

USER MANUAL

_AUDIOFUSE 16RIG

ARTURIA

_The sound explorers

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Thank you for purchasing AudioFuse 16Rig!

AudioFuse 16Rig brings a new set of powerful capabilities to the Arturia audio interface family, while retaining the same superior sound quality found in the entire Arturia Fuse lineup.

This manual covers the features and operation of the AudioFuse 16Rig. For additional information about Arturia's AudioFuse Control Center, the companion software for the AudioFuse 16Rig, please read the owner's manual for that software.

Important Safety Instructions

PRECAUTIONS INCLUDE, BUT ARE NOT LIMITED TO, THE FOLLOWING:

1. Read and understand all the instructions.
2. Always follow the instructions on the device.
3. Before cleaning the device, always remove the USB and DC cable. When cleaning, use a soft and dry cloth. Do not use gasoline, alcohol, acetone, turpentine or any other organic solutions; do not use a liquid cleaner, spray or cloth that's too wet.
4. Do not use the device near water or moisture, such as a bathtub, sink, swimming pool or similar place.
5. Do not place the device in an unstable position where it might accidentally fall over.
6. Do not place heavy objects on the device. Do not block openings or vents of the device; these locations are used for air circulation to prevent the device from overheating. Do not place the device near a heat vent at any location with poor air circulation.
7. When rackmounting the device, allow air circulation space above and below it for heat dissipation, and use caution when installing in the rack to avoid damage to the rack ears and mounting screws.
8. Do not open or insert anything into the device that may cause a fire or electrical shock.
9. Do not spill any kind of liquid onto the device.
10. Always take the device to a qualified service center. You will invalidate your warranty if you open and remove the cover, and improper assembly may cause electrical shock or other malfunctions.
11. Do not use the device with thunder and lightning present; it may cause electrical shock.
12. Do not expose the device to hot sunlight.
13. Do not use the device when there is a gas leak nearby.
14. Arturia is not responsible for any damage or data loss caused by improper operation of the device.

Specifications subject to change:

The information contained in this manual is believed to be correct at the time of printing. However, Arturia reserves the right to change or modify any of the specifications without notice or obligation to update the hardware that has been purchased.

IMPORTANT:

The product and its software, when used in combination with an amplifier, headphones or speakers, may be able to produce sound levels that could cause permanent hearing loss. DO NOT operate for long periods of time at a high level or at a level that is uncomfortable. If you encounter any hearing loss or ringing in the ears, you should consult an audiologist.

NOTICE:

Service charges incurred due to a lack of knowledge relating to how a function or feature works (when the product is operating as designed) are not covered by the manufacturer's warranty, and are therefore the owner's responsibility. Please study this manual carefully and consult your dealer before requesting service.

Introduction

Dear musician,

We'd like to thank you for purchasing the AudioFuse 16Rig, one of the most powerful pro audio interfaces on the market. Thanks to its great flexibility, and its powerful features, the AudioFuse 16Rig will serve you well whether you're just getting started or expanding your existing setup with the very best equipment.

As with the other interfaces in our Fuse lineup, the AudioFuse 16Rig has been constructed using components of the utmost quality so as to achieve recordings of the utmost quality. And we made it a point to ensure maximum flexibility and transparency, no matter how you might want to use it:

- Flexible connectivity for everything from mic-level and line-level analog audio to ADAT digital, MIDI, Word Clock, and even device-directed USB-A for external controllers
- Pristine audio quality from world-class conversion, and digitally recallable preamps
- Plug-and-play with macOS, one-time driver installation for Windows PCs
- USB type C connection: low latency, superb bandwidth, and backward compatibility with USB 2.0 (Arturia driver required for Windows systems)

This manual will help you make the most of the AudioFuse 16Rig. It will also refer occasionally to AudioFuse Control Center (AFCC), the powerful companion software we designed to work with the AudioFuse family of audio interfaces. You can use AFCC to adjust the front panel controls of the AudioFuse 16Rig, as well as accessing parameters and routing options that aren't controllable from the unit itself.

If you haven't already downloaded AFCC, you can find it on the [Downloads & Manuals page](#) of the Arturia support site. Select the "AudioFuse Control Center" link on the left side of the page and download the appropriate version for your computer. You'll also find the latest version of this manual under "AudioFuse 16Rig" on that page.

Be sure to visit arturia.com website for information about all of our other great hardware and software instruments. They have proven time and again to be the go-to solutions for musicians around the world.

Wishing you all the best in your musical endeavors,

The Arturia team

Table Of Contents

1. Welcome to the AudioFuse 16Rig!	4
1.1. Features of AudioFuse 16Rig:	5
1.2. Diving in (suggestions on using this manual)	6
1.3. Firmware 2.0: Immersive Monitoring and more	6
1.3.1. AudioFuse Control Center: Immersive Monitoring Screen	7
1.3.2. AudioFuse Control Center: Audio Settings	14
1.3.3. AudioFuse Control Center: other new features	15
1.3.4. Immersive Monitoring control on the AudioFuse 16Rig hardware	16
1.3.5. Other features	16
2. Setting Up The Hardware	16
2.1. What's in the box?	17
2.2. Mounting the hardware: tabletop or rackmount?	17
2.2.1. Configuring the AudioFuse 16Rig for Tabletop Use	18
2.2.2. Rackmounting the AudioFuse 16Rig	19
3. Registration and Software Setup	19
3.1. Registration	20
3.1.1. AudioFuse Control Center	20
3.1.2. AudioFuse Creative Suite	20
3.1.3. User Guide	20
3.2. Preparing the AudioFuse 16Rig to work with your computer: macOS or Windows?	21
3.2.1. macOS	21
3.2.2. Windows	22
4. Getting to Know the AudioFuse 16RIG	22
4.1. The Front Panel	23
4.1.1. Outputs 3 & 4	23
4.1.2. Inputs 1 & 2	23
4.1.3. Inputs 3 & 4	24
4.1.4. The Control Interface	24
4.1.5. Monitor and Phones	25
4.1.6. Front Panel USB	26
4.2. The Rear Panel	27
4.2.1. Line Inputs 1-16	27
4.2.2. Line Outputs 3-10 and Speaker Outputs	27
4.2.3. Digital Audio I/O	28
4.2.4. MIDI, USB, and Power	28
5. Putting Your AudioFuse 16Rig To Work	28
5.1. Recording line level devices	29
5.2. Recording a microphone	29
5.3. Recording a Guitar or Bass	30
5.4. Reamping a Guitar or Bass	31
5.5. Creating a Mix	32
5.6. Using Two Sets of Speakers	34
5.7. Connecting to ADAT Digital Audio Devices	34
5.8. Working with multiple ADAT Devices using Word Clock Sync	36
5.9. Connecting MIDI and USB devices	38
5.10. Using the AudioFuse 16Rig as a Control Voltage source	39
6. Getting Around	39
6.1. Physical Controls: How do I get from here to there?	40
6.1.1. The Encoder and Back Button	40
6.1.2. Quick Access Buttons	40
6.1.3. AudioFuse 16Rig Hardware Shortcuts	41
6.1.4. The Mute and Monitor buttons	41
6.2. Screen Layout: what will I see when I get there?	42
6.2.1. The Carousel	42
6.2.2. Pages	43
6.2.3. Scroll, Select, Activate	43
6.3. "Why is this function greyed out?"	44
6.4. Don't forget the software!	44
7. Menus and Functions	44
7.1. INPUTS	45

