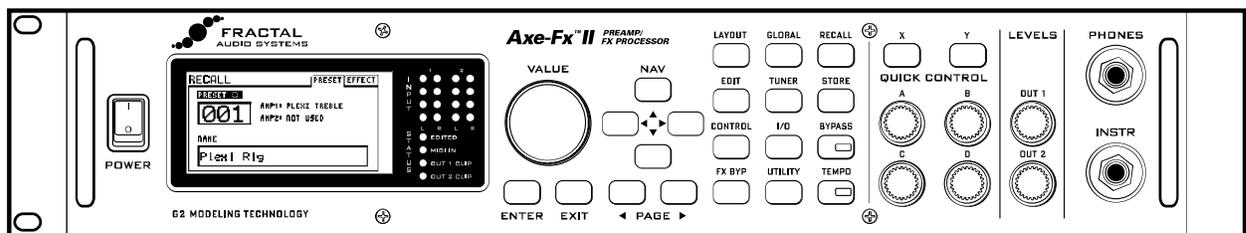


# Axe-Fx II™ PREAMP/ FX PROCESSOR



## Owner's Manual



# Legal Notices

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## Important Safety Instructions

	<p><b>WARNING:</b> To reduce the risk of fire or electric shock, do not expose this appliance to rain or moisture.</p> <p><b>CAUTION:</b> 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.</p>
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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 v6.00c**

# Certificate of Conformity

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

## ***Axe-Fx II Digital Guitar Preamplifier and Effects Processor***

- that is covered by this certificate and marked with CE label conforms to following standards:

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

***Issued in May 2011***

***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 25 years of experience spanning guitars, amps, effects, synthesizers, software, and beyond. He also works as a consultant to many greats including Dweezil Zappa, Adrian Belew, Steve Vai, Dream Theater, the Edge, Peter Dinklage, Neal Schon, King's X, 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 awesome 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 the closest competitor). For the player, this means better sound, smarter features, and improved performance.

We think that a time traveler from that date back in 2006 would be quite impressed with how things turned out. We are, and we hope you will be too. And now, on to the show...

*—Fractal Audio Systems, May 2011*

# Table of Contents

<b>Foreword</b> .....	<b>iii</b>
<b>Table of Contents</b> .....	<b>iv</b>
<b>What’s New</b> .....	<b>1</b>
<b>1 Introduction</b> .....	<b>4</b>
<b>1.1 What is the Axe FX II?</b> .....	<b>4</b>
<b>1.2 The Inventory/Grid Concept</b> .....	<b>6</b>
<b>1.3 Connectivity and More</b> .....	<b>7</b>
1.3.1 My Head Hertz... ..	7
<b>2 Overview</b> .....	<b>8</b>
<b>2.1 The Front Panel</b> .....	<b>8</b>
<b>2.2 The Rear Panel</b> .....	<b>10</b>
<b>2.3 Computer Integration</b> .....	<b>12</b>
2.3.1 Minimum Requirements.....	12
2.3.2 Software Installation.....	12
2.3.3 Capabilities.....	13
<b>3 Connections</b> .....	<b>15</b>
<b>3.1 Setting Levels</b> .....	<b>15</b>
<b>3.2 The PEDAL Jack</b> .....	<b>16</b>
<b>3.3 System Parameters</b> .....	<b>16</b>
<b>3.4 Connection Diagrams</b> .....	<b>17</b>
3.4.1 Axe-Fx II into Self-Powered Full-Range Speakers.....	18
3.4.2 Axe-Fx II into Studio Monitors .....	18
3.4.3 Axe-Fx II with Power Amp and Guitar Speakers .....	19
3.4.4 Axe-Fx II Effects Loop .....	20
3.4.5 Axe-Fx II Digital Audio Interconnection .....	20
3.4.6 Axe-Fx II Four Cable Method (“4CM”) .....	21
3.4.7 Direct to FOH plus Real Amps on Stage .....	22

3.4.8	Axe-Fx II as Effects Processor Only (with Guitar Amps)	23
3.4.9	Axe-Fx II as a Computer Audio Interface	24
3.4.10	Axe-Fx II and MFC-101	25
3.4.11	Axe-Fx II: One Possible “Big Rig”	26
<b>4</b>	<b>Basic Operation and Editing</b>	<b>27</b>
<b>4.1</b>	<b>Presets</b>	<b>27</b>
<b>4.2</b>	<b>The Grid</b>	<b>28</b>
4.2.1	Inserting and Removing Blocks	28
4.2.2	Shunts	29
4.2.3	Connector Cables	30
4.2.4	Moving Blocks on the Grid	32
4.2.5	Example Presets on the Grid	33
<b>4.3</b>	<b>Editing Sounds</b>	<b>35</b>
4.3.1	Quick Control	36
<b>4.4</b>	<b>X/Y Switching</b>	<b>36</b>
4.4.1	X/Y Quick Jump	37
<b>4.5</b>	<b>Bypassing a Block</b>	<b>37</b>
<b>4.6</b>	<b>Loading Effects from another Preset</b>	<b>37</b>
<b>4.7</b>	<b>Saving Changes</b>	<b>38</b>
4.7.1	Swapping the Locations of Two Presets	38
<b>5</b>	<b>Effects Guide</b>	<b>39</b>
<b>5.1</b>	<b>Amplifier [AMP]</b>	<b>39</b>
5.1.1	Basic Amp Parameters	41
5.1.2	Amp Dynamics Parameters	44
5.1.3	Advanced Amp Parameters	44
<b>5.2</b>	<b>Cabinet [CAB]</b>	<b>48</b>
<b>5.3</b>	<b>Chorus [CHO]</b>	<b>51</b>
<b>5.4</b>	<b>Compressor [CMP]</b>	<b>53</b>
<b>5.5</b>	<b>Crossover [XVR]</b>	<b>55</b>
<b>5.6</b>	<b>Delay [DLY]</b>	<b>56</b>
5.6.1	Mono Delay	57

## TABLE OF CONTENTS

5.6.2	Stereo Delay.....	58
5.6.3	Dual Delay.....	59
5.6.4	Ping-Pong Delay.....	60
5.6.5	Sweep Delay.....	60
5.6.6	Reverse Delay.....	60
5.6.7	Tape Delay.....	61
5.6.8	Delay Common Parameters.....	62
<b>5.7</b>	<b>Drive [DRV].....</b>	<b>63</b>
<b>5.8</b>	<b>Effects Loop [FXL].....</b>	<b>66</b>
<b>5.9</b>	<b>Enhancer [ENH].....</b>	<b>67</b>
<b>5.10</b>	<b>Feedback Send [SND] &amp; Return [RTN].....</b>	<b>67</b>
<b>5.11</b>	<b>Filter [FLT].....</b>	<b>67</b>
<b>5.12</b>	<b>Flanger [FLG].....</b>	<b>69</b>
<b>5.13</b>	<b>Formant [FRM].....</b>	<b>72</b>
<b>5.14</b>	<b>Gate/Expander [GTE].....</b>	<b>73</b>
<b>5.15</b>	<b>Graphic Equalizer [GEQ].....</b>	<b>74</b>
<b>5.16</b>	<b>Looper [LPR].....</b>	<b>74</b>
<b>5.17</b>	<b>Megatap Delay [MGT].....</b>	<b>76</b>
<b>5.18</b>	<b>Mixer [MIX].....</b>	<b>77</b>
<b>5.19</b>	<b>Multiband Compressor [MBC].....</b>	<b>78</b>
<b>5.20</b>	<b>Multi Delay [MTD].....</b>	<b>79</b>
5.20.1	Quad Tap Delay.....	80
5.20.2	Plex Delay.....	81
5.20.3	Plex Detune.....	83
5.20.4	Plex Shift.....	83
5.20.5	Band Delay.....	83
5.20.6	Quad Series Delay.....	84
5.20.7	Ten-Tap Delay.....	84
5.20.8	Rhythm Tap Delay.....	85
5.20.9	Diffusor.....	86
5.20.10	Quad Tape Delay.....	87
<b>5.21</b>	<b>Tremolo/Panner [PAN].....</b>	<b>87</b>

<b>5.22</b>	<b>Parametric EQ [PEQ]</b> .....	<b>88</b>
<b>5.23</b>	<b>Phaser [PHA]</b> .....	<b>89</b>
<b>5.24</b>	<b>Pitch Shifter [PIT]</b> .....	<b>91</b>
5.24.1	Detune .....	93
5.24.2	Fixed Harmony .....	94
5.24.3	Intelligent Harmony .....	94
5.24.4	Classic Whammy .....	97
5.24.5	Octave Divider .....	97
5.24.6	Crystals.....	98
5.24.7	Advanced Whammy.....	99
5.24.8	Arpeggiator .....	100
5.24.9	Custom Shifter .....	101
<b>5.25</b>	<b>Quad Chorus [QCH]</b> .....	<b>102</b>
<b>5.26</b>	<b>Resonator [RES]</b> .....	<b>104</b>
<b>5.27</b>	<b>Reverb [REV]</b> .....	<b>105</b>
<b>5.28</b>	<b>Ring Modulator [RNG]</b> .....	<b>108</b>
<b>5.29</b>	<b>Rotary Speaker [ROT]</b> .....	<b>108</b>
<b>5.30</b>	<b>Synth [SYN]</b> .....	<b>110</b>
<b>5.31</b>	<b>Tone Match [TMA]</b> .....	<b>111</b>
<b>5.32</b>	<b>Vocoder [VOC]</b> .....	<b>112</b>
<b>5.33</b>	<b>Volume/Pan [VOL]</b> .....	<b>113</b>
<b>5.34</b>	<b>Wahwah [WAH]</b> .....	<b>114</b>
<b>5.35</b>	<b>Input Noise Gate</b> .....	<b>115</b>
5.35.1	Input Impedance.....	115
<b>5.36</b>	<b>Output Mixer</b> .....	<b>116</b>
<b>5.37</b>	<b>Common Mix Parameters</b> .....	<b>117</b>
<b>6</b>	<b>Global Blocks</b> .....	<b>119</b>
<b>6.1</b>	<b>Introduction</b> .....	<b>119</b>
<b>6.2</b>	<b>Using Global Blocks</b> .....	<b>119</b>
6.2.1	Saving to a Global Block.....	120

## TABLE OF CONTENTS

6.2.2	Loading and Linking a Global Block.....	121
6.2.3	Loading Global Blocks without Linking .....	122
6.2.4	Unlinking Preset and Global Blocks .....	122
<b>7</b>	<b>Modifiers &amp; Controllers .....</b>	<b>124</b>
7.1	Introduction.....	124
7.2	Creating a Modifier .....	124
7.2.1	Transformations.....	126
7.2.2	Damping.....	127
7.2.3	Auto Engage.....	128
7.2.4	Program Change Reset .....	128
7.3	Control Sources.....	129
7.3.1	LFO1 & 2 .....	129
7.3.2	ADSR 1 & 2 .....	130
7.3.3	Sequencer .....	130
7.3.4	Envelope Follower .....	131
7.3.5	Pitch Detector .....	131
7.3.6	Manual Knobs .....	131
7.3.7	External Controllers .....	132
7.3.8	Modifier Power!.....	132
<b>8</b>	<b>Global Parameters.....</b>	<b>133</b>
8.1	Configuration Parameters .....	133
8.2	Output Parameters .....	134
8.3	Custom Scales .....	134
<b>9</b>	<b>Input/Output Parameters.....</b>	<b>135</b>
9.1	Input Parameters .....	135
9.2	Audio Parameters .....	135
9.3	MIDI Parameters.....	137
9.4	Control Parameters .....	139
9.5	Pedal Parameters.....	141
9.6	X/Y Quick-Jump Assign.....	141
<b>10</b>	<b>Utilities .....</b>	<b>142</b>

10.1	LCD Contrast.....	142
10.2	Preset Utilities.....	142
10.3	Status Meters.....	142
10.4	Reset System.....	142
10.5	IR Capture .....	143
10.6	Firmware.....	144
11	Tuner .....	145
12	Tempo .....	146
12.1	Setting the Tempo .....	146
12.2	Synchronizing Sound Parameters .....	146
12.3	Tempo to Use .....	147
12.4	Auto Delay .....	147
12.5	Metronome.....	147
13	Backing Up and Restoring.....	148
13.1	MIDI/SysEx Backup and Restore .....	148
13.1.1	Dumping to a computer.....	148
13.1.2	Restoring from a Computer .....	149
13.2	Onboard ROM Backup and Restore .....	149
13.3	Machine-to-Machine Transfers.....	150
14	Firmware Updates.....	151
14.1	Firmware.....	151
15	Troubleshooting.....	152
16	Appendix .....	154
16.1	Shortcuts Overview .....	154
16.2	60-Second Edit Guide .....	155
16.3	Understanding Preset Size Limits.....	156
16.4	Signal Flow, Global, and I/O Parameters .....	156

TABLE OF CONTENTS

**16.5 LFO Waveforms, Duty, and Phase ..... 157**  
    16.5.1 LFO Phase.....157

**16.6 Tempo Cross Reference ..... 158**

**16.7 Mono and Stereo..... 159**

**16.8 Mixology ..... 160**

**16.9 Humbuster™ Technology..... 161**

**16.10 Setting up a Wah Pedal..... 162**  
    16.10.1 Using the Onboard Pedal Jack .....162  
    16.10.2 Using an Expression Pedal on an MFC-101.....163

**16.11 Setting Up Spillover ..... 164**  
    16.11.1 Within a Single Preset .....164  
    16.11.2 Across Different Presets.....164

**16.12 Using Send and Return ..... 165**  
    16.12.1 Creating Feedback Loops .....165  
    16.12.2 Extending the Length of Effect Chains .....165

**16.13 Loading User Cab IRs..... 166**

**16.14 Glossary & Resources ..... 167**

**16.15 Axe-Fx II Bank & Preset Numbers Table ..... 170**

**16.16 Factory Default Settings..... 171**

**17 Specifications..... 173**  
    17.1 Midi Implementation Chart ..... 174

**Warranty ..... 175**

# 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. 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 stunningly recreates great guitar signal chains—stompboxes, amps, cabs, mics, studio effects, and more—with unprecedented power, flexibility, and control. As the successor to what many believe to be some of the best-sounding processors ever created, our new arrival takes "real amp tone and feel" to the next level, offering the latest word in our commentary on restoring digital to its rightful place as the superior solution for guitar processing. Never willing to rest on our laurels, we are proud to present the Axe-Fx II.

## Twice the Processing Power

Our philosophy is to never cut corners on processing power. Our new state-of-the-art algorithms required a powerful platform on which to operate, so the Axe-Fx II features two (count 'em) 600 MHz dual-core Analog Devices TigerSHARC™ Digital Signal Processors working in tandem. One processor is devoted solely to amp modeling while the other handles effects processing and system tasks. An ultra-high-speed bus interconnects the two processors, shuttling data back and forth. Mated to the processors is a high-speed, 64-bit external memory bus and twice the RAM of previous Axe-Fx products. A 200K gate FPGA provides peripheral functionality and system bus management.

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, the Axe-Fx II consumes less than 40W and is nearly silent in operation.

## G2 Amp Modeling™ with Virtual Vacuum Tube™ Technology

All this power would be useless without algorithms to take advantage of it. Years of research have yielded what we call **G2 Amp Modeling Technology**, comprising major breakthroughs in both preamp and power amp modeling.

First, we created our new **Virtual Vacuum Tube** technology, or **VVT**. VVT is a complete departure from the static waveshaping technology used by other products. It is a digital replica of a vacuum tube, complete with time, frequency, and level dependencies. Just like a real vacuum tube, our digital replica is truly dynamic and changes with your playing. This creates a level of realism, complexity, and response that other products just can't match.

Next, we rewrote the book on power amp modeling. **G2** models the entire power amp, including the phase inverter, power tubes, output transformer, power transformer, choke, filter caps, and more. The results are amazing: warm, yet tight bass, powerful midrange, and silky highs. In addition, the Amp block now includes an integrated, 8-band graphic equalizer for additional tone-shaping without extra blocks.

The current lineup of Axe-Fx II amp models are the result 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.

## WHAT'S NEW

### Improved Speaker Simulation

The Axe-Fx II features new dynamic speaker modeling technology in the Amp block. This alone improves the way amp and cab work together for realistic tone and feel. In addition, the Axe-Fx II features a new, higher resolution (2048) convolution cabinet emulator with 70 factory Impulse Responses (IRs) and 50 "USER CAB" memory locations. Factory IRs also include custom creations from the Redwirez and Ownhammer collections, plus contributions from loudspeaker design engineer and Axe-Fx early adopter, Jay Mitchell.

### 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. Ten block types—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 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. A built in FLASH ROM enables onboard backups of preset banks and system settings.

### Axe-Fx II/Computer Integration with Onboard USB

The new onboard "Audio Class 2.0 compliant" USB interface provides a range of 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 3<sup>rd</sup>-party interface. On USB 2.0 or better systems, recording capabilities include not only the main processed stereo outs but a pair of dry channels for easy re-amping.

### New I/O Capabilities and Less Noise

All rear analog inputs are now balanced like the onboard XLR outputs. The ¼" unbalanced outputs feature our new **Humbuster™** technology, which senses and subtracts the ground noise of equipment connected with a simple stereo-to-mono 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 Output 2, 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. 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

The Axe-Fx II features an all-new dedicated control port for connecting an MFC-101 MIDI Foot Controller via that unit's EXPANSION port. Readily available CAT5 (Ethernet) cable provides bi-directional communications over lengths of 100 meters or more, PLUS phantom power from the built-in power supply of the Axe-Fx II. (Traditional 5 or 7-pin MIDI and power are still supported for Luddites and 3rd-party pedalboards.)

## New FX Processing Features and Enhancements

The effect processing capabilities of the Axe-Fx II are also greatly enhanced.

First, we've made it easier to dial in "all-time favorites" on the Chorus, Delay, Flanger, and Phaser effects, with a TYPE control that instantly makes all other required settings for great sound in seconds. These include tape delay models, analog "bucket brigade" effects, "script" and "block" logo phasers, 'vibe, and many more. The sound and features of these effects have also been upgraded to provide extremely authentic representations of the specific characteristics of classic originals.

Many of the effects and algorithms have also been updated and enhanced. A dedicated 60-second looper block is now included. The Delay and Reverb blocks now feature many improvements including integrated parametric equalizers. Chorus, Flanger, and Wahwah modeling has been improved and also now includes optional built-in nonlinearity "circuit overdrive" functions. The Phaser has been expanded. Graphic Equalizers (including Global) are 10-band. There are many new modifier slots. Compressors now use millisecond values for attack and release times. Tempo parameters now offer 64 different rhythmic values. Filters allow far greater resonance. There are simply too many of such improvements to list!

## Global Blocks Allow Shared Sound Settings Across Multiple Presets

Those familiar with "Global Amps" from previous Axe-Fx products will appreciate Global Blocks. The new system has been improved and expanded to every block type in the inventory (except Tone Match). Those new to Axe-Fx products will appreciate how this feature allows centralized control of a preset collection. Save any "normal" block to a separate and independent global memory area then load this entry across multiple presets with a "link" to keep it instantly and seamlessly synchronized to the master. You can update the master from any linked instance as desired. Should you choose to remove a link, this leaves both the global and the normal entries fully intact and able to be edited independently of one another again.

## Improvements and Enhancements, plus More to Come...

Aside from the areas mentioned in this overview, there are many more exciting updates, enhancements, improvements, and additions in the Axe-Fx II. And while the Axe-Fx Standard and Axe-Fx Ultra essentially reached maturity in terms of their capacity for further development, the Axe-Fx II is well equipped for a future of free upgrades and updates in the tradition that established our commitment to product evolution.

# 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 is the highly anticipated sequel to the Axe-Fx Standard and Axe-Fx Ultra. 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 “wire up” your dream rig. 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 it 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 in fact has more capabilities like those of a typical 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 again 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 platform upon which you can create any number of incredible guitar tones, and it is 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 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 12x4 “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.

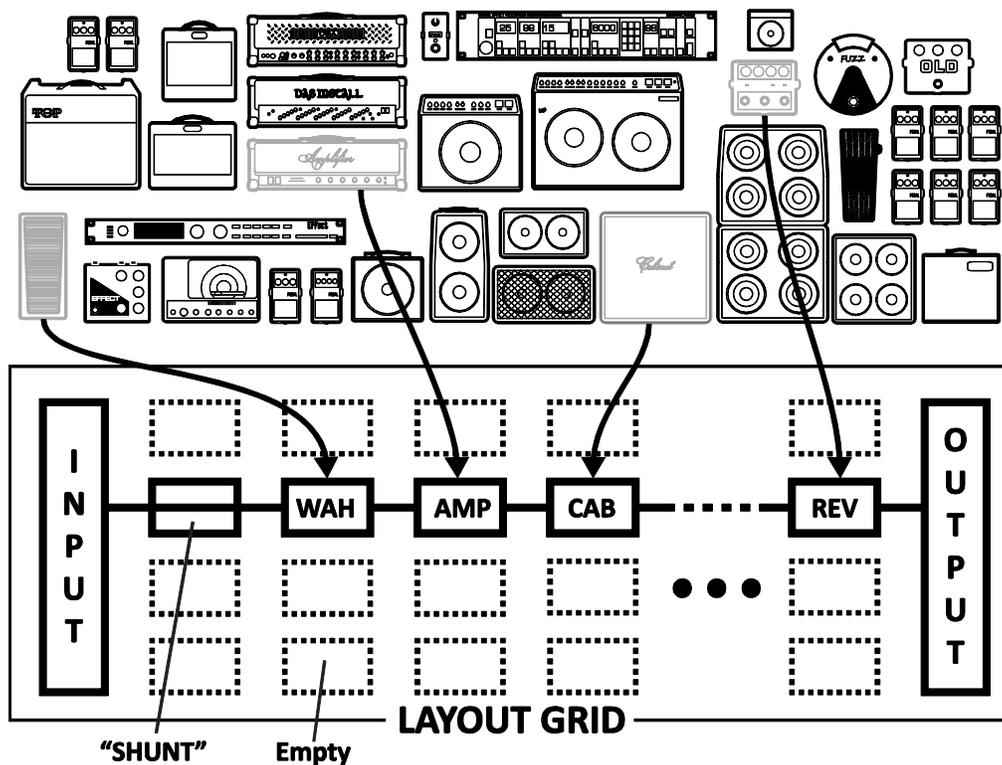


Figure 1-1 – The Inventory/Grid Concept

(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:

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

In addition to the blocks listed above, each preset also includes a programmable **Input Noise Gate** (p. 115) and an **Output Mixer** (p. 116). 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.119) that can be inserted and then kept synchronized across multiple presets.

Twenty-two different **Modifiers & Controllers** (p. 124) 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. 133), and **Section 9: Input/Output Parameters** (p. 135). Meanwhile, **Section 10: Utilities** (p. 142) can fill out your understanding of functions related to general use and maintenance.

Finally, **Sections 11 through 14** cover the **Tuner** (p. 145) and **Tempo** (p.146) functions, plus the basics of **Backing Up and Restoring** (p. 148) and **Firmware Updates** (p. 151).

### 1.3.1 My Head Hertz...

As even this brief introduction demonstrates, the Axe-Fx II contains an entire world of diverse possibilities. Only the precise terminology of audio engineering allows comparably diverse communities of casual and professional players, producers, engineers, and others to use and enjoy this powerful device. In case you want to familiarize yourself with specific audio terms or come up to speed on other topics, the **Appendix** is filled with useful material, including a **Glossary** (p. 166). It is followed by **Specifications** (p. 175) and your **Warranty** (p. 173).

## 2 Overview

The Axe-Fx II has a simple hardware interface with clearly labeled controls and jacks. Review the following to familiarize yourself with the hardware features of the unit.

### 2.1 The Front Panel

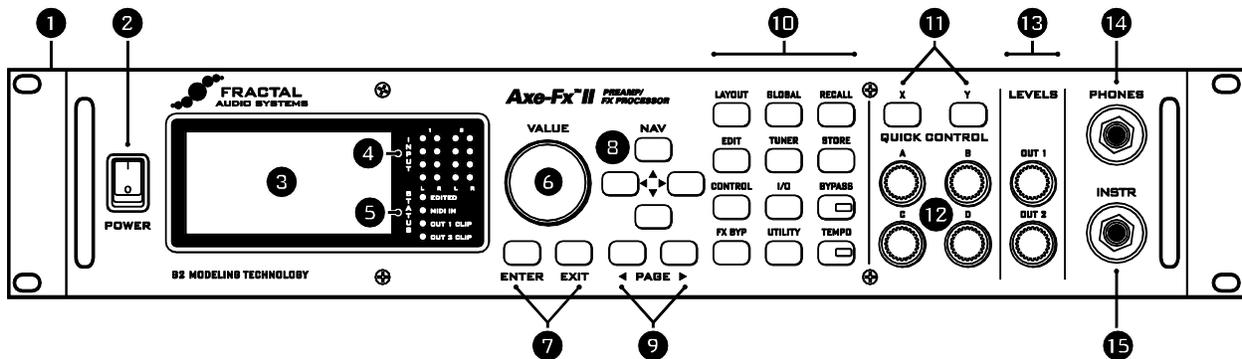


Figure 2-1

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. 115).
    - **OUTPUT** page contains a mixer for overall level control of a preset (p. 116).
  - ▶ **EDIT** – Select any block on the layout grid and press this button to open its **EDIT menu**.  
Press repeatedly to cycle through the 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. 124 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 the SAVE/LOAD GLOBAL BLOCKS screen (p. 119).
  - ▶ **GLOBAL** – This menu contains four pages: CONFIG, OUT1, OUT2, and SCALES. See p. 133 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. 145) and displays its menu. Press **EXIT** or **RECALL** to close.
  - ▶ **I/O** – Contains five pages used to configure the various input and output options of the Axe-Fx II.  
See p. 135 for details.
    - **INPUT** is used to set input levels for front and rear jacks.
    - **AUDIO** contains various settings and switches used to configure the signal path.
    - **MIDI** contains settings related to basic MIDI connectivity, preset changes, and SysEx.
    - **CONTROL** is where MIDI CC numbers or the pedal jack are assigned globally to various functions.
    - **PEDAL** allows you to configure the onboard pedal jack for various types of external hardware.
  - ▶ **UTILITY** – This menu contains various utility functions. See p. 142 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 output of the A/D directly to the input of the D/A converters, defeating all processing functions and lighting the **BYPASS LED**. 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. 146 for more information about Axe-Fx II tempo control.
- 11. X/Y** – On the Axe-Fx II, many effects offer two sets of parameters, called X and Y. These buttons select between them while editing. The X/Y state can be then switched remotely during performance for dual function blocks (see p. 36). Outside of editing, X and Y can be individually set up to instantly display your choice of EDIT menus for quick adjustment.

**12. QUICK CONTROL** – The four knobs marked A, B, C, and D each serve two different functions.

- While editing, they are used like the **VALUE** wheel to adjust parameters. Mapping is dynamic and indicated by a small A,B,C, or D next to the onscreen parameter. See p. 36 for details.
- In RECALL mode, these knobs work as “Manual knob” modifier sources, allowing control of assigned parameters right from the front panel. See p. 131.

**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.

## 2.2 The Rear Panel

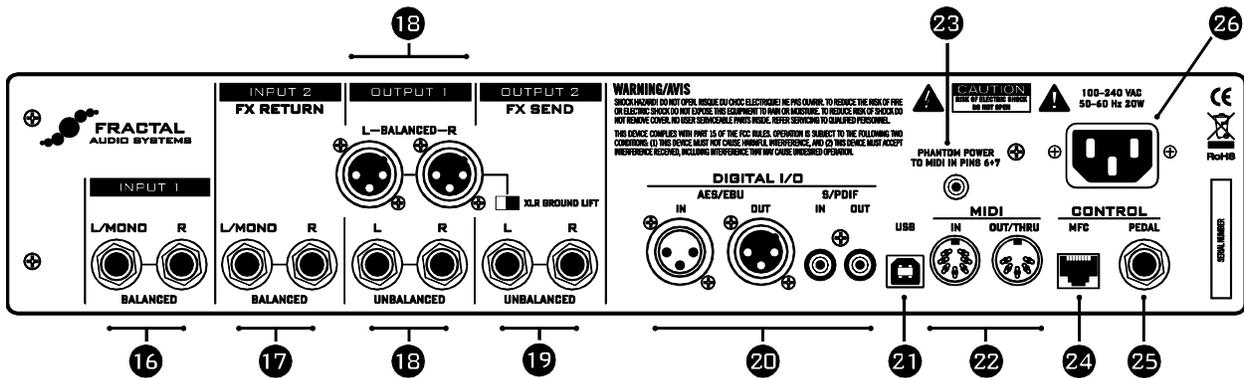


Figure 2-2

**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. 135)

**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. 66). 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") Jacks (“FX SEND”)** – The output from Connect to the inputs(s) of outboard equipment when using the **FX Loop** block (p. 66). 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 16.9 on p. 161.

- 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. 135). 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 great range of two-way audio and MIDI capabilities. See section 2.3 on p.12 for details. Like the digital jacks, USB Audio operates at 48k.
- 22. MIDI IN** and **MIDI OUT/THRU** jacks – The Axe-Fx II has a **MIDI IN** port for connecting a legacy MIDI Controller or Interface, plus a **MIDI Out/Thru** combo jack that transmits or forwards MIDI signals to other devices.
- 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 RJ-45 jack allows you to use a standard CAT5/Ethernet (non-crossover) cable to connect the Axe-Fx II and 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.



**Warning!** Connect the MFC jack only to the **EXPANSION** port of a Fractal Audio Systems MFC-101. Do not connect to an Ethernet device such as a computer, hub, switch, or router, as damage to one or both units could occur!

- 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 the Axe-Fx II with a “host” of great features.

### 2.3.1 Minimum Requirements

#### Windows Minimum Requirements:

- ▶ **OS:** Windows XP SP3 (x86 or x64), Vista SP2 (x86 or x64), Windows 7 SP1 (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:** 10.7.4 or later (Note: Many Axe-Fx II owners achieve satisfactory results with 10.5.7 or later).
- ▶ **CPU:** Intel Processor
- ▶ **Memory:** 512MB minimum
- ▶ **USB 2.0 Support required**

### 2.3.2 Software Installation

Although the Axe-Fx II drivers are fully class-compliant, software installation is still required on all platforms.

Step-by-step instructions are included with the installer. Both Mac and Windows versions can be downloaded from our web site at <http://www.fractalaudio.com/support>.

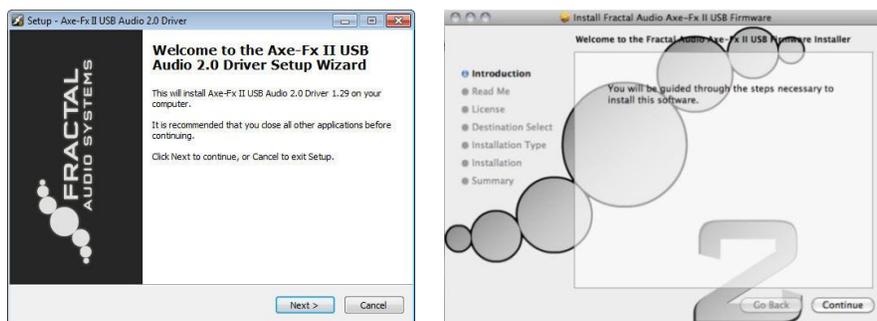


Figure 2-3 – Windows and Mac driver installers

### 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.

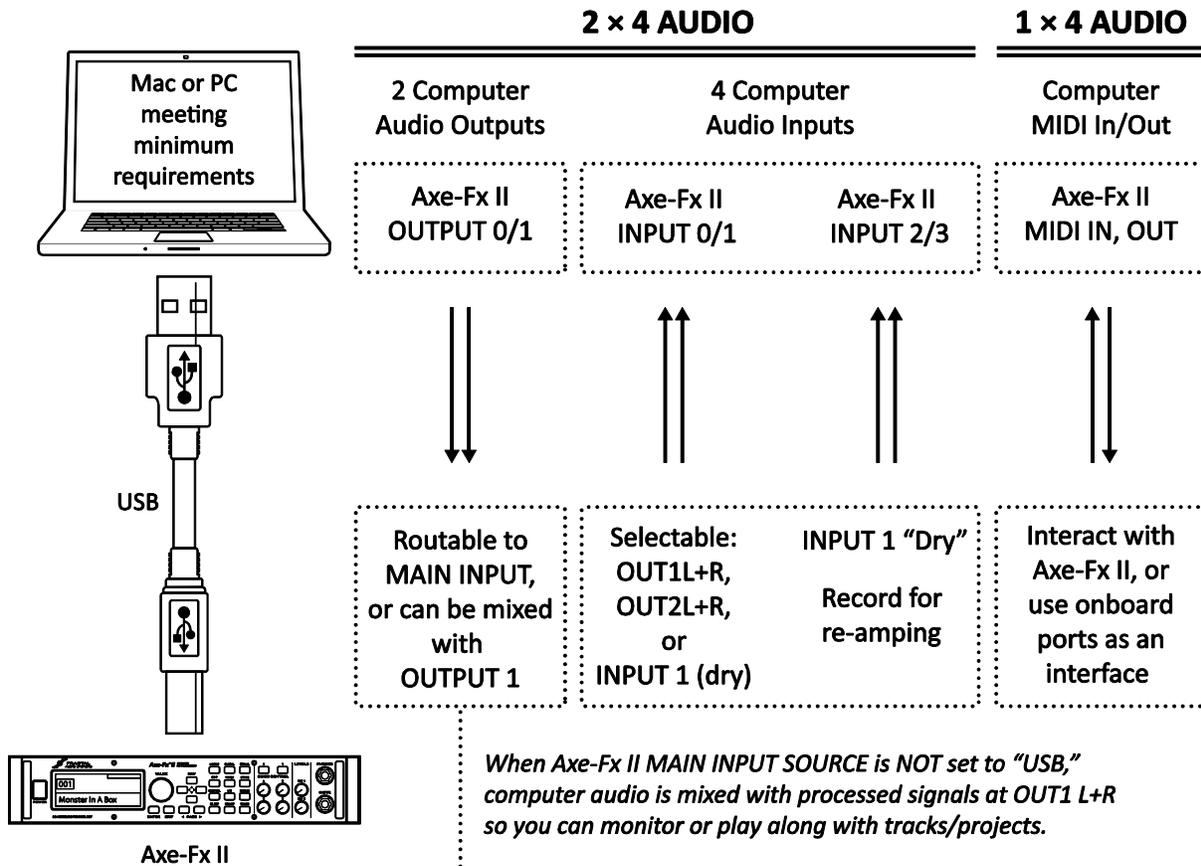


Figure 2-4 – USB Features

Audio and MIDI ports (shown in the lower half of the dotted-outline box 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 outputs, typically called OUT 0 and OUT 1, allow audio to be sent from the computer to the Axe-Fx II where it can be mixed with processed signal at the main outputs or routed through onboard effects.

