU UTILIZATION! REDUCE LOAD" will flash on the screen, and you'll need to take steps to get back below the limit. The most likely solution will be to change back whatever setting you had just made, but it is also possible to enter the grid and remove or edit other blocks to address the issue.

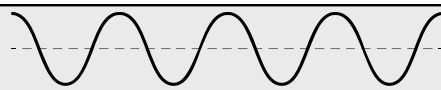
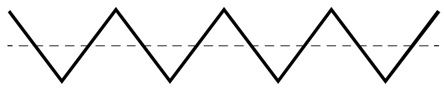
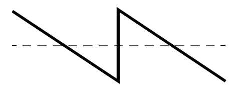
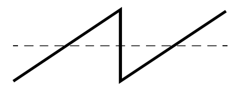
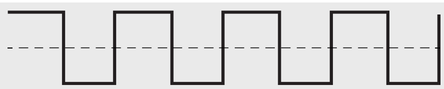
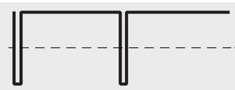
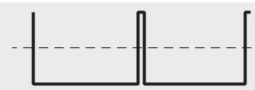
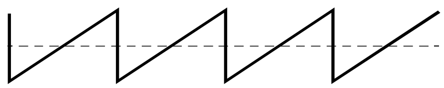
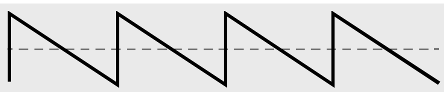
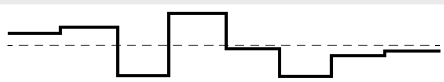
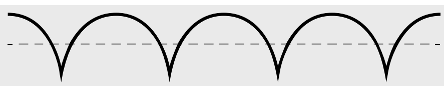
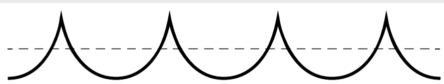
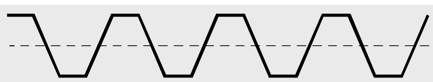
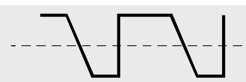
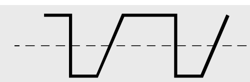
These warnings happen very infrequently, even for power users. The CPU limit is generally not an issue when creating musically viable settings. In the past, professional players have been able to replace entire rigs of gear—amps, pedals, and beyond—with a single Axe-Fx Ultra Preset. Although our G3 technology has resulted in amps, cabs, and certain other blocks that require additional power per instance, the onboard CPU power is DOUBLE that of the Ultra, so there is still a considerable net pickup.

The Axe-Fx II dedicates a small percentage of CPU resources to USB processing. Extremely large presets may run fine while USB is disconnected, but need some economizing as described above to run while USB is connected.

16.7 LFO Waveforms, Duty, and Phase

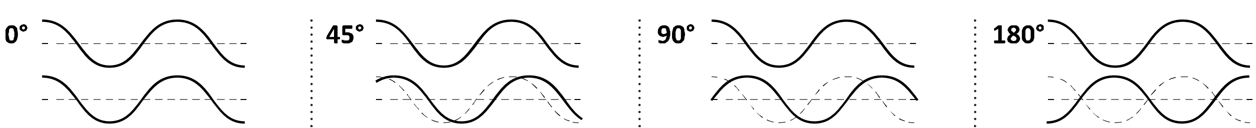
All of the modulation effects in the Axe-Fx II (Chorus, Flanger, Delay, Phaser, Tremolo, etc.) and the two Global LFOs share a common set of waveform types—“shapes” that define the way they change over time. These are represented below, together with an indication of how the **DUTY** parameter controls 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 “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	
TRAPEZOID			

16.7.1 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 0–100, the other is swinging from 100–0 (and vice versa). Any interim setting is also allowed. Phase has no effect on the Axe-Fx II RANDOM waveform.



16.8 Tempo Cross Reference

The tables below list the rhythmic values available in every tempo parameter on the Axe-Fx II. The first lists these *in the order they appear* as you turn the VALUE wheel, with their equivalency in BEATS. The heavy border shows a breaking point between common and uncommon tempos.

The second table (cross-referenced to the first by the INDEX values) lists time tempo time values sorted from *shortest to longest*. Its cell borders show - - - 16th, - - - 8th, and — quarter note boundaries.

Values in Actual Order, with Beat Equivalents

INDEX	VALUE	BEATS	INDEX	VALUE	BEATS
1	1/64th trip	0.042	40	22/64 (11/32)	1.375
2	1/64th	0.063	41	23/64	1.438
3	1/64th dot	0.094	42	25/64	1.563
4	1/32nd trip	0.083	43	26/64 (13/32)	1.625
5	1/32nd	0.125	44	27/64	1.688
6	1/32nd dot	0.188	45	28/64 (7/16)	1.75
7	1/16th trip	0.167	46	29/64	1.813
8	1/16th	0.25	47	30/64 (15/32)	1.875
9	1/16th dot	0.375	48	31/64	1.938
10	1/8th trip	0.333	49	33/64	2.063
11	1/8th	0.5	50	34/64 (17/32)	2.125
12	1/8th dot	0.75	51	35/64	2.188
13	1/4 trip	0.667	52	36/64 (9/16)	2.250
14	1/4	1	53	37/64	2.313
15	1/4 dot	1.5	54	38/64 (19/32)	2.375
16	1/2 trip	1.333	55	39/64	2.438
17	1/2	2	56	40/64 (5/8)	2.5
18	1/2 dot	3	57	41/64	2.563
19	1 trip	2.667	58	42/64 (21/32)	2.625
20	1	4	59	43/64	2.688
21	1 dot	6	60	44/64 (11/16)	2.75
22	2	8	61	45/64	2.813
23	3	12	62	46/64 (23/32)	2.875
24	4	16	63	47/64	2.938
25	4/3	5.333	64	49/64 (49/64)	3.063
26	5/4	5	65	50/64 (25/32)	3.125
27	5/64	0.313	66	51/64	3.188
28	7/64	0.438	67	52/64 (13/16)	3.25
29	9/64	0.563	68	53/64	3.313
30	10/64 (5/32)	0.625	69	54/64 (27/32)	3.375
31	11/64	0.688	70	55/64	3.438
32	13/64	0.813	71	56/64 (7/8)	3.5
33	14/64 (7/32)	0.875	72	57/64	3.563
34	15/64	0.938	73	58/64 (29/32)	3.625
35	17/64	1.063	74	59/64	3.688
36	18/64 (9/32)	1.125	75	60/64 (15/16)	3.75
37	19/64	1.188	76	61/64	3.813
38	20/64 (5/16)	1.25	77	62/64 (31/32)	3.875
39	21/64	1.313	78	63/64	3.938

Values from Shortest to Longest

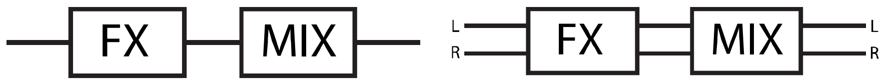
VALUE	AKA	INDEX	VALUE	AKA	INDEX
1/64 trip	1/96th	*1*	33/64		*49*
1/64		*2*	34/64 (17/32)		*50*
1/32 trip	1/48th	*4*	35/64		*51*
1/64 dot		*3*	36/64 (9/16)		*52*
1/32		*5*	37/64		*53*
1/16 trip	1/24th	*7*	38/64 (19/32)		*54*
1/32 dot	3/64th	*6*	39/64		*55*
1/16		*8*	40/64 (5/8)		*56*
5/64		*27*	41/64		*57*
1/8 trip	1/12th	*10*	42/64 (21/32)		*58*
1/16 dot	3/32nd	*9*	1 trip	2/3	*19*
7/64		*28*	43/64		*59*
1/8th		*11*	44/64 (11/16)		*60*
9/64		*29*	45/64		*61*
10/64 (5/32)		*30*	46/64 (23/32)		*62*
1/4 trip	1/6th	*13*	47/64		*63*
11/64		*31*	1/2 dot	3/4	*18*
1/8 dot	3/16th	*12*	49/64 (49/64)		*64*
13/64		*32*	50/64 (25/32)		*65*
14/64 (7/32)		*33*	51/64		*66*
15/64		*34*	52/64 (13/16)		*67*
1/4		*14*	53/64		*68*
17/64		*35*	54/64 (27/32)		*69*
18/64 (9/32)		*36*	55/64		*70*
19/64		*37*	56/64 (7/8)		*71*
20/64 (5/16)		*38*	57/64		*72*
21/64		*39*	58/64 (29/32)		*73*
1/2 trip	1/3rd	*16*	59/64		*74*
22/64 (11/32)		*40*	60/64 (15/16)		*75*
23/64		*41*	61/64		*76*
1/4 dot	3/8th	*15*	62/64 (31/32)		*77*
25/64		*42*	63/64		*78*
26/64 (13/32)		*43*	1	whole	*20*
27/64		*44*	5/4		*26*
28/64 (7/16)		*45*	4/3		*25*
29/64		*46*	1 dot	3/2	*21*
30/64 (15/32)		*47*	2	2 bars	*22*
31/64		*48*	3	3 bars	*23*
1/2		*17*	4	4 bars	*24*

16.9 Mono and Stereo

Here are some points about stereo and mono operation of the Axe-Fx II.