7.1.1. ANALOG INPUTS / ADAT INPUTS	45
7.1.2. ANALOG IN / FRONT LINE / FRONT INST / FRONT MIC	45
7.2. OUTPUTS	46
7.2.1. MONITOR	47
7.2.2. PHONES	47
7.2.3. ANALOG OUT 3 & 4	47
7.2.4. ANALOG OUT 5-10	48
7.3. MIXER	48
7.3.1. MIXER CHANNEL SETTINGS	49
7.3.2. MAIN MASTER	50
7.3.3. MAIN CHANNELS	50
7.4. CUE	51
7.5. MIDI	51
7.5.1. MIDI ACTIVITY	51
7.5.2. MIDI CONFIG	52
7.5.3. STANDALONE MIDI	53
7.6. SETTINGS	53
7.6.1. CLOCK SETTINGS	53
7.6.2. MONITOR SETTINGS	55
7.6.3. PREFERENCES	55
7.6.4. SYSTEM INFO	56
7.7. PRESETS	56
7.7.1. PRESET SAVE	57
7.7.2. PRESET LOAD	57
8. AudioFuse Control Center	57
8.1. The Upper ToolBar	58
8.1.1. AFCC Menu	58
8.1.2. Screen Selectors	59
8.1.3. Preset Controls	59
8.1.4. Import & Export Configuration Settings	62
8.1.5. Sample Rate and Clock Source	64
8.2. Global Settings	65
8.2.1. Audio Settings	65
8.2.2. Preferences	66
8.2.3. Firmware Update	67
8.3. Overview Screen	67
8.3.1. Analog Inputs	68
8.3.2. ADAT Inputs	68
8.3.3. Analog Outputs	70
8.3.4. Loopback and ADAT Outputs	71
8.4. Main Mixer Screen	72
8.4.1. Adding Channels	72
8.5. Cue Mixer Screen	74
8.6. Routing Matrix Screen	74
8.6.1. Sources	76
8.6.2. Destinations	77
8.7. MIDI Screen	77
8.7.1. MIDI Configuration	78
8.7.2. Standalone MIDI	79
8.7.3. MIDI Activity	79
8.7.4. Main Mixer MIDI Mapping	80
8.8. Advanced remote control	82
8.8.1. The Open HTTP API	83
8.9. Hover Tips	83
9. Specifications	84
9.1. System Requirements	85
9.2. Audio Specifications	85
9.3. Latency Performances	87
9.3.1. Analog to Analog roundtrip latency	87
9.3.2. USB loopback roundtrip latency	87
10. Appendices	87
10.1. USB Audio mapping	88

10.1.1. Audio Mapping at 44.1 / 48 kHz sampling rates:	88
10.1.2. Audio Mapping at 88.2 / 96 kHz sampling rates:	89
10.1.3. Audio Mapping at 176.4 / 192 kHz sampling rates:	90
11. Declaration of Conformity	90
11.1. FCC	91
11.2. CANADA	91
11.3. CE	91
11.4. ROHS	91
11.5. WEEE	91
12. Software License Agreement	92

1. WELCOME TO THE AUDIOFUSE 16RIG!



The AudioFuse 16Rig

When Arturia released the AudioFuse USB audio interface in 2017, its features and design were a revelation to the pro audio world. Every part of its analog and digital signal paths provided world-class quality, from high-performance preamps to premium A/D and D/A converters. Even with all these features, it fit into a compact, beautifully-engineered design that set a new price/performance standard.

Since then, Arturia has expanded its Fuse lineup with other rackmount and desktop interfaces – from specialized AudioFuse products to smaller MiniFuse interfaces perfect for smaller studios. All of these products continue our commitment to maximum features for the money, all with uncompromising sound quality.

The AudioFuse 16Rig is the newest Fuse, and it offers a unique and extensive combination of features never before seen in other Fuse interfaces – or in any other USB audio interface, for that matter. For the first time, there's a centralized and highly flexible audio interface for electronic musicians and other users of multiple keyboards and sound modules. It's set up to easily work with conventional MIDI gear, modular synthesizers requiring not only audio output but voltage control capability, and an ever-increasing set of controllers that require USB connections rather than 5-pin MIDI.

The AudioFuse 16Rig offers a full 16 balanced line-level inputs plus up to 16 channels of ADAT digital audio and two brand-new, digitally controlled mic preamps on the front panel. You also get 8 line-level outputs that are DC coupled for optional control of modular synthesizers, dedicated speaker outputs, Word Clock sync, MIDI, and USB connections to any Mac or Windows computer... as well as USB Hub ports for other gear and a USB Host port that lets you use an external controller even when the AudioFuse 16Rig isn't connected to a computer.