To pass unprocessed computer audio through the Axe-Fx II, set the **MAIN INPUT SELECT** (p. 135) to “ANALOG (IN1)” (the default setting) or “SPDIF/AES.” This allows you, for instance, to play along with backing tracks or use the Axe-Fx II as a high quality “soundcard.”

To process computer audio with the onboard effects, set the **MAIN INPUT SELECT** to “USB.” Signals received will arrive at the grid INPUT. This allows you, for instance, to re-amp a dry track, 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).

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

Four outputs, typically called IN 0, IN 1, IN 2, and IN 3, allow audio to be routed from the Axe-Fx II to the computer, recorded, processed, or monitored.

The signal source for the first pair of outputs is selectable. The **USB/DIGI OUT SOURCE** parameter of the I/O:AUDIO menu (p. 135) determines what is sent to the computer's AXE-FX II ASIO DRIVER IN 0/1 inputs.

- ▶ Selecting **OUTPUT 1 L+R** sends the main processed output of the Axe-Fx II to the computer.  
The same signal is, of course, still routed normally to the rear XLR and unbalanced jacks.
- ▶ Selecting **OUTPUT 2 L+R** sends the output of the FX loop block (p. 66) to the computer.
- ▶ Selecting **MAIN INPUT** sends the source selected under **MAIN INPUT SELECT** (p. 135) to the computer.
  - The same result can be obtained by using OUTPUT 1 with a “shunts-only” preset (p. 29).
  - Switching **INPUT 1 LEFT SELECT** (p. 135) 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 channels, available only on systems supporting USB 2.0 (see **Minimum Requirements** above), always provides a way to simultaneously record a “dry” track for reamping. The source for these channels is the unprocessed signal from the main input (i.e. the front **INSTR** jack or the rear **INPUT 1 L/MONO** jack, depending on both what is selected under **INPUT 1 LEFT SELECT** (p. 135)) and the rear **INPUT 1 R** jack).

***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 enables the Axe-Fx II and the computer to communicate and synchronize, with capabilities for program change, parameter automation, tempo sync, SysEx dump/receive, firmware update, and more. MIDI over USB is considerably faster than “legacy” MIDI<sup>1</sup>, and allows two-way communication with the computer over a single cable.

Axe-Edit, our free companion editor/librarian, can now be used without a 3<sup>rd</sup>-party MIDI Interface or drivers, a cause of common problems.

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<sup>1</sup> The transfer rate for the new connection is roughly faster than standard MIDI. Keep in mind that new preset, bank, and firmware sizes have also increased considerably. Despite this, there is a net increase in the speed of computer interaction when compared to previous products.

## 3 Connections

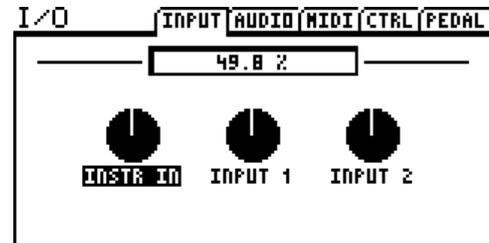


Before making connections, be sure to turn down the volume of your amps and switch off all power. Take extreme care NEVER to connect the SPEAKER outputs of an amplifier to any jack on the Axe-Fx II as this will damage one or both devices. If you're not sure, don't do it!

### 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.



All inputs are normalized, allowing the Axe-Fx II to operate with unity gain irrespective of the trim settings. The sole purpose of the input trims is to optimize A/D drive level for the best signal-to-noise ratio and distortion performance. Each input has its own dedicated A/D. The **INSTR** input jack parallels the rear inputs for improved signal-to-noise performance.

The front panel **OUTPUT LEVEL 1/2** knobs independently control the volumes at these rear panel jacks. Output 1 also adjusts the headphones level. Optimal levels will depend on what the Axe-Fx II is connected to.

To operate with unity gain, simply 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 levels result in clipping of attached equipment, turn down the front panel **OUTPUT LEVEL 1/2** knobs. At the minimum setting, volume is reduced but not 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 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 or CAB might be a natural starting point) or adjust the main GAIN slider of the preset's output mixer (p. 116).

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 global graphic equalizer to make a global adjustment.

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. 135). Block or preset adjustment may still be required.

### 3.2 The PEDAL Jack

The onboard **PEDAL** jack of the Axe-Fx II allows you to connect an expression pedal or footswitch for controlling sound functions. To use a pedal or switch at this jack, you must first configure its **TYPE** and, for continuous-type expression pedals, perform a simple calibration routine. See section **9.5** on p. **141** for details on 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. **124**, with a section specifically about External Controllers on p. **132**.

### 3.3 System Parameters

As you can see, the Axe-Fx II is equipped with a flexible array of input/output connector jacks. The diagrams above cover the layout of the hardware, 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 that certain "System Parameter" settings be made. For example, when "real," physical guitar cabs are used, the global "Speaker Simulation" switch will likely need to 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**. In the PDF version of this manual, available on our web site, you'll also find a diagram attached at the end showing inputs, routing, outputs, and the effects of system parameters.

Many of the basic setups shown here can be easily combined or expanded. Mono setups might be made stereo or vice versa. The MFC-101 MIDI Foot controller can be added for intelligent remote control. Computer connectivity, for example, can be added to any other setup, enabling you to use **Axe-Edit™**, our free software Editor/Librarian for the Axe-Fx II. Countless combinations are possible.

### 3.4 Connection Diagrams

Familiarize yourself with the capabilities of inputs, outputs, and control connections through this overview. The diagrams that follow in sections 3.4.1 through 3.4.10 illustrate several real-world applications.

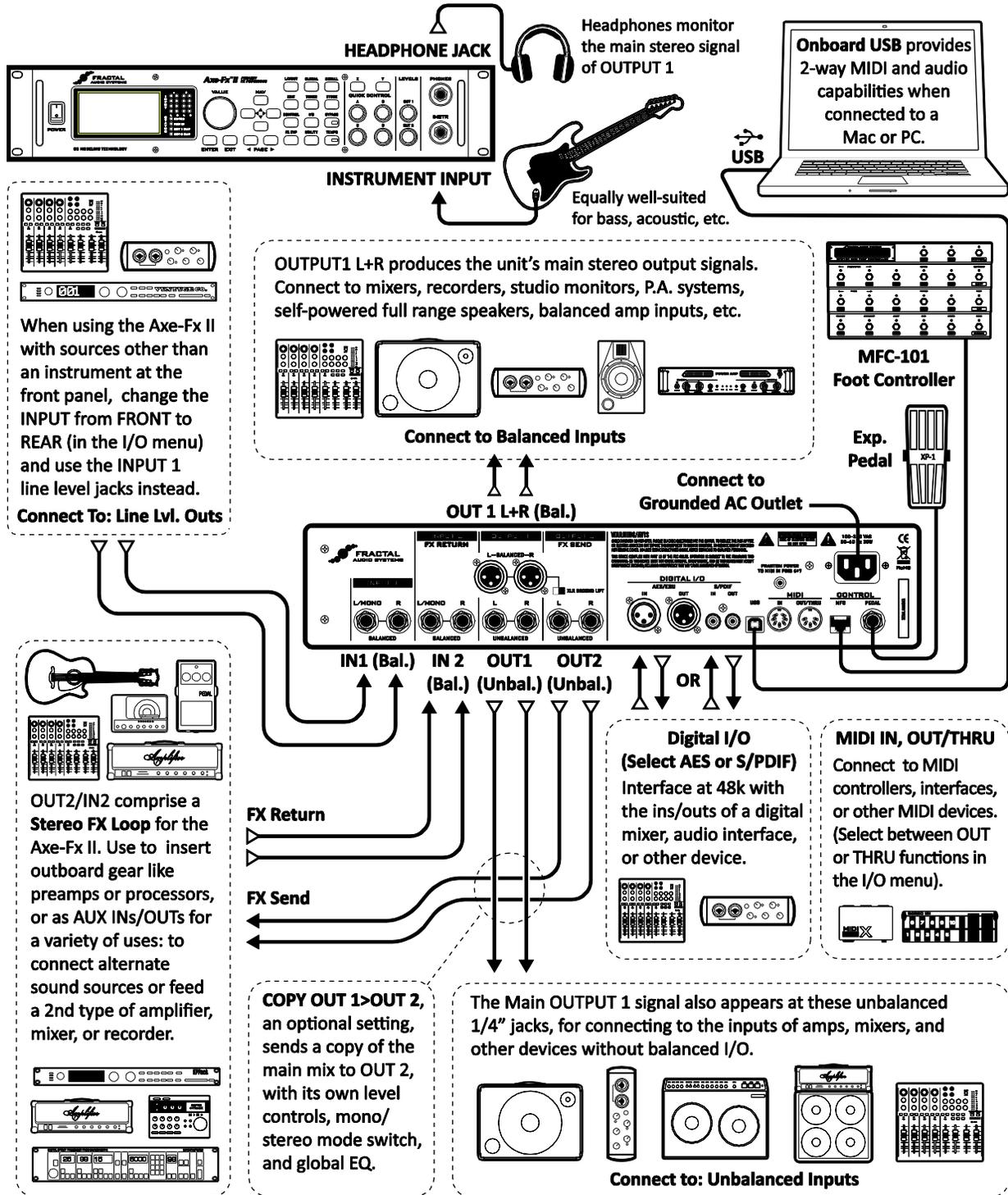


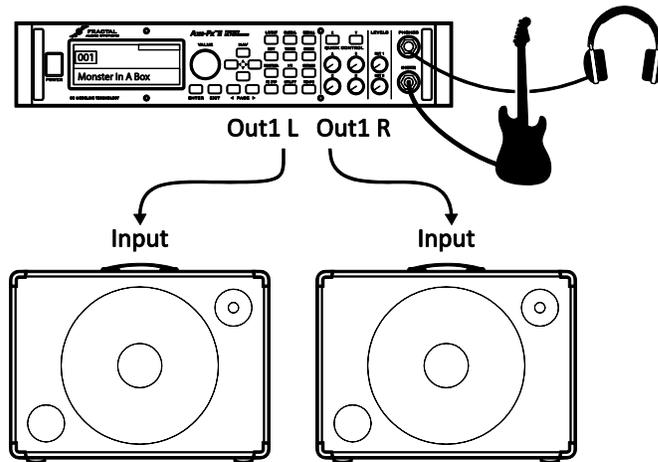
Figure 3-1 – I/O Overview

## CONNECTIONS

### 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.)



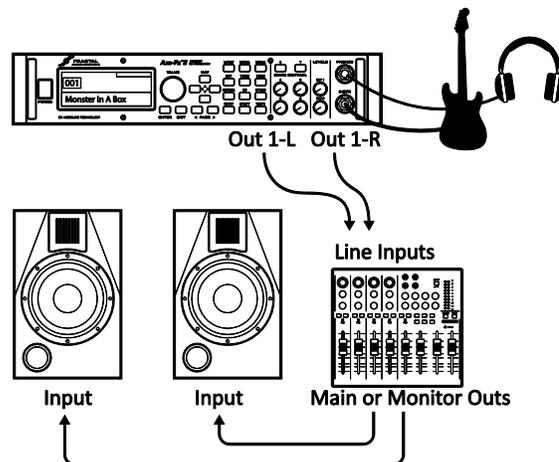
**Global Settings:** Default    **I/O Settings:** Default

**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.135) 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.)



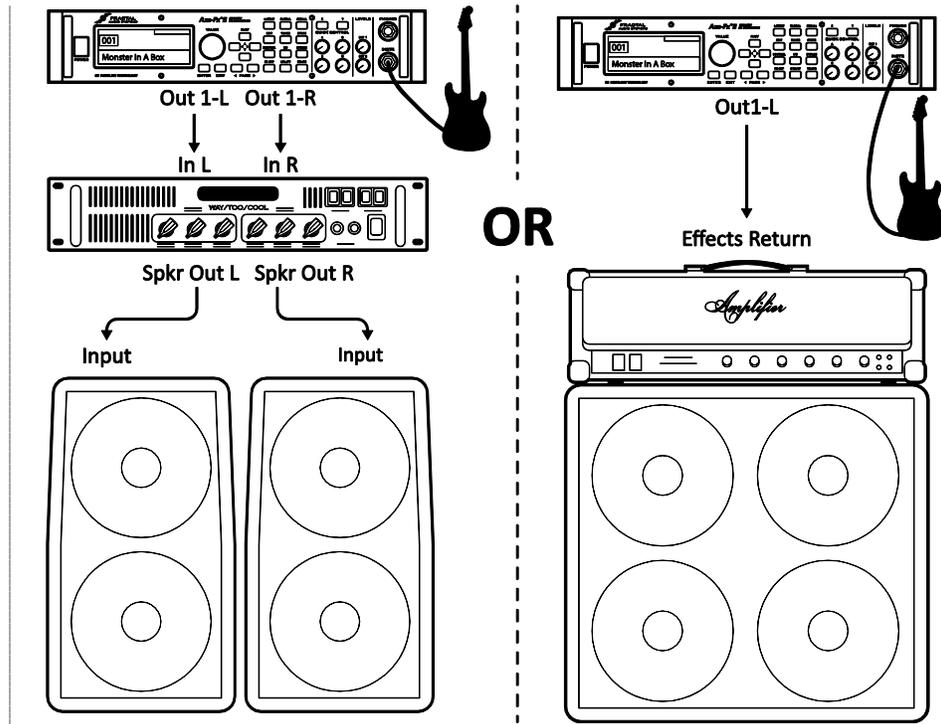
**Global Settings:** Default    **I/O Settings:** Default

**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.135) 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” power amp that does not color the tone or create a pronounced effect on the feel or dynamics. Solid State amps typically fit this bill.
- Power Amp simulations turned OFF when using the RETURN of the effects loop of a head or combo amplifier, or a more “guitar oriented” standalone power amp that adds noticeable sag/saturation/tonal color.
- In either case, it is completely safe and reasonable to try both setting ways 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. 133). 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.

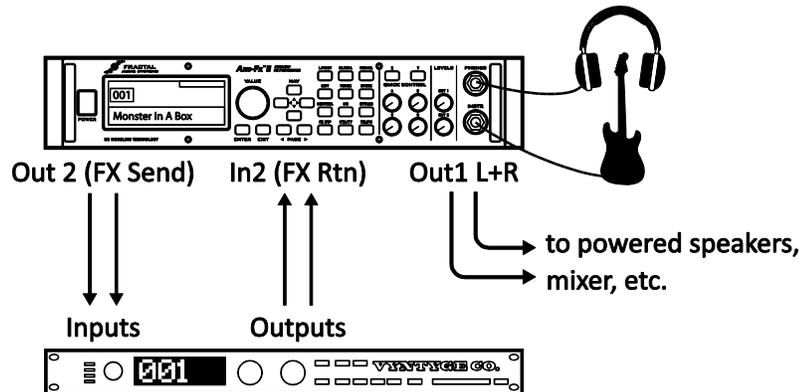
In any case, the settings required for this setup are not ideally suited for monitoring through the Axe-Fx II headphone jack, as what you hear through headphones will not sound like what you will hear through the speakers. Switching Power Amp and Speaker sims on and off, however, is a simple matter and can be done on an as-needed basis.

## CONNECTIONS

### 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



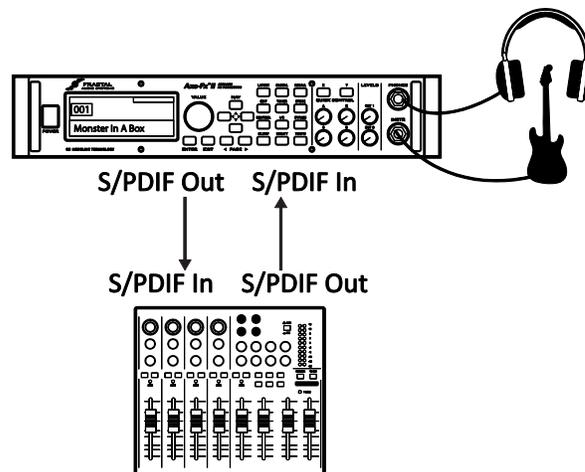
**Global Settings:** See Below **I/O Settings:** See Below

**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 66. 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



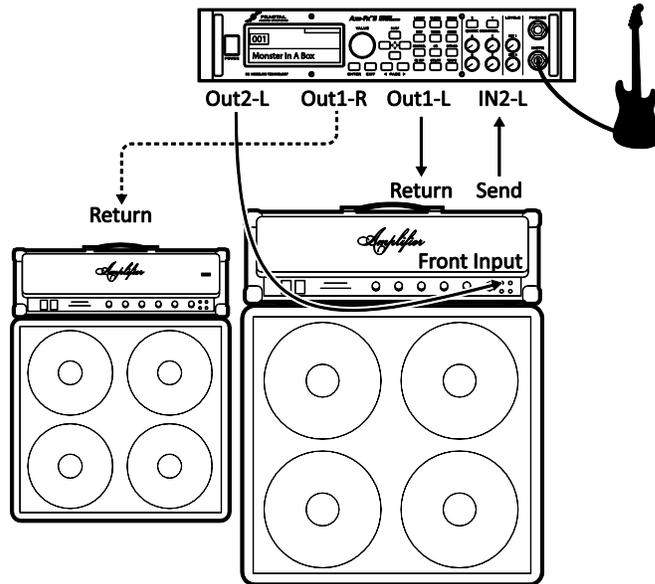
**Global Settings:** Default **I/O Settings:** See below.

**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 roundtrip. The required clock rate for both input and output signals is 48k. 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. 135). 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<sup>1</sup> and built-in or separate Guitar Speaker(s).
- ▶ Second Amp for Stereo (opt.)



**Global Settings:** Defaults OK, since special 4CM presets should not have AMP or CAB blocks!

**I/O Settings :** Set **OUT1 MODE** as required for stereo or mono. Adjust the two **OUTPUT BOOST PAD** parameters to lower the noise floor. See p.135 for details on these parameters. Otherwise, default settings.

**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 inside its effects loop, where many effects work best. Although a head and separate cab are shown, some 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. 66). 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 “roundtrip” to the amp’s preamp, and returns to the grid, where it is then processed by additional blocks—maybe chorus, delay, reverb, pitch shifter, etc. The final routing is via Axe-Fx II OUT1/L to the Return (Power Amp Input) of the amplifier and the cabs. 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. 161), 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.

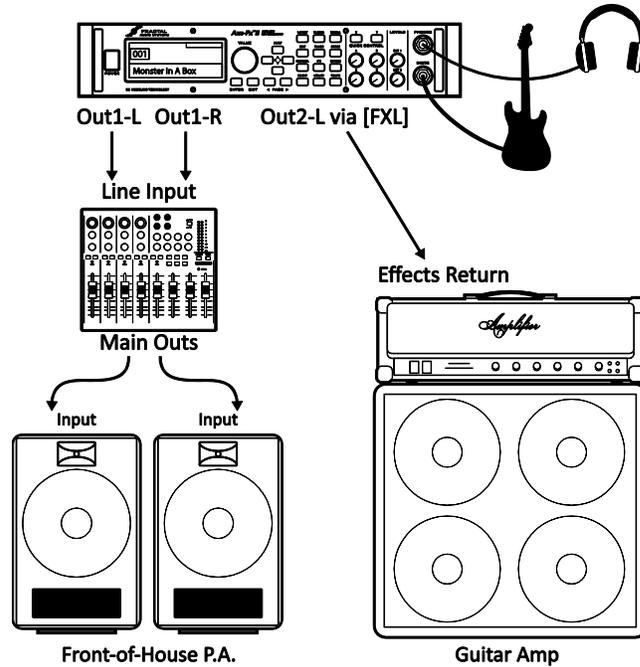
The no-amp/no-cab presets required for this setup are not well-suited for headphone monitoring, as what you hear in the headphones will not include any power amp or speaker simulations.

<sup>1</sup> Note: If your amp’s effects loop is the PARALLEL type, the chain of post-FX-loop blocks should 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, specially designed “dual-chain” presets also feed on-stage or “backline” amps. The first chain contains a fully simulated signal path. The second chain shares some effects with the first (those in front of the amps) but diverges from there.

The “backline” chain differs from the direct chain in several ways. First, its Amp has the SAG parameter set to “0.00” to disable power amp simulation *in that block only*. Since real guitar speakers are being used, no CAB block is inserted. Duplicates of effects which follow the amp in the direct chain follow. The chain ends in an FX LOOP block, which is routed via OUTPUT 2 to the FX Return of a tube amp. If a solid state or very neutral tube amp were used, the SAG of that second AMP block might be set higher for simulated tube power amp tone and dynamics.

An example of the type of preset used in this scenario appears below as it would appear in Axe-Edit.

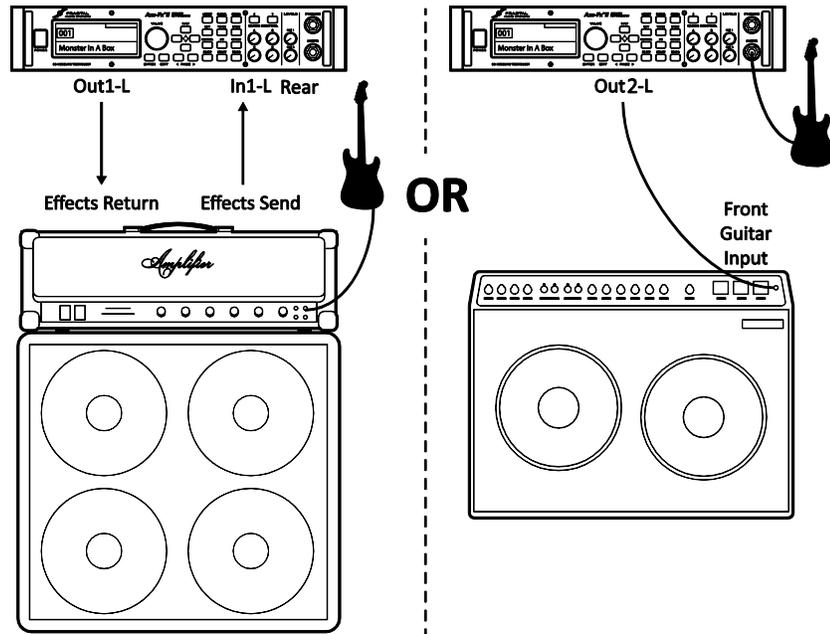


Figure 3-2 – A Dual Chain Preset

### 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. 161), 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.

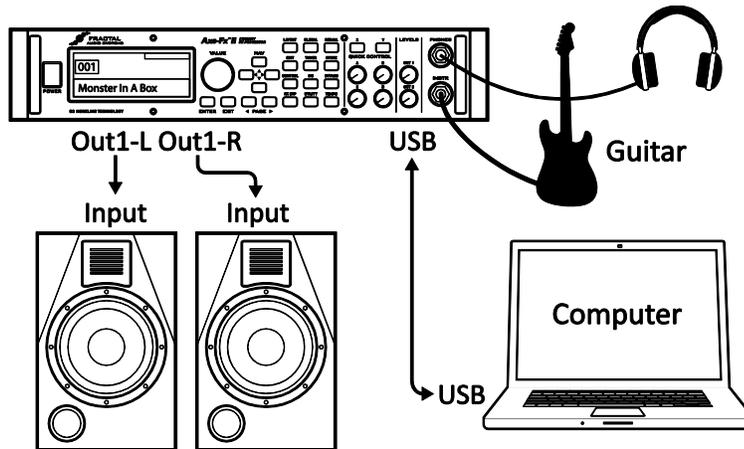
It is easy enough to extend each of the above configurations for use in stereo. With the Axe-Fx II in a loop, connect OUT1 R to the RETURN of a second amp, bypassing that unit’s preamp altogether, or run fully within the loops of two separate amps using IN1 L/R and OUT1 L/R. With the Axe-Fx II between 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.

## CONNECTIONS

### 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.<sup>2</sup>
- Route stereo audio from the computer through the Axe-Fx II, simultaneously using all the regular processing capabilities for your guitar so that 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.

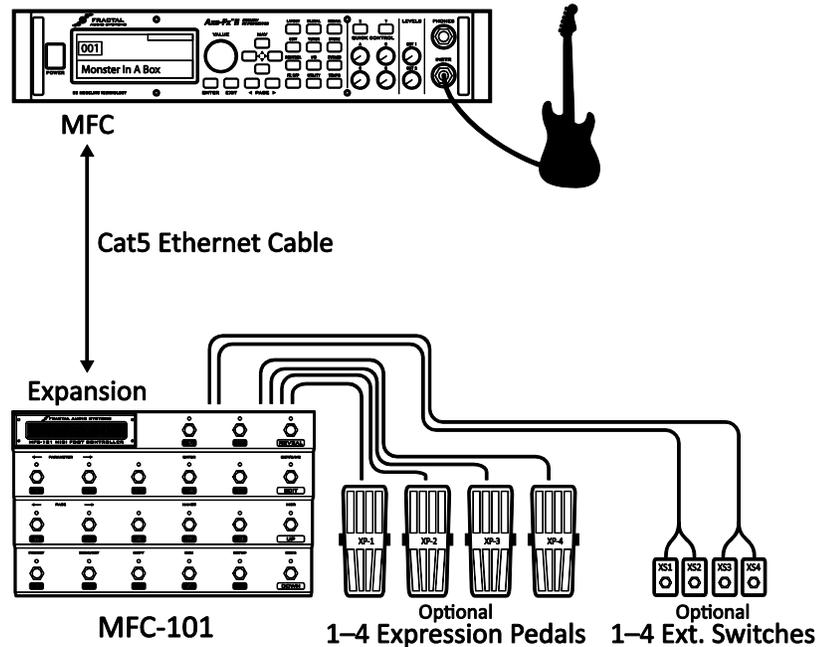
The Axe-Fx II can also be used in conjunction with any 3<sup>rd</sup>-party audio/MIDI interfaces.

<sup>2</sup> USB 2.0 required for more than two simultaneous channels from the Axe-Fx II to the computer.

### 3.4.10 Axe-Fx II and MFC-101

COMPONENTS:

- ▶ Guitar
- ▶ Axe-Fx II
- ▶ Ethernet Cable (CAT5)
- ▶ MFC-101 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.

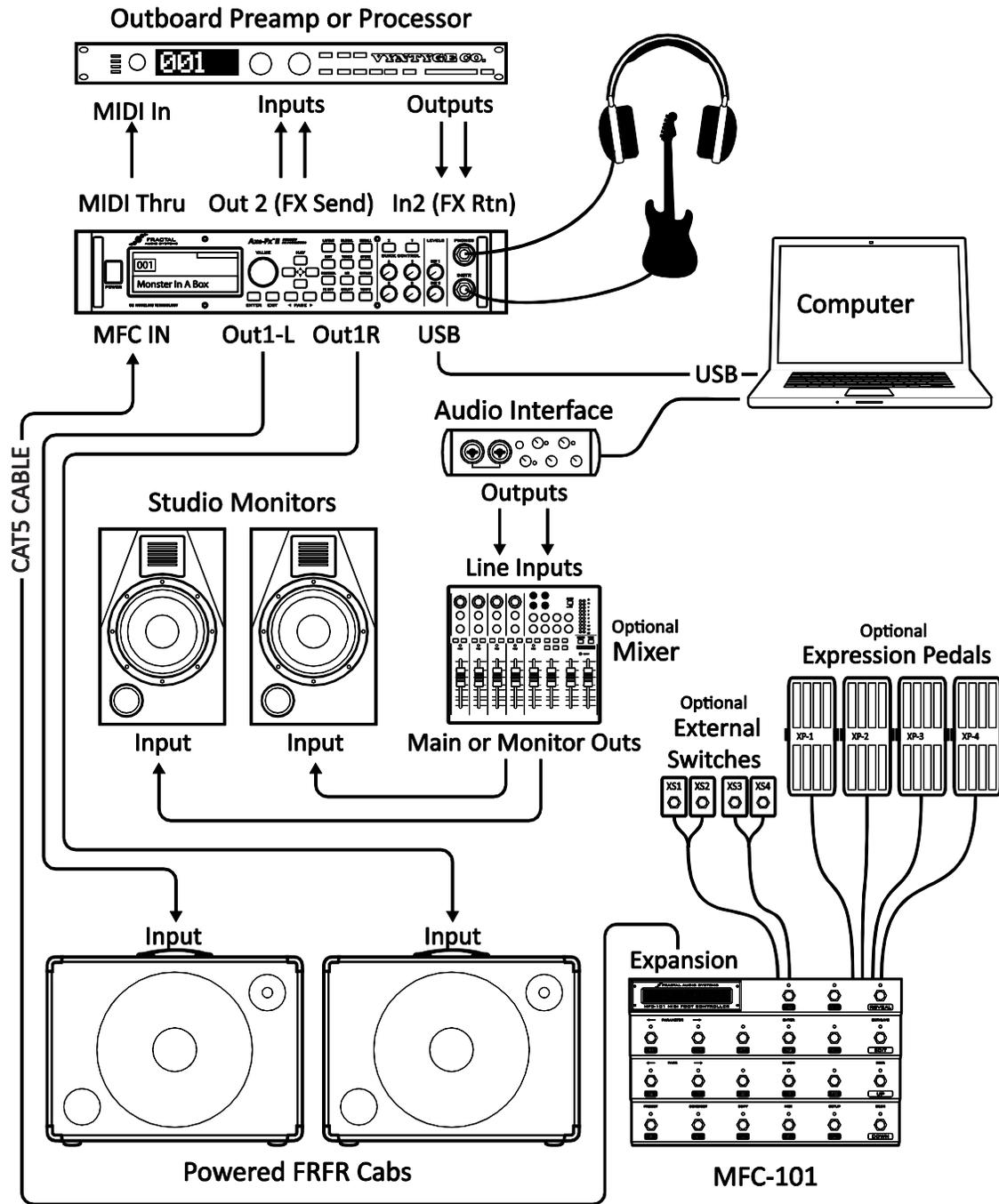
**Notes:** The Axe-Fx II provides a significant advantage over its predecessors in that the Fractal Audio MFC-101 MIDI foot controller can be connected to the new, dedicated rear-panel **MFC control port**. A single Ethernet (CAT5, non-crossover) cable carries bidirectional communications to support the many powerful features of MFC-101 “Axe-Fx Mode,” including automatic preset name and tuner display on the floor, “preset-aware” tri-state Instant Access Switch LED support, and much more.

Although it utilizes innovative connectors, the MFC-101 still transmits/receives MIDI to/from the Axe-Fx II, which can then forward messages to other devices via its MIDI THRU port. (Note: MIDI THRU is disabled by default. Turn it ON in the I/O menu under MIDI.)



**IMPORTANT:** 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 the MFC Control port connector to connect the two units. The Axe-Fx II provides required power from its internal power supply and transmits it over the Ethernet cable.

### 3.4.11 Axe-Fx II: One Possible “Big Rig”



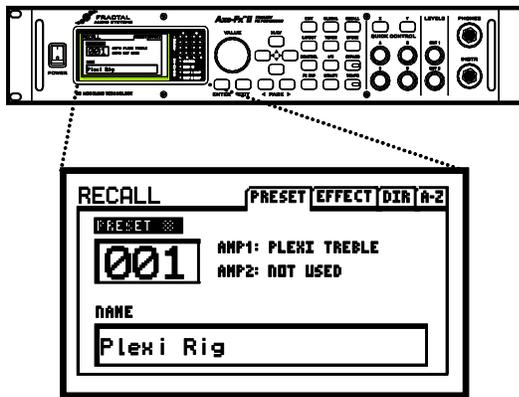
Here the Axe-Fx II is the centerpiece of a “big rig,” combining the capabilities of several other diagrams shown above. The main outs feed a pair of powered FRFR cabs, so you can design and monitor sounds and play live with the same sound. Meanwhile, USB provides all of the usual computer audio and MIDI integration features, with full-range studio monitors connected to a 3<sup>rd</sup>-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 ;)

# 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. 155 of the Appendix.

## 4.1 Presets

Each factory preset is a complete, pre-wired, pre-programmed setup for “end-to-end” sound—a guitar-in/stereo-out path with amps, cabs, and effects—perfectly mixed and possibly even equipped with one or more “modifiers” for automation or remote control. 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 saved in the Axe-Fx II.



The **AXE-FX II** has 384 editable presets. Press **RECALL** to enter normal play mode. The **VALUE** wheel or **NAV** buttons select presets.

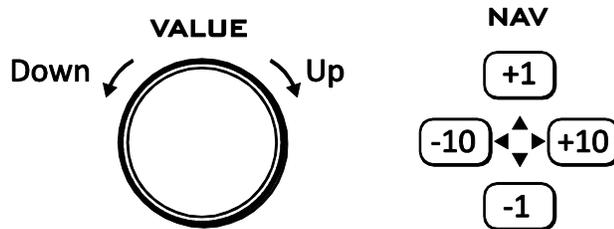


Figure 4-1

As stated in the illustration, you can load presets sequentially with the **VALUE** wheel or use the **NAV** buttons. The new **PRESET DIRECTORIES** allows direct access to all 384 presets in an ordered list. To open the directory, press **RECALL** and use the **PAGE** buttons to select the “**DIR**” (numeric) or “**A-Z**” (alphabetical).

To use the directory, just turn the **VALUE** wheel or use the **NAV** buttons to select the desired preset and press **ENTER** to load. You will be returned automatically to the main **RECALL PRESET** screen.



Presets can also be loaded using MIDI program change commands. Although it isn’t indicated in the display, the 384 presets are organized into three banks. Bank A contains presets 0–127, B contains 128–255, and C contains 256–383. MIDI Bank Select (CC#0) and Program Change commands allow you to specify/load any of the onboard presets remotely. (The MFC-101 MIDI Foot Controller, in fact, comes pre-programmed to select all 384 Axe-Fx II presets.) See p. 170 for a cross-reference on banks/program changes and preset numbers.

Any preset can be edited and saved at any time to any memory location. Let’s look at how to make changes.

## 4.2 The Grid

The grid, located on the EDIT page of the LAYOUT menu, is a 12 × 4 matrix into which effect “blocks” can be inserted and then connected together 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.

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

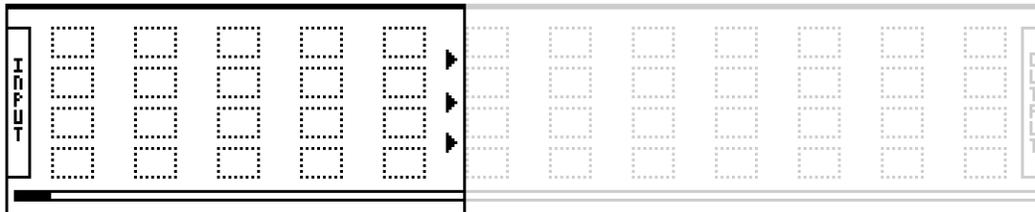
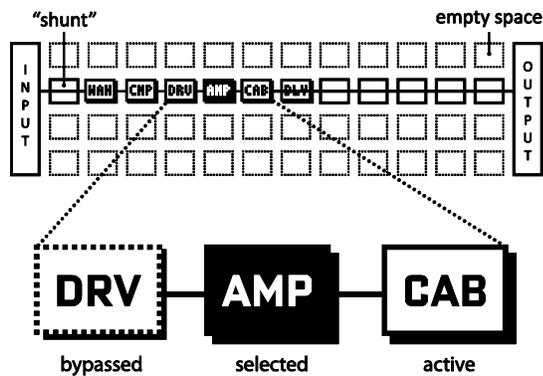


Figure 4-2 – In this image of an empty grid, off-screen areas are grayed.