1. **Each Grid row is Stereo –** Many new users don't initially realize that a single path through the grid is already in full stereo. You don't need two rows for this! There are four full-stereo paths, then, from the input to the output.

What looks like this in the grid… …is actually this.



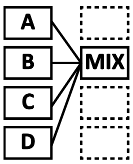
2. **Different blocks process in different ways** — Some blocks (AMP, CAB, Drive, and the pedal compressor) will sum the signal to mono at their inputs and process in mono, but the mono mix appears at both Left and Right outputs for the block. Other blocks—Ping-Pong Delay or Reverb for instance—sum the input to mono but then process and output stereo signals. Other blocks, for example the chorus or wah, have stereo inputs, stereo processing, and stereo outputs. The **Effects Guide**, beginning on p. **39**, covers every block and its processing in detail.
3. **Mono Summing vs. Splitting** – When you need to run the Axe-Fx II in MONO, several options determine how otherwise stereo signals will be processed:
   a. **Half-Stereo**: By leaving the Axe-Fx II in stereo and connecting only the left main output to a mono input, you get "half-stereo." This works fine, with the caveat that the right channel will not be heard! Panning will impart volume changes, and the Ping-Pong Delay and other wide effects will need adjustment to be heard somewhat as intended. Tone may also change if panned cabs or amps have been used.

   b. **Summed Mono**: By choosing "SUM L+R" mode for either OUTPUT 1 or OUTPUT 2, the two channels are added together and the resulting signal produced at both LEFT and RIGHT jacks. This has the advantage of including all sounds intended for both stereo channels, but short delay or phase differences between channels can result in strange artifacts or even total cancellation. This setting is best when you're confident that the presets you'll be playing have been designed for, or tested, in a SUMMED MONO listening environment. The Enhancer block and other short delays could be problematic (including very short reverbs or the room simulator in the CAB block). PHASE REVERSE switches should also be set to "OFF" when using this setting.

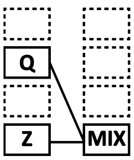
   c. Dual Mono: By choosing "COPY L>R" mode for either OUTPUT 1 or OUTPUT 2, you get a dual mono signal. The sound will be identical to that of half-stereo, with the same limitations, except it will be produced at both the left and right jacks. Use this setting when you want two mono outputs without the problems typically caused by summing L+R.
4. **Mono and Stereo at Once** – New for the Axe-Fx II is the ability to operate OUT2 as a summed MONO copy of the stereo OUT1 signal (or vice versa). Select COPY OUT1>OUT2, then choose the output mode for each pair of jacks that best suits your needs. See **Mono Summing** above for possible concerns.
5. The new **Global Blocks** feature (p. **131**) allows the Enhancer and other width-sensitive delay effects to be manipulated across multiple presets at once. It might be wise to employ this feature if you imagine needing to use a number of presets in either stereo or mono situations.

16.10 **Mixology**

The MIXER and FX LOOP blocks—and every preset's output mixer—all contain **4-channel mixers**. As on a real mixer, the **LEVEL**/**GAIN** parameters on the input channels determine what comes into the mixer. The overall mix may then be acted upon by master output **GAIN** or **LEVEL**, and, where applicable, master balance controls.



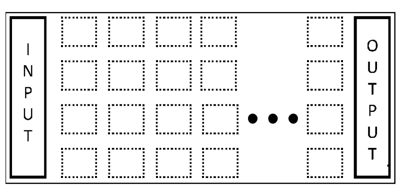
The four channels of an Axe-Fx II mixer are "wired" to the four grid rows in the column to the left of the mixer. In the first example at left, the level of Effect **A** would be controlled by the mixer's CHANNEL 1 **LEVEL/GAIN** control, because **A** is on grid row 1. Effect **B** would be controlled by Channel 2, because B is on row 2, and so on.

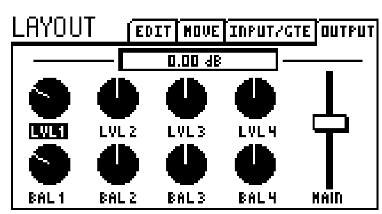


In the second example, the level of Effect Q would be controlled by the mixer's CHANNEL 2 **LEVEL/GAIN** control, because **Q** is on row 2. Effect **Z** would be controlled by Channel 4, because Z is on row 4. The location of the mixer never matters, only the location of blocks connected to its inputs!

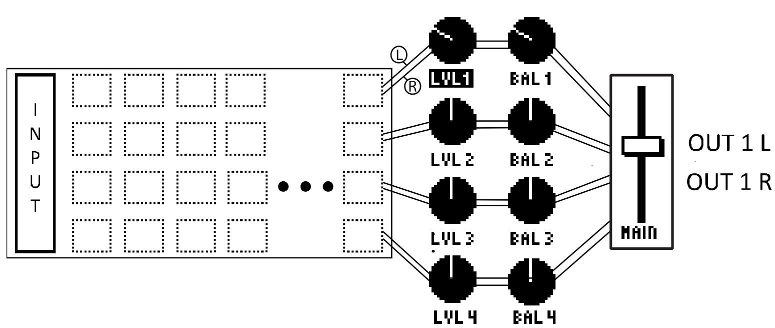
All mixer inputs are **stereo**. The **BALANCE** controls determine the left/right output balance of each channel. When the balance for an input channel is set fully left, only the left channel of the block connected at that channel will be heard at the mixer's output; its right channel is silenced. Remember that every BLOCK in the Axe-Fx II has Left and Right outs, even in cases where the signal is summed to dual mono (as is the case with the AMP, CAB, pedal type Compressor, and, depending on their settings, certain other blocks as well).

Knowing what you now know about Axe-Fx II mixers, take a look at the Output mixer of a preset, located on the MIX tab of its LAYOUT menu. On the surface, the output mixer looks like this:





But you can easily understand that these controls actually perform the functions illustrated below:



To review: channels 1-4 of the OUTPUT mixer are fed respectively by the stereo outputs of rows 1-4 of the grid. Each "channel" has a LEVEL that determines the gain of the incoming signal (+/- 20dB) and a BALANCE control that determines how the LEFT and RIGHT channels contribute to the final mix. A master GAIN slider offers +/-20 dB control over the final mixed output.

16.11 **Humbuster™ Technology**

All 1/4" outputs of the Axe-Fx II feature our new **Humbuster™** technology, which senses and subtracts the ground noise of connected equipment using a simple stereo-to-mono cable. This provides up to a 20 dB reduction in ground noise without resorting to dangerous "cheater plugs," expensive isolation transformers, or other noise reduction devices or methods. It is especially helpful when using the Axe-Fx II with a device like an amp head, which can both add and amplify ground hum, but it provides similar benefits when connecting to the 1/4" unbalanced inputs of any device: full-range cabs, mixers, and other processors or devices. Humbuster technology causes no signal loss or degradation.

To take advantage of this feature, you must connect the 1/4" output(s) of the Axe-Fx II to unbalanced 1/4" input(s) of another device using a special TRS-to-TS cable. The wiring requirements for this cable are shown below.

Figure 16-2 - Humbuster Cable

Axe-Fx Side	Cable	Remote Side
Tip	Wire 1 (hot)	Signal
Ring	Wire 2 (ground)	Ground
Sleeve	Shield	Ground

OUTPUT 1 L+R Unbalanced, and **OUTPUT 2 L+R** feature Humbuster noise reduction.

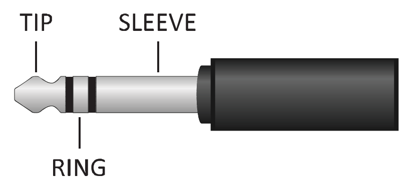
It is also possible to use these outputs with normal 1/4" cables.

16.12 **Setting up a Wah Pedal**

16.12.1 Using the Onboard Pedal Jack

Setting up a pedal for use as a wah (or a whammy, or a volume, etc.) is a four-step process.

1. Connect the pedal.
2. Calibrate the pedal.
3. Assign the pedal to an External Controller.
4. Assign the External Controller to the Wah parameter.



Here are the steps in greater detail:

1) Connect an expression pedal to the PEDAL 1 jack of the Axe-Fx II XL/XL+. (Note that the Axe-Fx II Mark 2 has only one un-numbered pedal jack. You'll need to adjust these instructions accordingly.) Expression pedals use Tip-Ring-Sleeve cables and typically have a linear resistance taper.
2) Calibrate the pedal.
   a. Press I/O
   b. Change to the PEDAL page.
   c. Select the PEDAL 1 CAL parameter.
   d. Press ENTER. Move the pedal through its full range of motion several times.
   e. Press ENTER.
3) Assign the PEDAL jack to an External Controller.
   a. Press I/O. Change to the CTRL page.
   b. Select the EXT 1 parameter.
   c. Change its value from "16" (the default) to PEDAL 1.
4) Assign **External 1** to control the Wah.
   a. Press LAYOUT to view the grid.
   b. Select or insert a WAH block.
   c. Press EDIT to open its edit menu.
   d. Find and select the **CONTROL** parameter.
   e. Press ENTER to view the MODIFIER screen.
   f. Change the **SOURCE** to EXT 1. Test the pedal to see the dot on the screen moves. If it doesn't move, check the above steps again, starting from #1, or try a different pedal/cable.
      i. OPTIONAL: to change the FEEL of the Wah, try adjusting the value of **MID**.
      ii. OPTIONAL: to make the Wah "smoother," adjust **DAMPING** to a value from 1–20 ms.
      iii. OPTIONAL: to set the Wah so it turns on and off automatically when you move the pedal, scroll down on the MODIFIER SCREEN and change the value for AUTO ENG to anything but "OFF." (See p. **140** for more details)
   g. Press EXIT to return to the WAH edit menu. Make other adjustments as desired, such as setting the range using the **MIN/MAX FREQUENCY**, or adjusting the **RESONANCE** and **TRACKING**.
   h. STORE the preset to save the new wah and modifier settings.