All of those functions are jam-packed into a compact single-rackspace chassis with user-configurable rack ears. Want the AudioFuse16Rig in your rack? Attach the ears for that. Want it on your tabletop in easy reach? Rotate the ears 90° and they become an elevated tabletop stand.

While much of the AudioFuse 16Rig's power is accessible directly from the hardware, you'll find more options within the AudioFuse Control Center software, which has been updated to add support for this newest AudioFuse. Be sure to download the AFCC from the [Downloads & Manuals page](#) of the Arturia support site.

There's no telling what you'll be able to create with the AudioFuse 16Rig as the center of your music-making... um... rig. (Yes, that's why we chose that name!)

1.1. Features of AudioFuse 16Rig:

- 16 balanced line-level analog input channels on TRS
- 2 high-quality digitally-recallable mic preamps on XLR/TRS combo with independent pad and 48V phantom power
- Inputs 1 & 2 have a Hi-Z setting for connecting electric guitars and basses
- Inputs 3 & 4 optionally accessible via front-panel 3.5mm stereo jack
- Inputs 1 through 4 can be manually switched between front and rear panel jacks
- 8 balanced line-level analog output channels on TRS, all DC coupled
- Dedicated stereo Speaker outputs on balanced TRS connectors
- Stereo headphone output (6.35 and 3.5mm TRS), designed for headphones up to 600 Ohms
- Comprehensive monitoring section with individual level control for speakers and phones
- A/B monitoring via a second speaker pair with front-panel level controls, using any pair of line outputs
- Outputs 3 & 4 are mirrored on the front panel, and can be used for reamping, extra headphone outputs, or as convenient line-level connections.
- NEW IN VERSION 2.0: Immersive Monitoring control for up to 16 channels of output including main, surround, sub, and height channels.
- Low latency (DSP powered) Main Mixer with MIDI control
- Dual ADAT I/O ports for 16 channels of I/O at 48 kHz or 8 channels at 96 kHz
- MIDI In and dual Out/Thru on 5-pin DIN
- Front and rear panel USB-A hub connections
- Front panel USB-A MIDI Host connector for hosting external USB MIDI devices
- Word clock input and output on BNC for precise digital sync
- USB-C interface compatible with PC and macOS, fully USB 2.0 compatible
- Full color front panel display with easy parameter access and metering
- Robust metal chassis in a single-space rack form factor
- Rack ears can be rotated for rackmount or tabletop use
- AudioFuse Control Center software offers graphical user interface with added control
- Full standalone operation (no computer needed)

1.2. Diving in [suggestions on using this manual]

In this manual, you'll be introduced to all of the AudioFuse 16Rig's functions, step by step. It's organized so that you can quickly find what you need, whether it's a detailed list of parameters for a given page or just figuring out the difference between clicking a knob and turning it.

The Table of Contents is laid out by section, with some general information at the beginning and end. Inside each section, you can click on topic and page references to quickly find what you need.

One important bit of advice: the more you learn in depth about the AudioFuse 16Rig, the more you can do with it – and you'll work more quickly and efficiently, so you don't lose focus on the important part of this process: making the music.

Take your time and go through the manual, either in order or by chapters that interest you, until you know what every function does. Pretty soon you'll find yourself flying around the interface, turning your creative ideas into great-sounding reality.

And above all: have fun! That's what music is all about.

1.3. Firmware 2.0: Immersive Monitoring and more

With Version 2.0 of the AudioFuse 16Rig firmware (and accompanying AudioFuse Control Center software), AudioFuse 16Rig adds a complete Immersive Monitoring system that supports speaker systems with up to 16 independently assignable output channels, complete with bass management, time delay adjustment, flexible LFE controls, independent A/V sync, and more.

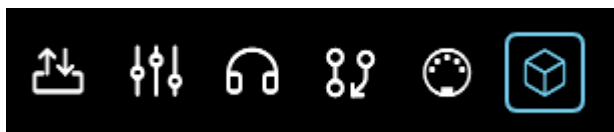
Now you can work in any surround layout with up to 16 channels, with your choice of main, surround, height, and low-frequency monitors. This makes AudioFuse 16Rig a highly flexible monitor solution for everything from stereo and 2.1 systems up to quadraphonic (4.0), 5.1 and 7.1 surround, as well as "3D" audio formats such as Dolby Atmos 7.1.4 or even 9.1.6.



NOTE: Immersive Monitoring can be used at sample rates of 44.1 kHz or 48 kHz only.

1.3.1. AudioFuse Control Center: Immersive Monitoring Screen

All of AudioFuse 16Rig's Immersive Monitoring features are accessible via the new **Immersive Screen**, accessed by clicking the Immersive (cube) icon on the Upper Toolbar:



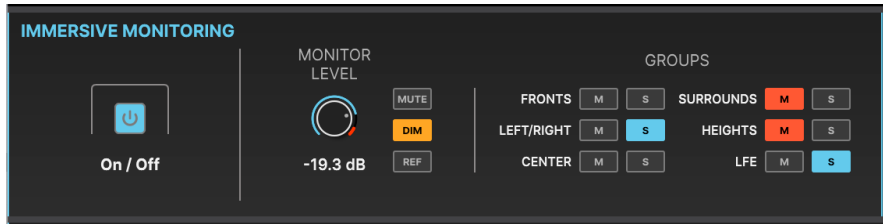
The Immersive Monitoring Screen selector icon

This brings up the Immersive Screen, as shown here:



The Immersive Monitoring Screen

1.3.1.1. On/Off, Monitor Level, and Speaker/Group Mute and Solo



Starting at top left, the first control is a button to turn Immersive Monitoring **On/Off**.

- When On, all of the features and settings on this Screen become active.
- When Off, AudioFuse 16Rig returns to its conventional monitor controls as described in the [Outputs \[p.46\]](#) section of this manual. This screen is grayed out.



Immersive Monitoring settings are stored as part of a saved Preset.

Next are the Monitor Level and Groups controls, which handle listening levels, muting and soloing.

- The **Monitor Level** knob displays the current overall monitor level. A blue ring around the knob shows its current level, from -90 dB (muted) to 0 dB.