### 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 blocks to insert plus options for SHUNT (see below) and NONE.

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

Press **FX BYP** to bypass a selected blocks.

Figure 4-3

#### To INSERT a BLOCK into an empty space...

- ▶ 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 98%. 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. 156 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.

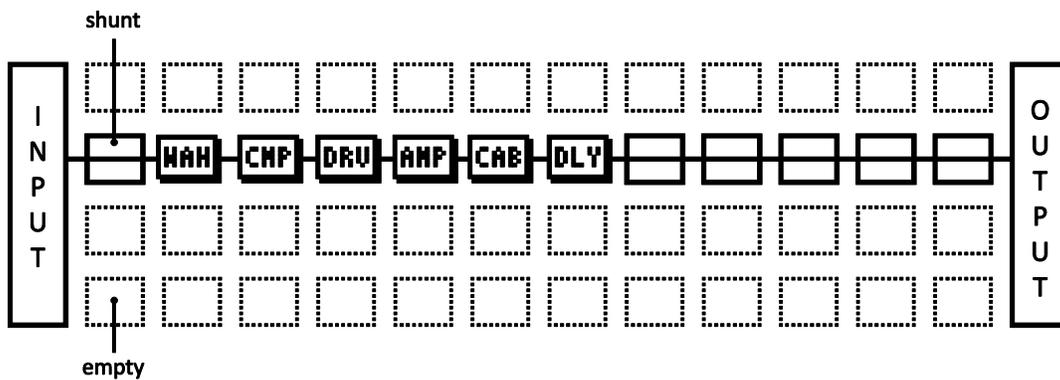


Figure 4-4 – Shunts are used to span distances between INPUT, BLOCKS, and OUTPUT.

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.”

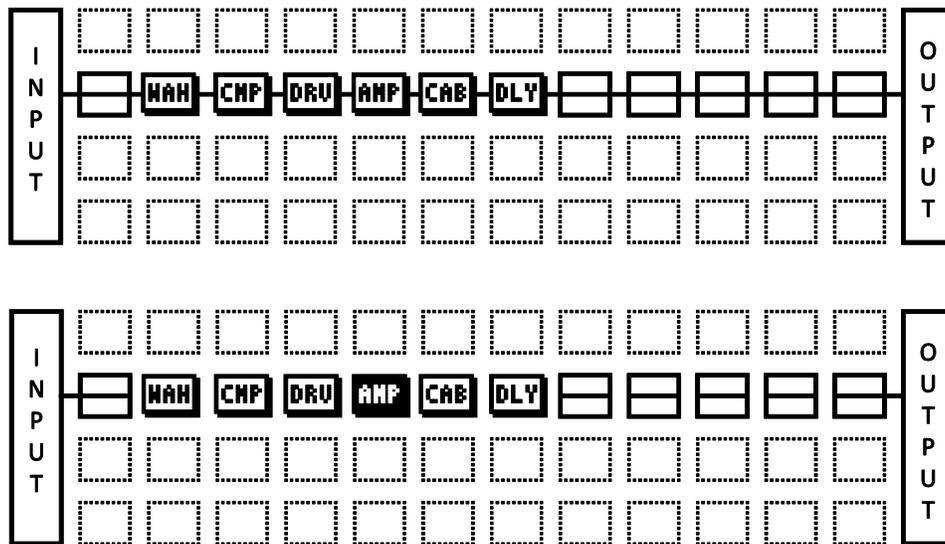


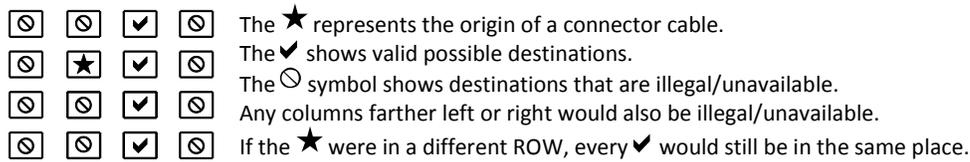
Figure 4-5 – Two otherwise identical presets shown WITH and WITHOUT connector cables between the blocks.

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.

**The Rules of Axe-Fx II Cables**

- ▶ No cables = No sound. Even one missing link will break the entire chain.
- ▶ Signal flows from LEFT to RIGHT.
- ▶ A cable MUST originate from a BLOCK or a SHUNT. Empty locations are not viable origins.

- ▶ If you try to connect to an EMPTY location, a SHUNT will be created there.
- ▶ You can ONLY connect to blocks in the next column to the right.



- ▶ 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.

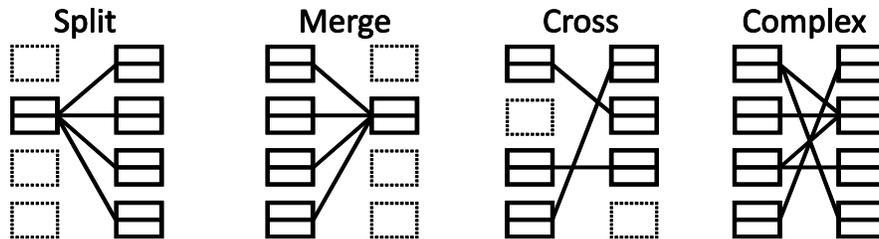


Figure 4-6 – Cables can CROSS one another without issue. Routings can be as COMPLEX as required.

### 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.

## BASIC OPERATION AND EDITING

- ▶ 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.

## 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.

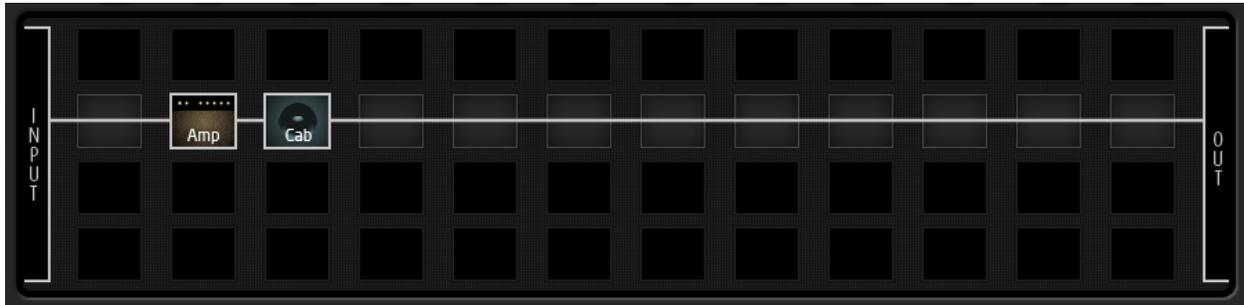


Figure 4-7 – Amp/Cab Preset

**EXAMPLE 2: Long Pedal Chain** – Here, effects are strung together to create a giant virtual rig. Compressor, Volume, Wahwah, 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.

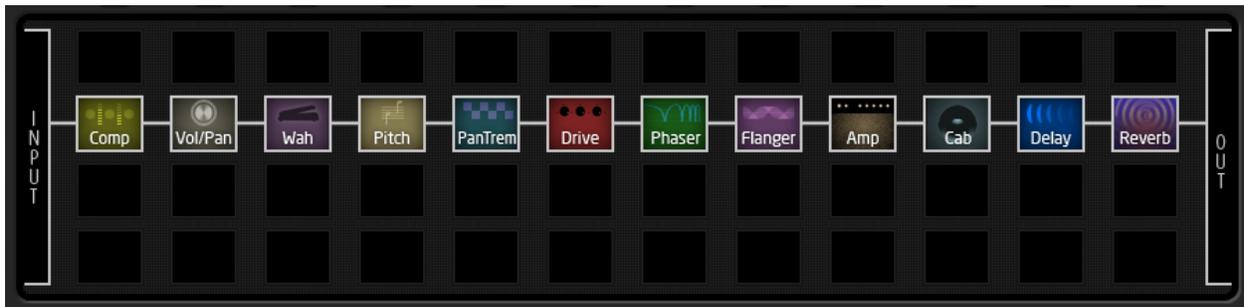


Figure 4-8 – A Pedal Chain

## BASIC OPERATION AND EDITING

**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. (Note: A single, stereo CAB block might also have been used, as shown in **Figure 5-4** on p. 48, but here, the decision was made to use each CAB block in “high resolution mono.”) 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.

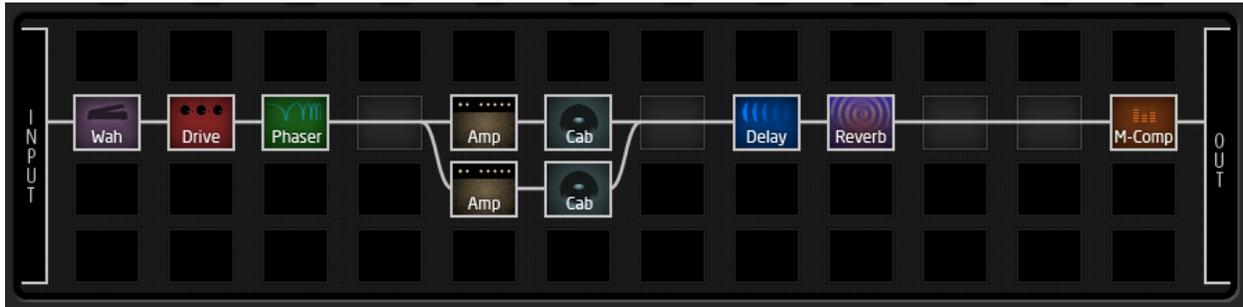


Figure 4-9 – Dual Amp Preset

**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 (its level controlled by an envelope for a “ducking” effect).

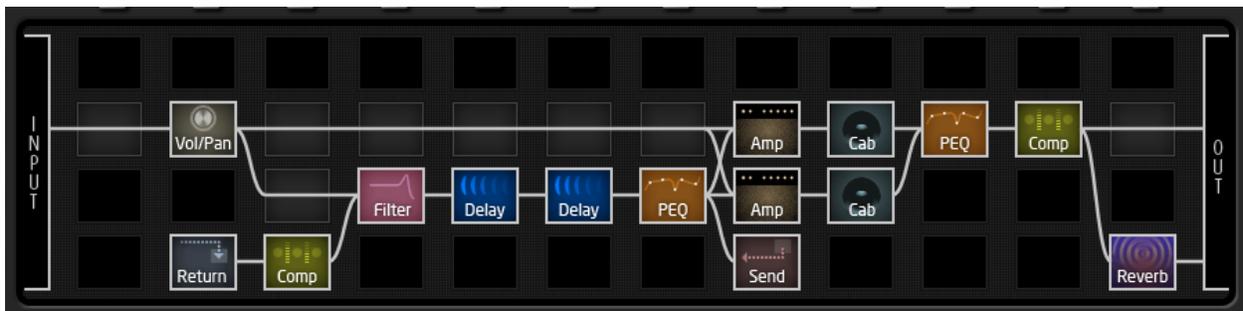


Figure 4-10 – Complex Preset

## 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](http://forum.fractalaudio.com) and join the discussion.

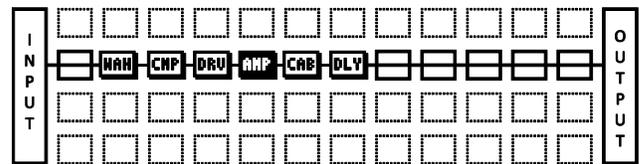
### 4.3 Editing Sounds

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...

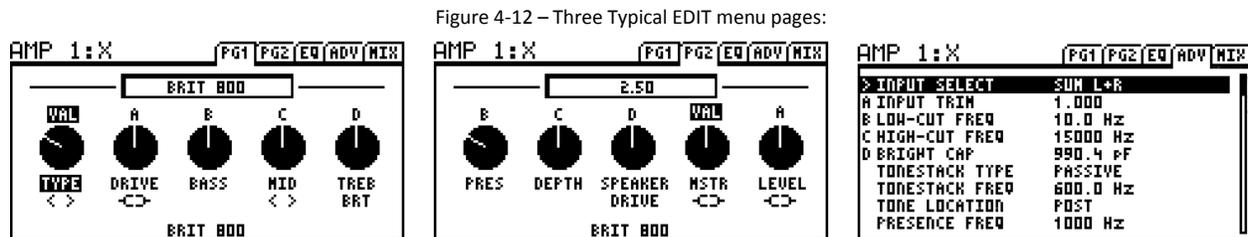
- ▶ Press **LAYOUT** to display the GRID.
  - If the GRID is not shown at first, press ◀ **PAGE** to reach the EDIT tab.
- ▶ Use the **NAV** buttons to select the desired block.
- ▶ Press **EDIT** to display the EDIT menu.
  - **NAV** buttons select parameters on the page.
  - The name of the selected 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.
- ▶ 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.



With a block **selected** in the grid,  
 Press **EDIT** for its edit menu.  
**LAYOUT** to go back ↵

Figure 4-11: Press EDIT with any block selected on the Grid

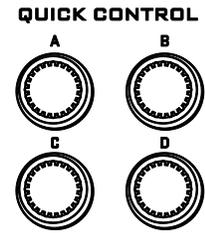
Pressing **EDIT** repeatedly will step through the EDIT menus for all blocks in a preset, sparing you a return trip to the grid. You can also press **EDIT** without entering the grid to go directly to editing a selected block in any preset.



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. 73), Parametric EQ (p. 88), and others, have specialized graphical displays designed to provide rich, intuitive editing experiences.

### 4.3.1 Quick Control

The **QUICK CONTROL A,B,C**, and **D** knobs on the front panel are dynamically mapped to four parameters in almost every block EDIT menu page<sup>1</sup>, whether knob or text-based. The current mapping is indicated by the appearance of a small A,B,C, or D above or to the left of the parameters on the screen. Assignments change dynamically as different parameters are selected for use with the **VALUE** wheel. Turn the corresponding lettered encoder knob to adjust a parameter.



The quick control knobs can also be used as Modifier SOURCES (see section 7 on p. 124). Their function in this capacity is disabled while editing as it might otherwise compete with menu operations.

## 4.4 X/Y Switching

One of the new features of the Axe-Fx II is X/Y switching, available on 10 different effect block types: Amp, Cab, Chorus, Delay, Drive, Flanger, Pitch Shifter, Phaser, Reverb, and Wahwah. Each instance of these blocks is equipped with two fully independent sets of parameters, “X” and “Y,” making it possible to have two completely different settings for one block. X or Y can be recalled instantly and seamlessly at the touch of a front panel button (while editing) or by using MIDI remote control (during performance). Each block with X/Y may be saved with either state selected.

The benefits of X/Y Switching are considerable. You might simulate amp channel switching (X as an American Clean and Y as a British Crunch). *This won't require the CPU overhead of two separate amp blocks!* Another application is mode switching effects. Instead of painstakingly tuning modifiers (p. 124), you can just dial in X, dial in the 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 under heavy feedback.

It is important to understand that **MODIFIERS** (see p. 124) 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 below).

Each available X/Y switch has its own dedicated global MIDI CC# assignment for remote switching<sup>2</sup>. This option is found in the I/O:CONTROL menu. See p. 139 for details.

### 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. If you accidentally double tap when you don't want to copy, press **EXIT** to cancel. So, to copy X to Y, double-tap **Y** and then press **ENTER**.

<sup>1</sup> Exceptions: Quick control functions work in PEQ blocks, but letters are not shown. Mixer type blocks do not support Quick Control.

<sup>2</sup> 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).

### 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. 141.

## 4.5 Bypassing a Block

Can you imagine a stompbox that can’t be stomped? Neither can we. Any effect<sup>3</sup> 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.



Engaged



Bypassed

*Note that accidentally double tapping or holding FX BYP will enter the Global Blocks menu instead (see p. 119). Press **EXIT** and try again if this happens.*

2. Use a MODIFIER (section 7) attached to the block’s **BYPASS MODE** parameter.<sup>4</sup>
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. 135)

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.117.

## 4.6 Loading Effects from another Preset

The **Recall Effect** function makes it possible to load block settings directly from one preset to another. 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. 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.

<sup>3</sup> The MIXER, FB SEND, and SHUNT blocks have no bypass functions.

<sup>4</sup> 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 384 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.
  - The **NAME** parameter allows you to edit preset names before saving:
    - Turn the **VALUE** wheel to change characters.
    - Press **X** or **Y** to jump to “handy” letters/symbols (space, 0, A, N, a, n).
    - Use **NAV** to move the cursor from one position to the next.
    - You can use up to 23 characters in a preset name.
- ▶ Press **ENTER** to Store.
- ▶ Press **ENTER** 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 the factory presets without overwriting any of them, 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 33 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 **Amplifier** block reproduces the sounds of an impressive array of vintage and modern guitar and bass amplifiers, with 70+ different “types” based on stock, custom and hybrid models. You get great amp sounds from the Axe-Fx II using only the basic drive and tone controls on PG1 and PG2 of the edit menu. In general most knobs behave exactly like those on actual amp when possible.

Advanced parameters offer deeper control, but don’t be put off by these at first: their values are “set right” for all models by default. That said, a deeper understanding of the AMP block offers deeper control, putting more tones at your fingers and enabling “mods” that would require a seasoned tech in the real world. (Even he/she would be unable to achieve many of the options possible with the virtual amps in the Axe-Fx II). It may be best if you can *listen* to the effects of the various controls while you read this section. The right amp with the right settings is the heart of the right tone.

**Note:** The **Cab** block (p. 48) is vital in creating an overall sound; if you’re not getting the sound you’re expecting from an amp, try different cab settings.

The “heart” of the amp is its capacity for ultra-realistic distortions, created using our proprietary dual-stage non-linearity generators. The preamplifier stage emulates the type of distortion generated by classic tube preamplifiers. The power amplifier stage emulates the type of distortion generated by a tube power amplifier and uses our proprietary **G2 Modeling Technology™**. The Axe-Fx II Amp also integrates a new built-in, 8-band graphic equalizer for additional tone control without extra blocks.

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

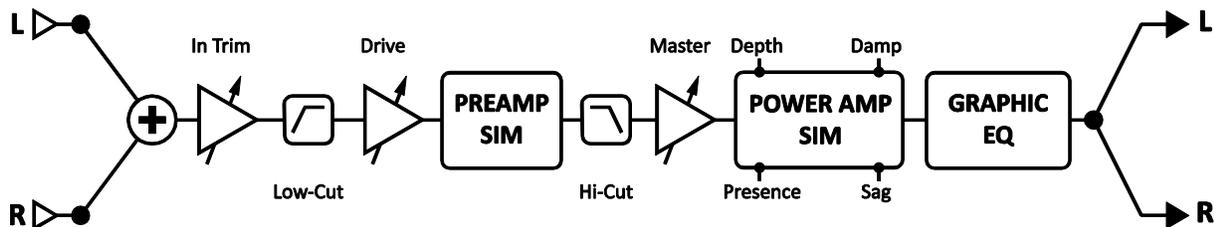


Figure 5-1 – A simplified diagram of the AMP block

The AMP block processes audio in mono. The INPUT SELECT and BALANCE parameters (see below) allow flexibility when combining the AMP with stereo effects.

### Amp X/Y Channel Switching

Each instance of the **Amplifier** 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 current **Bypass State** and any **Modifier** assignments). See **X/Y Switching** on p. 36 for more information.

## Amp Types

The following table details the different AMP types of the Axe-Fx II.

*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*

AMP TYPE	BASED ON	NOTES
<b>59 Bassguy</b>	1959 Fender® Bassman®	A low-to-medium gain amp designed for bass but more widely adopted by guitarists.
<b>65 Bassguy</b>	1965 Fender® Bassman®	The blackface version with a different circuit design.
<b>Vibrato Verb</b>	Fender® Vibroverb®	Based on a 40W combo that's great for clear or grinding cleans and gutsy blues.
<b>Deluxe Verb</b>	Fender® Deluxe Reverb®	Great, chimey tone with nice power amp breakup when you push the MASTER.
<b>Double Verb</b>	Fender® Twin Reverb®	Based on the "Vibrato" channel. Known for amazing clean sounds and nice breakup.
<b>Jr Blues</b>	Fender Blues Jr.	A gutsy little classic with dual EL84s.
<b>Class-A 15w TB</b>	Vox® AC-15®	The heart of this amp's tone comes from its power section and no negative feedback.
<b>Class-A 30w</b>	Vox® AC-30®	A combo that dominated the British Invasion. Gritty character, warm tone, great feel.
<b>Class-A 30 TB</b>	Vox® AC-30TBX®	Created in response to demand for "More Treble." Great highs + slightly reduced bass.
<b>Brit JM45</b>	Marshall® JTM45®	Made famous by Clapton and others; actually a modified Bassman® design.
<b>Plexi Normal</b>	Marshall® Super Lead 1959®	The classic amp head that gave rise to "the stack." Great for crunchy rhythm work.
<b>Plexi Treble</b>	1959® Treble Channel	Based on the "High Treble" channel of the legendary '59.
<b>1987x Normal</b>	Marshall® 1987x Vintage Series	Features what many consider to be an "essential" mod to the tonestack of this Plexi.
<b>1987x Treble</b>	1987x Treble Channel	The treble channel of the 1987x Vintage Series Plexi.
<b>Brit 800</b>	Marshall® JCM 800®	Based on the vaunted model 2204. Bring the Master up for true 80s tone.
<b>Brit 800 Mod</b>	Marshall® JCM 800®, modded	The 800 with a collection of the most popular mods to the amp.
<b>Hipower Normal</b>	Hiwatt® DR103 (Normal)®	Medium-gain, full sound amp with a unique tone-stack and a chimey, grinding tone.
<b>Hipower Brillnt</b>	Hiwatt® DR103 (Brilliant)®	A brighter model based on the amp's "Brilliant" channel.
<b>USA Clean 1</b>	Mesa Boogie™ MKIV™ (Rhythm 1)	A somewhat neutral, clean- sounding model that can pushed into warm clipping.
<b>USA Clean 2</b>	Mesa Boogie™ Triaxis™ (Green)	Based on the "Vintage Fat Rhythm (Mark I, Blackface)" channel.
<b>USA Rhy 1</b>	Mesa™ Boogie MKIV™ (Rhythm 2)	Based on THE California crunch rhythm sound. Rhythm Ch. 2 with "Fat" switch OFF.
<b>USA IIc+ Norm</b>	Mesa Boogie® Mark II™ Lead (Normal)	Based on a US-made amp famous for its smooth overdrive sound. This model has the pull bright OFF.
<b>USA IIc+ Bright</b>	Mesa Boogie® Mark II™ Lead (Bright)	Based on a US-made amp famous for its smooth overdrive sound. This model has the pull bright ON.
<b>USA Lead 1</b>	Mesa Boogie™ MKIV™ (Lead)	This model has a tight, focused, hi-gain sound. Great for fusion and rock leads.
<b>USA Lead 2</b>	MKIV® (Lead + Treble Shift ON)	Treble Shift gives this amp a slightly different character with a little more cut.
<b>Recto Org Vntg</b>	Mesa Boogie™ 2 Ch. Dual Rectifier® (Orange Ch.)	Based on the original Mesa Boogie 2-channel Dual Rectifier "Vintage" (Orange) Ch.
<b>Recto Red Vntg</b>	Mesa Boogie™ 2 Ch. Dual Rectifier® (Red Ch.)	The red channel of the same.
<b>Recto Org Mdrn</b>	Mesa Boogie™ 2 Ch. Dual Rectifier® (Orange Ch.)	Based on the modern version of the Dual Rectifier amp. Circuit changes made this version more aggressive.
<b>Recto Red Mdrn</b>	Mesa Boogie™ 3-Ch. Dual Rectifier® (Red Ch.)	The red channel of the same.
<b>Euro Blue</b>	Bogner® Ecstasy Blue Channel®	Based on the 20th Anniversary model. OD channel w/ BOOST + STRUCTURE OFF.
<b>Euro Red</b>	Bogner® Ecstasy Red Channel®	Same as above but with OD channel w/ BOOST + STRUCTURE ON.
<b>Shiver Cln</b>	Bogner® Shiva Clean Channel	Based on the 90W anniversary model. Powerful shimmering cleans.
<b>Shiver Ld</b>	Bogner® Shiva Lead Channel	Based on a sweet, rich- sounding amp with aggressive, English- style midrange punch.
<b>Euro Uber</b>	Bogner® Überschall	Based on "High Gain" channel of this 120W head. Heavy grinding lows and insane gain.
<b>Solo 99 Cln</b>	Soldano® X99® Preamp (Clean )	Based on the clean channel of a Soldano/Caswell midi-motorized preamp.
<b>Solo 100 Rhy</b>	Soldano™ SLO-100® (Normal Ch.)	Based on SLO-100®, noted for its hot-rod chrome chassis and aggressive rhythm tone.
<b>Solo 100 Ld</b>	Soldano™ SLO-100® (OD Ch.)	Based on the snarling lead channel of the above amp.
<b>Friedman BE</b>	Friedman Brown Eye	What many call "the ultimate modded Plexi" by Dave Friedman of Rack Systems.
<b>Friedman HBE</b>	Friedman Hairy Brown Eye	The BE amp's alternate voicing with a gain boost. A killer hi-gain tone in your arsenal.
<b>PVH 6160</b>	Peavey® EVH® 5150™	Based on the high- input lead channel of an amp named after the criminally insane.
<b>Mr Z 38 Sr</b>	Dr. Z Maz 38 SR®	Based on an amp popular with country and roots players.
<b>CA3+ Rhy</b>	CAE 3+ SE® (Ch 2)	Based on Channel 2 (Rhythm) of a preamp designed by Custom Audio Electronics®.
<b>CA3+ Ld</b>	CAE 3+ SE®(Ch 3)	Channel 3, (Lead) of the same.
<b>Wrecker 1</b>	Trainwreck™ Express	Based on the Trainwreck Express—designed and built by the late, great Ken Fischer.
<b>Corncob M50</b>	Cornford MK50II®	Based on a boutique British amp. Plexi-Meets-Modern tone with big cojones.
<b>Carolann Od-2</b>	CarolAnn OD2r®	The celebrated OD2r. Model fine-tuned by the highly respected Alan Phillips himself!
<b>Fryette D60 L</b>	Fryette D60®	Based on the Fryette Amplification D60 in the "Less" mode.
<b>Fryette D60 M</b>	Fryette D60®	Based on the Fryette Amplification D60 in the "More" mode.

<b>Brit Brown</b>	“The Brown Sound”	A faithful recreation of the legendary “Brown Sound” —The modded “#1” Marshall®.
<b>Citrus RV50</b>	Orange® Rockerverb®	Based on the dirty channel of the 50W head known for warmth and rich harmonics.
<b>Jazz 120</b>	Roland® JC-120®	The only solid-state-based model in our collection; a quintessential clean tone.
<b>Energyball</b>	ENGL Powerball®	Very high-gain German model. Lots of bass. Great for aggressive, drop-tuned riff work.
<b>ODS-100 Clean</b>	Dumble OD Special® (Clean)	Based on the Clean channel of a coveted but rare amp made famous by Robben Ford.
<b>ODS-100 Lead</b>	Dumble OD Special® (Drive Ch.)	The same amp. OD channel. Also played by the great Larry Carlton and many others!
<b>FAS Rhythm</b>	Created by Fractal Audio Systems	Combines the best features of the British and USA crunch models.
<b>FAS Lead 1</b>	Created by Fractal Audio Systems	Neutral high-gain lead with a tight midrange.
<b>FAS Lead 2</b>	Created by Fractal Audio Systems	Hot-rodged British lead sound with a tonestack by Custom Audio Electronics.
<b>FAS Modern</b>	Created by Fractal Audio Systems	A high-gain hybrid. Equally well-suited to modern rhythm and lead work.
<b>Das Metall</b>	Diezel™ VH4®	Based on a high-gain, boutique amp famous for its powerful, heavy, aggressive sound.
<b>Brit Pre</b>	Marshall® JMP-1 Preamp®	Based on a rack-mount preamplifier version of the Brit 900. Crunchy “ZZ” tone.
<b>Buttery</b>	Budda® Twinmaster	Based loosely on a late 90s specimen. Relies mostly on power amp distortion.
<b>Boutique 1</b>	Matchless Chieftain®	A medium-gain amp: thick, yet crisp, with a fair amount of power amp breakup.
<b>Boutique 2</b>	Matchless Chieftain® + Boost	Based on the same amp but with a boost for more gain and high-frequency emphasis.
<b>Cameron Ch 1</b>	Cameron CCV100	An amp its creator Mark Cameron calls “one pissed off amp.”
<b>Cameron Ch 2</b>	Cameron CCV100	The higher gain tone of the Cameron.
<b>SV Bass</b>	Ampeg SVT®	Based on a head used for decades by famous bassists the world over.
<b>Tube Pre</b>	Studio Tube Preamp	A completely neutral, low-gain tube pre useful for “warming up” various sources.
<b>FAS Brown</b>	“The Brown Sound”	The original BROWN model from the Axe-Fx Ultra.
<b>Big Hair</b>	FAS Original	Mids without mud. Revive the 80s metal scene. (Spandex not included.)
<b>Solo 99 Lead</b>	Soldano® X99® Preamp (Lead)	Based on the lead channel of a Soldano X99 preamp.
<b>Supertweed</b>	FAS Supertweed	A custom creation by Fractal Audio Systems. One heck of a tweed!
<b>Tx Star Lead</b>	Mesa Boogie® Lonestar™	This model is based on the lead channel of a Mesa Lonestar.
<b>FAS Wreck</b>	Trainwreck™ Express	The original WRECKER 1 model from the Axe-Fx Ultra.
<b>Brit JVM OD 1</b>	Marshall® JVM 410®	Based on the JVM410, channel OD1, Orange mode.
<b>Brit JVM OD 2</b>	Marshall® JVM 410®	OD 2 Channel of the same.
<b>FAS 6160</b>	PVH 6160	A modified version of the PVH 6160 – less fizzy than the original, with a bouncier feel.
<b>CALI LEGGY</b>	Carvin Legacy I	A model based on this signature amplifier.
<b>USA LEAD 1+</b>	Mesa Boogie™ MKIV™ (Lead)	The Lead channel with the MID GAIN switched on
<b>USA LEAD 2+</b>	Mesa Boogie™ MKIV™ (Lead)	The same, with both the Treble Shift and Mid Gain on.
<b>Prince Tone</b>	Fender® Princeton®	Based on a single-ended Fender Princeton with a single-ended power section.
<b>Blankship Leeds</b>	Blankship Leeds	Based on a boutique version of an 18W Marshall with a big sound at low power.

New amp types will be added from time to time. Please check our web site for owner’s manual updates.

### 5.1.1 Basic Amp Parameters

**TYPE** – Selects the amplifier simulation. The simulations are, for the most part, based on classic and modern amps, but they also include some hybrids and wholly original types that we created. For a complete list of current amp types provided, see table above.

**DRIVE** – Sets the preamp gain for more or less preamp distortion. Used in conjunction with the **MASTER** (see below), the **DRIVE** control and the amp **DRIVE** determine whether the sound will be clean, slightly broken up, moderately overdriven, or completely distorted. **DRIVE** matches actual amp knob positions within about 10%.

The Axe-Fx II faithfully reproduces the sound of the treble peaker circuit on the **DRIVE** control found on many amps. This can be heard when the low frequencies are attenuated more than the highs when the **DRIVE** is turned down (and vice versa). To make the most of this capability, place a **DRIVE** block in front of the **AMP**. Adjust the **DRIVE** control of the amp to achieve the desired amount of input treble peaking, and then adjust the **DRIVE** and **LEVEL** of the block out front to achieve the desired effect.

**BOOST** – Toggles the input boost for an additional 12 dB of input gain. Use this to add more gain to the amp simulation. To turn the **BOOST** switch ON or OFF, use the **NAV** keys to select the **TYPE** knob while editing an amp and press **ENTER**. The letters “BOOST” appear beneath the knob to indicate that the boost is engaged. The symbol < > appears when this setting is OFF. (See **BRIGHT** below for an illustrated example).

**BASS, MID, TREBLE** – While many amp sims use simple filters to approximate amp tone controls, the Axe-Fx II exactly replicates the frequency and phase response of a classic passive tonestack. Use your ears to set each control to the desired position. A different setting on the model may be required to achieve the same response as the actual amp. In most cases however, knobs will be accurate to within 10% of the actual amp.

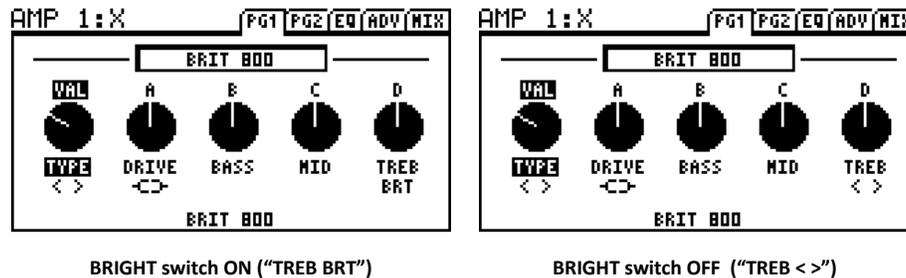
❗ Some of the amps simulated on the Axe-Fx II do not have the full complement of tone controls. Some, for example, have no mid control. To faithfully simulate the configuration of the original, set any superfluous controls to noon (or “0.00” if you are using the “ACTIVE” tonestack type; see below). Of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

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

**BRIGHT** – Many amplifiers contain a “treble peaker” on the volume control. On some amps this switch is included as a pull switch on the volume control or toggle switch. On others, the circuit is hard-wired.

Every amp **TYPE** on the Axe-Fx II includes a bright circuit. If the original amp has no bright circuit, the default state is off but **BRIGHT** can be turned on to apply circuit values most suited to an amp of that general type. If the amp has a hard-wired treble peaker, the default **BRIGHT** state is on. The effect may be subtle or quite pronounced; depending on the amp selected, and is also affected by the **BRIGHT CAP** setting (p.54).

To turn the bright switch ON or OFF while editing an amp, use the **NAV** keys to select the **TREBLE** knob and press ENTER. The letters “BRT” appear beneath the knob to indicate that the bright circuit is engaged. The symbol <> appears when the bright switch is OFF.