Remember that MODIFIER assignments are not global. You must repeat Step 4 for every preset in which you want the pedal to control the Wah (or use RECALL EFFECT; see p.**37**). The advantage of this, however, is flexibility: you will be able to use the same pedal to control multiple effects in multiple presets. See **Modifiers & Controllers** on p.**136** for more information.

16.12.2 Using an Expression Pedal on an MFC-101

The process for using an expression pedal connected to a Fractal Audio Systems MFC-101 MIDI Foot controller is almost the same as that required for the onboard PEDAL jack.

1. Connect the pedal to the MFC-101 **PEDAL 2** jack. (Pedal 1 is pre-programmed for VOLUME control.)
2. Calibrate the pedal.
3. With default settings on both MFC-101 and Axe-Fx II, no custom I/O parameters are required.
4. Assign the External Controller to the Wah parameter**.**

Here are the steps in greater detail:

Please note that this section assumes your MFC-101 has factory default settings. To return an MFC-101 to factory default settings, press the #11 footswitch and reboot the unit by disconnecting it from and reconnecting it to power. **WARNING! THIS WILL ERASE ALL USER SETTINGS ON THE MFC-101!**

1) Connect an expression pedal to the #2 onboard pedal jack of the MFC-101. (Pedal Jack #1 is pre-configured at the factory to control OUT1 VOLUME on the Axe-Fx II.)
2) Calibrate the pedal according to the instructions in Section 7.1 of the MFC-101 Owner's Manual. **Be sure to change to "XP2" at Step 5!**
3) Since Expression Pedal #2 on the MFC-101 is pre-configured to send MIDI CC# 16, and MIDI CC#16 is pre-configured to control EXTERNAL 1 on the Axe-Fx II, no special settings are required in the I/O menu for this tutorial.
   If you were using a different MIDI controller, or if you had XP2 assigned to a different CC#, you would use the CTRL page of the I/O menu to set the desired EXTERNAL CONTROLLER to the desired MIDI CC number.
4) Assign **External 1** to control the Wah:
   a. Press LAYOUT to view the grid.
   b. Select or insert a WAH block.
   c. Press EDIT to open its edit menu.
   d. Find and select the **CONTROL** parameter.
   e. Press ENTER to view the MODIFIER screen.
   f. Change the **SOURCE** to EXT 1. Test the pedal to see the dot on the screen moves. If it doesn't move, check the above steps again, starting from #1, or try a different pedal/cable.
      i. OPTIONAL: to change the FEEL of the Wah, try adjusting the value of **MID**.
      ii. OPTIONAL: to make the Wah "smoother," adjust **DAMPING** to a value from 1–20 ms.
      iii. OPTIONAL: to set the Wah so it turns on and off automatically when you move the pedal, scroll down on the MODIFIER SCREEN and change the value for AUTO ENG to anything but "OFF." (See p. **140** for more details)
   g. Press EXIT to return to the WAH edit menu. Make other adjustments as desired, such as setting the range using the **MIN/MAX FREQUENCY**, or adjusting the **RESONANCE** and **TRACKING**.
   h. STORE the preset to save the new wah and modifier settings.

Again, please recognize that MODIFIER assignments are not global. You must repeat Step 4 for every preset where you want the pedal to control the Wah. The advantage of this, however, is that in another preset you will be able to use the same pedal to control a Whammy, or the feedback of a delay, or any other parameter that allows the use of a Modifier. See Modifiers & Controllers on p. **136** for more information.

16.13 **Setting Up Spillover**

Spillover allows delay and reverb tails to ring out when an effect is bypassed or when you change presets. This method details how to set up spillover when using different presets. Firmware 9.0 added SCENES capability, which makes it easier to get perfect spillover within a single preset as detailed in the first section below. (See p. 186)

16.13.1 Within a Single Preset

The first case is the easiest to set up and requires almost no special settings. To enable tails to ring when an individual delay or reverb effect is **bypassed** (by a footswitch or scene change for instance), simply change its BYPASS MODE to "**MUTE FX IN.**" The explanation of Bypass Mode on p. **128** explains why and how this works.

Scenes allow blocks to be engaged or bypassed automatically one-by-one or in groups. This is one of the best and most popular ways to create presets with perfect spillover effects.

16.13.2 Across Different Presets

Setting up spillover that works across different presets is a bit more involved. The first step is to set **DELAY SPILL** on the CONFIG page of the GLOBAL menu (p. **145**) according to whether you want Delays, Reverbs, or "BOTH" to spill over when you change presets. ("Delay" does not include Multi-Delay or Megatap blocks).

Then, you need to ensure that the same delay or reverb blocks exist in both the preset you are changing from and the preset you are changing to. These need to be not only the same block but the same INSTANCE (i.e. you must use **Delay 1** and **Delay 1** vs. **Delay 1** and **Delay 2**).

The moment you change to a new preset, the current settings for its delay or reverb blocks "take over" processing the tails. If you change from a preset where delay has a time of 500 ms to one where the time is 100 ms, the tails will be "inserted" into the new effect and be heard as 100 ms echoes. For spillover to work *perfectly* then, the pair(s) of blocks in both "starting" and "landing" presets must have essentially identical settings and be placed in similar routing architectures. You would hear quite a sudden difference in the tail, for instance, if a delay was placed after a clean amp in the first preset and in front of a heavily overdriven amp in the second.

Bypass states and **BYPASS MODE** settings must also be considered. Switching from a preset where delay or reverb is engaged to one where it is bypassed with a **BYPASS MODE** setting of "MUTE FX OUT" will prevent the tails from being heard. Switching to a preset where the block is bypassed with a setting of "MUTE FX IN" however, will cleverly allow the tails to ring while material you play after the preset change will be heard *without* the effect.

Some smart shortcuts exist for setting up reverb and delay presets for spillover. If you create one preset as desired, you can save a copy to a new location and make changes only to other blocks. Using Global Blocks is another way to ensures that mix, level, and other important settings will be consistent (though block routing and sequence on the grid could still be a concern). Using the RECALL EFFECT function (p. **37**), you can "import" a delay or reverb block from another preset. This certainly beats pencil and paper as a means to transfer settings. Finally, Axe-Edit, our free companion editor/librarian to the Axe-Fx line, offers numerous conveniences, like copying/pasting blocks from one preset into various others and the ability to keep a "library" of effect block templates that can be inserted into any preset at any time.

16.14 **Using Send and Return**

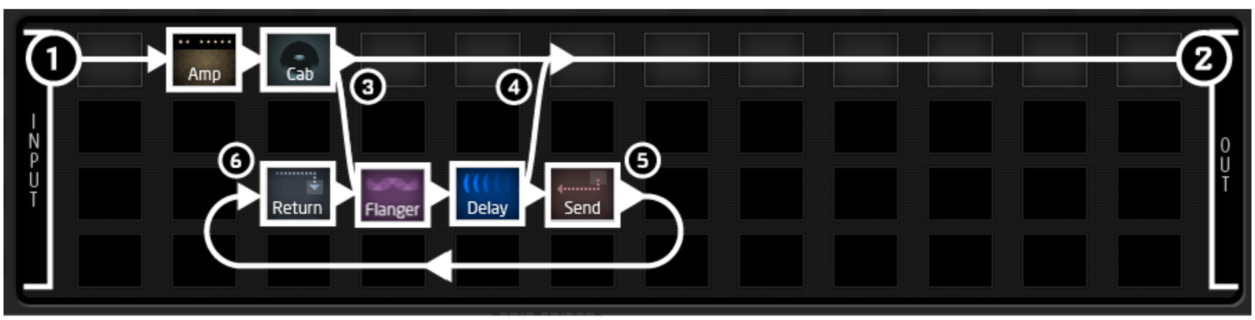
The **Feedback Send** and **Feedback Return** blocks (p.73) have two main uses: creating feedback loops and extending the length of effects chains beyond the size of the grid.

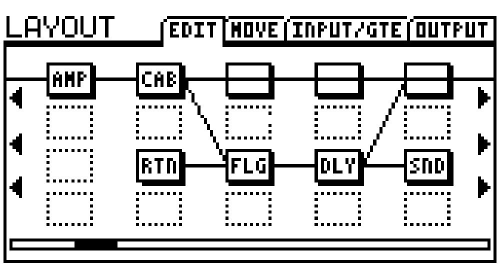
16.14.1 Creating Feedback Loops

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 exaggerated overlay of a preset illustration rendered in Axe-Edit.

The "Main" signal enters the grid at (1), passes through the AMP and CAB, and reaches the output (2), where it is heard at the speakers. Meanwhile, a tap of this main line at (3) carries signal to a Flanger and a Delay, set to 500 ms, no feedback, 100% wet. Nothing comes out of the Delay for 500 ms. Then, the first flanged echo emerges in the mix with the main line at (4), where it is heard at (2). This echo simultaneously enters the Send block (5) and is routed to the Return (6). From here it again passes through the Flanger—where the effect "stacks" on the last pass through—and hits the delay again. This loop of Return→ Flanger→ Delay→ Send→ Return→ would continue forever if the FB Return block didn't make it a bit quieter each time so that it gradually fades away. Each time the loop exits the Delay, we get to listen in on the current state of things as it is routed through (4) to the output (2).