When one or more Channel **Trim** controls are set high enough that there's a danger of overall output level going above 0 dBFS (digital clipping), a red ring will appear on the **Monitor Level** knob, showing where the Level control will automatically "stop" to prevent clipping.

- The **Mute** and **Dim** buttons are joined by a **Ref** button, which quickly toggles the **Monitor Level** to a preset reference level of your choice. You can set this level in the [Audio Settings tab of the Settings pop-up \[p.14\]](#).
- The **Groups** buttons allow muting and soloing of the various types of channels used in the current monitor configuration: Fronts, Left/Right, Center, Surrounds, Heights, and LFE. When working with surround and immersive formats such as Dolby Atmos, it's very handy to be able to listen to the various elements of the sound field in isolation.

1.3.1.2. Bass Management, LFE, and A/V Sync



These controls enable/disable and adjust several critical functions of a multichannel mix.

- **Bass Management (BM)** routes lower frequencies of certain channels away from speakers not designed to handle them and into the subwoofer (**LFE/SUB**). For example, this lets the small Left/Right speakers in a 2.1 desktop monitoring system focus on reproducing high frequencies, routing lower frequencies to a subwoofer. This section allows Bass Management to be toggled on/off, and to trim the level of the BM Mix being sent to the LFE channel via the **BM Mix To Sub** knob.
- **LFE** stands for *Low-Frequency Effects*. It's the name for the channel of a multichannel mix that is devoted to low-frequency sounds in a video soundtrack such as explosions and earthquakes. There's a button to turn on a **+10 dB** LFE boost, in case it's needed by your encoding software or playback system, and to adjust the **Cutoff** for which sounds are too high-frequency to be directed to LFE. This knob is actually a 6-position selector: Off, and 80 to 130 Hz in 10 Hz increments.

(These two settings are often handled by your encoding software or the subwoofer itself, but in case they aren't, we give you the option to control them here.)



Note that while LFE and Bass Management mixes are often sent to the same subwoofers, they are not the same thing! Bass Management is a rerouting of low frequencies in music that would otherwise be sent to small speakers that might have trouble reproducing them, whereas LFE is a specific channel in a mix that is *dedicated* to low-frequency content.

- **A/V Sync** is a global delay setting so that an entire mix can quickly be time-delayed to line up with video as it appears on a screen during production. There's an **On/Off** button and an **Add Delay** knob with a range of 0 to 20 ms.

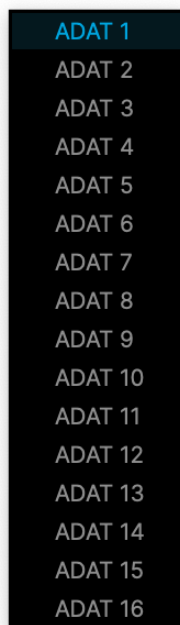
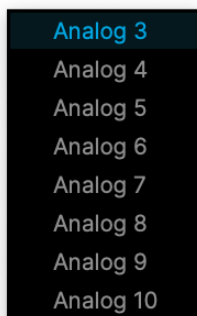
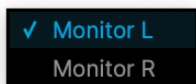
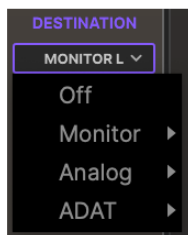
1.3.1.3. Channel Features

Each of the 16 Channels available for Immersive Monitoring has the same set of controls. These are:

- **Channel Name:** click to rename each channel for your own setup.
- **Speaker Type:** A pop-up menu allows you to choose the type of speaker in the monitor setup. While this distinction doesn't change the functionality of a channel in most cases, it does assign channels to appropriate mute/solo groups.

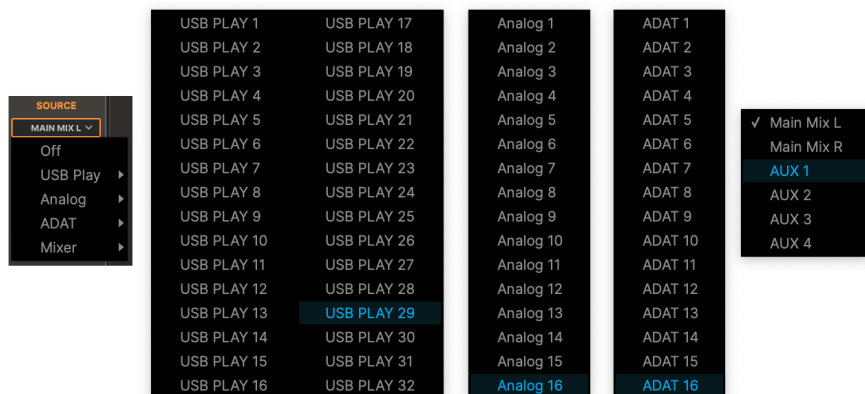


- **Destination** lets you select the physical output of the AudioFuse 16Rig that will play the audio on the channel. You can choose any line-level analog output or ADAT output on the interface, as shown in these pop-up menus:



If you need more analog outputs to accommodate larger layouts like 7.1.4 or 9.1.6, they can be easily added with the AudioFuse X8 Out expander.

- **Source** lets you select where the channel audio will be coming from. This is a set of nested pop-up menus, first to choose a type of input and then a specific channel.



- **Trim** lets you adjust the level of a specific channel with respect to other channels in the mix, so you can level-match the speakers in your monitoring system. The knob has a range of ± 6 dB.
- **Delay** lets you add delay to a channel to help time-align it with other channels in the mix. This is a drag-to-scroll number that is displayed in either milliseconds of delay (**ms**) or meters of distance (**m**). Click the icon to choose ms or m. The control range is 0 to 20 ms, or 0 to 6.85 m.

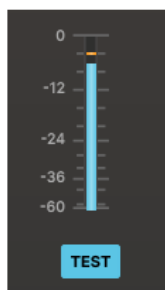
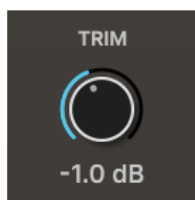


You can also click on the parameter and enter a precise number from your keyboard!