**FAT** – 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.

**PRESENCE/HI-CUT** – 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. If **DAMPING** (which determines the amount of negative feedback) is set to “0.00,” **PRESENCE** becomes a simple high-shelf equalizer at the output of the power amp, and its label changes to “**HI-CUT**.” This allows you to control the high-frequency response of the power amp for models that don’t have any negative feedback. 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 might darken the sound undesirably.

*Note: Unlike previous Axe-Fx models, the virtual presence circuit on the Axe-Fx II now operates from 0-10 instead of +/-5. Set it as you would on a regular amp.*

**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.

**MASTER** – The almighty **Master Volume** is a very important control. It determines the amount of distortion contributed by the power amp simulator, and its setting is key to an amp’s sound. As the Master is turned up, the entire character of the amp will change. The tone controls will have less influence on the sound, and the sound will have more “bloom” and touch sensitivity. To prevent volume jumps when changing models, Virtual “master” settings 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 qualities of different **DRIVE** and **MASTER** settings and how to find the great tones offered by different combinations.

- If a real amp doesn’t have a Master, begin with the Axe-Fx **MASTER** at or close to maximum when using that **TYPE**. Lower settings can also be used, and can sometimes reduce “harshness.”
- For “Cranked Amp” sound, set the **MASTER** to 3 o’clock and then slowly bring the drive up until the desired tone is achieved. At high Master settings, less drive is usually required, especially for high-gain types. If the Master Volume is set high, reducing the drive control is usually required for “best” results.
- Amps designed for preamp distortion will typically sound better with the **MASTER** set low to prevent the tone becoming muddy or excessively 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** to achieve desired power amp 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.

**LEVEL** – This is a copy of the **LEVEL** control on the MIX page for easy volume adjustment without page-flipping. It has no effect on the inherent sound of an amp.

**POWER AMP VOICING** – Voices the amp to a variety of tonal styles, taking the guesswork out of mix engineering by automatically adjusting the overall tone. Choose “Neutral” for the raw amp sound or choose one of the other voicings for a quick “mix-ready” tone.

## Speaker Page

These parameters shape the virtual speaker impedance curve and the resulting resonances in the virtual power amp. Amp-Speaker interaction causes 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 **DAMPING** is greater than 0. This is because negative feedback flattens the response curve.

**LF RESONANCE, LF RES FREQ, LF RES Q** – Guitar loudspeakers have a low-frequency resonance, typically about 100 Hz. This shifts up slightly when the speaker is mounted in an enclosure. This resonance causes an increase in the power amplifier response due to the finite output impedance of the power amp.

**MID RESONANCE, MID RES FREQ, MID RES Q** – While most speakers don’t have a third resonance, this parameter allows you to fine-tune the edge-of-breakup profile for “hyper-realistic” tones. For authentic response, set MID RES flat (0.00).

**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.

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

### Amp EQ Page

The amplifier includes a built-in, 8-band graphic equalizer, eliminating the need to follow the amp with a separate EQ block for additional tone-shaping.

## 5.1.2 Amp Dynamics Parameters

**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.

**!** **IMPORTANT:** Turning this control fully counterclockwise defeats power amp simulation for an individual AMP block so you can use it as a PREAMP into an external tube power amp without globally disabling power amp simulation (see section 8.1 on p. 133 or the “FOH + Real Amps” diagram on p. 22).

**DYNAMICS** – This controls the amount of dynamics processing and models the interaction between the power amp, power supply and loudspeaker under high power-level conditions.

**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 DRIVE** – Simulates core saturation in the output transformer. Higher values simulate more core saturation and, therefore, more transformer distortion.

## 5.1.3 Advanced Amp Parameters

**INPUT SELECT** – The AMP 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 setting).

**INPUT TRIM** – Allows you to adjust the relative gain of the preamp. Increasing the value will cause the amp to have more gain than designed and vice versa. 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 Drive control because the Drive control interacts with the surrounding circuitry and changes the frequency response as it is varied.

**DEFINITION** – This simple control shapes the high-end definition of tone as it enters the amp block.

**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. You can increase the high-cut frequency to achieve a brighter tone. Conversely you can reduce this value to achieve a darker or less brittle tone.

**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.

**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.

**STONE 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 amplifier). This parameter defaults to an appropriate value whenever you change the amp **TYPE**, but it can then be changed as desired.

**PRESENCE FREQ** – Alters the center frequency of the amp’s **PRESENCE** control. This parameter defaults to an appropriate value whenever you change the amp **TYPE**, but it can then be changed as desired.

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

**GRID MODELLING** – Enables or disables grid modeling in the virtual power amp tubes. Turning this to OFF can reduce subjectively undesirable distortion.

**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.

**DAMPING** – 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, **DAMPING** 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.

**MAIND IMP. (SAG)** – This is a duplicate of the **SUPPLY SAG** parameter on the **AMP DYNAMICS** page.

**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.

**TRIODE1 PLATE FREQ, TRIODE2 PLATE FREQ** – This parameter sets the cutoff frequency of the plate impedance for the next-to-last triode 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 this parameter. The triode 2 plate capacitor is also exposed.

**TRANSFORMR LF, TRANSFORMER HF** - Duplicates of controls found on the amp’s **SPKR** page.

**TRANSFORMER MATCH** – Duplicate of the same parameter on the **DYNAMICS** page.

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

**MV 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 AC 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.

**SAT SWITCH** – The **Saturation Switch** engages a popular mod between the preamp and the tonestack for more aggressive distortion character.

**PRESENCE/DEPTH TYPE** – Selecting “PASSIVE” models the typical passive circuit used in actual tube amps. “ACTIVE” types use an idealized circuit that may be less authentic but useful in some circumstances.

**TRIODE HARDNESS** – This parameter controls how sharply the triodes enter saturation and can be used to simulate softer or harder tubes. The default value is 5.0 and is set to this value whenever the type is changed.

The effect of this is subtle and most apparent at edge of breakup. Lower values give softer saturation, higher values give a more aggressive breakup.

**POWER AMP LO CUT, POWER AMP HI CUT** – These filters are provided for shaping the tonal color of the virtual power amp. Note that in addition to applying a voicing filter, the **POWER AMP VOICING** parameter (**Basic Amp Parameters**, beginning on p. 41) will also automatically set values for **POWER AMP LO** and **HI CUT**.

**BIAS EXCURSION** – Controls how much the grid voltage droops when the grids conduct.

**EXCURSION TIME, RECOVERY TIME** – control the time constants associated with bias excursion.

**DYNAMICS, DYNAMICS TIME** – These control the dynamics processor in the amp block, affecting the feel of the amp by setting how much and how quickly the level compresses in response to higher input.

### AMP Mix Page

The **Amplifier** block also has a **MIX** page with **LEVEL**, **BALANCE**, and **BYPASS MODE** parameters. See **Common Mix Parameters** on p. 117 for more information.

### AMP block Resolution

Presets which use a single amp block run in high-resolution mode, utilizing an entire processor. This provides greater 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. Note that switching between presets with differing number of amp blocks may introduce an additional delay as a “soft reset” of the amp blocks must be done whenever changing the resolution.

## 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 contains 70 built-in “factory” cabinet simulations, plus 50 memory locations you can use to load custom Impulse Response (“IR”) files. The Cab block also offers the ability to apply room and microphone simulations, plus tone-shaping of low and high frequencies.

Factory cabs include custom creations by Fractal Audio System, selections from 3<sup>rd</sup>-party libraries by RedWirez and Ownhammer, and contributions from loudspeaker design engineer Jay Mitchell. Visit <http://www.redwirez.com> and <http://www.ownhammer.com> to purchase additional IRs you can load into your Axe-Fx II.

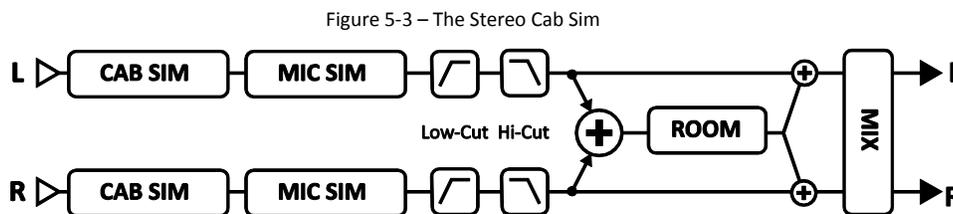
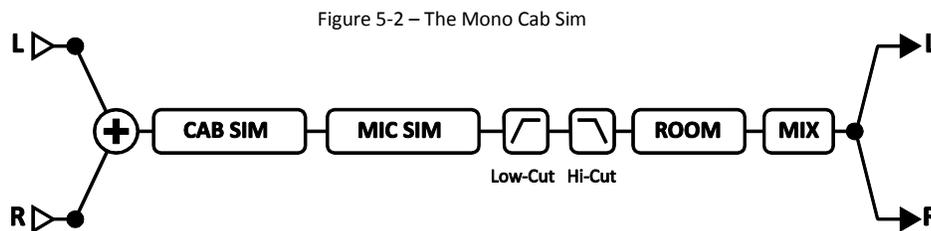
Each Axe-Fx II preset can use two fully independent **Cab** blocks.

### Cab X/Y Channel Switching

Each instance of the Cab 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 current **Bypass State** and any **Modifier** assignments). See **X/Y Switching** on p. 36 for more information.

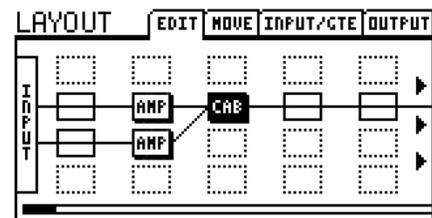
### Cab Parameters

**MODE** – Selects between “MONO HI-RES,” “MONO LO-RES,” and “STEREO” modes. In HI-RES mode, resolution is at maximum (1 × 2048). In LO-RES mode, resolution is halved (1 × 1024). In STEREO mode, the processing is stereo, and the resolution is lo-res (2 × 1024).



To use a stereo Cab block with two amps, set the balance control for AMP 1 fully left, the balance control for AMP 2 fully right, and set the Cab block **MODE** set to “STEREO.”

Figure 5-4 – Two amps into a CAB block set to STEREO mode



**CAB (TYPE)** – Sets the cabinet type by selecting from 70 “FACTORY” and 50 “USER” IRs. Cabinet types are listed in the table below. Those marked (OH) are exclusive “blends” created from selections in the Ownhammer collection. Those marked (RW) are exclusive “blends” created from selections in the RedWirez collection. Those marked (JM) were created by Jay Mitchell. Those not marked are creations of Fractal Audio.

1. 1×6 Oval	24. 2×12 Top Boost Blue (RW)	47. 4×12 German V30 (RW)
2. 1×8 Tweed	25. 2×12 Top Boost Silver (RW)	48. 4×12 German Boutique (RW)
3. 1×10 Gold	26. 2×12 Boutique (RW)	49. 4×12 PVH 6160 (RW)
4. 1×10 Blue	27. 2×12 Jazz (RW)	50. 4×12 Uber T75 (RW)
5. 1×12 Tweed	28. 2×12 Gold 30 Far-Field (JM)	51. 4×12 Uber V30 (RW)
6. 1×12 Black	29. 2×12 G12-65 Far-Field (JM)	52. 4×12 Uber T75+V30 (RW)
7. 1×12 Blue	30. 2×12 Boutique	53. 4×12 Citrus V30 (RW)
8. 1×12 EVL (RW)	31. 2×12 Doubleshow (JM)	54. 4×12 Mill 12K (OH)
9. 1×12 Studio	32. 4×10 Tweed Bass	55. 4×12 SLM Blue (OH)
10. 1×12 EMI Open Back (JM)	33. 4×10 Bassguy P10 (RW)	56. 4×12 SLM H65 (OH)
11. 1×12 Boogafunk Blue (OH)	34. 4×12 Basketweave G12H30 (RW)	57. 4×12 SLM H75 (OH)
12. 1×12 Boogafunk E12L (OH)	35. 4×12 Basketweave G12L (RW)	58. 4×12 SLM M75 (OH)
13. 1×12 Tweed Blue (RW)	36. 4×12 Basketweave G12M20 (RW)	59. 4×12 SLM V30 (OH)
14. 1×12 Tweed Deluxe (RW)	37. 4×12 Basketweave G12M25 (RW)	60. 4×12 20W
15. 1×12 Brit Blue (RW)	38. 4×12 1960A G12M (RW)	61. 4×12 25W
16. 1×12 Brit G12H30 (RW)	39. 4×12 1960B T75 (RW)	62. 4×12 V30
17. 1×15 Blues	40. 4×12 1960B K120 (RW)	63. 4×12 German
18. 1×15 Thunderbolt	41. 4×12 1960B V30 (RW)	64. 4×12 30W (Ultra)
19. 2×12 Black	42. 4×12 Hi-Power (RW)	65. 4×12 Cali
20. 2×12 Brit	43. 4×12 Recto V30 (RW)	66. 1×15 L.A. Bass
21. 2×12 Doubleverb D120 (RW)	44. 4×12 Recto V30 (OH)	67. 4×10 Aluminum Bass
22. 2×12 Doubleverb C12N (RW)	45. 4×12 Solo V12 (RW)	68. 8×10 SV Bass (RW)
23. 2×12 Blue	46. 4×12 Solo S12X (RW)	

When the **MODE** of the CAB block is set to “STEREO”, independent **CAB L** and **CAB R** type parameters appear.

### User Cabs

The Axe-Fx II allows you to load up to 50 cabinet IR files of your own choosing. A variety of 3<sup>rd</sup>-party libraries exist, and it is possible (though not a trivial endeavor) to create your own. For step-by-step instructions on loading User Cab IRs, see section **16.13** of the Appendix on p. **166**. You can also create your own User Cab IRs, using a built in utility. See **IR Capture** on p. **143** for details.

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 the file is loaded into the Axe-Fx II (using Axe-Edit or a 3rd-party SysEx utility), but not after.

**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. When the **MODE** of the CAB block is set to STEREO, **SPKR SIZE** is not offered.

**MIC (TYPE)** – Selects the microphone simulation type used. There are eight different types based on classic guitar cabinet microphones. When the **MODE** of the CAB block is set to STEREO, independent **MIC L** and **MIC R** parameters appear.

*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®)

It is also possible to set the MIC TYPE to “NONE.” When the **MODE** of the CAB block is set to “STEREO”, independent **MIC L** and **MIC R** type parameters appear.

**PROXIMITY** – Available only when the Cab **MODE** is set to “STEREO”, simulates the classic proximity effect, causing an increase in bass or low frequency response as proximity is increased.

**LEVEL L, LEVEL R** – Allows independent control over left and right output channels. These parameters appear only when the Cab **MODE** is set to “STEREO”.

**PAN L, PAN R** – Allows independent control over left and right output pan. Adjust this control to obtain the desired amount of stereo separation. These parameters appear only when the Cab **MODE** is set to 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.

**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.

**LOW-CUT/HI-CUT** – Adjusts the cutoff points of first order high-pass and low-pass filters. Increase the low-cut if the sound is too “bassy” or “boomy.” Decrease the high-cut frequency for a darker cab tone.

**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.

### Cab Mix Parameters

The **Cab** block **MIX** page also has **LEVEL, BALANCE,** and **BYPASS MODE** parameters.

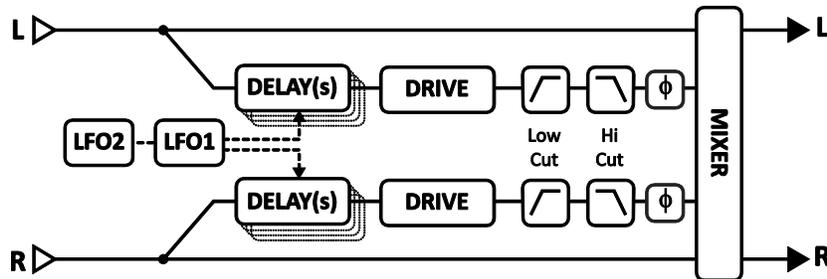
See **Common Mix Parameters** on p.117 for more information.

## 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



### Chorus X/Y Channel Switching

Each instance of the **Chorus** 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 current **Bypass State** and any **Modifier** assignments). See **X/Y Switching** on 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, and 8-VOICE STEREO. Rate, Depth, Level, Balance, Bypass Mode, and Global Mix are not affected.

**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 used 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 1.0–20.0 ms. Lower values create a more unified sound while higher values approach the feel of double tracking.

**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. Some might say “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 is the most commonly used waveform, but the trapezoid waveform is also of interest as it was used in a classic “dimension” chorus processor. Set the LFO phase to 90 degrees to duplicate the sound of this particular unit.

*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 16.5 on p.157 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 *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%).

The Chorus block also has a **MIX** page with **LEVEL**, **BALANCE**, and **BYPASS MODE** parameters. See **Common Mix Parameters** on p.117 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).

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

### Pedal-Type and Common Compressor Parameters

**TYPE** – The Axe-Fx II contains two different compressor types: STUDIO and PEDAL. The STUDIO type simulates the behavior of popular high-end “Feed Forward” stereo studio compressors. The PEDAL type simulates classic stompbox “Feedback” compressors.

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

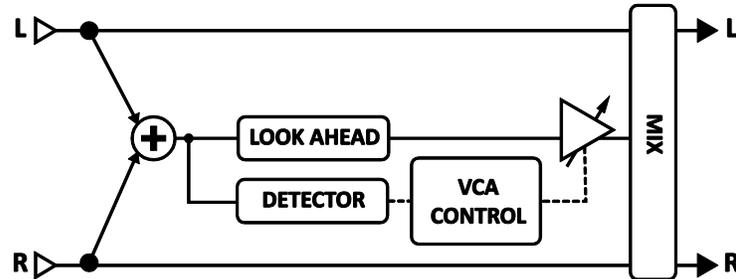
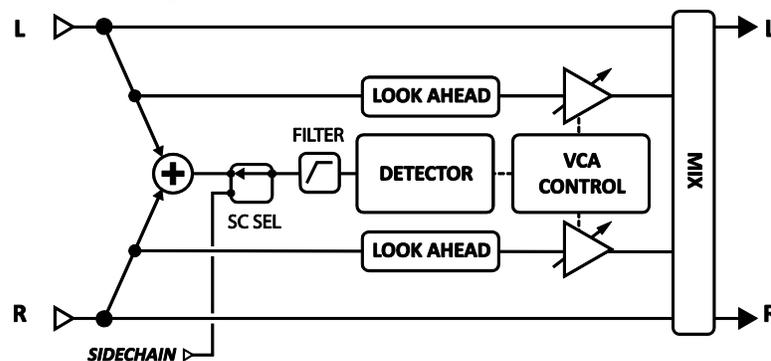


Figure 5-7 – The Compressor Block's "Studio" Type



**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.

**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”), PEAK, or RMS + PEAK detection. RMS detection is smoother 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 the best attributes of both: the speed of a peak detector and the smoothness of an RMS detector.

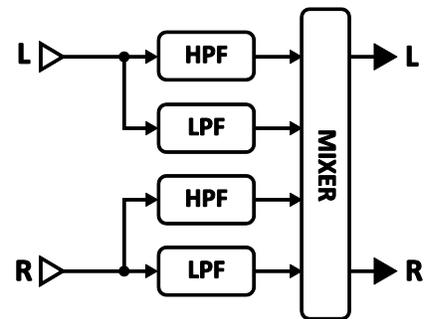
**FILTER** – Sets the frequency of a high-pass filter on the input of the compressor’s detector stage. Raising the filter frequency can help prevent low frequencies from “pumping” the entire mix. Does NOT filter the output.

**SCSEL** – 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, the main inputs, or the isolated left or right channel of the compressor’s input signal.

## 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.117 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. 79), the one **Megatap Delay** block (p. 76), and the new **Looper** block (p. 74).

### Delay X/Y Channel Switching

Each instance of the Delay 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 current **Bypass State** and any **Modifier** assignments). See X/Y Switching on 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 purpose and parameters of each configuration are listed below.

**Table 1: Delay Block Types and Configurations**

TYPE		Configuration
<b>Digital Mono</b>	Full-range, a pristine modern delay (Default Mono).	<b>MONO DELAY</b>
<b>Analog Mono</b>	Frequency response and character of an analog delay.	
<b>Mono Tape</b>	Frequency response and character of a tape delay.	
<b>Vintage Digital</b>	Uses bit-depth reduction for a lo-fi vibe.	
<b>2290 w/ Mod</b>	Based on a former industry-standard unit.	
<b>Digital Stereo</b>	Full-range, a pristine modern delay (Default Stereo).	<b>STEREO DELAY</b>
<b>Analog Stereo</b>	Frequency response and character of an analog delay.	
<b>Stereo Tape</b>	Frequency response and character of a tape delay.	
<b>Ambient Stereo</b>	Ultra-wide echoes.	
<b>Ducking Delay</b>	“Ducking” automatically lowers delay volume when you play harder, resulting in a less “cluttered” mix.	
<b>Dual Delay</b>	A default setting for the dual delay.	<b>DUAL DELAY</b>
<b>Ping-Pong Delay</b>	A default setting for the ping-pong delay.	<b>PING PONG</b>
<b>Sweep Delay</b>	A default setting for the sweep delay.	<b>SWEEP</b>
<b>Reverse Delay</b>	A default setting for the Reverse delay.	<b>REVERSE</b>

**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%.

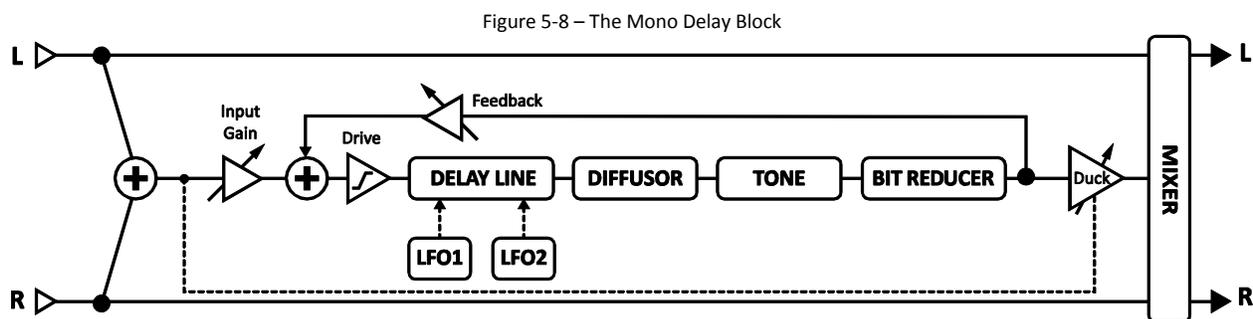
**MSTR FDBK** – Master Feedback scales any and all feedback parameters on PG2 of the Delay. Note that the range of this control is 0–200%, making it possible (easy, in fact) to “overload” the feedback loop.

**MIX** – This is a copy of the **LEVEL** 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.



**TIME** – Sets the time of the delay line (time between repeats). 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.

**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.

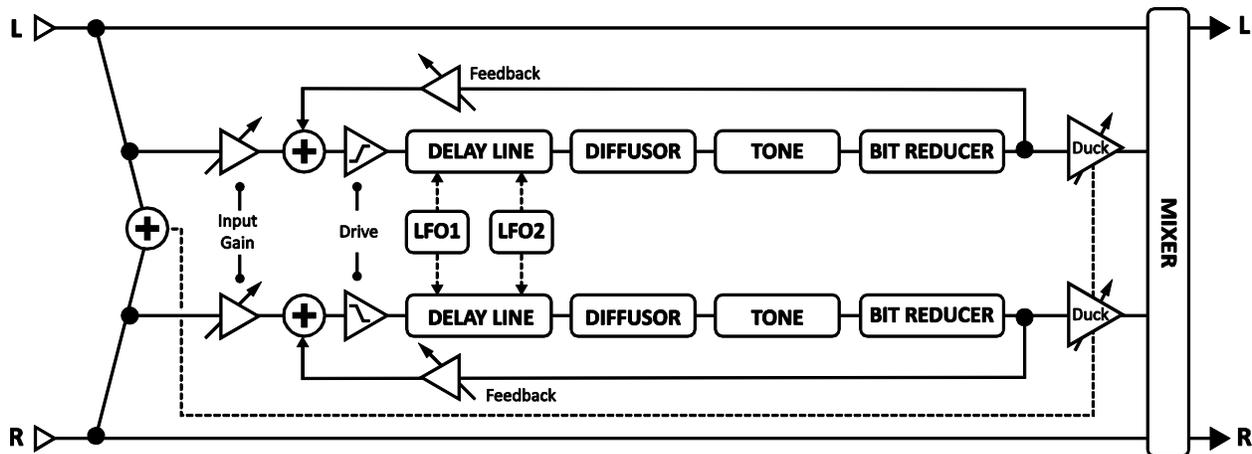
**BIT REDUCTION** – This control makes it possible to create the lo-fi sounds of vintage digital delays. The number shown is the number of bits to be *subtracted* from 24-bit full scale. To create a 16-bit delay, for example, set **BIT REDUCTION** to “8” (24 – 8 = 16). Bit Reduction is often used with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

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 (time between repeats). 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.

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

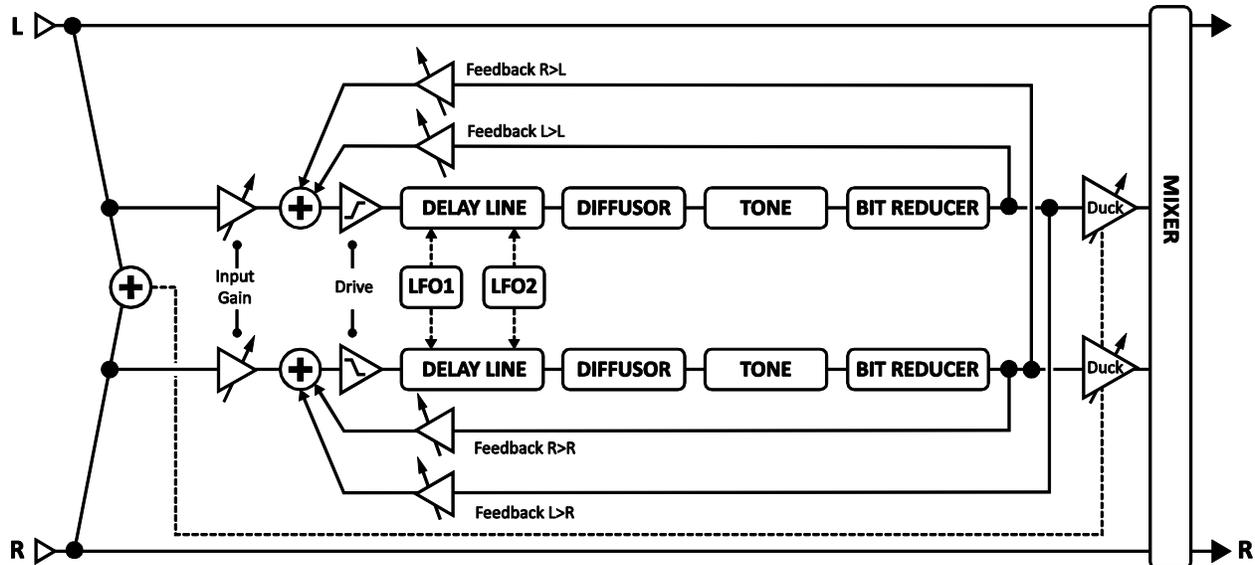
**DRIVE** – Sets the amount of distortion in the delay path.

**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.

**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.57) 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.

**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 Mono 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%).

### 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 16.5 on p. 157 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.*

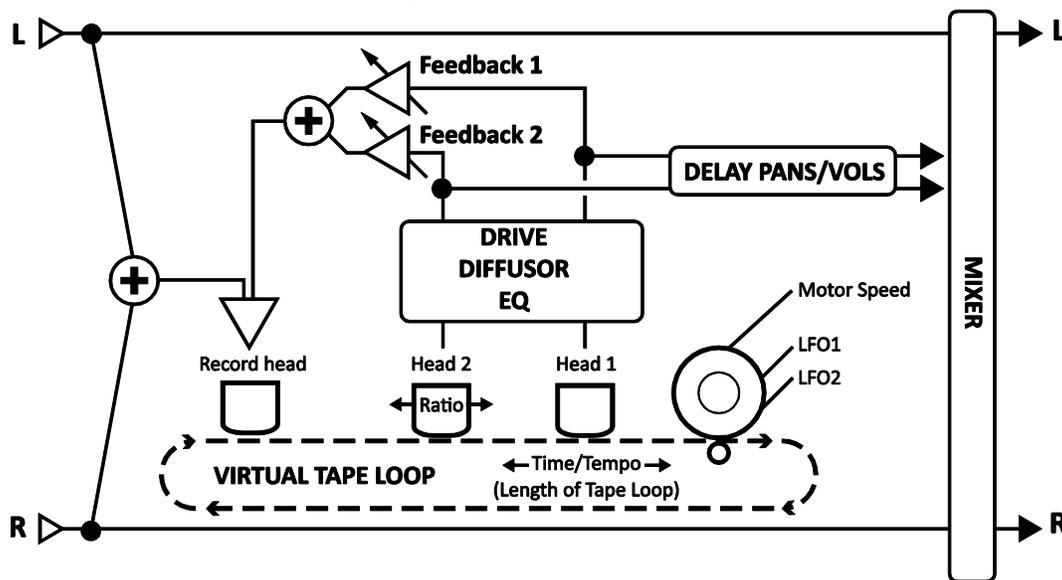
**TEMPO** – Locks the TIME parameter in rhythmic relation to the global tempo. See the **TEMPO** section under the **MONO DELAY** type (above) for more about the relationship between BPM and delay time in milliseconds.

**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 “warp” delayed material.

*NOTE: The effect of MOTOR SPEED is cumulative with that of the onboard LFOs (see Delay Common Parameters on p. 61). 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 length of the virtual tape loop 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.

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

## 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 16.5 on p. 157 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 16.5 on p. 157 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 THRSOLD** – 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 contains some cool parameters that didn't fit anywhere else.

**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 a powerful equalizer inside the feedback loop.

**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.117 for more information.

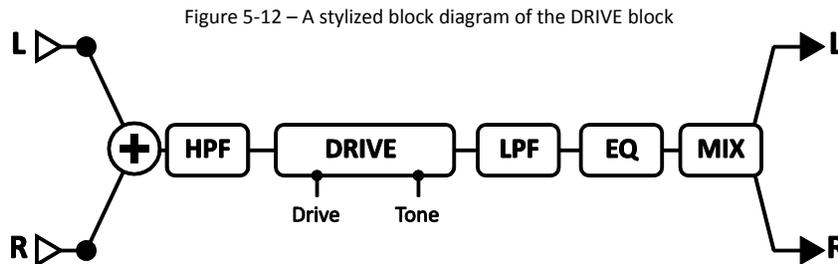
### A Word on “Spillover”

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. 164.

## 5.7 Drive [DRV]

The **Drive** block replicates 22 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.



### Drive X/Y Channel Switching

Each instance of the Drive 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 current **Bypass State** and any **Modifier** assignments). See section 4.4 on p. 36 for more about X/Y switching.

### 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
<b>Rat Dist</b>	Based on the ProCo™ Rat Distortion.
<b>PI Fuzz</b>	Based on the Big Muff® Pi Fuzz.
<b>Tube OD</b>	Based on the Chandler™ Tube Driver that actually contained a 12AX7.
<b>Super OD</b>	Based on the Boss™ Super Overdrive.
<b>Treb Boost</b>	Based on a classic Treble Booster.
<b>Mid Boost</b>	A custom mid-boost overdrive.
<b>T808 OD</b>	Based on the Ibanez™ TS-808® Tube Screamer.
<b>Fat Rat</b>	A modified version of the Rat Dist. A bit fuller and smoother.
<b>T808 MOD</b>	Captures the most popular 808 mods.
<b>Octave Dist</b>	An octave distortion based on the Tycobrahe® Octavia®.
<b>Plus Dist</b>	Based on the MXR™ Distortion Plus.
<b>Hard Fuzz</b>	A hard-clipping, 60s-style fuzz.
<b>FET Boost</b>	A gentle, smooth clipping booster with tone controls.
<b>Tape Dist</b>	Simulates the clipping of an overdriven reel-to-reel tape deck.
<b>Full OD</b>	Based on the Fulltone™ Fulldrive OD Pedal.
<b>Blues OD</b>	Based on the Marshall™ Bluesbreaker®.
<b>Shred Dist</b>	Based on the Marshall™ Shredmaster®.
<b>M-Zone Dist</b>	Simulates the Boss™ Metalzone™, popular for extreme gain settings.

<b>Bender Fuzz</b>	Based on the classic Tonebender circuit.
<b>BB Pre</b>	Based on the Xotic® Pedals BB Preamp®.
<b>Face Fuzz</b>	Based on Dallas Arbiter Fuzz Face®.
<b>Master Fuzz</b>	Based on the Maestro Fuzztone, aka Satisfaction fuzz.
<b>Bit Crusher</b>	Based on a black box we found lying in the trash outside <i>Studio Harshclip</i> .

**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 to occur.*

**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 for the distortion generation.

**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.” Bit Reduction is often used in conjunction with audio filtering like that which is possible using the **HI CUT**, **LO CUT**, and **EQ** parameters of the drive block. Tip: It is *supposed* to sound nasty!

**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.117.

**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].

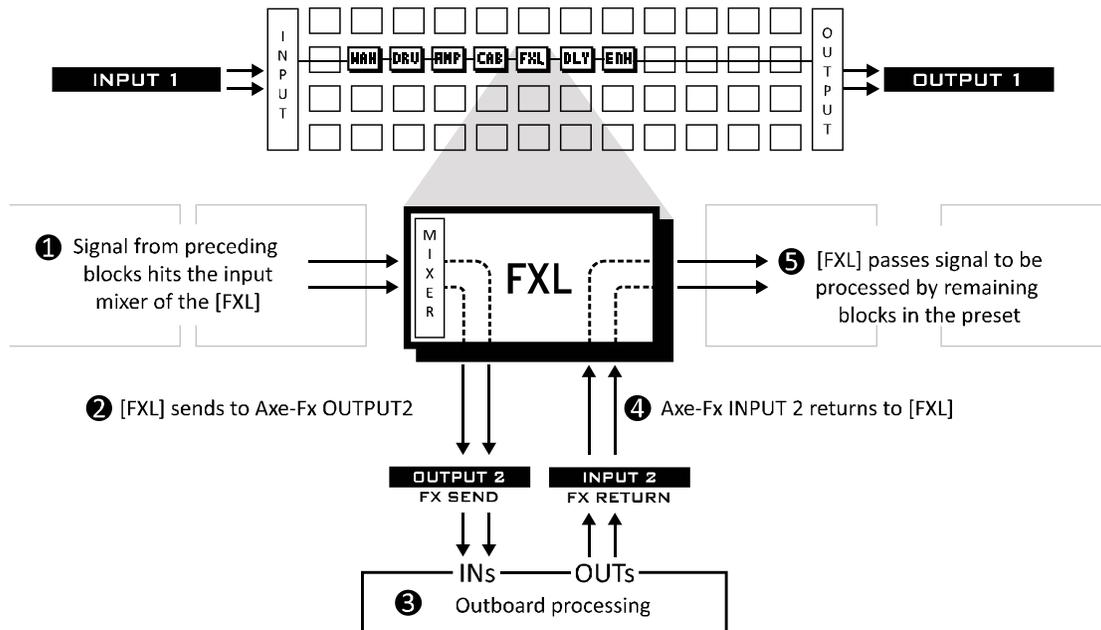


Figure 5-13 – Effects Loop Routing

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) onstage 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 enter the grid at any point. Instruments with more than 2 outputs (rare but not unheard of) can thus be utilized.