So to review, the send "transports" signal to the return, where **LEVEL** controls the amount of feedback.





At left, the significant columns of this preset are shown as they would appear on the display of the Axe-Fx II. Notice that on the Axe-Fx II the connector from SEND to RETURN is *not visible*!

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.

16.14.2 Extending the Length of Effect Chains

The 4x12 grid will suffice for the vast majority of long, complex routings. However, certain “Axe-aholics” require a way to break this virtual sound barrier and exceed the number of columns available in the grid. The Feedback Send and Return can be used for just this purpose. Place a **Send** block at the end of your first long chain and place the **Return** block at the beginning of another, setting the return **MIX** to “100%” and the level to 0 dB. Continue through other effects to the output as illustrated below.

The real preset below with 19 effects hits just over 90% on the CPU load meter. Should’ve gone for 20!
(The CAB block **MODE** here is set to MONO LORES by the way. That’s a great way to get the most out of your CPU!)

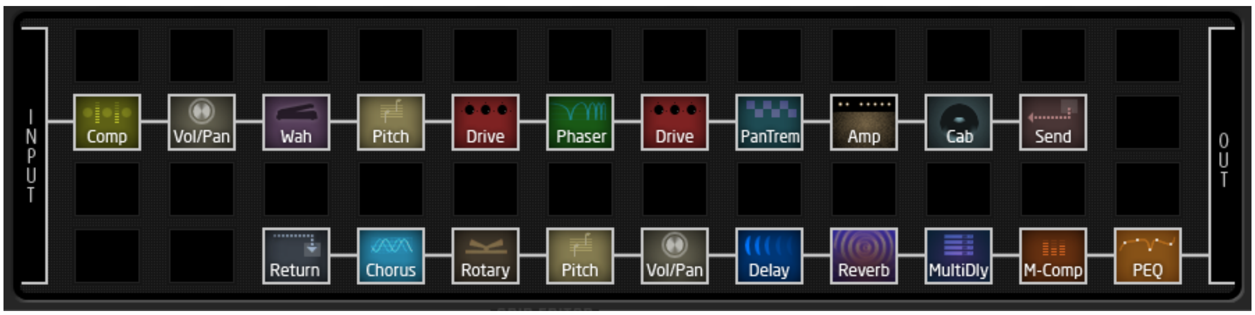


Figure 16-3 – A Giant Preset with SEND/RETURN

16.15 **Scenes**

Note: A separate "MINI MANUAL" also covers SCENES. Find it on our web site support page at:
http://www.fractalaudio.com/downloads/manuals/axe-fx-2/Axe-Fx-II-Scenes-Mini-Manual-1.02.pdf

In addition to amp and cab modeling, an Axe-Fx II preset can contain a number of pre- or post-effects—an entire virtual guitar rig with tremendous flexibility and control. For flexibility and control in "old-school" rigs, a device called a *switcher* is used to switch effects into or out of the signal path. Switchers also have presets, which reduce tap-dancing by providing single-stomp access to different sets of pedals.

Until now, simulating this capability on the Axe-Fx has required multiple presets or fancy foot controller programming. The multi-preset approach, however, was not without certain drawbacks. Setup and maintenance could be challenging, changes were not always gapless, and it took care to get levels and spillover right.

To solve this we created **SCENES**. Every Axe-Fx II preset now contains eight **SCENES**. Each scene stores the BYPASS setting for every block in the preset, the X/Y selection for those blocks that support it, the preset's MAIN output level, and the MAIN output of the FX LOOP block. Scene changes are seamless and instantaneous, with perfect spillover requiring almost no effort. Scenes can easily be selected from the front panel or with a MIDI foot controller such as the MFC-101.

In the example below, three scenes of an Axe-Fx II preset are shown. AMP and CAB remain ON in all three scenes. In Scene 1 ("S1," top) DELAY and REVERB are engaged. In Scene 2 (mid), DELAY is switched off, while COMPRESSOR and PHASER are simultaneously switched ON. In Scene 3 (bottom), COMPRESSOR, DRIVE, CHORUS, MULTITAP DELAY and REVERB are on. X-Y states and preset MAIN output level can also be set differently in each scene.

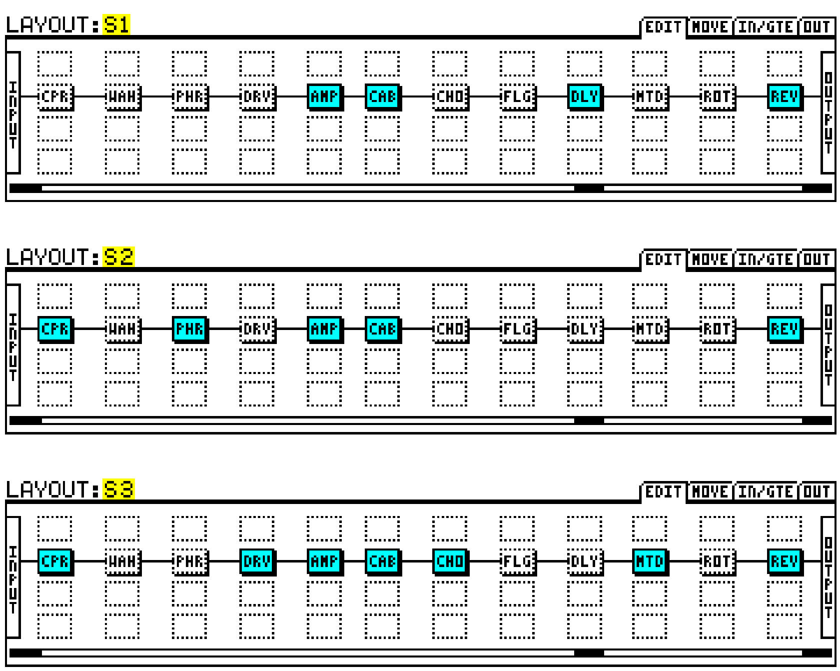


Figure 16-4 - Block Bypass Changes Across Three Scenes

16.15.1 Selecting Scenes

The current scene is shown on RECALL:PRESET screen, and on all pages of the LAYOUT menu.

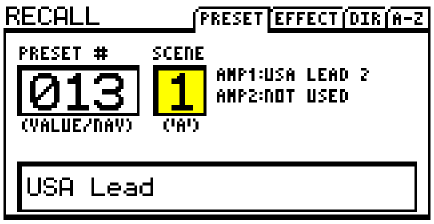


Figure 16-5 - "Scene 1" shown on the RECALL:PRESET screen

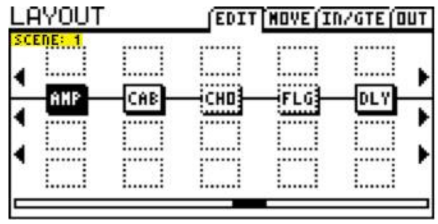
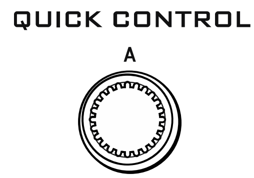


Figure 16-6 "Scene 1" shown on the LAYOUT:EDIT (grid) screen.

To select a scene, turn the "A" QUICK CONTROL knob on the front panel of the Axe-Fx. Scenes can also be selected via MIDI. See SCENES AND MIDI on p. 186.



16.15.2 SETTING UP SCENES

To Set Up Scene Bypass States…

- Press LAYOUT to display the EDIT (grid) page.
- Select the desired scene by turning the "A" QUICK CONTROL knob.
- Use the NAV and FX BYP buttons to set up the scene as desired.

Note: All blocks in a preset will be engaged by default in scenes 2–8.

To set the X/Y STATES for a given block in a given scene…

- Select the scene as described above.
- Navigate to the desired block and enter its EDIT menu.
- Press X or Y to set the desired state.

Note: X/Y is supported for Amp, Cab, Chorus, Delay, Drive, Flanger, Pitch, Phaser, Reverb & Wah.

To set the MAIN output level for a scene…

- Press LAYOUT.
- Page to the OUT screen.
- Set the level using the MAIN output level fader.

Note: MAIN output level controls everything you hear, including the level of effect "tails."

If you're using Axe-Edit, you may enjoy the ability to copy and paste scenes, found by right- or control-clicking the SCENE switches.

16.15.3 SAVING SCENES

To save scene changes, simply save the preset. When using the "A" QUICK CONTROL knob to select scenes, you can make changes across multiple scenes before saving. If you're using MIDI to select scenes, you may need to save changes to the current scene before selecting a new one, depending on the setting of SCENE REVERT.

16.15.4 SPILLOVER IN SCENES

BYPASS MODE (on the MIX page of most block edit menus) determines how effects sound when bypassed by a scene change. Familiarize yourself with each of the options by reviewing **Bypass Mode Parameters** (Section 5.37 on page 128). The Bypass Mode option "MUTE FX IN" creates natural spillover when a delay or reverb is bypassed. See also the section entitled **Spillover Within a Single Preset** on p. 181.

16.15.5 SCENES AND MIDI

MIDI offers three ways to select scenes: Direct Selection, Increment/Decrement, and Program Change Mapping.