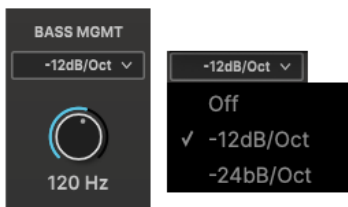
- **Test** plays a pink noise signal over the selected channel so you can audition and verify signal routings, etc.



This noise signal can be quite loud, so mind your speakers and your ears!



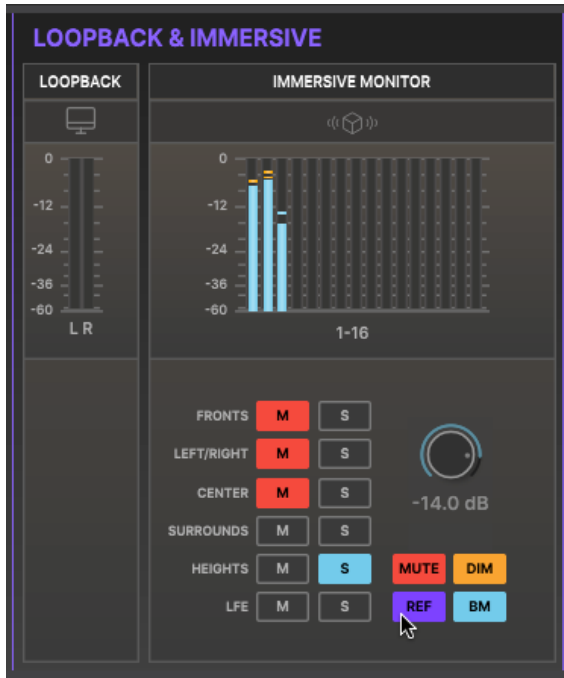
- **BASS MGMT** controls the Bass Management settings for that channel. The knob sets the crossover frequency below which the audio is routed to the **LFE/SUB** channel, and the slope of the filter that rolls off the bass frequencies (either 12 or 24 dB/octave).



i The **LFE/SUB** channel will not have a Bass Management control section, as this is where Bass Management audio is sent.

1.3.1.4. AudioFuse Control Center: Immersive Monitoring on the Overview Screen

The **Loopback** section of the [Overview Screen \[p.67\]](#) now includes a compact set of meters and basic controls for Immersive Monitoring:



- A set of **level meters** for up to 16 channels
- **Mute** and **Solo** buttons for all channel groups
- **Monitor Level** knob
- Buttons for **Mute**, **Dim**, **Ref**, and **BM** (Bass Management on/off)

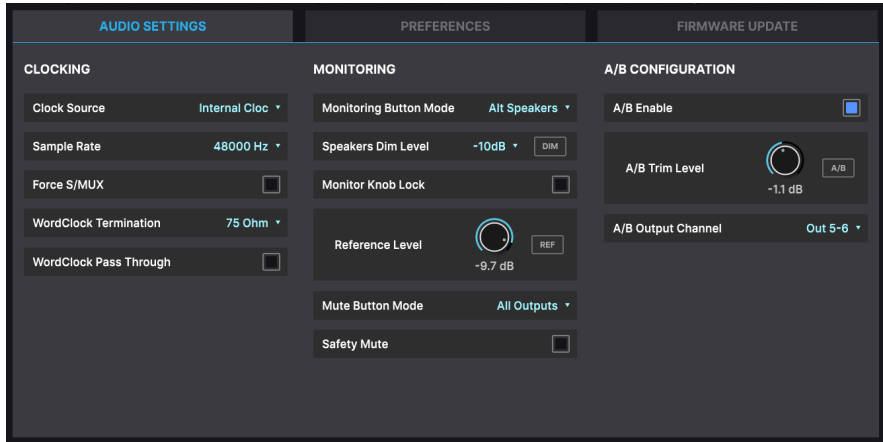
This allows you to quickly work with the main functions of the Immersive Monitoring system without having to leave the Overview Screen. The Immersive Screen is used to configure your monitor system, and once your setup is saved, it will only be used for system adjustments.



Note: because the Ref button instantly resets the level and is not an 'on/off' control, it does not remain illuminated - it momentarily flashes purple when clicked.

1.3.2. AudioFuse Control Center: Audio Settings

The Audio Settings menu now has settings for these new features, as well as a few entirely new ones.



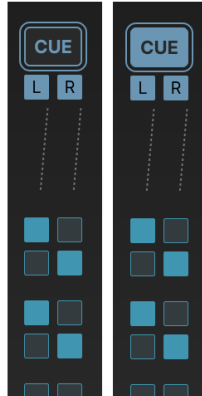
These changes are all in the center column (**MONITORING**). They include:

- An expanded pop-up menu for Monitoring Button Mode. You can now set the Monitoring Button (with the speaker icon) on the AudioFuse 16Rig's front panel to perform the following functions, indicated by a specific color for the icon's LED backlight:
 - **Mono** (green)
 - **Dim** (yellow)
 - **Alt Speakers** (dark blue)
 - **Reference Level** (purple - *new*)
 - **Bass Management** (light blue - *new*)
- **Monitor Knob Lock**, which deactivates the front-panel **MONITOR** knob, although you can still control levels via AudioFuse Control Center. This feature is useful for users with external hardware monitor controllers, who don't want the interface's output levels to change by accident. (It's also handy to prevent accidental level adjustments by that overeager producer or hearing-damaged client messing around in your studio.)
- **Reference Level** setting, from -90 to 0 dB. This is a preferred listening level that you can instantly return to with one click. It's useful for film/TV mixers that need to work at calibrated loudness levels.
- **Mute Button Mode**: This allows you to determine what the front panel **MUTE** button does. There are two options:
 - **Monitoring Only** (default): the **MUTE** button turns off all signals routed to any analog or ADAT outputs that are specified as Channel Destinations in the Immersive Monitoring system (when it's on), or to the Monitor 1/2 outputs (when it's off). All other signal routings, e.g. sends to outboard effects, remain active.
 - **All Outputs**: every analog and ADAT output is muted, regardless of where it is routed. > In all cases, the **PHONES** output always remains active.