The **FX Loop** block has a **MIX** page with **LEVEL**, **BALANCE**, and **BYPASS MODE** parameters. See **Common Mix Parameters** on p. 117 for more information.

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

## 5.9 Enhancer [ENH]

The **Enhancer** increases apparent stereo width through multiband frequency based separation of left and right channels.

**WIDTH** – Sets the spread

**INVERT** – Depth sets the amount of separation

**LOW CUT, HI CUT** –

**LEVEL** – Sets the output level

The Enhancer has no mix parameters or bypass mode modifier switch.

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

The **Feedback Send** and **Feedback Return** blocks allow you to route sound from any point in the preset routing to any other point, bending the rule that signal may only flow from left to right. No connector 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 of a “feedback loop,” almost invariably involving some type of delay. The MIX of the delay in the loop is normally set to “100%,” since direct signal recirculation results in instability and undesirable phase issues.



**Warning:** Use caution with the Feedback blocks, as you can easily program an unstable loop and cause internal clipping and/or very high sound levels, which may damage your hearing. 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 instability, return the LEVEL control to minimum and analyze your routing for possible causes of instability.

See **Using Send and Return** on p. 165 for creative application ideas.

Each Axe-Fx II preset can use one **FB Send** and one **FB Return** block.

The **Feedback Send** block has **SEND LEVEL** and **OUTPUT LEVEL** controls. The latter controls the amount of signal that passes through the block.

The **Feedback Return** block has **MIX**, **LEVEL**, **BALANCE**, and **BYPASS MODE** parameters. See **Common Mix Parameters** on p. 117 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.

## EFFECTS GUIDE

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** – Sets the type of filter. All the standard filter types are available. A NULL type has a flat frequency-response characteristic and can be selected if you want to use the block as a simple gain element.

**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.

**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.117 for more information.

## 5.12 Flanger [FLG]

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). A great example of tape flanging can be heard in the bridge of the *Doobie Brother's* song

"Listen to the Music." 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.

### Flanger X/Y Channel Switching

Each instance of the Flanger 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 current **Bypass State** and any **Modifier** assignments). See X/Y Switching on 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 nominal delay time of the delay line. This changes the character of the effect. Low values give a phaser-like sound whereas high values are more 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.

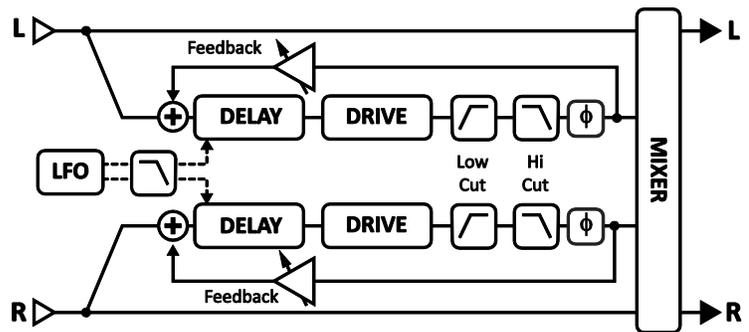


Figure 5-14 - The Flanger Block

**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 16.5 on p.157 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.

**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.

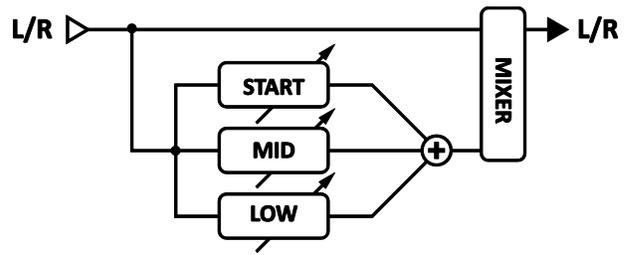
### Flanger Mix Parameters

The **Flanger** block also has a **MIX** page with **LEVEL**, **BALANCE**, and **BYPASS MODE** parameters. See **Common Mix Parameters** on p.117 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 the fuss and muss.



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.

Vowel	Example
AAA	ba <u>t</u>
EEE	ma <u>t</u> e
III	fe <u>t</u>
OHH	to <u>e</u>
OOO	so <u>o</u> n
EHH	be <u>t</u>
AHH	po <u>t</u>
AWW	fa <u>u</u> lt
UHH	gu <u>n</u>
ERR	gi <u>r</u> l

### Formant Mix Parameters

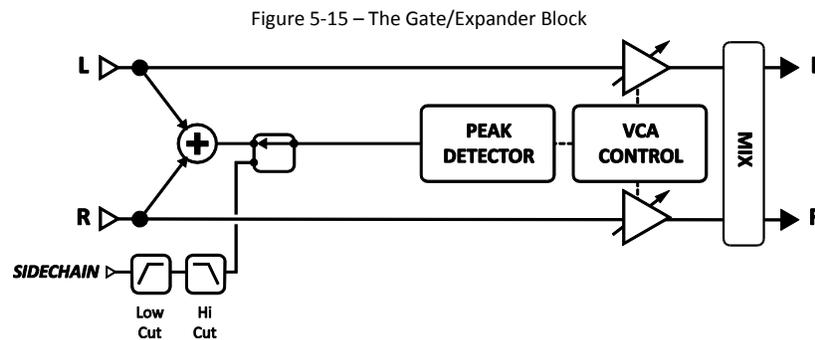
The **Formant** block has a **MIX** page with **MIX**, **LEVEL**, **BALANCE**, **BYPASS MODE**, and **GLOBAL MIX** parameters.

See **Common Mix Parameters** on p.117 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**.

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



**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 which goes to 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.117.

## 5.15 Graphic Equalizer [GEQ]

The **Graphic Equalizer** is a 10-band equalizer with band centers at 31, 63, 125, 250, 500, 1000, 2000, 4000, 8000, and 16,000 Hz. Each band can boost or cut up to 12 dB. Simply select the desired band with the **NAV** buttons and use **VALUE** to vary the gain.

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

The AMP block (p.39) has a built-in, 8-band 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. 134), 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.117.

## 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. Looper CC# assignments appear on the CONTROL page of the I/O menu (p. 139).

### 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.

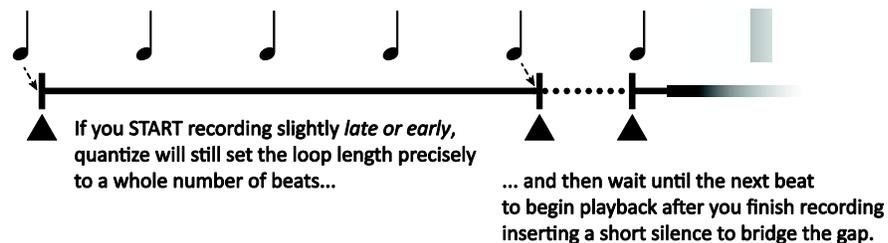
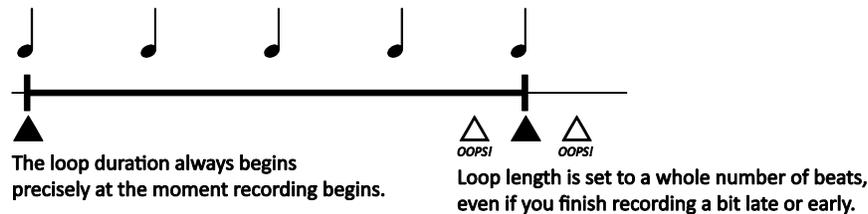
**MONO** – Recording and playback are in mono. Maximum loop length is 60 seconds. Undo is not possible.

**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 60 seconds. Undo works.

**QUANT** – With quantize enabled, loop length is perfectly set to some whole number of beats) with reference to the pulses of the Axe-Fx GLOBAL TEMPO (see p. 146)).



**THRSH** – The Axe-Fx Looper can start recording automatically when the input exceeds a set level. Turn this feature ON or OFF using this parameter.

**THRSH LEVEL** – This sets the level which must be exceeded at the input of the Looper for recording to start automatically.

**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.

### Looper Mix Parameters

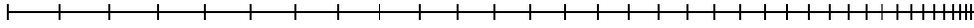
The **Looper** has a **MIX** page with **LEVEL**, **BALANCE**, and **BYPASS MODE** parameters.

See **Common Mix Parameters** on p. 117 for more information.



**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. 117 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 one stereo mix or one mono mix. The mixer lets you fine-tune level blends or use **modifiers** (p. 124) to crossfade between different effects or chains. Each pair of gain and balance controls corresponds to a row in the grid. For a more thorough description of how Axe-Fx II mixers work, see section 16.8, *Mixology*, on p. 160.

### 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 fully independent **Mixer** blocks.

## 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.

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** – Determines whether the selected band will use RMS (“Root Mean Square”), PEAK, or RMS + PEAK detection. RMS detection is smoother and is generally used to even out levels over a long period of time. Peak detection, commonly used with guitar, is useful for fast limiting. RMS + Peak combines the best attributes of both: the speed of a peak detector and the smoothness of an RMS detector.

**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.

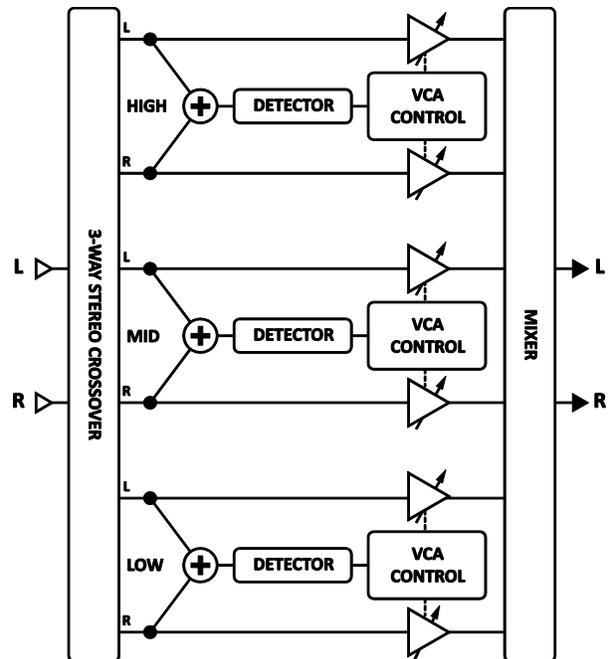


Figure 5-17 – Stylized view of the Multiband Compressor

## 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.

### 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.

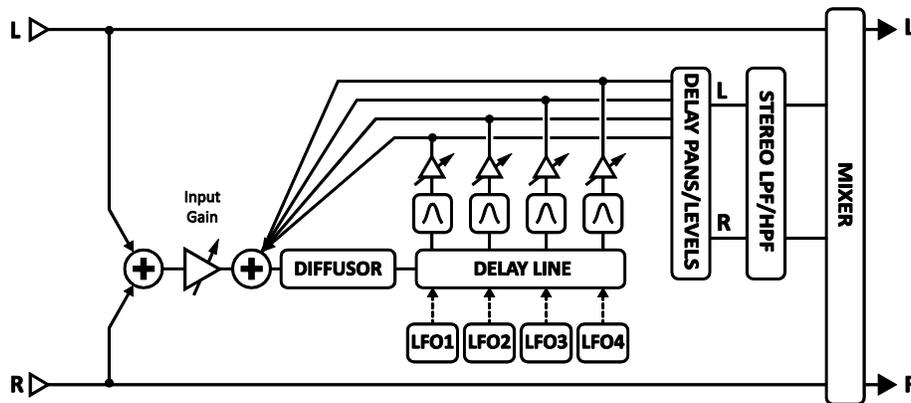
### Mix Parameters

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.117 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** – Sets the level of the selected tap in the overall feedback blend. Note that the sum of the four feedback values cannot exceed 100%.

**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 diffusor 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 THRESHLD – (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.

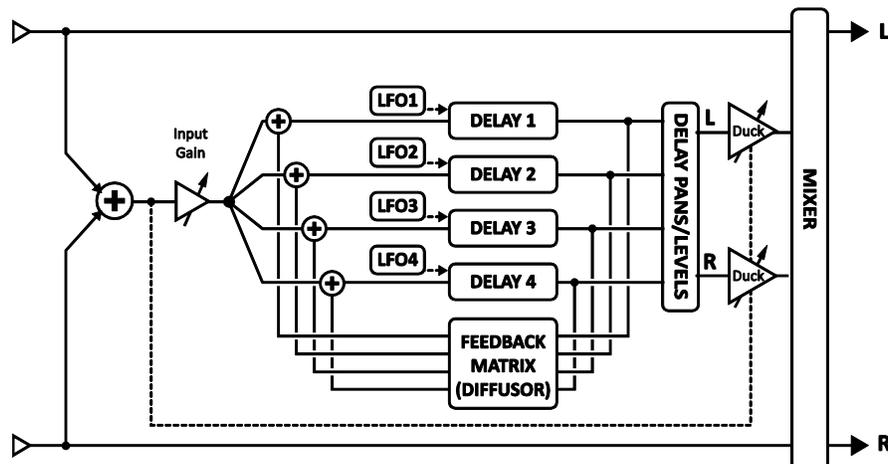
**LFO 1,2,3,4 TEMPO** – Synchronizes the rate of the LFO for the selected tap in relation to the global tempo. For example, if the tempo is set to “1/4” and the global tempo is 120 BPM, the LFO rate will automatically be set to 2 Hz ( $\text{BPM}/60 = \text{Hz}$ ). To ignore the global tempo, set these 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

## 5.20.2 Plex Delay

In terms of delay effects, a multiplexor, 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.

**FEEDBACK 1,2,3,4** – Sets the level of the selected tap in the overall feedback blend. Note that the sum of the four feedback values cannot exceed 100%.

**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.

**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 greatly the volume can be reduced by the ducker. 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 THRESHLD** – (**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.

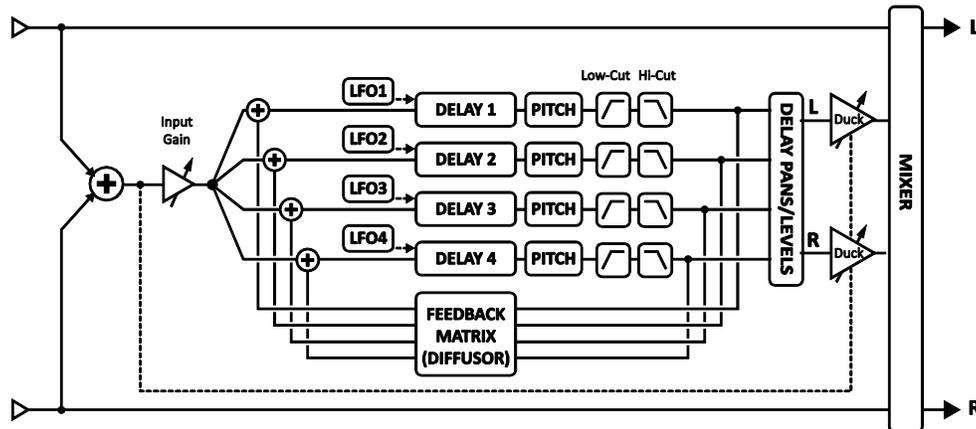
**LFO 1 TEMPO** – Synchronizes the rate of the LFO in relation to the global tempo.

**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 (5.20.2 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:

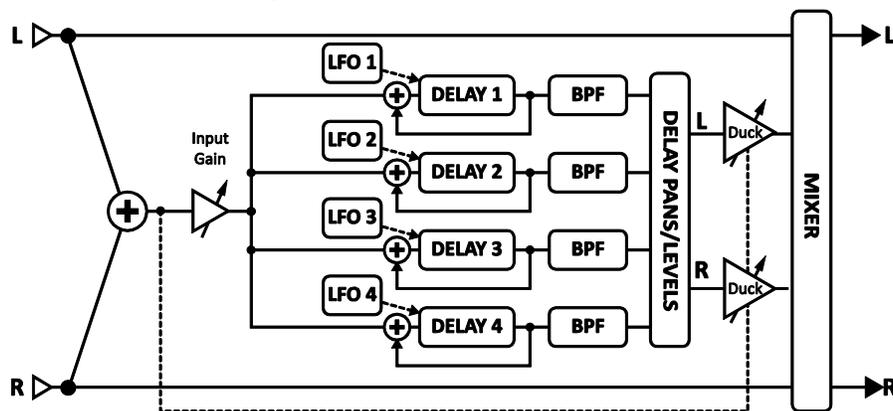
**DIRECTION** – This determines whether small granules of audio in the pitch shifter are played back forward or reversed. To understand how this works, imagine a word whose individual letters have been mirror-imaged but are still in the correct left-to-right order (“Axə-Ɔx”). In this case, the letters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

Aside from the fact that its four delay lines are parallel to one another, and that it lacks a diffusor, the **Band Delay** Multi Delay type is identical to the Quad Tap (5.20.1, above).

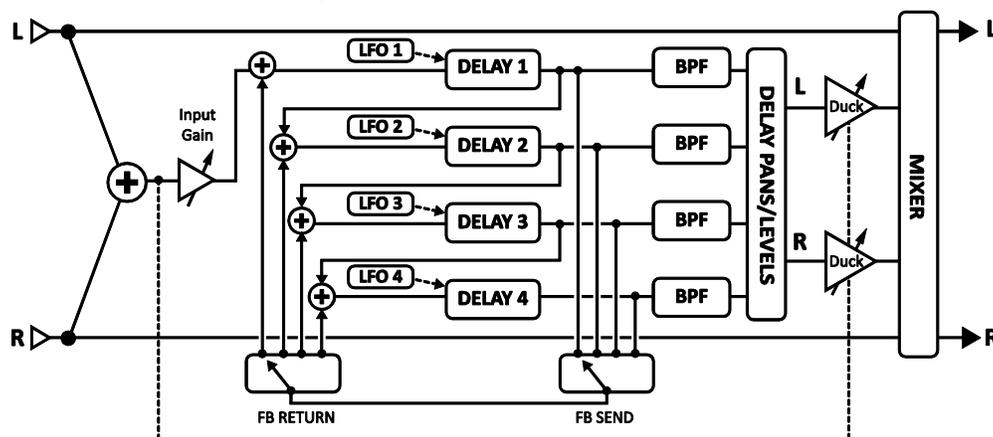
Figure 5-21 – The Band Delay Multi Delay Type



### 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.

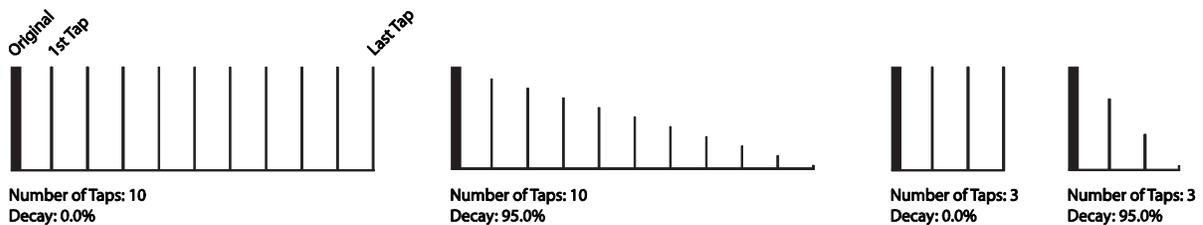
**MONO/STEREO** – Sets the mode of the Ten-Tap Delay. In mono mode, twice as much delay per tap is possible.

**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 full-left and the right channel full-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 (5.20.7 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.

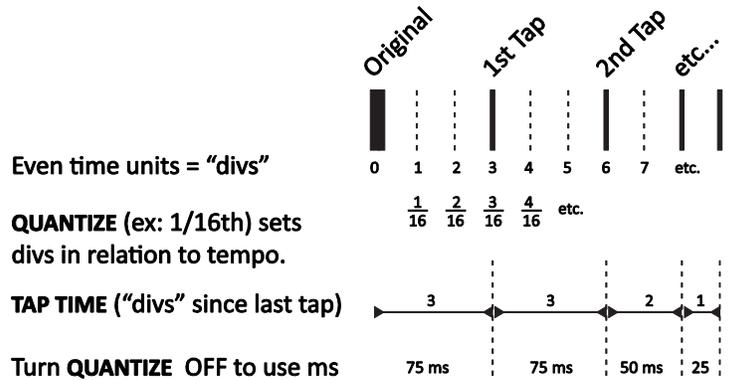


Figure 5-23 – The concept of Rhythm Tap "divs"

**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<sup>th</sup> note, and all tap times will sound some whole number of 16<sup>ths</sup> 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 **diffusor** 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.61.

## 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.

Each Axe-Fx II preset can use two fully independent **Tremolo/Panner** blocks.

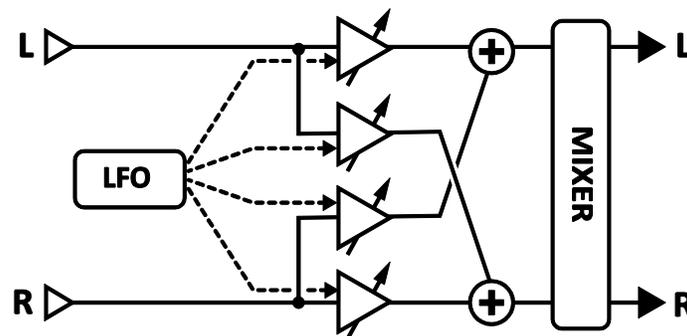


Figure 5-24 – The Pan/Tremolo Block

### Parameters

**EFF 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 16.5 on p. 157 for more information on LFO waveform shapes, duty, and phase.

**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.117 for more information.

## 5.22 Parametric EQ [PEQ]

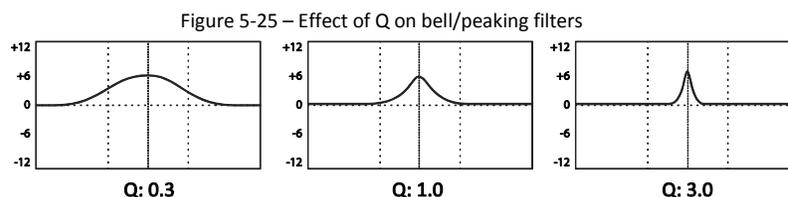
The 5-band **Parametric Equalizer** is one of the most precise and flexible tone-shaping tools in the Axe-Fx II. It lets you select the exact frequencies you want to focus on, adjust how much you want to boost or cut, and specify how the change should affect neighboring frequencies. The bands include 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.

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:



Q exerts a different effect for BLOCKING or SHELIVING 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.
- **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.

### 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.117 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.

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

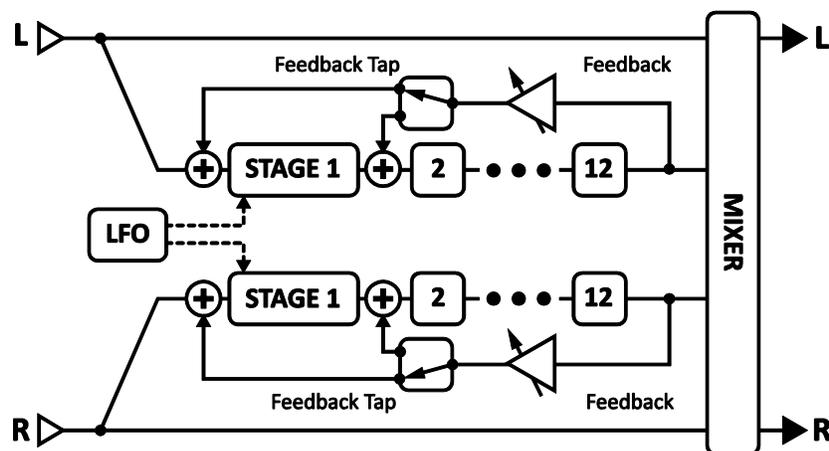


Figure 5-26 – The Phaser Block

## **Phaser X/Y Channel Switching**

Each instance of the Phaser 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 current **Bypass State** and any **Modifier** assignments). See X/Y Switching on p. 36 for more information.

### **Basic Parameters**

**TYPE** – This control instantly sets other phaser parameters for different useful sound settings. Types include: DIGITAL MONO, DIGITAL STEREO, SCRIPT 45, SCRIPT 90, BLOCK 90, CLASSIC VIBE, and STEREO 8-STAGE.

**RATE** – Sets the frequency of the LFO that controls the “sweep.” Set fully counterclockwise to sync to Global LFO1 (p. 129). 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.

**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.

**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.

### **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 16.5 on p.157 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 the “SCRIPT 90” type requires 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. 117 for more details on these.

## 5.24 Pitch Shifter [PIT]

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 different 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 32-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.

### Pitch Shifter X/Y Channel Switching

Each instance of the Pitch 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 current **Bypass State** and any **Modifier** assignments). See X/Y Switching on 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 MONO** – In this mode the pitch information is detected and analyzed at the input of the pitch block. The “MONO” detector is optimized for speed and accuracy when playing single notes.
- **LOCAL POLY** – As with “LOCAL MONO,” pitch information is detected and analyzed at the input of the pitch block. In this mode however, the detector is optimized for polyphonic sources. The pitch tracking is necessarily slower but is more stable when used with chords. Complex chord types can still cause instability; simple combinations of two or three notes work best.

Both of the local modes allow the detector to track the pitch of a delay or effect tail even after you’ve stopped playing. They also include 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** – When set to “ON,” the Pitch Shifter will track the pitch of the incoming note and adjust internal pitch shifting techniques accordingly for the best performance at any moment. Setting **PITCH TRACK** to OFF ignores the pitch data and uses fixed shifting techniques. With pitch tracking OFF, the sound can waver or flutter depending upon the note(s) played. With pitch tracking ON, the sound is typically smoother, but complex chords may cause pitch instabilities.

**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.

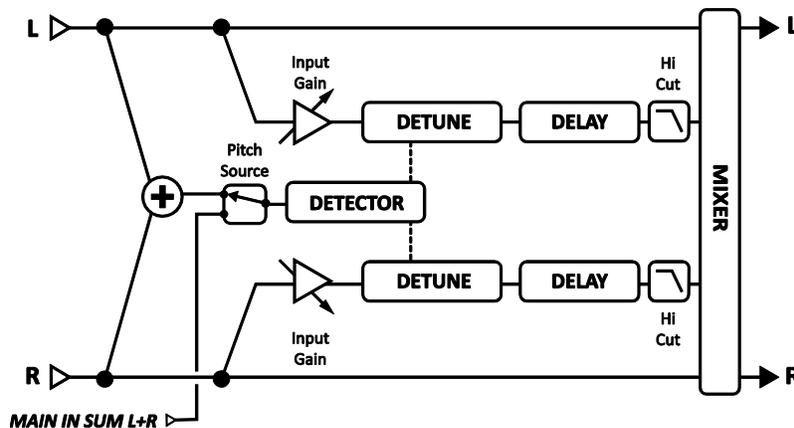
### Pitch Shifter Mix Parameters

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. 117 for more information on these controls.

#### 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.

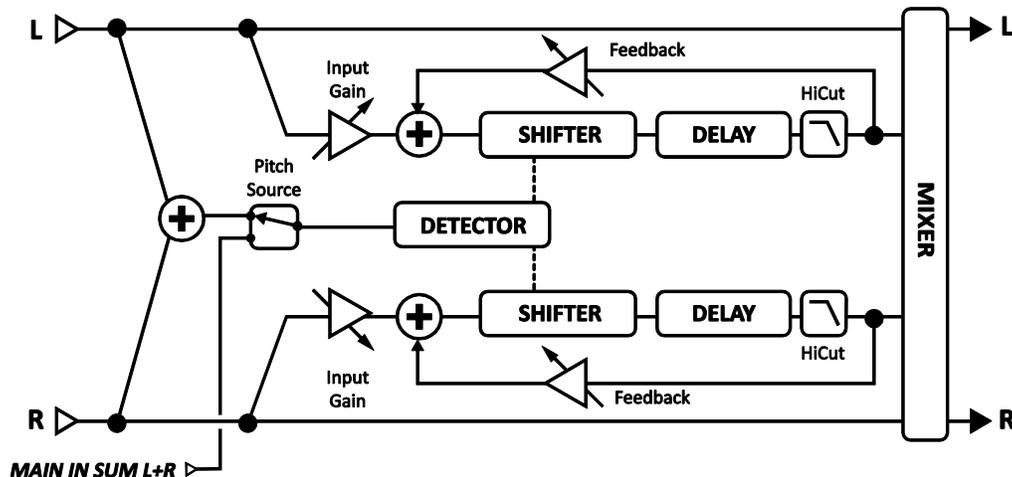
**VOICE 1 PAN, VOICE 2 PAN** – Sets the panning of the selected voice.

**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.”

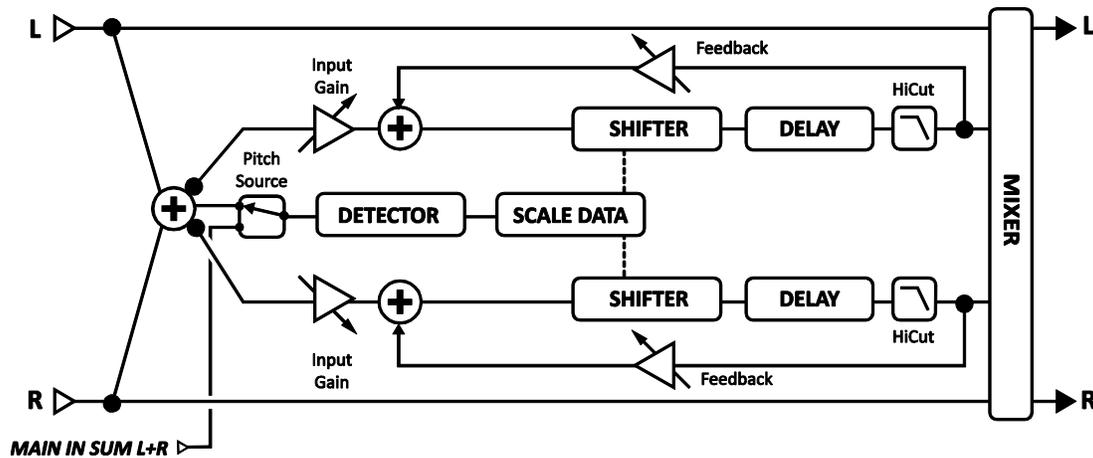
**VOICE1 FEEDBACK, VOICE 2 FEEDBACK** – Sets how much of the output signal is returned to the input for “shift-upon-shift” effects. Using both feedback and delay (above) results in cool-sounding cascades.

### 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	
BLUES SCALE	1	b3	4	b5	5	b7		
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. 124), typically assigned to some source controlled by a pedal. In comparison to the Advanced Whammy (p. 99) 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

**CONTROL** – Adjusts the output pitch. Attach to a controller for dynamic control of the pitch. See the Wah pedal tutorial on p. 162 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.

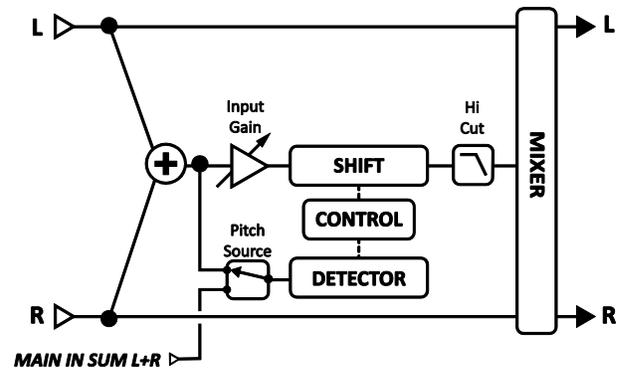


Figure 5-30 - The Classic Whammy Pitch Shifter Type

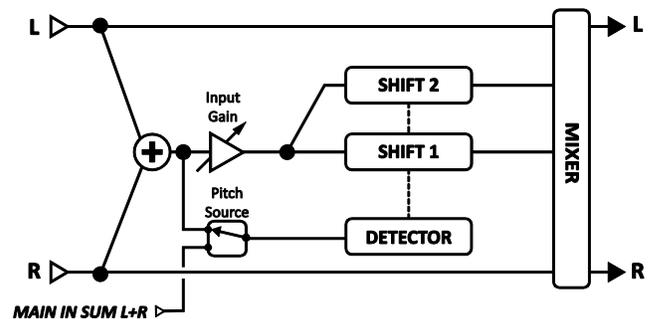
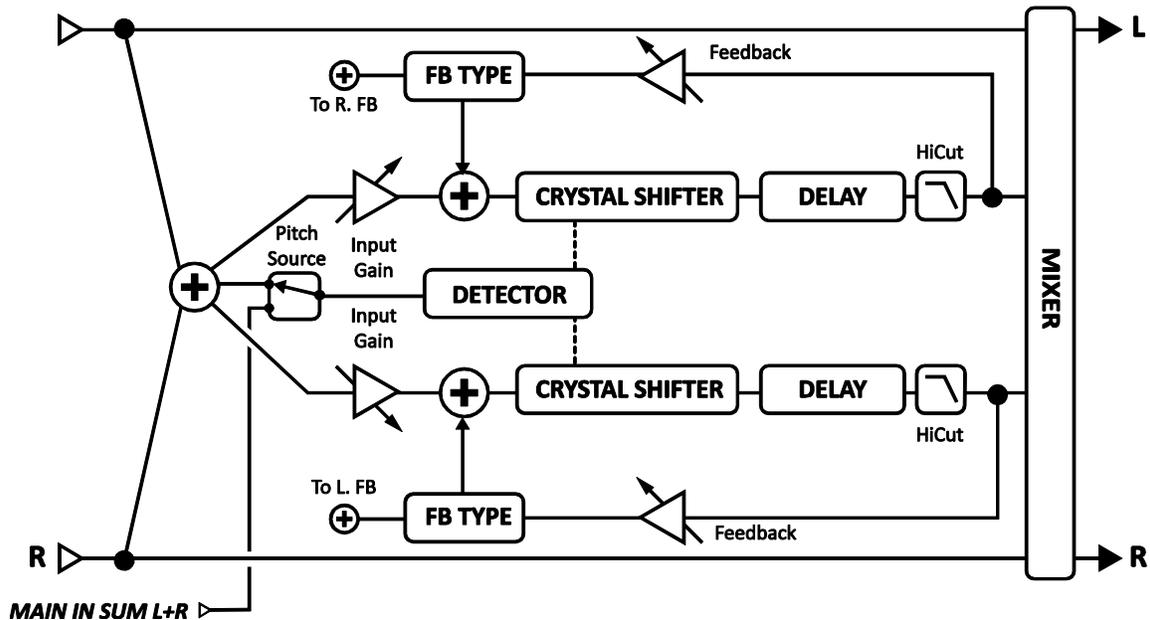


Figure 5-31 - The Octave Divider Pitch Shifter Type

## 5.24.6 Crystals

The **Crystals** Pitch Shifter is similar to the **Fixed Harmony** mode (p. 94) 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 (“Axə-Ɔx”). 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 and tiny 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. 97) 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.