<u>Direct Scene Selection…</u>

Scenes can be selected directly using MIDI Control Change messages. SCENE SELECT in the I/O:CTRL menu determines which CC# is used. The default is CC#34. The value of the CC message determines which scene is selected. For the math-minded, the relationship between CC value and scene can be described as [('cc value' mod 8) + 1]. For the rest of us, you can just use "scene number minus one," consult the table at the end of this section, or use the MFC-101, where none of this matters except knowing which scene you want to select (see below).

Note: CC values of 0 and 127—or a switch on the Axe-Fx "PEDAL" jack—will load scenes 1 and 8.

<u>Scene Increment/Decrement…</u>

MIDI CCs can also be used to step up or down through scenes. The entries SCENE INCR and SCENE DECR in the I/O:CTRL menu designate CCs for each action . The defaults are CC#123:INCREMENT and CC#124:DECREMENT. Values from 64-127 trigger the action while values from 0-63 are ignored.

<u>Program Change to Scene Mapping…</u>

The Axe-Fx II MIDI Program Change Map (See the Axe-Fx II Owner's Manual, Section 9.3) has been updated so that SCENE as well as PRESET can be mapped for each incoming program change message. **Ignore Redundant PC** (under I/O: MIDI) must be **ON** for seamless scene changes via PC.

To use program changes for scene selection:

- Set MAPPING MODE to "CUSTOM."
- Select an entry for MAP FROM PRESET.
- Select the desired entries for MAP TO PRESET and MAP TO SCENE.

Like other I/O and Global parameters, the **CUSTOM Map** is dynamic and does not need to be saved.

Note: Custom Program Change mapping (with or without scenes) is <u>incompatible</u> with MFC-101 "TotalSync," with Axe-Edit, and with certain Bank Dump and other MIDI Activities. Please set MAPPING MODE back to "NORMAL" if you require compatibility with these features.

<u>Scene Revert…</u>

This option on the MIDI page of the I/O menu selects between one of two behaviors for scene recall via MIDI.

With **SCENE REVERT** "OFF" (default) modifications to the current Scene are "semi-permanent" across scene changes. So if you tweak Scene 1, switch to Scene 2, then back to scene 1, your tweaks will remain intact—until you load an entirely new preset or reload the current one.

With **SCENE REVERT** “ON”, modifications are lost when you change the scene without saving first. 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.

Just be aware of the fact that with **SCENE REVERT “**OFF”, you might easily forget about changes to one scene and save the preset while a different scene is loaded. When setting up SCENES, it is recommended to set up and save each scene one-by-one, and to test thoroughly.

A caution for those using a 3rd party MIDI controller: pre-programmed IA switch states which “disagree” with the states you saved per SCENE will cause audio to glitch, IA LED states to not match the effect states heard, or worse…

16.15.6 MFC-101 Scene Features

MFC-101 firmware version 2.15 added a number of features to make working with Axe-Fx II scenes faster and easier. It is possible to directly assign any scene (SCENE 1, 2, 3 etc.) or to select “**SCENE INCR**”, “**SCENE DECR**”, or **“SCENE 1/2" toggle** as the function for any Axe-Fx Mode IA switches. Changing scenes (whether on the Axe-Fx II front panel or remotely) also updates MFC-101 LEDs for all **Axe-Fx Mode** IA Switches. See your MFC-101 manual for more information on how to assign Axe-Fx Mode IA Switch functions. (OK, here’s a crash course:

1. Press **EDIT**
2. Press **MIDI** (#4)
3. Press PAGE > (#7) 7x to select **IA01 Axe-Fx SCENE1** (You may have changed this default already.)
4. Use **UP** and **DOWN** to select the desired IA switch by its number (ex: IA07)
5. Use User **PARAMETER >** (12#) to select, followed by **UP**/**DOWN** to set the desired Axe-Fx function.
6. Press **EDIT** to SAVE/EXIT.

16.15.7 Table of CC# Values for Scene Select

As described above, scenes may be set using a CC# message (#34 is the default but this may be changed under I/O:CTRL:SCENE SELECT). The *value* of the incoming control change message determines which scene is selected.

VAL	SCN	VAL	SCN	VAL	SCN	VAL	SCN	VAL	SCN	VAL	SCN	VAL	SCN	VAL	SCN	VAL	SCN
0	1	**17**	1	**16**	1	**32**	1	**48**	1	**64**	1	**80**	1	**96**	1	**112**	1
1	2	**18**	2	**17**	2	**33**	2	**49**	2	**65**	2	**81**	2	**97**	2	**113**	2
2	3	**19**	3	**18**	3	**34**	3	**50**	3	**66**	3	**82**	3	**98**	3	**114**	3
3	4	**20**	4	**19**	4	**35**	4	**51**	4	**67**	4	**83**	4	**99**	4	**115**	4
4	5	**21**	5	**20**	5	**36**	5	**52**	5	**68**	5	**84**	5	**100**	5	**116**	5
5	6	**22**	6	**21**	6	**37**	6	**53**	6	**69**	6	**85**	6	**101**	6	**117**	6
6	7	**23**	7	**22**	7	**38**	7	**54**	7	**70**	7	**86**	7	**102**	7	**118**	7
7	8	**24**	8	**23**	8	**39**	8	**55**	8	**71**	8	**87**	8	**103**	8	**119**	8
8	1	**25**	1	**24**	1	**40**	1	**56**	1	**72**	1	**88**	1	**104**	1	**120**	1
9	2	**26**	2	**25**	2	**41**	2	**57**	2	**73**	2	**89**	2	**105**	2	**121**	2
10	3	**27**	3	**26**	3	**42**	3	**58**	3	**74**	3	**90**	3	**106**	3	**122**	3
11	4	**28**	4	**27**	4	**43**	4	**59**	4	**75**	4	**91**	4	**107**	4	**123**	4
12	5	**29**	5	**28**	5	**44**	5	**60**	5	**76**	5	**92**	5	**108**	5	**124**	5
13	6	**30**	6	**29**	6	**45**	6	**61**	6	**77**	6	**93**	6	**109**	6	**125**	6
14	7	**31**	7	**30**	7	**46**	7	**62**	7	**78**	7	**94**	7	**110**	7	**126**	7
15	8	**32**	8	**31**	8	**47**	8	**63**	8	**79**	8	**95**	8	**111**	8	**127**	8

16.16 **Modifier Power!**

This list used to appear at the end of the MODIFIERS chapter. It was moved here to make space there for updated content. Here is a list of ideas for the MODIFIER feature of the Axe-Fx II:

1. Eliminate conspicuous chorus “pulsing” with subtle **RATE** modification. Try an LFO or Envelope.
2. Modify the INPUT GAIN of a Delay or Reverb and create an “EFFECTS SEND” pedal to feed the effects.
3. Create a “Power Saturation” pedal to increase an amp’s **MASTER** while compensating its **LEVEL.**
4. “Double-whammy” goes UP and DOWN…at the same time. (2 PITCH blocks, one pedal).
5. Create a groovy FILTER by assigning **FREQUENCY** to a “QUANTIZED” TRI wave LFO sync’d to tempo.
6. Ducking reverb. Run the effect 100% wet (parallel to the dry) then use ENVELOPE to lower its level dynamically.
7. Place a low-pass or peaking FILTER in front of a DRIVE and control its frequency for “foot tone fuzz.”
8. Crossfade between two different signal paths by controlling different MIXER channels inversely.
9. Touch-Wahs are old news. Try an envelope-controlled FORMANT, PHASER, FLANGER, or RINGMOD.
10. Create a Ray-Gun. Modify Global LFO1 **DEPTH** with a pedal (0-100%). Assign LFO1 to the **CONTROL** parameter of a Whammy (Pitch Block) set to +/- 1 8va.
11. Reclaim pedalboard real estate with a “Wildcard” footswitch. (Hi Dweezil!) Set it to the CC# of an External Controller. Then assign this EXT source to the **BYPASS MODE** modifiers of different effects across different presets, or anything else you can imagine...
12. Map SCENE CONTROLLER 1 to DRIVE to increase the number of amp sounds available within a single preset. Scene 1 might be fairly clean, Scene 2 light crunch, and scene 3 heavy drive. Try SCENE CONTROLLER 2 on MASTER!
13. Use the SEQUENCER to craft your own LFO shapes for a Phaser (FREQUENCY), FLANGER (TIME), FILTER (FREQ), or Tremolo (use the VOLUME block). Applying some DAMPING in the modifier to change sudden steps into smooth transitions. Assign another modifier set to an external source to SEQUENCER:RUN and you’ll have a way to start/stop the new curves with your foot.
14. Traditional Rotary speakers have a SPEED footswitch. It’s easy to assign ROTARY:RATE to any EXTERNAL source connected to a latching footswitch. Use MIN and MAX on the modifier to set just how slow or how fast you want each setting to be.
15. The MIXER block allows modifiers on every GAIN and BALANCE parameter. With SCENE controllers, this becomes an extremely powerful way to route signals and meet diverse needs within a single preset. For example, two amps could be panned hard left and right in SCENE 1, then dead center with a few dB knocked off in SCENE 2, etc.
16. Use the SEQUENCER in traditional fashion to change SHIFT of a SYNTH block for some “On the Run” action (or create your own using this table (Hint, the front panel will only allow you to enter values with 0.1 precision, but Axe-Edit allows you to enter 0.01 values ;-)
17. What great ideas do you have? Share them at http://forum.fractalaudio.com

16.17 **Glossary & Resources**

4CM: See “Four Cable Method.”