- **Safety Mute**, a new feature that automatically mutes the outputs when you change a preset, or when you turn off the Immersive Monitor System. This is important because in stereo mode, the **MONITOR** knob only controls the main stereo outputs, so there is no longer any volume control for immersive speakers. Safety Mute is On by default, but can be switched off if you are confident about handling levels in your immersive speaker system without this auto-mute feature.

1.3.3. AudioFuse Control Center: other new features

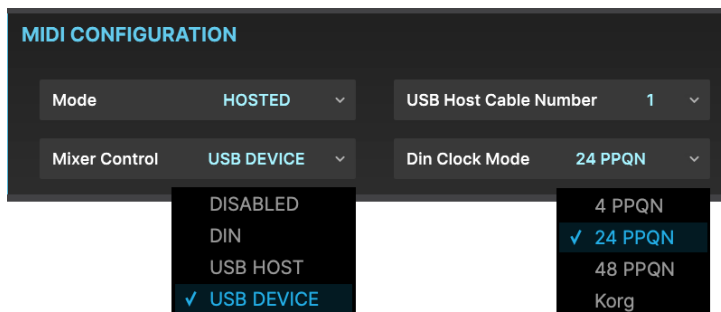
1.3.3.1. Cue Mix on the Routing Matrix Screen



While the functions of the [Routing Matrix Screen \[p.74\]](#) haven't changed, Version 2.0 has added a couple of worthwhile features:

- As shown above, the **CUE** icon at the top of its column in the Routing Matrix has been turned into a button. You can now simply click on it to quickly turn the Cue mix on and off.
- The CUE L/R signals can now be routed to more than one destination, just like any other signal.

1.3.3.2. New MIDI Features



Two MIDI Configuration options on the MIDI Screen have been expanded:

- The **Mixer Control** pop-up now adds a **USB DEVICE** option, so you can control the AudioFuse 16Rig's Main Mixer from your DAW.
- **Clock Output** now has a menu of choices rather than simply offering 24 PPQN. The choice of 4, 24, or 48 PPQN, or Korg sync, gives you the same options found on many Arturia devices.

1.3.4. Immersive Monitoring control on the AudioFuse 16Rig hardware

The Carousel of menu items on the AudioFuse 16Rig hardware display now has an Immersive menu, indicated by a Cube icon.

There are two pages to the Immersive section: one with a set of level meters and buttons for **Mute**, **Dim**, **Ref**, and **BM**, and one with a set of **Group Mute and Solo** buttons.

When the Immersive Monitoring system is On, the front-panel **MONITOR** knob controls Immersive Volume.

1.3.5. Other features

You'll find other nice extras as you use AudioFuse 16Rig with Version 2: - There are now pop-up screens that appear momentarily on the front panel when certain features are accessed (like changing the volume of the **MONITOR** or **PHONES**) - You can specify in Preferences whether your unit stays off or auto-restarts after a power cut - The shutdown process is a bit quicker

2. SETTING UP THE HARDWARE

The first thing to do when you unbox the AudioFuse 16Rig is to set it up for rackmount or tabletop use. Follow these instructions carefully so you don't damage your interface.

2.1. What's in the box?

- The AudioFuse 16Rig
- Rack/stand ears and rear supports with screws
- Power supply with adapters for worldwide use
- USB cable: USB-C to USB-C
- USB cable: USB-C to USB-A
- Registration card with serial number and Unlock Code
- Quick Start Guide

2.2. Mounting the hardware: tabletop or rackmount?

The two bright orange rack ears can be attached to the sides of the AudioFuse 16Rig to either put it in a rack or place it on a desk or table. The screw holes are drilled so as to help you avoid putting the ears on the wrong sides, but you should still be careful. Just remember that the left ear has the Arturia logo on it!

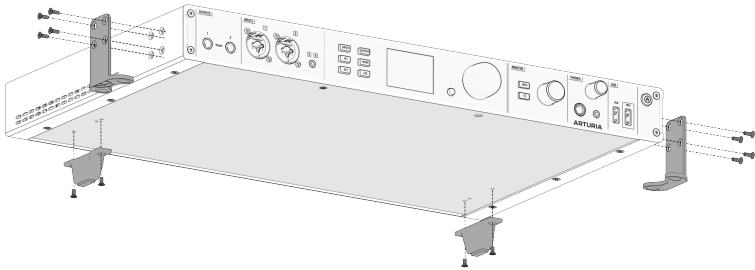


The unit comes with mounting screws especially chosen for this purpose. Don't use other screws, and carefully follow the directions in this manual, which match the ones printed on the inside of the box lid. You don't want your new interface to sit all wobbly on your table, do you?



IMPORTANT: Note that the hole spacing is different between the left and right rackmount ears. Therefore, please make sure that the Arturia 'A' logo is on the left side of the interface.

2.2.1. Configuring the AudioFuse 16Rig for Tabletop Use



AudioFuse 16Rig configured for tabletop use

In addition to the rack ears and screws, your AudioFuse 16Rig should also have a pair of rectangular black plastic feet for the rear of the box, and a set of two semi-circular rubber pads. You'll need these for tabletop use.

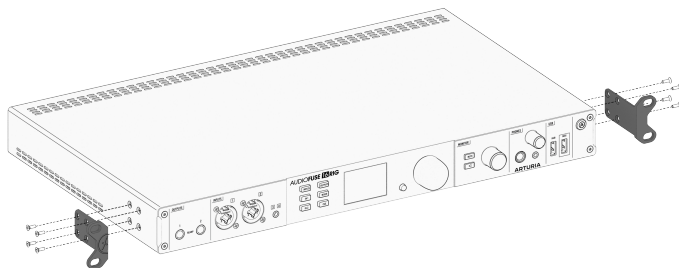
1. Stick the rubber pads onto the rack ears at the spot that will become the "feet". This will keep the feet from scratching whatever you rest them on.
2. Attach the rubber pads so that the straight edge aligns with the 90° bend in the rack ear, and then press the circular portion of the pad firmly onto the rack ear in the direction of the two large screw holes. (See the diagram above or on the box lid.)
3. Be sure to mount the rack ears so they are pointing under the AudioFuse 16Rig (toward its center), not pointing away.
4. The two rear feet have tiny plastic guide pins that help them seat properly on the bottom panel. Once you line those up with the guide holes in the bottom of the unit, it will be easy to fasten the feet into place with the provided screws.
5. When the feet are properly installed, the AudioFuse 16Rig will be supported in four places: the left and right rack ears (under the front) and the left and right feet (under the back).