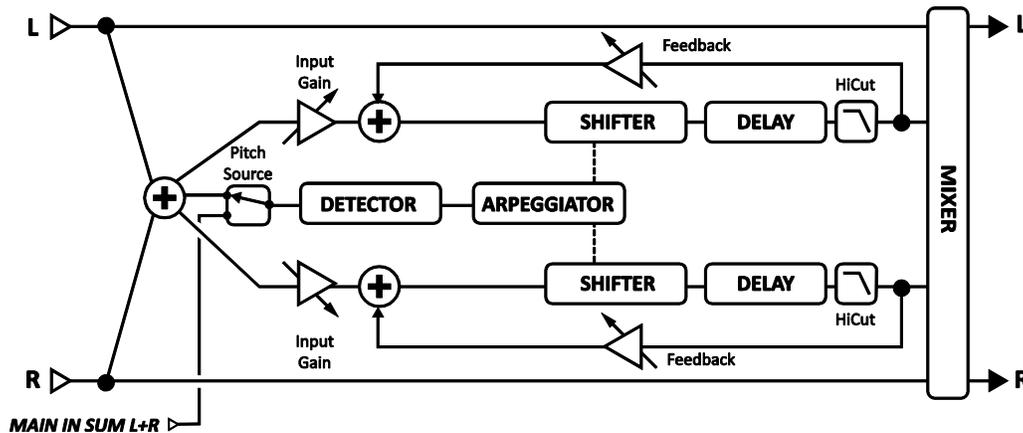
**STOP** – Sets the stop pitch shift amount in semitones. This is the amount of pitch shift applied when the CTRL knob is in the maximum position.

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. 94) 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. 131) 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. 76) 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. 94) except that it uses the custom scales tables stored in the global memory. See section 8.3 on p. 134 for more information on setting up global scales.

**VOICE 1 SCALE, VOICE 2 SCALE** – Selects the custom scale to use for each voice.

## 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.

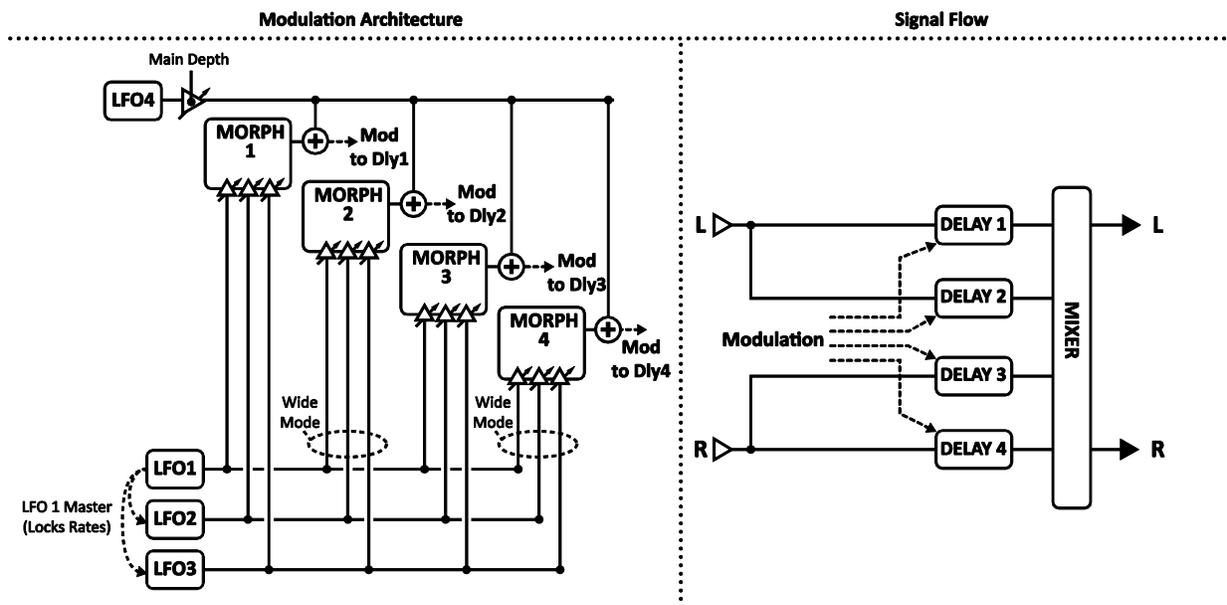


Figure 5-35 – The Quad Chorus Block

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 depth of LFO4 for 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 is 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. 117 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.

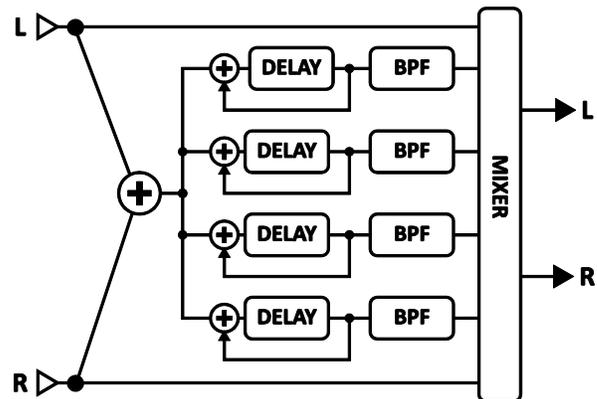


Figure 5-37 — The Resonator Block, Shown in its MONO INPUT MODE.

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

### 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.

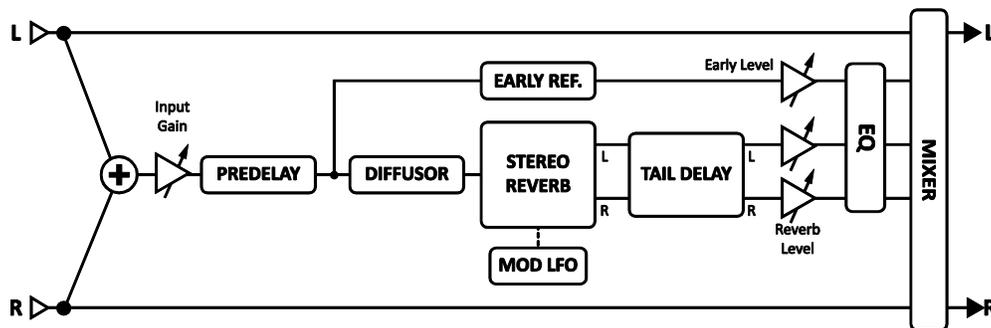
See **Common Mix Parameters** on p. 117 for more details on these.

## 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



### Reverb X/Y Channel Switching

Each instance of the Reverb 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 current **Bypass State** and any **Modifier** assignments). See X/Y Switching on p. 36 for more information.

### Basic Parameters

**TYPE** – Selects the reverb type. There are seven basic types and several variations. Changing the **TYPE** will set values for other reverb parameters as well, but these can then all be freely modified manually.

Type	Notes
<b>Room</b>	Simulates an actual room. This is the type to use when you want the most natural, realistic reverb. Also great on vocals and percussion.
<b>Hall</b>	Similar to the Room reverb but simulates a concert hall. It is a little less smooth and has some response peaks, lending it a unique character. Use this when you want your sound to stand out a little.

<b>Chamber</b>	Simulates the sound of large, boxy chamber. Useful when you want a bright, resonant reverb sound.
<b>Plate</b>	Simulates the sound of a vintage reverb plate characterized by smooth yet bright sound that is especially popular for vocals. The Axe-Fx II Plate simulation captures the "shiny" sound without all the complicated setup.
<b>Cathedral</b>	Surround your tone with heavenly reverb in this incredible simulation of a grand space.
<b>Spring</b>	Simulates the classic electro-mechanical reverb effect that has traditionally been placed inside guitar combo amps as well as standalone reverberators.
<b>Cavern</b>	Carlsbad has nothing on this monster. A vast, cavernous space.

**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).

**INPUT GAIN** – Sets input level into the reverb. Primarily for use with a pedal or other control to set the “Send” to the reverb (allowing reverb to ring out even after the pedal is pulled back). In other situations this should normally be set to “100%.”

**MIX** and **LEVEL** are duplicated from the MIX page.

### Advanced Parameters

**TYPE, TIME** – Duplicated from the BASIC page (p.105).

**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 is increased, the reverb will become somewhat darker as high frequencies are absorbed.

**COLOR** – Adjusts the high-frequency absorption for a darker or brighter reverb tail.

**INPUT 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.

**DIFFUSION TIME** – Controls the length of the input diffusor. Lower values simulate a small diffusion space. Higher values simulate a large space.

**WALL DIFFUSION** – Controls how quickly the reverb tail’s density builds. Lower values cause discrete echoes to be heard for a longer time. Higher values cause the echo density to build rapidly.

**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. The overall smoothness of the reverb tail is a function of this parameter along with the **DIFFUSION** and **SIZE** parameters. Large **SIZE** values will make the individual echoes more apparent, as will lower values of **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. Adjust to taste based on the program material.

**PRE DELAY** – Adds extra delay before the reverb starts. The **SIZE** control 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.

**TAIL DELAY** – This delays the reverb tail independently of the reverb’s “early reflections.” It can simulate the sound of spaces in which reflections from surfaces at different distances reach the listener at different times.

**EARLY LEVEL** – Adjusts the relative volume level of the early reflections. This control has no effect for the Spring Reverb type.

**REVERB LEVEL** – Adjusts the relative volume level of the reverb tail.

**MOD DEPTH** – This parameter controls the depth of modulation of the reverb tail. For a more musical sound, experiment with this and the **MOD RATE** control. The reverb contains an LFO that modulates the delay lines that make up the reverb. This modulation helps fill out the soundstage and makes the reverb sound fuller. The modulation is best used on pitched instruments. For non-pitched instruments like drums, it may sound better with the modulation off. Set the depth to zero to defeat the modulation. To hear the effect of the modulation, turn the mix to max and adjust the depth and rate. Then set the mix to the desired amount.

**MOD RATE** – Controls the rate of modulation of the reverb tail.

**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.”

### 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 has a **MIX** page with **MIX, LEVEL, BALANCE, BYPASS MODE,** and **GLOBAL MIX** parameters.

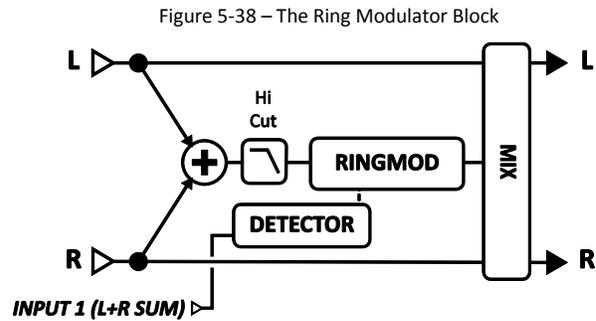
See **Common Mix Parameters** on p. 117 for more details on these.

### 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. 164.

## 5.28 Ring Modulator [RNG]

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 be dynamically controlled by the pitch of the input.



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. 117 for more details on these.

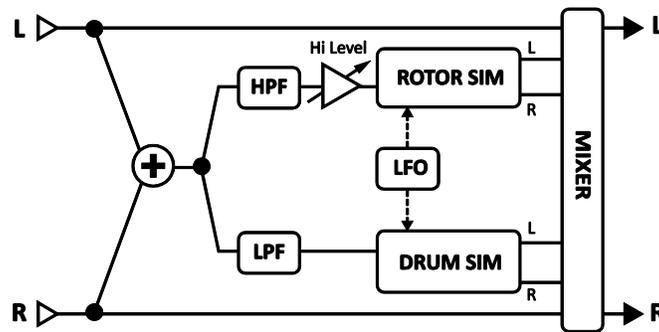
## 5.29 Rotary Speaker [ROT]

A Hammond B3 Organ without a Leslie cabinet is like a BLT without the lettuce and tomato. Guitar players also discovered the wonderful, spinning, 3D sound of the Leslie and its brethren. The classic unit contains a spinning drum with a slot in it, 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 Leslie sound is ubiquitous. Drum-only versions have been produced by other manufacturers; Stevie Ray Vaughan's "Cold Shot" offers an example of the sound of this variant.

The Axe-Fx II **Rotary Speaker** simulator reproduces all these classic sounds and offers more control. Also, it doesn't weigh 300 lbs. or require four guys to move it up a flight of stairs. Isn't technology great?

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

Figure 5-39 – The Rotary Speaker Block



## 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.

**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.

**MIC WIDTH** – Sets the placement of the (neutral-sounding) virtual mics, determining the stereo width of the effect. Because the spinning is cool in stereo, you may be tempted to run full-wide, but believe it or not, the great Leslie sounds are often recorded in mono!

The **Rotary Speaker** has **MIX**, **LEVEL**, **BALANCE**, and **BYPASS MODE** controls. See **Common Mix Parameters** on p. 117.

## 5.30 Synth [SYN]

The Axe-Fx II **Synth** block contains a three-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.

### Parameters

The synth has three voices, each with the following parameters:

**TYPE** – Sets the waveform to Sine, Triangle, Square, Sawtooth, Random, White Noise, or Pink Noise.

**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 up to +/- 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. 117 for more details on these.

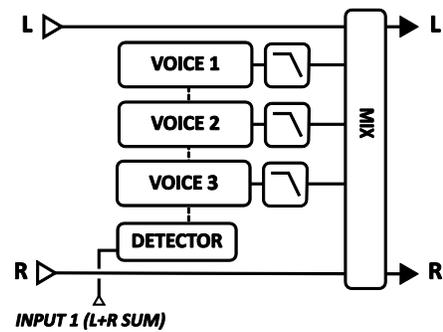
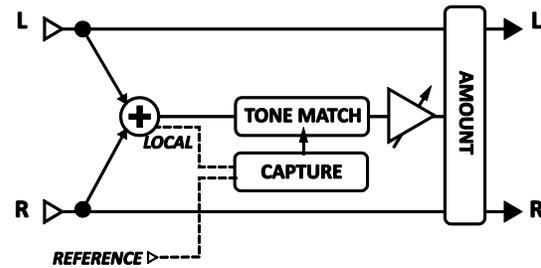


Figure 5-39 — The Synth Block

## 5.31 Tone Match [TMA]

The **Tone Match** 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).

The Tone matching process is covered by a separate mini-manual available [on our web site](#). Here is a summary:



1. **CREATE** a starting point preset on the Axe-Fx II, dialing it in as close as possible to the desired tone. Don't forget to include the Tone Match block, positioned where it can “hear” the designated reference signal and the output of your starting preset.
2. **CONFIGURE** the Tone Match block input options and connect or route the reference signal to the Axe-Fx.
3. **CAPTURE** the reference and local signals (using X and Y buttons) and watch as frequency plots are drawn in the display.
4. **COMPARE** the reference and local signals (pressing ENTER) to produce the final Tone Match.
5. **COMPLETE** the process by saving the preset. Tone Match data is stored inside the preset, which can be copied, exported or shared without needing to attach anything extra.

Optionally, Tone Match Data can be exported for use as an Axe-Fx II **User Cab IR** (see the “User Cabs” subsection of 5.2 on p 48 for more.)

**REF SOURCE** – This specifies where the Tone Match block will pick up the Reference Signal.

**REF CHAN, LOCAL CHAN** – The tone Match block can analyze and process only in mono. These parameters determines whether left, right, or a sum of left and right channels will be monitored by the tone match block for the “reference” and “local” signals.

**AVG TIME** – Determines how long of a time window is used during signal capture. Setting this to max invokes PEAK HOLD mode, which uses the maximum value observed for any frequency instead of averaging.

**AMOUNT** – Scales the Tone Match between full and flat, determining the magnitude of its effect.

**SMOOTHING** – This reduces the prevalence of peaks and valleys in the IR to ease a “notchy” sound.

### Tone Match Mix Parameters

The **PROCESS** page also has **LEVEL**, **BALANCE**, and **BYPASS MODE** parameters. See **Common Mix Parameters** on p. 117 for more information.

The Global Blocks feature is not available for the Tone Match block.

## 5.32 Vocoder [VOC]

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 Vocoder. 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.

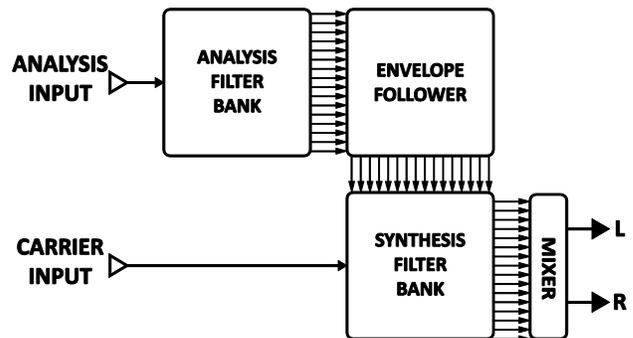


Figure 5-40 — The Vocoder Block

**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. 117 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. 162).

**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, 15A, and 10A 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. 117 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 *smoooooth* feel.

A wah is actually a very straightforward device. The signal is passed through a resonant filter whose frequency is controlled by a pedal. The Wahwah may be placed before distortion for a subtler, classic sound or after distortion for a more prominent and aggressive sound.

The wah is stereo-in/stereo-out.

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

### Wahwah X/Y Channel Switching

Each instance of the WAH 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 current **Bypass State** and any **Modifier** assignments). See X/Y Switching on p. 36 for more information.

### Parameters

**TYPE** – Selects between a low-pass, bandpass, or peaking filter. A bandpass filter gives a more dramatic sound whereas the other filters are more subtle.

**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.

**DRIVE** – This simulates overdriving the circuit of the wah pedal.

**TRACK** – Sets the "Q" tracking of the filter. As the frequency is increased, the resonance will be decreased by an amount proportional to this value. If this is zero, the resonance of the filter will be constant at all frequencies. Classic wah pedals usually have a resonance that decreases with frequency due to design limitations. This control can be used to mimic those pedals.

**CONTROL** – Sets the position of the wah. Normally you would use a MODIFIER to assign this parameter for real-time control, but simply dialing it in will give you a great "parked" wah sound.

## 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 turn to the INPUT/GTE page.

The Noise Gate is always active but can be defeated by turning the THRESH control fully counterclockwise.

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

**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.

**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.

### 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.”

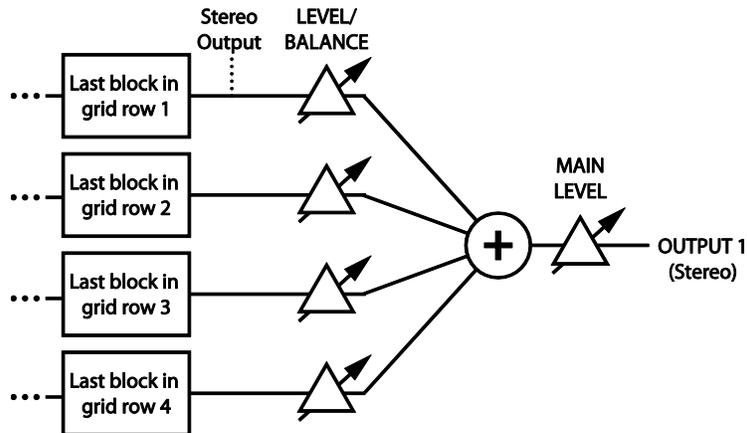
Normally you will want to leave this on AUTO, but you may also select any of the following values manually. This setting is saved with the preset.

- 1MΩ+ 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, located on the OUTPUT tab of the **LAYOUT** menu. 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.

Figure 5-41 – The Output Mixer



**LEVEL 1–4** – Think of these as input faders. Each is connected to one row of the grid as shown above.

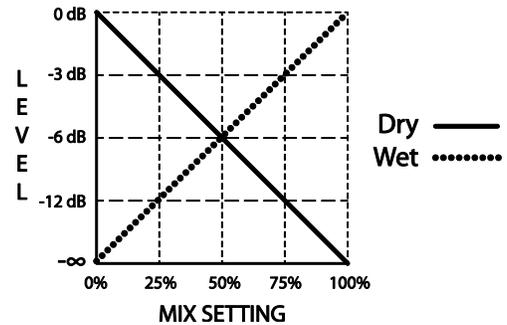
**BAL 1–4** – Each of these is connected to one row of the grid as shown above and determines the left-right balance for the incoming signals. See **BALANCE** under **Common Mix Parameters** (below) for more information.

**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.

## 5.37 Common Mix Parameters

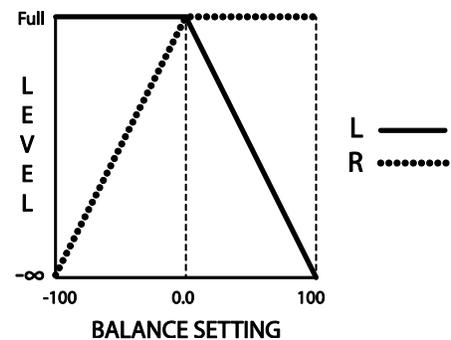
Almost every block in the Axe-Fx II has a **MIX** page with parameters to determine how the output of that block contributes to the overall preset signal. On some blocks, these controls appear on other pages. Take a moment to familiarize yourself with these important controls and the differences between 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.

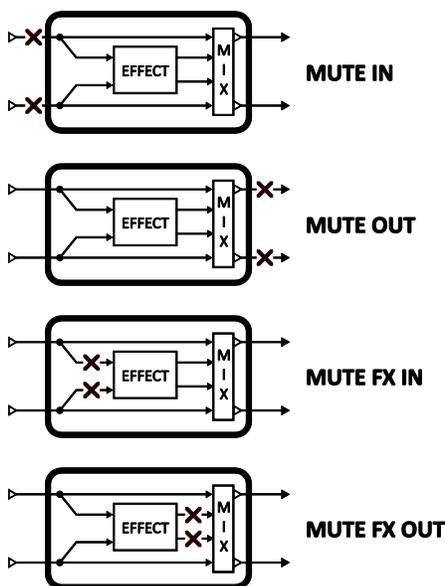


**LEVEL** – As you would expect, the **LEVEL** controls set the output level of a block. Almost all **LEVEL** controls have a range from -80.00 to +20.00 dB. Exceptions include the Compressor and Filter (+/-20.00 dB) and the Drive, which ranges from 0-10 dB.

**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. 133).

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:

<b>Delay</b>	<b>Formant</b>	<b>Pitch</b>	<b>Ring Mod</b>
<b>Chorus</b>	<b>Megatap Delay</b>	<b>Quad Chorus</b>	<b>Rotary</b>
<b>Feedback Return</b>	<b>Multi Delay</b>	<b>Resonator</b>	<b>Synth</b>
<b>Flanger</b>	<b>Phaser</b>	<b>Reverb</b>	<b>Vocoder</b>

# 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 of the 60+ block type instances on the Axe-Fx II can become global blocks, with 10 global memories for each of them. And, should you choose to REMOVE a link between a block and its global counterpart, this leaves both the global and the normal 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 Match, which is not supported). 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.”

<b>AMP 1</b>	<b>AMP 2</b>	<b>CAB 1</b>	<b>CAB 2</b>	<b>CHORUS 1</b>	<b>CHORUS 2</b>
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 #3
Global AMP 1 #4	Global AMP 2 #4	Global CAB 1 #4	Global CAB 2 #4	Global CHORUS 1 #4	Global CHORUS 2 #4
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	
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	

**AND SO ON...**

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 block *instance* you are using in a preset. Instance refers not only to the type, but to the possible block *number*, as well. 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.

## MODIFIERS & CONTROLLERS

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 (p. 148).

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.

#### SAVE/LOAD GLOBAL BLOCK

```
Global #
VAL  1      CABINET 1
SAVE TO & LINK WITH GLOBAL
LOAD FROM & LINK WITH GLOBAL
LOAD FROM GLOBAL (NO LINK)
UNLINK
ENTER: EXECUTE  EXIT: CANCEL
```

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, 4x12 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”).

```
CABINET 1:G1  PG1 PG2/MIX
MONO HIRES
VAL
MODE ROOM LEVEL ROOM SIZE MIC SPACING LOW CUT HI CUT
```

- ▶ **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.

### 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.

**IMPORTANT:** Presets shared between users will either need to be first stripped of their Global Block links or delivered with an accompanying “System” dump containing the Global Block settings. Otherwise, the “local globals” on the new system will overrun the settings of any linked blocks when the preset is first loaded. A future version of Axe-Edit should allow “one-stop” global link removal.

### 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.

**REMEMBER!** *Global links are not exported with individual presets, and presets in a bank will link to the local global settings stored in the System area of the Axe-Fx II. To back up presets with Global Blocks, you must also back up the System.*

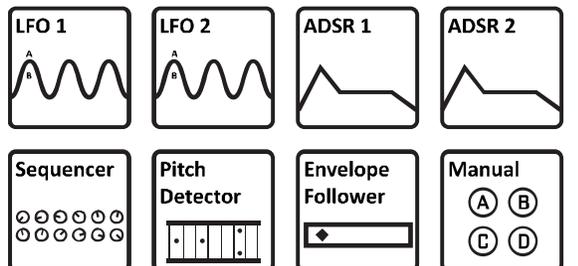
# 7 Modifiers & Controllers

## 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.

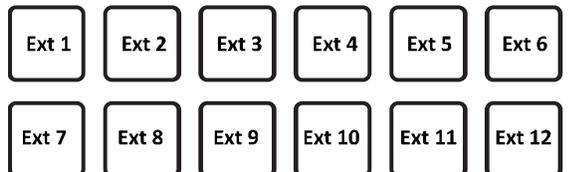
Figure 7-1



### 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, and 4 Manual knobs. 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, or a **MIDI CC#** (0–127) in the CTRL page of the I/O menu (p. 139). These assignments are global, but you can use external controllers for different things in different presets.

## 7.2 Creating a Modifier



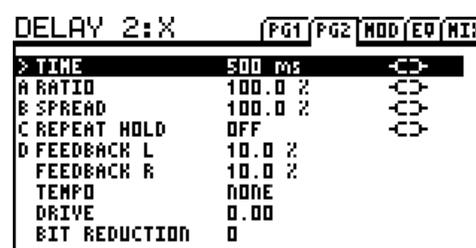
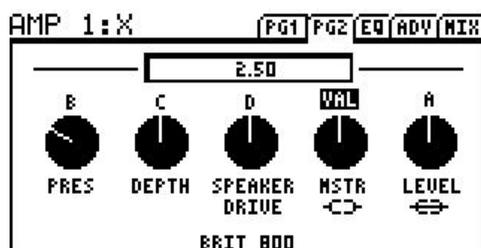
OPEN



FILLED

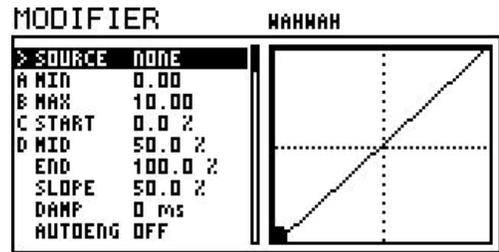
The process of creating a modifier begins at the parameter you want to control. Parameters that can be controlled are marked with a special symbol (left top). Look for it beneath a “soft” knob or to the right of a text parameter. If a modifier is already present, the symbol will have a line through it (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 (**←C→**) 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.



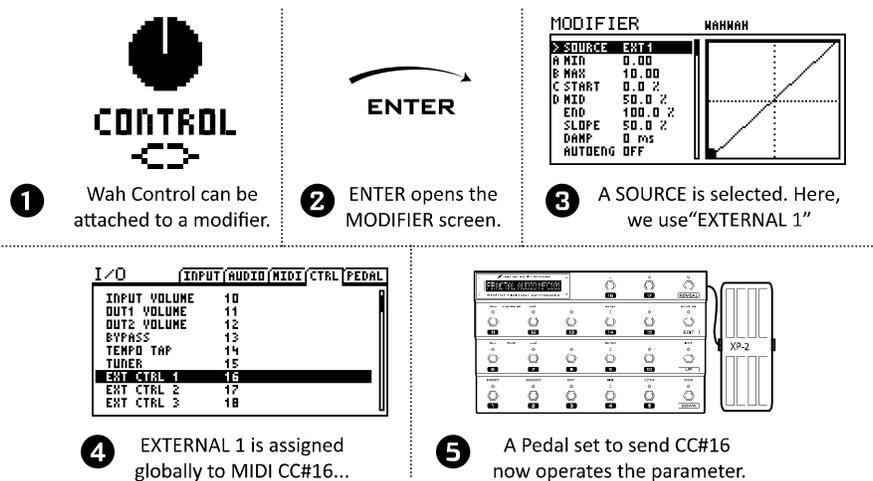
**To Remove a Modifier...**

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.

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).

## MODIFIERS & CONTROLLERS

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.

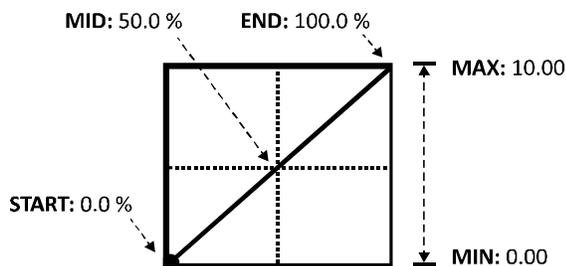
The **START**, **MID**, **END**, and **SLOPE** settings are used to re-map the ways in which parameters respond to changes in the source.

The **MIN** and **MAX** parameters “peg” the 0-100% scale of a modifier to actual parameter values. **MIN** and **MAX** are set using the same units as the parameter being modified. 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.

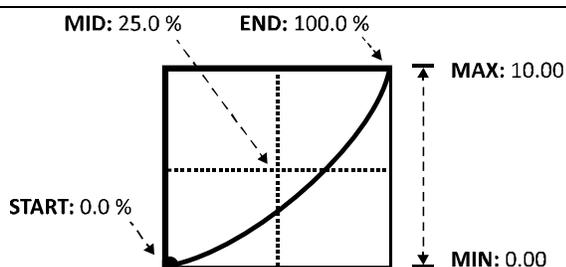
The following examples illustrate how **START**, **MID**, **END**, **SLOPE**, **MIN**, and **MAX** work together.

#### Example 1: Creating a Custom Curve

For the first example, let’s imagine a **VOLUME** parameter being controlled by a pedal (via EXT1/MIDI CC#16).

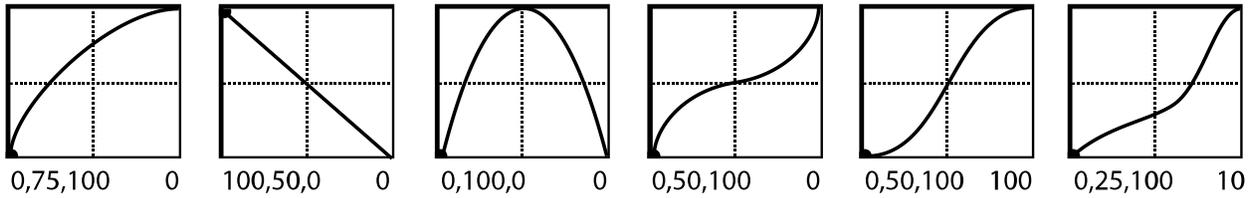


The default settings for **START**, **MID**, and **END** create a perfectly linear relationship between the source and the target. As the pedal is depressed, the volume increases in direct proportion. **MIN** and **MAX** are set to their extreme limits, so volume goes from 0.00-10.00 (OFF to FULL). But linear response is generally unsatisfactory for volume control because of the non-linear characteristics of our hearing. Ears aren’t very good at math...



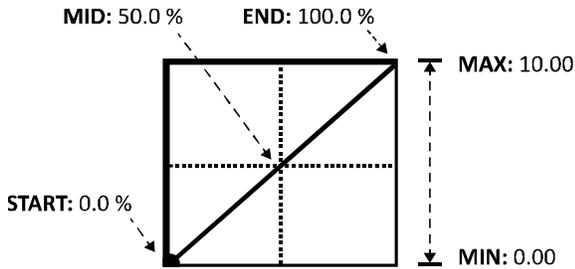
As we lower the value of **MID**, the response starts to take on a more comfortable curve, closer to the classic log or audio taper typically used for volume control. **MIN** and **MAX** are still set to “0.0” and “10.0” so that volume goes from off to full, but the way it *swells* has changed.

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.

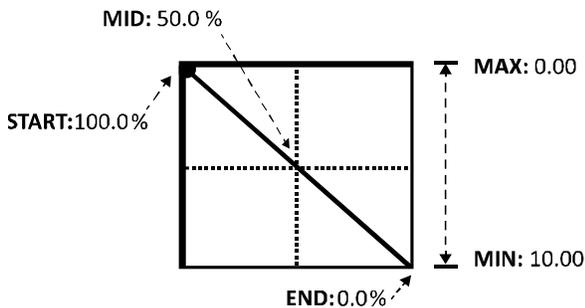


**Example 2: Setting MIN and MAX**

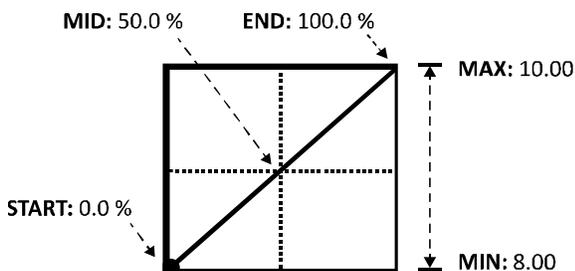
**MIN** and **MAX** allow a modifier’s range to be “pegged” to precise values by setting the y-axis scale of the modifier graph using the same units as the parameter being modified. This ultimately determines the relationship between the source and the target parameter. The following examples illustrate **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 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 their modifier sources. Their values may be re-mapped as we saw above, but changes are still essentially simultaneous; every little tremble of your foot is

## MODIFIERS & CONTROLLERS

reflected by a little tremble in the sound. **DAMPING** allows you to increase the hysteresis in this system, adding a little “viscosity” or “elasticity” so changes in the parameter can’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.

### 7.2.3 Auto Engage

Though 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 goes above a certain value. 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. Another example might be a pedal that controls the speed of a Rotary Speaker and then bypasses the effect when pulled all the way back.