A/D, D/A Converter: Analog-to-Digital or Digital-to-Analog Converter.

ADSR: Technically, this stands for Attack, Decay, Sustain, Release but used as a noun it refers to an Envelope Generator which, when “triggered,” produces a control signal that can be used to change parameter values in a way that is predictable over time. Envelopes are typically “one-shot,” meaning they play through and stop. But when set to loop, they can behave more like LFOs (below).

AES/EBU: Audio **E**ngineering **S**ociety/**E**uropean **B**roadcasting **U**nion. The name applied to a professional audio interface used for transferring digital audio between devices. AES for short. AES and S/PDIF both supply the same audio data with slight differences in the frame bits.

Aliasing: Aliasing in digital audio refers to the phenomenon that happens when we try to reproduce frequencies higher than one-half the sampling rate. See http://www.earlevel.com/main/1996/10/20/what-is-aliasing for an excellent description.

Axe-Edit: The companion editor/librarian software for the Axe-Fx line. Download it from our web site at http://www.fractalaudio.com

Balanced/Unbalanced: Balanced refers to audio signals designed to be carried over a three-conductor cable, which minimizes unwanted noise and interference. Cables designed to carry balanced signals are called Balanced Cables and usually use the XLR or Tip-Ring-Sleeve (TRS) connector end-types.

BPM: Beats per minute. A measure of musical tempo. The typical human heartbeat is about 60–80 BPM.

CPU: Central Processing Unit.

dB: Decibels. The unit for measuring sound intensity, or loudness. You’ll see this on level or volume parameters. There are abundant resources about the science behind loudness, but simply making a few adjustments can get your dB sense going.

Deg: Degrees (360ths of a circle) are used on the Axe-Fx II to specify differences in stereo LFO phase.

DSP: Digital Signal Processor/Processing.

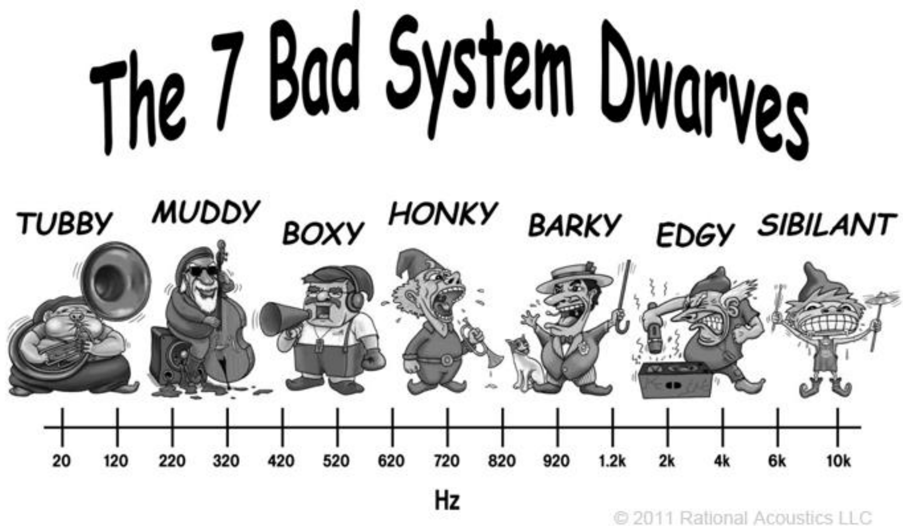
Feedback: When an output is connected to an input, feedback occurs. The connection might change mediums, as it does when sound travels from an amp’s speaker through the air to excite guitar strings connected to that amp’s input. It can also be direct, such as when some of the output of a flanger or phaser is routed back to the effect’s input. Feedback is also sometimes known in the effects world as “regeneration” and, less aptly, “resonance.”

Four Cable Method (4CM): A rig design where the Axe-Fx II is used both “in front of” and inside the effects loop of a tube amp. See p. **21** for a diagram and details.

Fractal-Bot: Fractal-Bot is an application for Mac or Windows computers which enables you to easily transmit firmware updates or other files to your Axe-Fx II. http://www.fractalaudio.com/fractal-bot.php

FRFR: "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 or de-emphasis. It is invariably an approximation.

Hz: Hertz. The number of times something happens in one second. 4 Hz = 4x/second. Low values for Hz are useful to describe RATES—the "speed" of a slow phaser's sweep might be 0.33 Hz, for example (once every three seconds). Hz are also the units used to plot low and high-frequency sounds. At the top of the frequency scale, units of kilohertz (kHz or just "k")—thousands of cycles per second—are more useful. You will see both Hz and kHz on equalizers, filters, and other effects that deal with sound as a "spectrum" of frequencies. With a little time, you'll learn to match the numbers with their effects on a sound. Check the web for more info, such as *The Guitar Player Book* (2007, free on Google Books), or get yourself this cool t-shirt from Rational Acoustics:



http://www.rationalacoustics.com/store/goodies/7-bad-system-dwarves-t-shirt.html

I/O: Stands for "Input/Output."

IR: Impulse Response. An Impulse Response file is a collection of data representing sound measurements taken from a speaker cabinet or system and used by the Axe-Fx II to enable the Cabinet block to emulate a particular speaker cabinet. A test signal is played through the actual speaker, recorded, and used to generate a profile utilized by the Axe-Fx II to reproduce the measured response.

Latency: In terms of effect processors, latency is an unwanted delay between what you play and what you hear. The Axe-Fx II latency is so low that it is equivalent to standing just a few feet away from a tube amp.

LFO: Low Frequency Oscillator. An LFO creates control signals used to periodically change sounds in real time. The back-and-forth sweep of a flanger or phaser, the sharp-flat wobble of a chorus, and the loud-soft pulse of tremolo are each the result of an LFO at work.

MIDI: Musical Instrument Digital Interface.

ms: Milliseconds. Thousandths of a second. 500 ms is one half-second. 100 ms is 1/10th of a second.

pF: Picofarads. You'll only see it on one parameter, the amp's BRIGHT CAP value, where it affects treble response.

Phantom Power: By leveraging an unused pair of copper wires inside of a MIDI or other cable, the Phantom Power system allows a single cable to carry both MIDI data and power between the Axe-Fx II and a connected floor unit.

Phase: This term is used to describe the position of one waveform relative to another. When two similar waves are in phase, their peaks and valleys will line up exactly, reinforcing one another. Waves that are out of phase have their peaks and valleys in opposition, so while one is headed up, the other is headed down. An LFO Phase control settings allows left and right sweeps to be either synchronized or offset from one another. An Audio Phase control, (such as the one in the Enhancer block or the PHASE REVERSE parameters of the Delay, Chorus, Flanger, and other effects) flips the polarity of an audio signal. (Some might say this turns the waveform upside down.) Combining two identical audio signals of opposite phase results in complete silence, explaining why you need to be careful when summing to mono any preset with blocks that invert phase or shift time alignment.

Resonance: Resonance is an increase in amplitude around certain frequencies. It results in the intensification or prolongation of certain components of a sound. In the Axe-Fx II, it is most commonly used in descriptions of the "Q" parameter of a filter or parametric equalizer. At lower values, Q determines the slope of the effect, and at higher settings, Q impacts the width and height of a peak that forms around the cutoff or center frequency.

RJ45: RJ45 is a type of standard connector cable end used in computer networking. Both Ethernet and its ruggedized counterpart EtherCON utilize RJ45 plugs and jacks.

S/PDIF: Sony/**P**hilips **D**igital **I**nter**F**ace

Semitones/Cents ("ct" or "cts"): Used to measure musical pitch. A semitone is one half-step, a.k.a. 1/12th of an octave—the difference heard from one guitar fret to the next. A cent is 1/100th of a semitone—extremely small in terms of your ability to hear a difference. You'll see these units on the Axe-Fx II Pitch Shifter and Synthesizer.

SysEx: Short for System Exclusive. A type of MIDI data that can be understood only by the particular make and model of MIDI device that created it. On the Axe-Fx II, it is used for presets, banks, system backups, and User Cab IR files, and to allow real-time control of the unit via a connected MFC-101 or computer running Axe-Edit (q.v.).

16.18 **Axe-Fx II Bank & Preset Numbers Table**

The following table shows the BANK SELECT and PROGRAM CHANGE commands needed to recall an Axe-Fx II preset via MIDI. DISPLAY OFFSET (p. **150**) has the potential to add +1 to all of the preset numbers (non-bold). Remember that any Bank Select message *stays in effect* until a new one is received of the Axe-FxII is rebooted. The Axe-Fx II Mark II has only three preset banks.