Now your unit is elevated above the tabletop and perfectly level and solid, so you could place other gear on it, such as a small controller or tablet.



Keep in mind that the AudioFuse 16Rig does generate heat that needs to be able to escape the chassis. Don't block the vents and leave room for air circulation, and don't stack another item that generates heat on top of it (such as a laptop).

2.2.2. Rackmounting the AudioFuse 16Rig



AudioFuse 16Rig configured for rackmounting

Attach the rack ears in the configuration shown above, and the AudioFuse 16Rig is ready to slot into your rack. In this configuration, you won't need the rear feet or rubber pads, so leave them in the box in case you want to rack mount the unit some day.



We recommend rackmounting the AudioFuse 16Rig with an open rackspace above or below it, to assist with keeping the unit cool during operation.

3. REGISTRATION AND SOFTWARE SETUP

Now that we know a bit more about the AudioFuse 16Rig, it's time to get it ready to work with your computer.

3.1. Registration

Once you've set up your hardware, the next part of the process is to register it with Arturia.

_Register
your instrument

To register, fill out the form with your Serial & Unlock Code.

Serial Number *

XXXX-XXXX-XXXX-XXXX

[Where can I find my Serial Number?](#)

Unlock Code *

XXXXXXXX

[I don't have an Unlock Code](#)

Register

Get access to

- Free downloads & latest updates
- Technical support
- The Sound Explorers community

You can find your Serial Number & Unlock Code at the back of your product for hardware and at the back of your license card for software.

If you got a software license from [arturia.com](#), your license is already registered, find it in the Arturia Software Center.

[Need help? See our FAQ](#)

Have your registration card handy and go to www.arturia.com/register to log into your Arturia account or create a new one. Then follow the online directions to register your unit.

Registering your AudioFuse 16Rig gives you access to a lot of important materials:

3.1.1. AudioFuse Control Center

You should download and install **AudioFuse Control Center** (AFCC) immediately. It's free, and is vital for the smoothest possible operation of your interface. It gives you access to controls that aren't available via the front panel, lets you lay out mixes, routings, and monitoring setups in a large and easy-to-use graphical interface. It will also notify you of firmware updates – and let you install the updates onto the hardware. See below for hints on a successful installation process.

3.1.2. AudioFuse Creative Suite

Great-sounding computer audio hardware deserves great-sounding computer audio software! The **AudioFuse Creative Suite** is a set of audio effects plug-ins, selected from Arturia's award-winning FX Collection, that are included with your purchase of a Fuse interface. These plug-ins include faithful emulations of vintage preamplifiers, compressors, time-based and modulation effects, reverbs, and more. Download and activate the Creative Suite plug-ins to jumpstart your music creation process.

3.1.3. User Guide

You don't need to register your AudioFuse 16Rig to download this manual, but the registration process will help you make sure you have the latest edition.

3.2. Preparing the AudioFuse 16Rig to work with your computer: macOS or Windows?

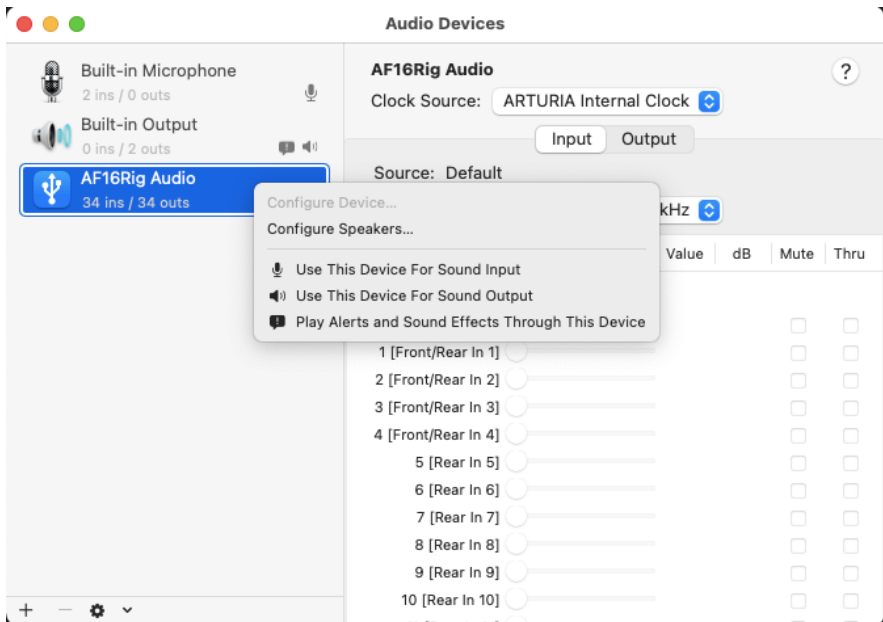
Once you've downloaded and installed AudioFuse Control Center, you'll need to check a couple of settings on your computer to ensure the best communication with your interface.

3.2.1. macOS

Connect the AudioFuse 16Rig to its power supply, plug it into your Mac with the appropriate USB cable, and power it up.

Launch **Audio MIDI Setup** (you'll find it in Applications/Utilities) and the AudioFuse 16Rig should already be there.