To use Auto Engage, set the following two parameters, found toward the bottom of the MODIFIER screen:

**OFF VAL** – Sets the threshold that the value of the current **SOURCE** must cross for auto-engage to occur.

When the value goes above this limit, the effect is engaged. Note that incoming MIDI CC values of 0–127 are scaled to 0–100% for purposes of this determination.

**AUTOENG** – Sets the time that the source must remain above or below the OFF VALUE for the bypass state to toggle. Fast = 100 ms; Med = 333 ms; Slow = 1000 ms. Use slower settings to “loosen” Auto Engage so your effect doesn’t snap off every time your source dips. Set to OFF to disable Auto Engage.

Try it! You’ll quickly find that Auto Engage comfortably eliminates the need for expression pedal “toe” switches.

### 7.2.4 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.

You can either set the stored value before turning **PC RST** to ON or change it afterwards. The parameter box above a modified knob always shows the “manual” value so you can freely set it to save it. (Knob and filter “graph” parameters with modifiers are animated as their sources change, so you won’t see these elements update in the display as you turn the value wheel.) Text-only parameter values remain directly editable for this purpose. When **BYPASS MODE** is assigned a modifier, you cannot change the stored setting with the **FX BYP** button; this control is disabled to ensure that a remote control and the actual bypass state remain in sync.

Internal controllers are not subject to **PC RST** because their values are updated immediately when a preset loads. (This is true even for an LFO or Sequencer that is “stopped” or an ADSR that has not yet been triggered.)

## 7.3 Control Sources

The Axe-Fx II provides 22 different control sources for use in Modifiers. These are detailed below.

### 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. Adjust 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%.

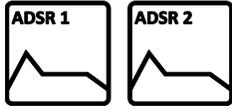
**DUTY** – Varies the duty cycle, or “symmetry,” of the Triangle, Square, and Trapezoid waveforms.

**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. 157 of the Appendix for a guide to LFO Waveforms, Duty, and Phase.

### 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.

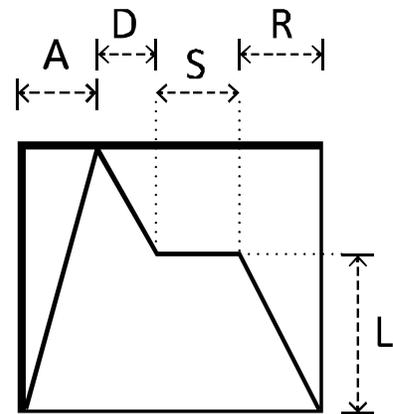
**ATTACK** – The envelope begins at zero and increases to 100% over the duration of the attack time (A).

**DECAY** – When the attack phase completes, the envelope descends for the duration of the decay time (D) until it reaches the sustain level (L).

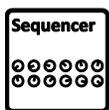
**SUSTAIN, LEVEL** – After the decay, the envelope remains at the sustain level (L) for the duration of the sustain time (S).

**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. 132) 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.

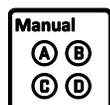
**GAIN** – The gain control works like a classic “sensitivity” control to set the relationship between incoming levels and outgoing control signals. By boosting the input of the envelope follower, GAIN allows a weaker input signal to exert a greater effect.

### 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



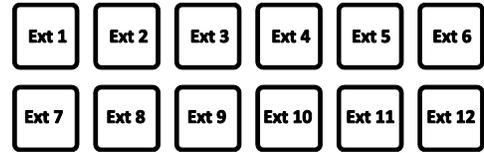
New to the Axe-Fx II are four front panel Quick Control knobs, detailed in section 4.3.1 on p. 36. When the unit is in EDIT mode, they function as hands-on controls to facilitate editing. When the unit is in RECALL mode, however, these knobs double as “MANUAL” control sources that can be used for making sound adjustments without menu diving. Press the front panel **CONTROL** button to find the MANUAL page for the current preset. The value of each knob is saved with each preset.

## MODIFIERS & CONTROLLERS

 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 (return to the grid or press RECALL).

### 7.3.7 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.



I/O	INPUT	AUDIO	MIDI	CTRL	PEDAL
EXT CTRL 1				16	
EXT CTRL 2				17	
EXT CTRL 3				18	
EXT CTRL 4				19	
EXT CTRL 5				20	
EXT CTRL 6				21	
EXT CTRL 7				22	
EXT CTRL 8				23	
EXT CTRL 9				24	

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. 137)—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%.

### 7.3.8 Modifier Power!

With the free space at the end of this section, here are some fun MODIFIER ideas.

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 Sample-and-Hold FILTER by assigning **FREQUENCY** to a RANDOM-type LFO sync’d to tempo.
6. Ducking reverb! Run the effect 100% wet (parallel to the dry) then use ENVELOPE to lower its level.
7. Place a low-pass 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 synth-like vibrato effect for guitar. Modify Global LFO1 **DEPTH** with a pedal (0-100%). Assign LFO1 to the **CONTROL** parameter of an Advanced Whammy (Pitch Block) set to +/-3 semitones.
11. **Bonus!** Reclaim pedalboard real estate with a “Wildcard” footswitch. (Hi Dweezil!) Set it to a CC# and assign this to an External Controller. Then assign this EXT source as needed to the bypass (mode) modifiers of different effects or settings across different presets.

# 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. 148). Default assignments for all system parameters are listed beginning on p. 171.

## 8.1 Configuration Parameters

The Configuration page (“CONFIG”) of the GLOBAL menu contains parameters that affect all presets at once.

PARAMETER	Description																
<b>POWER AMP</b> (OFF/ON)	This parameter 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.																
<b>CABINET</b> (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.																
<b>DELAY SPILL</b> (OFF/DELAY/ REVERB/BOTH)	Setting this to ON allows delays and reverbs tails to carry 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. Note that you must have nearly identical delay or reverb blocks in both the start and destination presets for spillover to work correctly. See section 16.8 on p. 160 for a quick-start guide to setting up spillover.																
<b>AMP GAIN</b> (+/- 12 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 in output between guitars.																
<b>REV MIX</b> (+/- 50%)	This parameter 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.”																
<b>FX MIX</b> (+/- 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: <table border="0" data-bbox="438 1654 1234 1759"> <tr> <td>Delay</td> <td>Formant</td> <td>Pitch</td> <td>Ring Mod</td> </tr> <tr> <td>Chorus</td> <td>Megatap Delay</td> <td>Quad Chorus</td> <td>Rotary</td> </tr> <tr> <td>Feedback Return</td> <td>Multi Delay</td> <td>Resonator</td> <td>Synth</td> </tr> <tr> <td>Flanger</td> <td>Phaser</td> <td>Reverb</td> <td>Vocoder</td> </tr> </table>	Delay	Formant	Pitch	Ring Mod	Chorus	Megatap Delay	Quad Chorus	Rotary	Feedback Return	Multi Delay	Resonator	Synth	Flanger	Phaser	Reverb	Vocoder
Delay	Formant	Pitch	Ring Mod														
Chorus	Megatap Delay	Quad Chorus	Rotary														
Feedback Return	Multi Delay	Resonator	Synth														
Flanger	Phaser	Reverb	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.																

## 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
<b>BANDS 1–10</b> (+/- 12 dB)	<p>Output 1 and Output 2 are individually equipped with 10-band graphic equalizers for fine-tuning of the tone across all presets. This comes in handy when using the Axe-Fx II in different environments, in which acoustics may vary, or when you change amps/speakers with no opportunity to adjust your presets.</p> <p>Output 1 settings apply to <i>all instances</i> 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.</p> <p>One exception to note: when <b>COPY OUT 1 TO OUT 2</b> is enabled in the I/O:AUDIO menu, the global Graphic EQ setting for OUT1 is NOT copied to Output 2, which is processed by its own global graphic equalizer in the usual way.</p> <p>Output 2 settings apply to <i>all instances</i> 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. 135 for more on routing).</p>
<b>GAIN</b> (+/- 12 dB)	<p>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.</p>

## 8.3 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
<b>Custom Scale Number</b> (1–32)	This selects from among the 32 global custom scales available to edit using the 12 parameters below.
<b>Input Pitch/Output Shift</b>	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. 148). Default assignments for all system parameters are listed beginning on p. 171. See also the diagram “Using Send and Return” on p. 165 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
<b>INSTR IN</b> (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.
<b>INPUT 1</b> (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.
<b>INPUT 2</b> (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
<b>MAIN INPUT SOURCE</b> (ANALOG (IN 1)/ USB/ SPDIF-AES)	This selects which one of several input sources 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 <b>INPUT 1 LEFT SELECT</b> (below). When “USB” is selected, the Axe-Fx II will process signals sent from the <b>AXE-FX II OUT 0</b> and <b>AXE-FX II OUT 1</b> audio outputs of the connected computer. When “SPDIF/AES” is selected, the digital inputs of the Axe-Fx II will be used.
<b>INPUT 1 LEFT SELECT</b> (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.
<b>INPUT 1 MODE</b> (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: <b>MAIN INPUT SOURCE</b> to ANALOG (IN1), <b>INPUT 1 LEFT SELECT</b> to FRONT, and <b>INPUT 1 MODE</b> to LEFT ONLY.

## INPUT/OUTPUT PARAMETERS

<b>INPUT 2 MODE</b> (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.
<b>OUTPUT 1 MODE</b> (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. <b>159</b> of the Appendix for more detail.
<b>OUTPUT 1 BOOST/PAD</b> (0-18 dB)	Firstly, 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 this control, increase the boost/pad amount 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.
<b>OUTPUT 1 PHASE</b> (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. Note that this inverts BOTH channels of the stereo pair.
<b>OUTPUT 2 MODE</b> (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. <b>159</b> of the Appendix for more on using the Axe-Fx II in mono.
<b>OUTPUT 2 BOOST/PAD</b> (0-18 dB)	See OUTPUT 1 Boost/Pad, above.
<b>OUTPUT 2 PHASE</b> (NORMAL/INVERT)	See OUTPUT 1 Phase, above.
<b>COPY OUT 1 TO OUT 2</b> OFF/ON	This switch causes OUT 2 to be fed by a copy of OUT 1 instead of the signal from the FX LOOP block. Its use is suited to a wide range of applications such as feeding amps and a recorder at the same time, sending identical signals to Front of House and Stage monitors while retaining separate front panel knob level control over each, using the Axe-Fx II simultaneously with both Stereo and Mono systems (see <b>OUTPUT 1/2 MODE</b> above), and more.
<b>SPDIF/AES SELECT</b> 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.
<b>SPDIF/AES SOURCE</b> OUTPUT1/ OUTPUT2/ INPUT	This selects the signal that will feed <i>both</i> the digital outs..

## 9.3 MIDI Parameters

The MIDI page of the I/O menu contains parameters related to MIDI channel, thru, and program changes.

PARAMETER	Description
<b>MIDI CHANNEL</b> 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.
<b>MIDI THRU</b> 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 <b>merged</b> with regular outbound MIDI—SysEx dumps or Realtime Tuner/Tempo Sysex for example. Thru is disabled during firmware update.
<b>PROG CHANGE</b> ON/OFF	This parameter determines whether the Axe-Fx II will process or ignore incoming MIDI program change messages.
<b>MAPPING MODE</b> 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.
<b>MAP FROM/MAP TO</b> 0–127, 0–383	These two parameters work in pairs to specify which preset will be loaded for each incoming program change message when <b>MAPPING MODE</b> is set to “CUSTOM.” For example, if you select “15” for <b>MAP FROM</b> and then set a <b>MAP TO</b> value of “100,” incoming Program Change #15 actually will load Axe-Fx II Preset 100 (or 101 if display offset is on; see below).
<b>SYSEX ID</b>	This is <b>00 01 74</b> for the Axe-Fx II and cannot be modified.
<b>DISPLAY OFFSET</b> 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. <i>Note: Display Offset requires a corresponding setting in a connected MFC-101 MIDI Foot Controller.</i>
<b>IGNORE REDUNDANT PC</b> 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.
<b>SEND REALTIME SYSEX</b> 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.

## INPUT/OUTPUT PARAMETERS

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### 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, 3<sup>rd</sup>-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. **132**) 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.

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## 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. 124), 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” for the onboard 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 remote 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 set up and transmitting correctly.

Unless otherwise noted, all functions interpret MIDI CC values of 0–63 as “OFF” and 64–127 as “ON.”

PARAMETER	Description
<b>INPUT VOLUME</b> 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.
<b>OUT 1 VOLUME</b> 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.
<b>OUT 2 VOLUME</b> 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.
<b>BYPASS</b> NONE/PEDAL/0-127	Operates the front panel <b>BYPASS</b> button (p. 8) remotely.
<b>TEMPO TAP</b> NONE/PEDAL/0-127	Provides the ability to set the Global Tempo (see p. 146) through a remote switch. <b>IMPORTANT:</b> 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!
<b>TUNER</b> NONE/PEDAL/0-127	Provides a way to enter or exit the TUNER function remotely.
<b>EXT CTRL 1–12</b> 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. 132).
<b>LOOPER REC, PLAY, ONCE, DUB, REV, BYPASS</b>	Each of the controls of the Looper block (p. 74) may be remote controlled.

## INPUT/OUTPUT PARAMETERS

**GLOBAL BYPASS**  
NONE/PEDAL/0-127

**Global Bypass** is used to *engage* all bypassed blocks in the currently loaded preset. It retains and can then *re-bypass* these same blocks when triggered a second time. This lets you prepare a collection of dual-state presets without remembering which individual switches are needed to toggle their blocks. In one song you might have a Drive block for switching between rhythm and lead work. Another may require Echo and Flanger to be switched on during the bridge. Global Bypass allows a single footswitch on your pedalboard to perform all of these functions and more.

**VOLUME INCR,  
VOLUME DECR**  
NONE/PEDAL/0-127

These two options provide a convenient way to **permanently** increase or decrease the MAIN output volume of the currently loaded preset. Each time VOL INCR is triggered by a CC# value greater than 63, the master volume is increased by 1.0 dB and the preset is saved. This is the same for VOLUME DECR, except it decreases volume. The MAIN volume can be found on the OUTPUT 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 will 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, MIXER, SHUNT and TONE MATCH 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 Default** Settings 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 on the rear panel of the Axe-Fx II. Simply connect the device, set its type, and perform a calibration if using a continuous-type expression pedal. The PEDAL function may be assigned to any of the items listed in the CONTROL page of the I/O menu (p. 139)

Any type of external switch may be used, as long as its contacts make and break the connection between tip and sleeve on a regular 1/4" 1/4" guitar cable. 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.

PARAMETER	Description
<b>PEDAL TYPE</b> 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.
<b>PRESET INCR</b> OFF/ON	When this is turned ON, a connected at the <b>PEDAL</b> jack may be used to increment presets.
<b>PRESET START,</b> <b>PRESET END</b>	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 preset, it starts from the beginning again.
<b>PEDAL CAL</b>	To calibrate an expression pedal connected to the onboard <b>PEDAL</b> jack, first select this menu choice, then: <ul style="list-style-type: none"> <li>▶ Press <b>ENTER</b>.</li> <li>▶ Move the pedal through its full range of motion several times.</li> <li>▶ Press <b>ENTER</b> again when finished.</li> </ul> <p>Switches, unlike pedals, do not need to be calibrated.</p>

## 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 detailed in **Section 13** beginning on p. **148**

## 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 meter scale is -80 to 0 db.

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. **156** for more information on this subject.

## 10.4 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.

## 10.5 IR Capture

Impulse Responses allow the Axe-Fx to simulate the sonic properties of a real speaker system. The CAB block (p. 48) is pre-loaded with “Factory” IRs for 70 different speakers, while 50 “User” IR slots allow you to load custom impulses.

This utility allows you to measure the frequency response of a real guitar cab (or other speaker) and save the result as a custom Impulse Responses (IR) for use in the CAB block. To use the utility, you must connect the Axe-Fx II to a power amp and speaker so the test tones may be played, connect a mic and preamp to the input of the Axe-Fx II so the resulting response may be captured, perform the test, name, and save the IR to a USER slot.

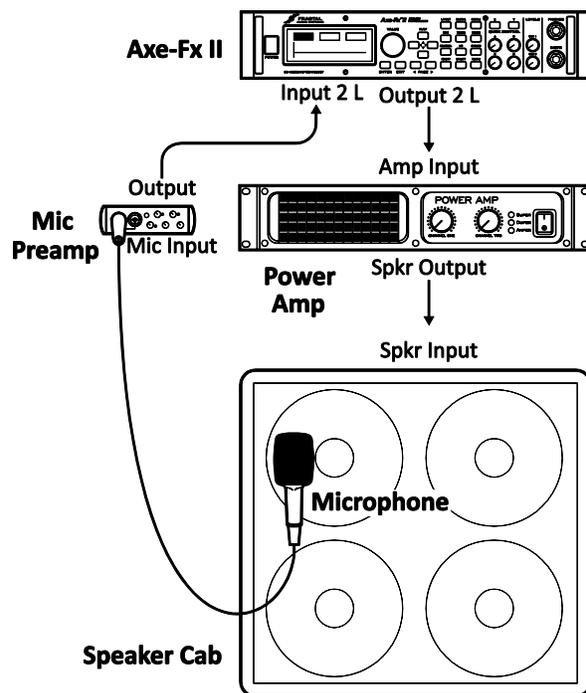
### Components & Connections

To capture your own IRs with the Axe-Fx II, you’ll also need several common pieces of pro audio equipment. Remember that every component in the chain contributes to the sound: the amp, the cab, the mic, and the preamp.

A sonically neutral solid-state **power amp** is best. High wattage is not important as an accurate impulse can be captured at relatively low levels, but take care to match the amp to the impedance of the connected cab.

Select a **microphone** suited for guitar cab recording and position it according to your preference of tones. The web is filled with excellent info on mic placement, distance, angle, etc., but don’t be afraid to experiment. Remember that microphones have tonal “color” which contributes significantly to an IR. As when recording, being able to hear what the mic “hears” by placing the cab in isolation will help produce desired results quickly. Mic position and distance are also critical, and it may take several attempts to find your “sweet spot.”

A **preamp** brings the signal of the mic up to line level for processing. This also contributes to the tone, but even the simple preamp in an inexpensive mixer can produce excellent results. Do not apply dynamics processing (e.g. compression) as this will misrepresent the response. With an optional mixer, multiple microphones can be



## UTILITIES

combined to produce a tone that blends their best characteristics. Note however that it is also possible to capture mics individually and then blend them in a preset using multiple CAB blocks or the “stereo” cab mode. Leave your guitar connected and monitors powered on so you can audition results when the capture is finished. Refer to the diagram here and make connections as follows:

1. Mic the cab, which should be in a quiet, isolated place where background noise won't be a problem.
2. Connect the mic to the preamp input and connect the preamp output to the **INPUT 2 Left (FX RETURN)** of the Axe-Fx II.
3. Connect **OUTPUT 2 Left (FX SEND)** of the Axe-Fx II to the input of your power amp input.
4. Connect the power amp's speaker out to your speaker cabinet. Turn the preamp and power amp ON.

### Capturing the Response

5. Once you've made required connections, press **UTILITY** on the Axe-Fx II and page to “**IR CAP**”.
6. Press **ENTER** with the onscreen [**TEST**] button selected to generate a sample “chirp” for setting levels. The test signal should sound loud enough to measure the speaker without background noise interference, but not loud enough to be uncomfortable or overdrive the power amp or speaker. The front panel **INPUT2** LED meters of the Axe-Fx II will indicate the level of the signal received. Adjust the levels of the power amp and preamp until an optimal level is obtained (the red LED indicates a level of -6db and may flicker on during “healthy” peak levels). If signal is especially low, you may need to increase the virtual input trim of the Axe-Fx II Input 2 (found under I/O: INPUT).
7. When levels are set correctly, navigate to the onscreen [**CAPTURE**] button and press **ENTER**.
8. Wait quietly for the capture to finish (~20 seconds), then use the **NAV** keys and **VALUE** wheel to select a location (“**Save to #**”) and a name, then navigate to the onscreen [**SAVE**] button, and press **ENTER** twice to store the impulse.

To audition your saved IR, change the CAB block of any preset to the USER cab number where you saved the custom IR. You can combine multiple IRs using multiple CAB blocks or the “Stereo” cab mode. If you store the IR in a user slot which is already in use in the currently loaded preset, you will be able to audition immediately upon saving.

It is also possible to use this utility to capture the responses of systems other than a power amp/mic/preamp. Possibilities might include the factory cabs of the Axe-Fx Ultra, a particular tape delay, or any other line-in/line-out device.

The PRESET utilities (p. 142) allow dumping individual User Cab IRs to MIDI or USB for backup. Remember also that all user cab IRs are included with a dump of the Axe-Fx II System (see chapter 13).

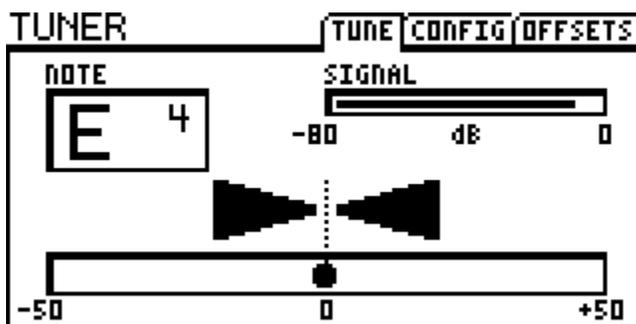
## 10.6 Firmware

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. To enter update mode, navigate to this menu and press **ENTER**.

See Section 14 on p. 151 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. 139) to be remotely engaged/disengaged using MIDI or the onboard Pedal Jack. As of firmware version 6.0, the Axe-Fx tuner can be used to tune droptune or bass guitars as well.



## Configuration Parameters

PARAMETER	Description
<b>CAL</b> 430.0 – 450.0 Hz	This calibrates the tuner by setting the frequency of A4 (in the 8va above middle C).
<b>MUTE</b> OFF/INPUT/OUTPUT	This setting determines whether the main inputs or outputs will be MUTED when the tuner is engaged.
<b>USE OFFSETS</b> OFF/ON	Determines whether the settings on the OFFSET page (see below) are applied or ignored.

## Offset Parameters

PARAMETER	Description
<b>E1, B2, G3,</b> <b>D4, A5, E6</b> +/-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.<sup>1</sup>

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.

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. 139).

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 rhythmic 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.

<sup>1</sup>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) value 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 features a built in metronome. Two parameters control its function.

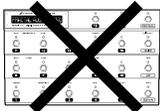
**METRONOME** – Turns the metronome on or off.

**METRO LEVEL** – Sets the level of the metronome from +/- 20db

# 13 Backing Up and Restoring

The Axe-Fx II can create external backups to a computer or other MIDI recorder, or onboard backups to its flash ROM memory. Both methods are described below, as are details on how to restore each type of backup. It is a wise practice to back up your Axe-Fx II regularly.

## 13.1 MIDI/SysEx Backup and Restore



**IMPORTANT!** If you have an MFC-101 connected to your Axe-Fx II, whether at the MIDI IN or MFC Control port, **power it OFF or DISCONNECT it** prior to performing any dump. Failure to do so may cause the MFC-101 to process incoming SysEx data resulting in unpredictable behavior.

### 13.1.1 Dumping to a computer

The **PRESET** page of the **UTILITY** menu contains various utilities for dumping the memory of the Axe-Fx II. Dumps are transmitted as **MIDI System Exclusive data**, or “SysEx”, which can be recorded for backup, sharing, editing, or other purposes.

Before executing a SysEx a dump, you must connect the Axe-Fx II to a computer. This might be via a legacy MIDI interface or via USB. Some means of recording the dump must be provided, typically a 3<sup>rd</sup> party SysEx utility such as **MIDI Ox** for PC (<http://www.midiox.com>) or **Snoize SysEx Librarian** for Mac (<http://www.snoize.com/SysExLibrarian>). The details of configuring and operating these programs are beyond the scope of this manual, but help can be obtained through their authors or at our forum (<http://forum.fractalaudio.com>). Besides these options, **Axe-Edit**, our own MIDI Librarian/Editor, provides a range of options. Axe-Edit is supported by its own documentation.

Once you have the computer connected and a SysEx utility ready to receive a dump, select the appropriate entry on the **UTILITY:PRESET** page and press **ENTER** to begin transmission. There are options for dumping PRESET, BANK, or SYSTEM to either MIDI (via the onboard MIDI OUT/THRU port) or USB (via the AXE-FX II MIDI IN port of a connected computer). MIDI data appears at the MIDI OUT/THRU port regardless of the setting for **MIDI THRU** in the MIDI page of the I/O menu (p. 137)

- ▶ **DUMP PRESET TO MIDI** dumps the currently loaded preset, including any unsaved changes, to the MIDI OUT port of the Axe-Fx II. There is no onscreen confirmation when a PRESET dump initiates or completes.
- ▶ **DUMP BANK (A, B, or C) TO MIDI** dumps a batch of 128 presets to the MIDI OUT port of the Axe-Fx II. 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. 137) is turned ON). An onscreen meter (“Bank Transfer in Progress”) indicates the progress as a bank is transmitted.

**NOTE:** It takes approximately 9-10 minutes for a bank dump to complete via MIDI, so the meter moves very slowly.

- ▶ **DUMP SYSTEM TO MIDI** dumps all of the system settings of the Axe-Fx II. A SYSTEM dump specifically includes:
  - All **GLOBAL** and **I/O** parameter settings

- All **TUNER** settings
- All **USER CAB** IRs
- All **GLOBAL BLOCKS**

An onscreen meter (“Bank Transfer in Progress”) indicates the progress as the SYSTEM is transmitted.

- ▶ **DUMP USER CAB TO MIDI** dumps the currently loaded User Cab IR for backup. If the “stereo” mode or multiple CAB blocks are used in the current preset, the dump will be the “left” IR of the “CAB 1” block.

### Dumping to USB

Each of the “**DUMP... TO USB**” options performs the same function as the corresponding “**DUMP...TO MIDI**” choice, except that the SysEx data is transmitted to the **AXE-FX II MIDI IN** port that appears in your computer when the Axe-Fx II is installed/connected via USB.

### 13.1.2 Restoring from a Computer



**WARNING!** When the Axe-Fx receives a BANK or SYSTEM SysEx file, it overwrites local data and settings without warning. Be certain you are sending the correct files, and consider making a backup of current settings before transmitting anything to your Axe-Fx II.

To restore a SysEx backup file with a 3<sup>rd</sup> party MIDI utility or sequencer:

- ▶ Create the connection between the Axe-Fx II and computer. This might be via legacy MIDI interface or USB.
- ▶ Transmit the desired SysEx file from the application to the Axe-Fx II. It is not necessary to ready the Axe-Fx in any way.
  - When the Axe-Fx receives an individual preset, it does NOT store it. Rather, it loads it into the “edit buffer” which must then be saved using normal methods (p. 37)
  - When the Axe-Fx receives a BANK or SYSTEM dump, it processes and stores the data immediately, overwriting local presets and settings without warning.

Axe-Edit, mentioned above in the section on creating backups, can also open, inspect, modify and transmit SysEx files. See our web site for more information about Axe-Edit. <http://www.fractalaudio.com/products-axe-edit.html>

## 13.2 Onboard ROM Backup and Restore

The Axe-Fx II contains an internal FLASH ROM where PRESET BANKS and SYSTEM SETTINGS may be stored for speedy recovery without a computer. The contents of the FLASH ROM are not overwritten or modified during firmware updates or when you restore presets, banks, or the system from SysEx 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.

## BACKING UP AND RESTORING



**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** 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. 137] is turned ON). A meter indicates progress. It takes less than 10 seconds for a backup to complete.
- ▶ **BACKUP SYSTEM** 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
  - All **USER CAB** IRs
  - All **GLOBAL BLOCKS**
- ▶ **RESTORE USER BANK A, B, or C** 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. 137) 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
  - All **USER CAB** IRs
  - All saved **GLOBAL PRESET** settings
- ▶ **RESTORE FACTORY BANK A,B, or C** 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.

## 13.3 Machine-to-Machine Transfers



**Warning:** When the Axe-Fx receives a machine-to-machine transfer, it overwrites local data and settings without warning.

Preset, Bank or System data can be transferred directly from one Axe-Fx II to another. Connect the MIDI OUT of the first Axe-Fx to the MIDI IN of the second and use any of the DUMP TO MIDI entries in the UTILITY menu.

When the 2<sup>nd</sup> Axe-Fx receives an individual preset, it does NOT store it. Rather, it loads it in the “edit buffer” which must then be saved using normal methods (p. 37).

When the 2<sup>nd</sup> Axe-Fx receives a BANK or SYSTEM dump, it processes and stores the data immediately, overwriting local presets and settings without warning.

It is not possible to connect one Axe-Fx II to another via USB.

# 14 Firmware Updates

## 14.1 Firmware

As detailed in the UTILITIES section (p. 143), firmware is the onboard software that gives the Axe-Fx all of its features, functions and capabilities. Through its capability for firmware updates, the unit has great potential to evolve, with additions, enhancements and the occasional bug fix.

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. To enter update mode, navigate to this menu and press **ENTER**.

### With Axe-Edit

Axe-Edit, the companion editor/librarian for the Axe-Fx II, includes a firmware update utility with step-by-step instructions. Choose "UPDATE FIRMWARE" from the Settings menu to initiate the process. Axe-Edit includes its own support documentation.

### With a 3<sup>rd</sup> Party MIDI Utility or Sequencer

- ▶ Connect the Axe-Fx II and the computer. This will typically be done using USB but legacy MIDI also works.
- ▶ Launch your MIDI Utility and ready it to send the firmware System Exclusive file.
- ▶ On the Axe-Fx II, press **UTILITY** and 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 lowering the rate at which your application sends SysEx data. This setting is typically called something like "Speed of sending MIDI", "Delay between buffers", or "Pause between played messages".

If you accidentally enter firmware upgrade mode, press EXIT or reboot the Axe-Fx II.

New firmware updates are made available through our website <http://www.fractalaudio.com/support>. Please always review the README file and instructions included with the download.

# 15 Troubleshooting

Fractal Audio Systems offers support through its web site at [www.fractalaudio.com](http://www.fractalaudio.com)

You can also get answers to most questions in our online forum at [forum.fractalaudio.com](http://forum.fractalaudio.com)

The Axe-Fx Wiki at [wiki.fractalaudio.com/axefx2](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 input and output ports created 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 CAT5 Ethernet cable and without the need for a wall-wart AC adapter. 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 the Wah?**

**A:** See the tutorial on page **162**.

**Q: My Axe-Fx II is behaving erratically or “froze up” while I was using it. What should I do?**

**A:** Reboot the unit. If the trouble persists across a reboot, try holding RECALL while you boot the unit to reset the “edit buffer.” This will load an EMPTY preset, but won’t affect any of your saved settings.

**Q: One 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. **135**.)
- ▶ 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. **125**) or change its **EXT CTRL INIT VALUE** from 0% to 100% (p. **137**).

**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 single 10” 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. 133) 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. 134) may also have an unusual setting.

**Q: Many of my presets suddenly sound hollow and thin. What could cause this?**

A: These terms are sometimes used to describe the sound of phase 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. Try the headphone jack for comparison purposes. See p. 15 for more on how to set levels.

**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. Adjusting the output levels of your presets, or reducing the global BOOST/PAD setting for the offending output, should help. See p. 15 for more on how to set levels.

**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.

**Q: I am pressing FX BYP, but the state of the selected block will not toggle. What gives?**

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 guitar, the amp, or perhaps the settings of remote-controllers. It is also possible, however, that the preset in question 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. 137.

# 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 Shortcuts Overview

The Axe-Fx II has several shortcuts and hidden features. These are summarized below.

### IN THE AMP BLOCK

- With the TREBLE knob selected, press ENTER to engage or disengage the BRIGHT switch.
- With the DRIVE knob selected, press ENTER to engage or disengage the BOOST switch.

### 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.

### 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.

### 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. 141).

## 16.2 60-Second Edit Guide

The following is provided as a quick start-up or reminder about editing on the Axe-Fx II.

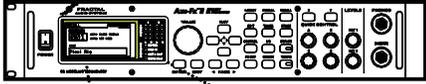
Figure 16-1 – 60-Second Edit Guide

PRESETS

GRID & BLOCKS

EDITING

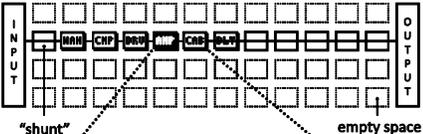
MODIFIERS



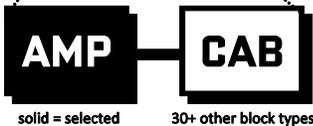
The **AXE-FX II** has 384 editable presets.  
Press **RECALL** to enter normal play mode.  
The **VALUE** wheel or **NAV** buttons select presets.



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



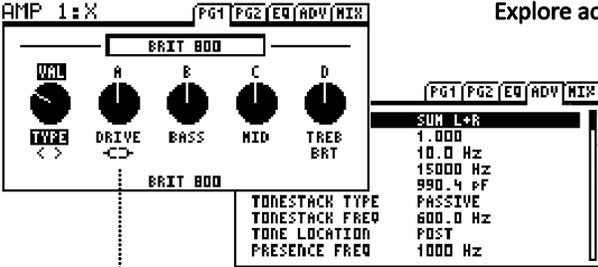
**NAV** and **VALUE** select/change grid spaces/blocks.  
Press **ENTER** to confirm or **EXIT** to cancel changes.  
Press **FX BYP** to bypass\* selected blocks.  
\*Dotted outline = bypassed



To *create* cables, press **ENTER** at the origin\* block then **NAV** to the destination\* and press **ENTER** again.  
To *remove* existing cables, use the same process.  
\*Origin must be a non-empty block. Blocks flash during cable creation/removal. Destination must be in the next column to the right.

**AMP** solid = selected      30+ other block types!

Press **EDIT** for block edit menus  
**LAYOUT** to go back ↴

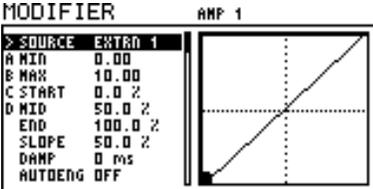


Use **NAV** to select parameters  
Use **VALUE** or **A,B,C,D** to make changes.  
Explore across menus with **◀ PAGE ▶**

**AMP** +9 other block types\* have 2 fully independent sets of controls.  
**X/Y** buttons change the "channel."  
\* Amp, Cab, Chorus, Delay, Drive, Flanger, Phaser, Pitch, Reverb, Wah

**↵** = Press **ENTER** for modifiers  
**EXIT** to go back ↴

**MODIFIERS** attach **control sources** to **parameters** for customizable automation and remote control, as in the example of a pedal-controlled wah.



As with other menu pages, use **NAV** to select from available parameters and use **VALUE** to make changes.

**LAYOUT** returns to the GRID • **RECALL** returns to play mode • **STORE+ENTER+ENTER** saves changes.

## 16.3 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.

**EXCESS CPU USAGE!  
REDUCE LOAD.**

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 G2 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.