PC #	BANK A (CC#0 = 0)	BANK B (CC#0 = 1)	BANK C (CC#0 = 2)	BANK D* (CC#0 = 3)	BANK E* (CC#0 = 4)	BANK F* (CC#0 = 5)	PC#	BANK A (CC#0 = 0)	BANK B (CC#0 = 1)	BANK C (CC#0 = 2)	BANK D * (CC#0 = 3)	BANK E * (CC#0 = 4)	BANK F * (CC#0 = 5)
0	0	128	256	384	512	640	**64**	64	192	256	448	576	704
1	1	129	257	385	513	641	**65**	65	193	257	449	577	705
2	2	130	258	386	514	642	**66**	66	194	258	450	578	706
3	3	131	259	387	515	643	**67**	67	195	259	451	579	707
4	4	132	260	388	516	644	**68**	68	196	260	452	580	708
5	5	133	261	389	517	645	**69**	69	197	261	453	581	709
6	6	134	262	390	518	646	**70**	70	198	262	454	582	710
7	7	135	263	391	519	647	**71**	71	199	263	455	583	711
8	8	136	264	392	520	648	**72**	72	200	264	456	584	712
9	9	137	265	393	521	649	**73**	73	201	265	457	585	713
10	10	138	266	394	522	650	**74**	74	202	266	458	586	714
11	11	139	267	395	523	651	**75**	75	203	267	459	587	715
12	12	140	268	396	524	652	**76**	76	204	268	460	588	716
13	13	141	269	397	525	653	**77**	77	205	269	461	589	717
14	14	142	270	398	526	654	**78**	78	206	270	462	590	718
15	15	143	271	399	527	655	**79**	79	207	271	463	591	719
16	16	144	272	400	528	656	**80**	80	208	272	464	592	720
17	17	145	273	401	529	657	**81**	81	209	273	465	593	721
18	18	146	274	402	530	658	**82**	82	210	274	466	594	722
19	19	147	275	403	531	659	**83**	83	211	275	467	595	723
20	20	148	276	404	532	660	**84**	84	212	276	468	596	724
21	21	149	277	405	533	661	**85**	85	213	277	469	597	725
22	22	150	278	406	534	662	**86**	86	214	278	470	598	726
23	23	151	279	407	535	663	**87**	87	215	279	471	599	727
24	24	152	280	408	536	664	**88**	88	216	280	472	600	728
25	25	153	281	409	537	665	**89**	89	217	281	473	601	729
26	26	154	282	410	538	666	**90**	90	218	282	474	602	730
27	27	155	283	411	539	667	**91**	91	219	283	475	603	731
28	28	156	284	412	540	668	**92**	92	220	284	476	604	732
29	29	157	285	413	541	669	**93**	93	221	285	477	605	733
30	30	158	286	414	542	670	**94**	94	222	286	478	606	734
31	31	159	287	415	543	671	**95**	95	223	287	479	607	735
32	32	160	288	416	544	672	**96**	96	224	288	480	608	736
33	33	161	289	417	545	673	**97**	97	225	289	481	609	737
34	34	162	290	418	546	674	**98**	98	226	290	482	610	738
35	35	163	291	419	547	675	**99**	99	227	291	483	611	739
36	36	164	292	420	548	676	**100**	100	228	292	484	612	740
37	37	165	293	421	549	677	**101**	101	229	293	485	613	741
38	38	166	294	422	550	678	**102**	102	230	294	486	614	742
39	39	167	295	423	551	679	**103**	103	231	295	487	615	743
40	40	168	296	424	552	680	**104**	104	232	296	488	616	744
41	41	169	297	425	553	681	**105**	105	233	297	489	617	745
42	42	170	298	426	554	682	**106**	106	234	298	490	618	746
43	43	171	299	427	555	683	**107**	107	235	299	491	619	747
44	44	172	300	428	556	684	**108**	108	236	300	492	620	748
45	45	173	301	429	557	685	**109**	109	237	301	493	621	749
46	46	174	302	430	558	686	**110**	110	238	302	494	622	750
47	47	175	303	431	559	687	**111**	111	239	303	495	623	751
48	48	176	304	432	560	688	**112**	112	240	304	496	624	752
49	49	177	305	433	561	689	**113**	113	241	305	497	625	753
50	50	178	306	434	562	690	**114**	114	242	306	498	626	754
51	51	179	307	435	563	691	**115**	115	243	307	499	627	755
52	52	180	308	436	564	692	**116**	116	244	308	500	628	756
53	53	181	309	437	565	693	**117**	117	245	309	501	629	757
54	54	182	310	438	566	694	**118**	118	246	310	502	630	758
55	55	183	311	439	567	695	**119**	119	247	311	503	631	759
56	56	184	312	440	568	696	**120**	120	248	312	504	632	760
57	57	185	313	441	569	697	**121**	121	249	313	505	633	761
58	58	186	314	442	570	698	**122**	122	250	314	506	634	762
59	59	187	315	443	571	699	**123**	123	251	315	507	635	763
60	60	188	316	444	572	700	**124**	124	252	316	508	636	764
61	61	189	317	445	573	701	**125**	125	253	317	509	637	765
62	62	190	318	446	574	702	**126**	126	254	318	510	638	766
63	63	191	319	447	575	703	**127**	127	255	319	511	639	768

16.19 **Factory Default Settings**

GLOBAL CONFIG	
Modeling Version	NEWEST
Power Amp Modeling:	ON
Cabinet Modeling:	ON
Spillover:	BOTH
Global Reverb Mix (offset):	0 %
Global Effects Mix (offset):	0 %
Global Noisegate Threshold Offset	0 %
Global Amp Gain:	0 %
IR Capture Mode	ULTRA-RES
GLOBAL OUT 1	
EQ:	All bands flat (0.00)
GAIN:	Flat (0.00)
GLOBAL OUT 2	
EQ:	All bands flat (0.00)
GAIN:	Flat (0.00)
GLOBAL SCALES	
All scales (1-32), all degrees (A–G#):	-24 semitones
TUNER	
Calibration:	A4=440.0 Hz
Mute:	OFF
Use Offsets:	OFF
Offsets (E,A,D,G,B,E):	0.0 cts
I/O INPUT	
Instrument Input Level:	49.8%
Input 1 Level:	49.8%
Input 2 Level:	49.8%
I/O AUDIO	
Main Input Source:	ANALOG (IN 1)
Input 1 Left Select:	FRONT
Input 1 Mode:	LEFT ONLY
Input 2 Mode:	LEFT ONLY
Output 1 Mode:	STEREO
Output 1 Boost/Pad:	0 dB
Output 1 Phase:	NORMAL
Output 2 Mode:	STEREO
Output 2 Boost/Pad:	0 dB
Output 2 Phase:	NORMAL
Output 2 Echo:	OFF
S/PDIF / AES/EBU Select:	S/PDIF
USB / S/PDIF Out Source:	OUTPUT 1
USB Buffer Size	1024
I/O MIDI	
MIDI Channel:	1
MFC Port **(XL Only)**	DISABLED
MFC echo to MIDI OUT **(XL Only)**	OFF
MIDI Thru **(Mark II Only)**:	OFF
Receive Program Change:	ON
Mapping Mode:	NONE
(All 127 custom map entries)	1:1 (ex: 1=1, 2=2, 127=127)
Scene Revert	OFF
SysEx ID:	00 01 74 (cannot be changed)
Display Offset:	0
Ignore Redundant Program Change:	OFF
Send Realtime SysEx Messages:	ALL (= Tempo and Tuner)
MIDI Program Change Offset:	0
USB Adapter Mode:	OFF
Ext. Controllers 1-12 initial value:	0%

I/O CONTROL

Default CC assignments appear in the table below:

Function	CC#
Input Volume	10
Out 1 Volume	11
Out 2 Volume	12
Bypass	13
Tempo	14
Tuner	15
External Control 1	16
External Control 2	17
External Control 3	18
External Control 4	19
External Control 5	20
External Control 6	21
External Control 7	22
External Control 8	23
External Control 9	24
External Control 10	25
External Control 11	26
External Control 12	27
Looper Record	28
Looper Play	29
Looper Once	30
Looper Dub	31
Looper Rev	32
Looper Bypass	33
Looper Half	120
Looper Undo	121
Metronome	122
Scene Select	34
Scene Increment	123
Scene Decrement	124
Volume Increment	35
Volume Decrement	36
Amp 1 Bypass	37
Amp 2 Bypass	38
Cab 1 Bypass	39
Cab 2 Bypass	40
Chorus 1 Bypass	41
Chorus 2 Bypass	42

Function	CC#
Compressor 1 Bypass	43
Compressor 2 Bypass	44
Crossover 1 Bypass	45
Crossover 2 Bypass	46
Delay 1 Bypass	47
Delay 2 Bypass	48
Drive 1 Bypass	49
Drive 2 Bypass	50
Enhancer Bypass	51
Filter 1 Bypass	52
Filter 2 Bypass	53
Filter 3 Bypass	54
Filter 4 Bypass	55
Flanger 1 Bypass	56
Flanger 2 Bypass	57
Formant 1 Bypass	58
FX Loop Bypass	59
Gate/Expander 1 Bypass	60
Gate/Expander 2 Bypass	61
Graphic EQ 1 Bypass	62
Graphic EQ 2 Bypass	63
Graphic EQ 3 Bypass	64
Graphic EQ 4 Bypass	65
Megatap Delay Bypass	66
Multiband Comp 1 Bypass	67
Multiband Comp 2 Bypass	68
Multi-Delay 2 Bypass	69
Multi-Delay 2 Bypass	70
Parametric EQ 1 Bypass	71
Parametric EQ 2 Bypass	72
Parametric EQ 3 Bypass	73
Parametric EQ 4 Bypass	74
Phaser 1 Bypass	75
Phaser 2 Bypass	76
Pitch Shifter 1 Bypass	77
Pitch Shifter 2 Bypass	78
Quad Chorus 1 Bypass	79
Quad Chorus 2 Bypass	80