## 16.4 Signal Flow, Global, and I/O Parameters

If you are viewing the electronic version of this manual, the final pages (unnumbered) show a stylized view of Axe-Fx II signal and MIDI flow, with indications of the relative placement and functions of GLOBAL and I/O parameters. These are not included in the hard copy version because of printing constraints.

## 16.5 LFO Waveforms, Duty, and Phase

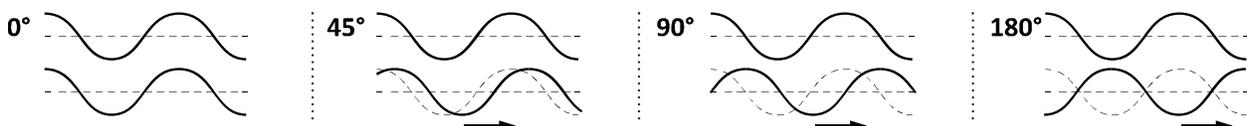
All of the modulation effects in the Axe-Fx II (Chorus, Flanger, Delay, Phaser, Tremolo, etc.) and the two Global LFOs share a common set of waveform types—“shapes” that define the way they change over time. These are represented below, together with an indication of how the **DUTY** parameter controls 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	NA
TRAPEZOID			

### 16.5.1 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.

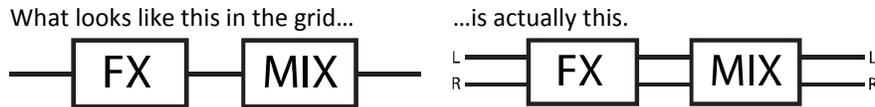




## 16.7 Mono and Stereo

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.

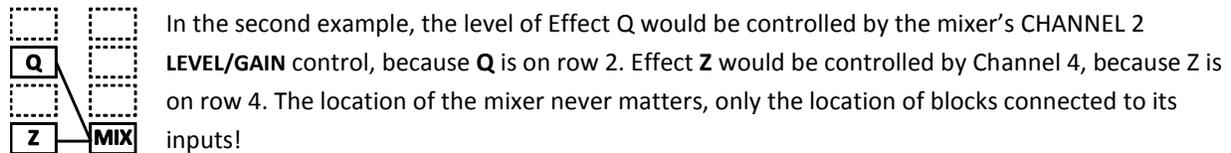
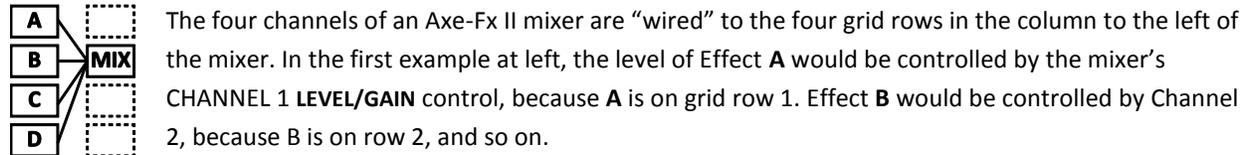


2. **Different blocks process in different ways** — Some blocks (AMP, CAB, Drive, and the pedal compressor) will sum the signal to mono at their inputs and process in mono, but the mono mix appears at both Left and Right outputs for the block. Other blocks—Ping-Pong Delay or Reverb for instance—sum the 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.

- The new **Global Blocks** feature (p. 119) 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.

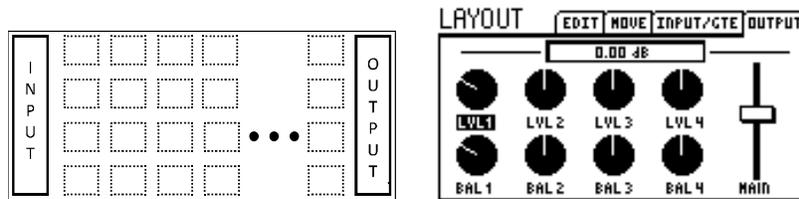
## 16.8 Mixology

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.

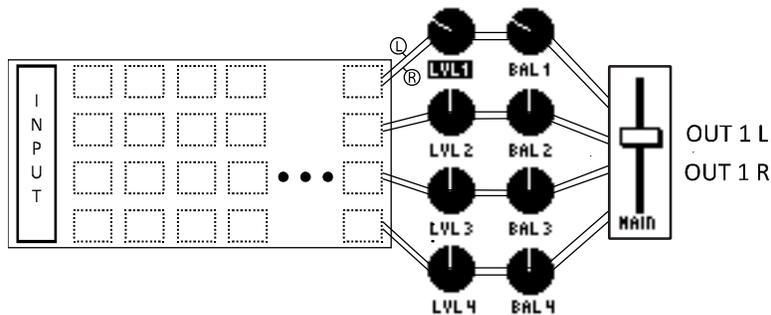


All mixer inputs are **stereo**. The **BALANCE** controls determine the left/right output balance of each channel. When the balance for an input channel is set fully left, only the left channel of the block connected at that channel will be heard at the mixer’s output; its right channel is silenced. 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.

## 16.9 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.

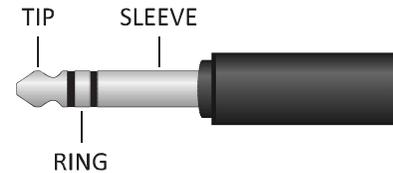
It is of course also possible to use these outputs with normal 1/4" cables.

## 16.10 Setting up a Wah Pedal

### 16.10.1 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 onboard pedal jack of the Axe-Fx II. 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 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.
- 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. 128 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. 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. 124 for more information.

## 16.10.2 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.
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 on Section 7.1 (p. 26) 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. 128 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. 124 for more information.

## 16.11 Setting Up Spillover

Spillover allows delay and reverb tails to ring out when an effect is bypassed or when you change presets.

### 16.11.1 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** (with a footswitch for instance), simply change its BYPASS MODE to “**MUTE FX IN.**” The explanation of Bypass Mode on p. 117 explains why and how this works.

### 16.11.2 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. 133) 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 ensure 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.

## 16.12 Using Send and Return

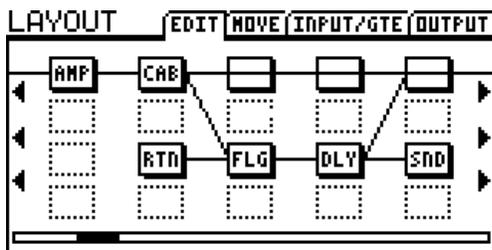
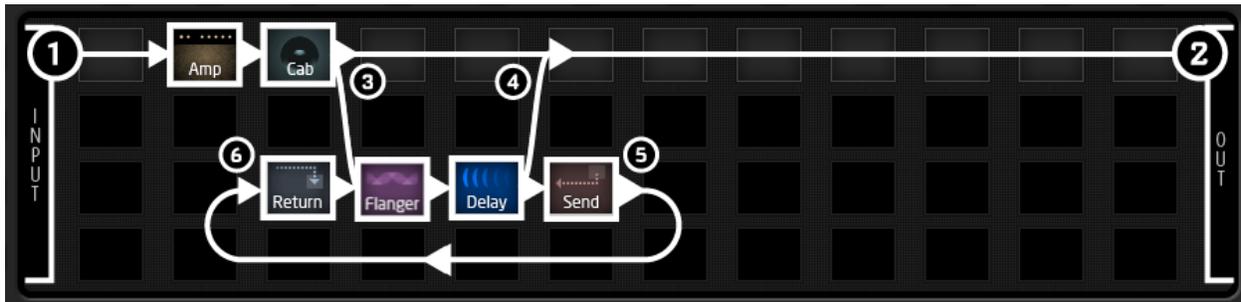
The **Feedback Send** and **Feedback Return** blocks (p.67) have two main uses: creating feedback loops and extending the length of effects chains beyond the size of the grid.

### 16.12.1 Creating Feedback Loops

Feedback loops allow you to combine effects in interesting ways and then route some of 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 “ports” send 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.

### 16.12.2 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.



Figure 16-3 – A Giant Preset with SEND/RETURN

Oh, and the above example isn't a fake. This real preset with 19 effects hits just under 91% on the CPU load meter. Should've gone for 20!

## 16.13 Loading User Cab IRs

The Axe-Fx II has 50 user memory spots into which you can load “Impulse Response” or “IR” files containing the information required to reproduce the sound of a particular speaker cabinet. Commercial IR files are readily available, and the Factory Cabs include several instances of cabs created using selections from the RedWirez and Ownhammer collections. It is also possible, though not a trivial endeavor, to create your own IR files.

Axe-Fx II Impulse Responses are transferred into the unit as MIDI System Exclusive data. You can use Axe-Edit, our free editor/librarian, or any 3rd-party MIDI utility to do this. Axe-Edit includes its own documentation. This section focuses on using 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 set to “MONO HI-RES” or “MONO LO-RES.”
  - 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 User IR SysEx file.
  - a. Remember to select the appropriate MIDI devices in your application's SETTINGS area.  
For a USB connection to the Axe-Fx II, this will be **AXE-FX II MIDI IN** and **AXE-FX II MIDI OUT**.
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. 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. There is no need to STORE or otherwise commit the change; it is permanent once complete.

Any User Cab IRs you have loaded into the Axe-Fx II are included when you backup or restore its **System** area. See p. 148 for details.

## 16.14 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 Engineering Society/European Broadcasting Union. 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.

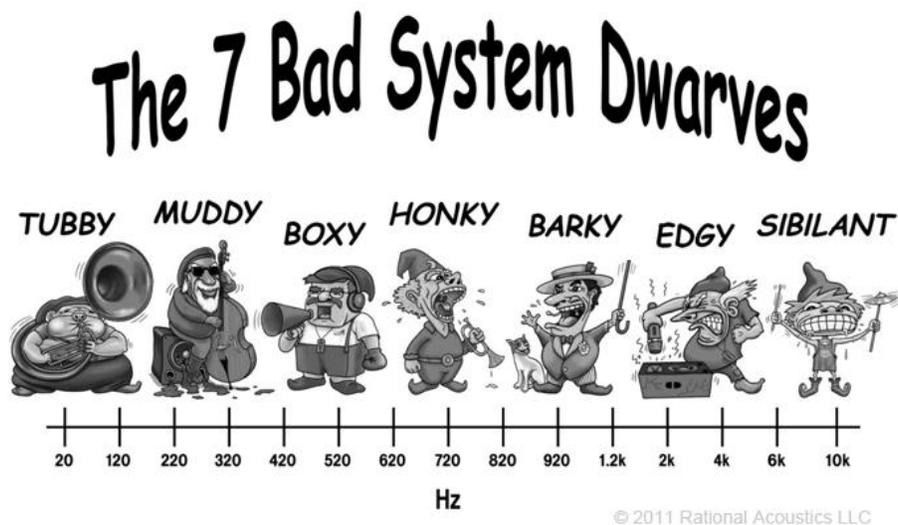
**Feedback:** When an output is connected to an input, feedback occurs. The connection might change mediums, as it does when sound travels from an amp’s speaker through the air to excite guitar strings connected to that amp’s input. It can also be direct, such as when some of the output of a flanger 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.

## APPENDIX

**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.  $4 \text{ Hz} = 4x/\text{second}$ . 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.

**S/PDIF:** Sony/Philips Digital InterFace

**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.).

## 16.15 Axe-Fx II Bank & Preset Numbers Table

The following table shows the BANK and PROGRAM CHANGE commands needed to recall an Axe-Fx II preset via MIDI. Bank select commands persist until another valid bank select is received or the unit is rebooted. Remember that DISPLAY OFFSET (p. 137) can add one (1) to all of the preset numbers (non-bold).

PC #	BANK A (CC#0 = 0)	BANK B (CC#0 = 1)	BANK C (CC#0 = 2)	PC#	BANK A (CC#0 = 1)	BANK B (CC#0 = 1)	BANK C (CC#0 = 2)
<b>000</b>	000	128	256	<b>064</b>	064	192	320
<b>001</b>	001	129	257	<b>065</b>	065	193	321
<b>002</b>	002	130	258	<b>066</b>	066	194	322
<b>003</b>	003	131	259	<b>067</b>	067	195	323
<b>004</b>	004	132	260	<b>068</b>	068	196	324
<b>005</b>	005	133	261	<b>069</b>	069	197	325
<b>006</b>	006	134	262	<b>070</b>	070	198	326
<b>007</b>	007	135	263	<b>071</b>	071	199	327
<b>008</b>	008	136	264	<b>072</b>	072	200	328
<b>009</b>	009	137	265	<b>073</b>	073	201	329
<b>010</b>	010	138	266	<b>074</b>	074	202	330
<b>011</b>	011	139	267	<b>075</b>	075	203	331
<b>012</b>	012	140	268	<b>076</b>	076	204	332
<b>013</b>	013	141	269	<b>077</b>	077	205	333
<b>014</b>	014	142	270	<b>078</b>	078	206	334
<b>015</b>	015	143	271	<b>079</b>	079	207	335
<b>016</b>	016	144	272	<b>080</b>	080	208	336
<b>017</b>	017	145	273	<b>081</b>	081	209	337
<b>018</b>	018	146	274	<b>082</b>	082	210	338
<b>019</b>	019	147	275	<b>083</b>	083	211	339
<b>020</b>	020	148	276	<b>084</b>	084	212	340
<b>021</b>	021	149	277	<b>085</b>	085	213	341
<b>022</b>	022	150	278	<b>086</b>	086	214	342
<b>023</b>	023	151	279	<b>087</b>	087	215	343
<b>024</b>	024	152	280	<b>088</b>	088	216	344
<b>025</b>	025	153	281	<b>089</b>	089	217	345
<b>026</b>	026	154	282	<b>090</b>	090	218	346
<b>027</b>	027	155	283	<b>091</b>	091	219	347
<b>028</b>	028	156	284	<b>092</b>	092	220	348
<b>029</b>	029	157	285	<b>093</b>	093	221	349
<b>030</b>	030	158	286	<b>094</b>	094	222	350
<b>031</b>	031	159	287	<b>095</b>	095	223	351
<b>032</b>	032	160	288	<b>096</b>	096	224	352
<b>033</b>	033	161	289	<b>097</b>	097	225	353
<b>034</b>	034	162	290	<b>098</b>	098	226	354
<b>035</b>	035	163	291	<b>099</b>	099	227	355
<b>036</b>	036	164	292	<b>100</b>	100	228	356
<b>037</b>	037	165	293	<b>101</b>	101	229	357
<b>038</b>	038	166	294	<b>102</b>	102	230	358
<b>039</b>	039	167	295	<b>103</b>	103	231	359
<b>040</b>	040	168	296	<b>104</b>	104	232	360
<b>041</b>	041	169	297	<b>105</b>	105	233	361
<b>042</b>	042	170	298	<b>106</b>	106	234	362
<b>043</b>	043	171	299	<b>107</b>	107	235	363
<b>044</b>	044	172	300	<b>108</b>	108	236	364
<b>045</b>	045	173	301	<b>109</b>	109	237	365
<b>046</b>	046	174	302	<b>110</b>	110	238	366
<b>047</b>	047	175	303	<b>111</b>	111	239	367
<b>048</b>	048	176	304	<b>112</b>	112	240	368
<b>049</b>	049	177	305	<b>113</b>	113	241	369
<b>050</b>	050	178	306	<b>114</b>	114	242	370
<b>051</b>	051	179	307	<b>115</b>	115	243	371
<b>052</b>	052	180	308	<b>116</b>	116	244	372
<b>053</b>	053	181	309	<b>117</b>	117	245	373
<b>054</b>	054	182	310	<b>118</b>	118	246	374
<b>055</b>	055	183	311	<b>119</b>	119	247	375
<b>056</b>	056	184	312	<b>120</b>	120	248	376
<b>057</b>	057	185	313	<b>121</b>	121	249	377
<b>058</b>	058	186	314	<b>122</b>	122	250	378
<b>059</b>	059	187	315	<b>123</b>	123	251	379
<b>060</b>	060	188	316	<b>124</b>	124	252	380
<b>061</b>	061	189	317	<b>125</b>	125	253	381
<b>062</b>	062	190	318	<b>126</b>	126	254	382
<b>063</b>	063	191	319	<b>127</b>	127	255	383

## 16.16 Factory Default Settings

### GLOBAL CONFIG

Power Amp Simulation: ON  
 Cabinet Simulation: ON  
 Spillover: BOTH  
 Global Amp Gain: +0%  
 Global Reverb Mix (offset): +0%  
 Global Effects Mix (offset): +0%

### 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  
 Copy Out 1 to Out 2: OFF  
 S/PDIF / AES/EBU Select: S/PDIF  
 USB / S/PDIF Out Source: OUTPUT 1

### I/O MIDI

MIDI Channel: 1  
 MIDI Thru: OFF  
 Receive Program Change: ON  
 Mapping Mode: NONE  
 (All 127 custom map entries) 1:1 (ex: 1=1, 2=2, 127=127)  
 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  
 Ext. Controllers 1-12 initial value: 0%  
 USB Adapter Mode: OFF

## APPENDIX

### I/O CONTROL

Default CC assignments appear in the table below:

Function	CC#	Function	CC#	Function	CC#
Input Volume	10	Crossover 1 Bypass	45	Reverb 1 Bypass	83
Out 1 Volume	11	Crossover 2 Bypass	46	Reverb 2 Bypass	84
Out 2 Volume	12	Delay 1 Bypass	47	Ring Modulator Bypass	85
Bypass	13	Delay 2 Bypass	48	Rotary 1 Bypass	86
Tempo	14	Drive 1 Bypass	49	Rotary 2 Bypass	87
Tuner	15	Drive 2 Bypass	50	Synth 1 Bypass	88
External Control 1	16	Enhancer Bypass	51	Synth 2 Bypass	89
External Control 2	17	Filter 1 Bypass	52	Tremolo 1 Bypass	90
External Control 3	18	Filter 2 Bypass	53	Tremolo 2 Bypass	91
External Control 4	19	Filter 3 Bypass	54	Vocoder Bypass	92
External Control 5	20	Filter 4 Bypass	55	Volume/Pan 1 Bypass	93
External Control 6	21	Flanger 1 Bypass	56	Volume/Pan 2 Bypass	94
External Control 7	22	Flanger 2 Bypass	57	Volume/Pan 3 Bypass	95
External Control 8	23	Formant 1 Bypass	58	Volume/Pan 4 Bypass	96
External Control 9	24	FX Loop Bypass	59	Wahwah 1 Bypass	97
External Control 10	25	Gate/Expander 1 Bypass	60	Wahwah 2 Bypass	98
External Control 11	26	Gate/Expander 2 Bypass	61	Tone Match Bypass	99
External Control 12	27	Graphic EQ 1 Bypass	62	Amp 1 X/Y	100
Looper Record	28	Graphic EQ 2 Bypass	63	Amp 2 X/Y	101
Looper Play	29	Graphic EQ 3 Bypass	64	Cab 1 X/Y	102
Looper Once	30	Graphic EQ 4 Bypass	65	Cab 2 X/Y	103
Looper Dub	31	Megatap Delay Bypass	66	Chorus 1 X/Y	104
Looper Rev	32	Multiband Comp 1 Bypass	67	Chorus 2 X/Y	105
Looper Bypass	33	Multiband Comp 2 Bypass	68	Delay 1 X/Y	106
Looper Half	120	Multi Delay 2 Bypass	69	Delay 2 X/Y	107
Looper Undo	121	Multi Delay 2 Bypass	70	Drive 1 X/Y	108
Looper Metronome	122	Parametric EQ 1 Bypass	71	Drive 2 X/Y	109
Global Bypass	34	Parametric EQ 2 Bypass	72	Flanger 1 X/Y	110
Volume Increment	35	Parametric EQ 3 Bypass	73	Flanger 2 X/Y	111
Volume Decrement	36	Parametric EQ 4 Bypass	74	Phaser 1 X/Y	112
Amp 1 Bypass	37	Phaser 1 Bypass	75	Phaser 2 X/Y	113
Amp 2 Bypass	38	Phaser 2 Bypass	76	Pitch 1 X/Y	114
Cab 1 Bypass	39	Pitch Shifter 1 Bypass	77	Pitch 2 X/Y	115
Cab 2 Bypass	40	Pitch Shifter 2 Bypass	78	Reverb 1 X/Y	116
Chorus 1 Bypass	41	Quad Chorus 1 Bypass	79	Reverb 2 X/Y	117
Chorus 2 Bypass	42	Quad Chorus 2 Bypass	80	Wahwah 1 X/Y	118
Compressor 1 Bypass	43	Resonator 1 Bypass	81	Wahwah 2 X/Y	119
Compressor 2 Bypass	44	Resonator 2 Bypass	82		

### I/O PEDAL

Pedal Type:	CONTINUOUS
Preset Increment:	OFF
Preset Start:	0
Preset End:	0
Pedal Cal:	(No Setting)

### I/O XY

X Quick Jump:	AMP 1
Y Quick-Jump:	AMP 1

# 17 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 "CAT5"
	 <b>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.</b>

## PHANTOM POWER:

Connector:	Female 2.5mm jack
	 <b>WARNING: Do NOT connect adapters with a rating higher than 1A (1000ma).</b>

## 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:	10 lbs. (4.5 kg)
Input Voltage:	100–240 VAC, 47 – 63 Hz (universal input)
Power Consumption:	<40 W
Backup Battery Life:	>10 years
Backup Battery Type:	CR2450 Lithium

## 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.***

**SPECIFICATIONS**

## 17.1 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	<b>Default Changed</b>	<b>1</b> <b>1-16</b>	<b>1</b> <b>1-16</b>	
Note Number	<b>True Voice</b>	<b>X</b>	<b>X</b>	
Velocity	<b>Note ON</b> <b>Note OFF</b>	<b>X</b> <b>X</b>	<b>X</b> <b>X</b>	
After Touch	<b>Keys</b> <b>Channels</b>	<b>X</b> <b>X</b>	<b>X</b> <b>X</b>	
Pitch Bend		<b>X</b>	<b>X</b>	
Control Change		<b>X</b>	<b>O</b>	CCs are GLOBALLY soft-assigned to functions via the I/O:CTRL menu. These include INPUT volume, OUTPUT 1 and OUTPUT2 master volume, Global Bypass, 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	<b>True Number</b> <b>Bank Select</b>	<b>O</b> <b>O</b>	<b>O</b> <b>O</b>	Bank Select and Program Change messages may be used to recall the 384 presets of the Axe-Fx II. The unit also supports custom program change mapping utilizing a FROM→TO table with 127 entries. BANK select messages are persisted until a subsequent valid Bank Select is received.  Selecting a preset via the Axe-Fx II front panel will also transmit the corresponding bank select and program change number messages
System Exclusive	<b>Fractal Audio</b> <b>Real-Time</b> <b>Non-Real-Time</b>	<b>O</b> <b>O</b> <b>X</b>	<b>O</b> <b>X</b> <b>X</b>	The list of parameters that can be controlled/edited via SysEx is too large to list here. Real-time SysEx is used to transmit Tempo and Tuner.
System Common	<b>Song Position</b> <b>Song Select</b> <b>Tune Request</b>	<b>X</b> <b>X</b> <b>X</b>	<b>X</b> <b>X</b> <b>X</b>	
System Real-Time	<b>Clock</b> <b>Commands</b>	<b>X</b> <b>X</b>	<b>O</b> <b>X</b>	Axe-Fx II Global Tempo syncs automatically to MIDI Beat Clock.
Auxiliary Messages	<b>Local ON/OFF</b> <b>All Notes OFF</b> <b>Active Sense</b> <b>Reset</b>	<b>X</b> <b>X</b> <b>X</b> <b>X</b>	<b>X</b> <b>X</b> <b>X</b> <b>X</b>	

O = YES, X=NO

# Warranty

Fractal Audio Systems warrants that your new Fractal Audio Systems product shall be free of defects in materials and workmanship for a period of one (1) year from the original date of purchase.

During the warranty period, Fractal Audio Systems shall, at its sole option, either repair or replace any product that proves to be defective upon inspection by Fractal Audio Systems.

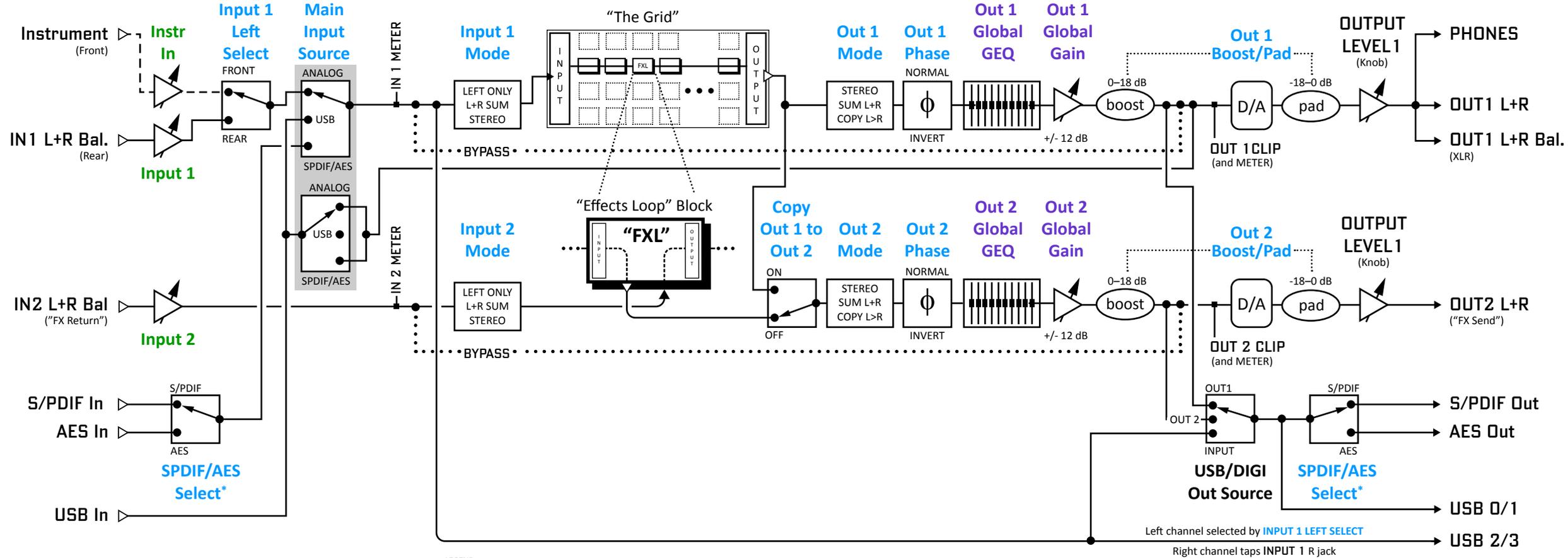
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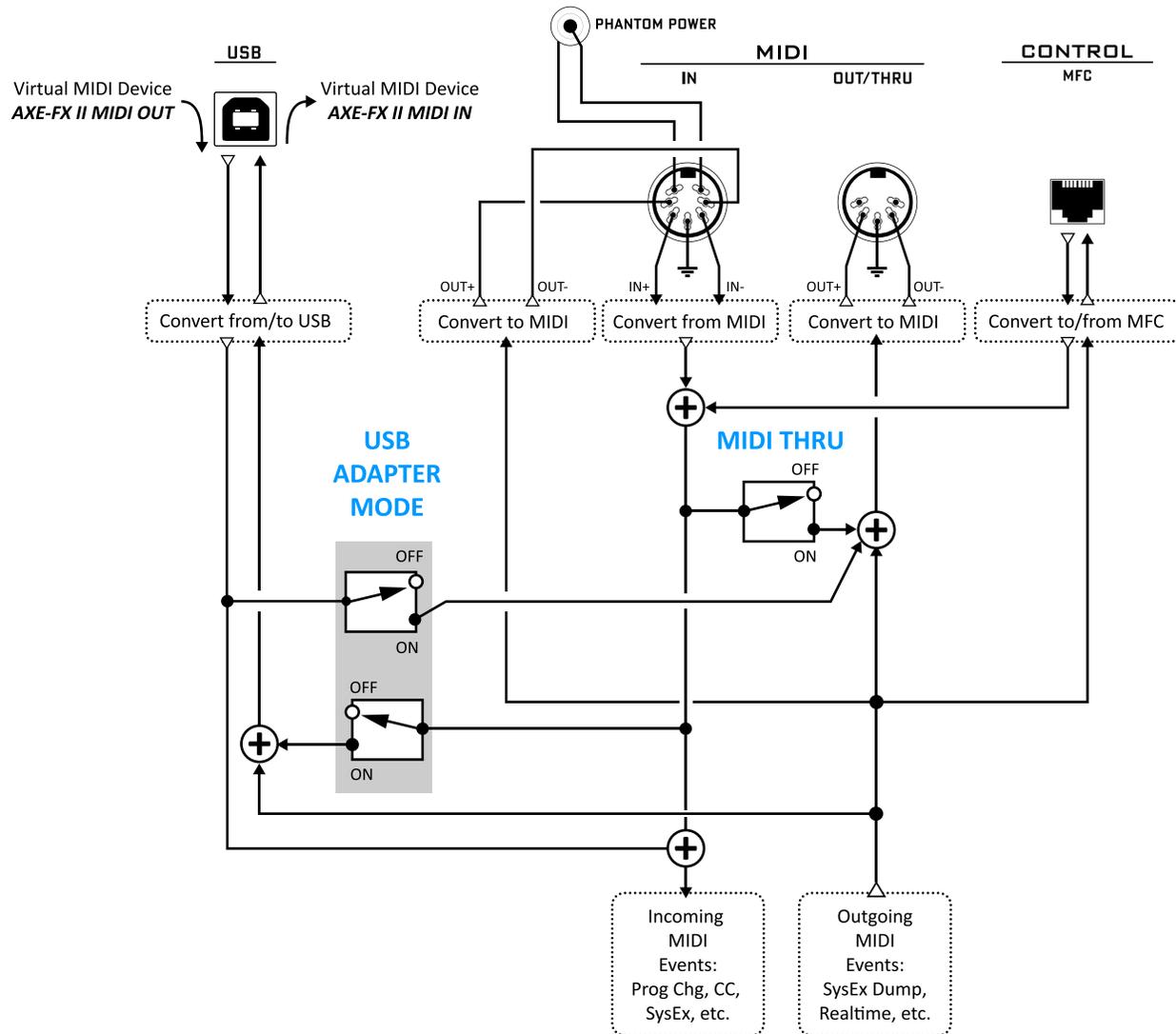
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# Axe-Fx II Input/Output Hardware & Parameters Guide



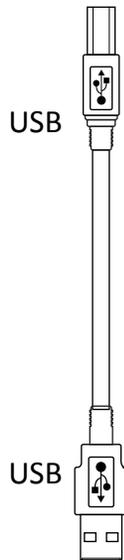
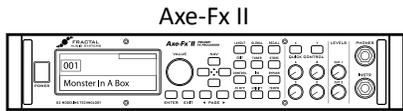
Left channel selected by **INPUT 1 LEFT SELECT**  
 Right channel taps **INPUT 1 R** jack

# Axe-Fx II MIDI Routing and Parameters



- Outgoing MIDI generated by the Axe-Fx II is sent to all outputs (“dumps” are to MIDI *or* USB.)
  - The **MIDI OUT** port.
  - The MIDI out pins of the bidirectional **MIDI IN** port.
  - The **MFC** port.
  - The *AXE-FX II MIDI INPUT* of a connected computer.
- Incoming MIDI is always processed by the Axe-Fx II, whether received from:
  - The *AXE-FX II MIDI OUTPUT* of a connected computer.
  - The **MIDI IN** port.
  - The **MFC** port.
- When the **I/O:MIDI:MIDI Thru** parameter is set to “ON”, MIDI received at the **MIDI IN** port or **MFC** port is forwarded to the **MIDI OUT** port (where it is merged with regular outbound messages).
- When the **I/O:MIDI:USB Adapter Mode** parameter is set to “ON,” two things happen:
  - MIDI from the *Axe-Fx II MIDI Output* of a connected computer is forwarded to the MIDI OUT port of the Axe-Fx II (where it is merged with regular outbound messages).
  - Incoming data at the **MIDI IN** port or the **MFC** port of the Axe-Fx II is forwarded to the *AXE-FX II MIDI INPUT* of a connected computer.

# Axe-Fx II USB Features



"4x2" AUDIO						"1x1" MIDI	
Computer to Axe-Fx II			Axe-Fx II to Computer			2-Way	
1	2		1	2	3	4	
<b>Route the Input</b>			<b>Select an Output Source</b>		<b>Dry Outs for Re-Amping</b>		
L+R are routed based on <b>I/O:AUDIO:MAIN INPUT SOURCE</b> . "USB" routes to the grid for processing. Other options mix computer audio with processed signals at the <b>OUTPUT 1 L+R</b> jacks.			I/O: AUDIO: <b>USB DIGI OUT SOURCE</b> selects between: "OUT1L+R" (The Grid) "OUT2L+R" (Typically FX Loop) or "INPUT 1" (Dry). <i>Same signal also feeds SPDIF/AES outs.</i>		LEFT taps the front <b>INSTR</b> jack or rear <b>INPUT 1 L/MONO</b> jack, based on <b>I/O:AUDIO: INPUT 1 LEFT SELECT</b>	RIGHT taps the rear <b>INPUT 1 R</b> jack	
				<b>Requires USB 2.0 or better</b>			
Win	AXE-FX II ASIO DRIVER OUT 0	AXE-FX II ASIO DRIVER OUT 1	AXE-FX II ASIO DRIVER IN 0	AXE-FX II ASIO DRIVER IN 1	AXE-FX II ASIO DRIVER IN 2	AXE-FX II ASIO DRIVER IN 3	
Mac	AXE-FX II AUDIO OUT LEFT	AXE-FX II AUDIO OUT RIGHT	AXE-FX II AUDIO IN LEFT	AXE-FX II AUDIO IN RIGHT	AXE-FX II AUDIO IN 2 LEFT	AXE-FX II AUDIO IN 2 RIGHT	AXE-FX II MIDI IN

Play audio tracks into the Axe-Fx II for processing (such as re-amping) or send them directly to the main outs so you can jam along with backing tracks, perform on a soft synth, or monitor a project in your DAW/sequencer.

Record the processed output of the Axe-Fx II to your sequencer or other audio application. Alternately, capture the signal from the FX LOOP block, or, on systems which don't support USB 2.0, record the dry signal from input 1.

USB 2.0 (or better) systems also allow you to record the dry unprocessed signal from input 1 (stereo) at the same time you're capturing the main processed outputs. Use these raw tracks for re-amping or reference.

Send MIDI from your sequencer to the Axe-Fx II, or connect to **Axe-Edit** for editing, firmware updates and more. Receive SysEx backups on the computer, or use the HW MIDI IN/OUT ports like a standalone interface for 3rd party gear.