Function	CC#
Resonator 1 Bypass	81
Resonator 2 Bypass	82
Reverb 1 Bypass	83
Reverb 2 Bypass	84
Ring Modulator Bypass	85
Rotary 1 Bypass	86
Rotary 2 Bypass	87
Synth 1 Bypass	88
Synth 2 Bypass	89
Tone Matching	99
Tremolo 1 Bypass	90
Tremolo 2 Bypass	91
Vocoder Bypass	92
Volume/Pan 1 Bypass	93
Volume/Pan 2 Bypass	94
Volume/Pan 3 Bypass	95
Volume/Pan 4 Bypass	96
Wahwah 1 Bypass	97
Wahwah 2 Bypass	98
Amp 1 X/Y	100
Amp 2 X/Y	101
Cab 1 X/Y	102
Cab 2 X/Y	103
Chorus 1 X/Y	104
Chorus 2 X/Y	105
Delay 1 X/Y	106
Delay 2 X/Y	107
Drive 1 X/Y	108
Drive 2 X/Y	109
Flanger 1 X/Y	110
Flanger 2 X/Y	111
Phaser 1 X/Y	112
Phaser 2 X/Y	113
Pitch 1 X/Y	114
Pitch 2 X/Y	115
Reverb 1 X/Y	116
Reverb 2 X/Y	117
Wahwah 1 X/Y	118
Wahwah 2 X/Y	119

I/O PEDAL (Unlike the XL, the Axe-Fx II Mark II has only one pedal jack)

Pedal 1,2 Type:	CONTINUOUS
Preset Increment/Decrement	OFF

Note: The above settings allow you to connect an ext. switch to perform PRESET UP or PRESET DOWN functions.

Preset Start:	0
Preset End:	0
Pedal 1,2 Cal:	(No Setting)

I/O XY

X Quick Jump:	AMP 1
Y Quick-Jump:	AMP 1

17 Specifications

17.1 Axe-Fx II XL/XL+ Specifications

FRONT PANEL INPUT	
Connector:	1/4” phone jack, unbalanced.
Impedance:	1 MegaOhm (less if Input Impedance is active)
Max. Input Level:	+16 dBu (conditioned for guitar use)
REAR INPUTS	
Connector:	1/4” phone jack, balanced.
Impedance:	1 MegaOhm
Max. Input Level:	+20 dBu
A/D CONVERSION	
Bit Depth:	24 bits
Sample Rate:	48 kHz
Dynamic Range:	> 110 dB
Frequency Response:	20 – 20kHz, +0 / -1 dB
Crosstalk:	< -60 dB over full bandwidth
ANALOG OUTPUTS	
Connectors:	1/4” phone jack unbalanced (hum-canceling), XLR balanced (for main output)
Impedance:	600 ohm
Max Output Level:	+20 dBu
Dynamic Range:	> 110 dB
Frequency Response:	20 – 20kHz, +0 / -1 dB
Crosstalk:	< -60 dB over full bandwidth
DIGITAL I/O	
Connectors:	RCA Coaxial Type for S/PDIF I/O, XLR for AES I/O
Format:	S/PDIF – 24 bit
Sample Rate:	48 kHz fixed
USB Audio Clock	48 kHz fixed
MIDI INTERFACE	
Input Connector:	7-pin DIN (pins 6 & 7 connected to phantom power in jack)
Out Connector:	5-pin DIN
Thru Connector:	5-pin DIN
PEDAL INTERFACE	
Connectors:	2 × 1/4” TRS phone jack
Format:	Switch: Momentary or Latching; Pedal: 10–100kΩ max, linear taper expression type.
FASLINK™ INTERFACE	
Connectors:	1 × XLR Female **WARNING: Connect ONLY to MFC-101 FASLINK connector or XA-2 FASLINK Adapter.**
MFC INTERFACE:	
Connector:	RJ-45 Ethernet/EtherCON **IMPORTANT: Do NOT connect to a computer or router/switch/hub!** **This jack is only intended for connection to an MFC MIDI Foot Controller.**
PHANTOM POWER:	
Connector:	Female 2.5mm jack **WARNING: Do NOT connect adapters with a rating higher than 1A (1000ma)**
GENERAL	
Finish:	Powder coated steel chassis with anodized aluminum faceplate
Display:	160x80 dot matrix graphic LCD
Dimensions:	19” × 3.5” × 14.25” (483 × 88 × 362 mm)
Weight:	14.75 lbs (6.7 kg)
Input Voltage:	100–240 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

Specifications subject to change without notice.

17.2 Axe-Fx II Mark II Specifications

FRONT PANEL INPUT	
Connector:	1/4” phone jack, unbalanced.
Impedance:	1 MegaOhm (less if Input Impedance is active)
Max. Input Level:	+16 dBu (conditioned for guitar use)

REAR INPUTS	
Connector:	1/4” phone jack, balanced.
Impedance:	1 MegaOhm
Max. Input Level:	+20 dBu

A/D CONVERSION	
Bit Depth:	24 bits
Sample Rate:	48 kHz
Dynamic Range:	> 110 dB
Frequency Response:	20 – 20kHz, +0 / -1 dB
Crosstalk:	< -60 dB over full bandwidth

ANALOG OUTPUTS	
Connectors:	1/4” phone jack unbalanced (hum-canceling), XLR balanced (for main output)
Impedance:	600 ohm
Max Output Level:	+20 dBu
Dynamic Range:	> 110 dB
Frequency Response:	20 – 20kHz, +0 / -1 dB
Crosstalk:	< -60 dB over full bandwidth

DIGITAL I/O	
Connectors:	RCA Coaxial Type for S/PDIF I/O, XLR for AES I/O
Format:	S/PDIF – 24 bit
Sample Rate:	48 kHz fixed
USB Audio Clock	48 kHz fixed

MIDI INTERFACE	
Input Connector:	7-pin DIN (pins 6 & 7 connected to phantom power in jack)
Out/Thru Connector:	5-pin DIN

PEDAL INTERFACE	
Connector:	1/4” TRS phone jack
Format:	Switch: Momentary or Latching; Pedal: 10–100kΩ max, linear taper expression type.

MFC INTERFACE:	
Connector:	RJ-45 Ethernet/EtherCON
	IMPORTANT: Do NOT connect this to a computer or router/switch/hub! **This jack is only intended for connection to an MFC MIDI Foot Controller.**

PHANTOM POWER:	
Connector:	Female 2.5mm jack
	WARNING: Do NOT connect adapters with a rating higher than 1A (1000ma)

GENERAL	
Finish:	Powder coated steel chassis with anodized aluminum faceplate
Display:	160x80 dot matrix graphic LCD
Dimensions:	19” × 3.5” × 14.25” (483 × 88 × 362 mm)
Weight:	14.75 lbs (6.7 kg)
Input Voltage:	100–240 VAC, 47 – 63 Hz (universal input)
Power Consumption:	<40 W
Backup Battery Life:	>10 years
Backup Battery Type:	CR-2450

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

Specifications subject to change without notice.

17.3 **Midi Implementation Chart**

NOTE: MFC-101 Presets and Instant Access Switches have the capability to send custom MIDI data, entered freeform as hex code, which can be used for many applications not supported "natively" (ex: Note On/Off).

Function		Transmitted	Received	Remarks
Basic Channel	**Default** **Changed**	**1** **1-16**	**1** **1-16**	
Note Number	**True Voice**	**X**	**X**	
Velocity	**Note ON** **Note OFF**	**X** **X**	**X** **X**	
After Touch	**Keys** **Channels**	**X** **X**	**X** **X**	
Pitch Bend		**X**	**X**	
Control Change		**X**	**O**	CCs are GLOBALLY soft-assigned to functions via the I/O:CTRL menu. These include INPUT volume, OUTPUT 1 and OUTPUT2 master volume, Scene Select, Tap Tempo, Tuner Invoke, 12 "EXTERNAL" control nodes (assignable as modifiers to one or more parameters on a per-preset basis), all LOOPER functions, the BYPASS switch of every block instance AMP1, AMP2, CAB1, CAB2, etc.), and the X/Y switches of the 20 block types that support this function (AMP1, AMP2, etc.).
Program Change	**True Number** **Bank Select**	**O** **O**	**O** **O**	Bank Select (CC#0) and Program Change messages may be used to recall the presets of the Axe-Fx II. The unit also supports custom program change mapping utilizing a FROM→TO table with 128 entries. BANK select messages are persisted until a subsequent valid Bank Select is received. An OFFSET may be applied using MIDI OC OFFSET on the MIDI page of the I/O Menu. Selecting a preset via the Axe-Fx II front panel will also transmit the corresponding bank select and program change number messages.
System Exclusive	**Fractal Audio** **Real-Time** **Non-Real-Time**	**O** **O** **X**	**O** **X** **X**	The list of parameters that can be controlled/edited via SysEx is listed in the Owner's Manual under Factory Default Settings. Real-time SysEx is used to transmit Tempo and Tuner.
System Common	**Song Position** **Song Select** **Tune Request**	**X** **X** **X**	**X** **X** **X**	
System Real-Time	**Clock** **Commands**	**X** **X**	**O** **X**	Axe-Fx II Global Tempo syncs automatically to MIDI Beat Clock.
Auxiliary Messages	**Local ON/OFF** **All Notes OFF** **Active Sense** **Reset**	**X** **X** **X** **X**	**X** **X** **X** **X**	

0 = YES, X=NO

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