To set up the AudioFuse 16Rig as your default audio device:



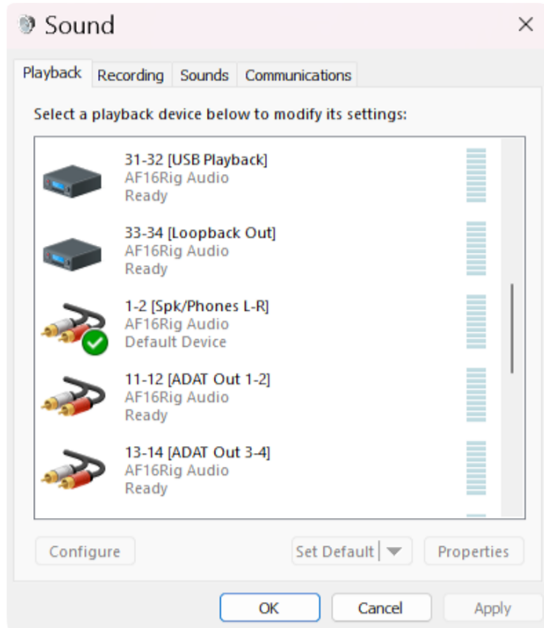
Audio MIDI Setup on macOS

1. Open the Audio Devices window and find "AF16Rig Audio" on the list of devices at left.
2. Highlight it, right-click on it, and a popup menu will allow you to choose it for Sound Input and Sound Output. (You probably won't want to use it for Alerts and Sound Effects.)

3.2.2. Windows

When you install AudioFuse Control Center on your Windows PC, it will also install the appropriate Arturia USB Audio Drivers. Once that's done, you can plug in and power up the AudioFuse 16Rig.

To set up the AudioFuse 16Rig as your default audio device:



Control Panel Sound Default Select on Windows

1. Go to the Control Panel and click Sound.
2. Find the AF16Rig in the device list and click Set Default and OK.

4. GETTING TO KNOW THE AUDIOFUSE 16RIG

As you've seen for yourself when you took the AudioFuse 16Rig out of the box, this interface has a whole lot going on!

Let's take a tour around the AudioFuse 16Rig, and in the process we'll learn a lot about what it can do.

4.1. The Front Panel

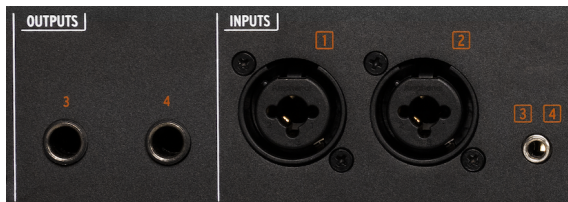


The AudioFuse 16Rig front panel

The AudioFuse 16Rig is designed not only for easy operation, but also for connecting gear in a big hurry. While large professional studios will often have patchbays near the console so that all their gear can be quickly connected, that's not practical for a home setup. (For one thing, you'll need to buy twice as many cables as you have already!)

You're going to want to try out new synths and controllers and effects boxes all the time, even when the bulk of your studio gear is permanently hooked up. Wouldn't it be nice if there was an audio interface that makes switching gear in and out easy?

That's the idea behind the front panel connections of the AudioFuse 16Rig.



The Outputs and Inputs on the front panel

4.1.1. Outputs 3 & 4

On the far left of the front panel, there are a pair of TRS outputs that duplicate Outputs 3 & 4 on the rear panel. They can be used for extra headphone outputs, running a line-level signal to external gear that's not normally hooked up, or *reamplification* - the process of sending a clean guitar signal back out to an amp or effects chain and recording the results back into the interface.

4.1.2. Inputs 1 & 2

Next we have a pair of inputs on combination TRS/XLR jacks. These can be used for line-level or instrument-level signals from synths or guitars, or for high-quality mic recording via built-in digitally-controlled analog preamps.

One of the problems that musicians with more synthesizers than microphones discover when buying audio interfaces is that they're often forced to pay for preamps they don't need. The AudioFuse 16Rig gives you two excellent preamps, which is plenty for many users' needs – and the rest of the budget is spent on a whopping 16 high quality line-level inputs. You'll find the rest of the inputs on the rear panel – see below.)

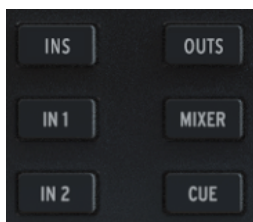
i There are also TRS jacks on the rear panel for Inputs 1 & 2. The AudioFuse 16Rig has an auto-sensing feature, so the rear-panel jacks can remain plugged in at all times, and are automatically ignored when the front-panel jacks are used. However, if you want to leave something plugged into the front panel as well, those inputs have a **REAR** switch to manually route signals from the rear panel.

4.1.3. Inputs 3 & 4

For those occasions when you want to run audio from a miniature desktop synth or even a smartphone into your mix, there's a 3.5mm minijack on the front panel that sends a stereo signal to Inputs 3 & 4.

i There is also a **REAR** button to activate the TRS jacks on the rear panel if you have devices connected to both the front and rear inputs. It is accessible from the menu system in the hardware and from the AudioFuse Control Panel software.

4.1.4. The Control Interface

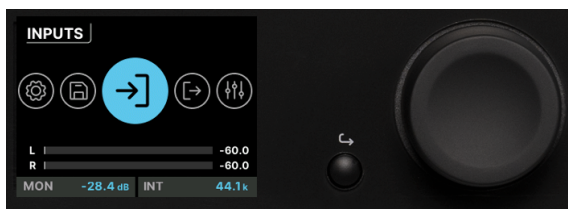


Direct Access Buttons

The center section of the front panel is where the AudioFuse's display and controls are located.

Six front-panel buttons give you quick access to vital features. They are:

- **INPUTS:** Displays metering for all input signals
- **IN 1:** Instant access to controls for Input 1
- **IN 2:** Instant access to controls for Input 2
- **OUTPUTS:** Displays metering for all output signals
- **MIXER:** Takes you to the AudioFuse 16Rig's Main mixer
- **CUE:** Takes you to the AudioFuse 16Rig's secondary (Cue) mixer



The display, encoder, and Back button

Next to these buttons is the color LCD that displays all relevant information while using the AudioFuse 16Rig.

To the right is the main encoder for parameter control, which can be turned or pressed (clicked). A small Back button next to it moves you back up the hierarchy of screen menus when needed.

4.1.5. Monitor and Phones



Monitor and Phones controls and front-panel USB ports

Next is the **Monitor** knob, which controls the level coming out of the rear-panel Speaker outs. It's accompanied by a **MUTE** button that when clicked, turns red and silences the speakers.

Below the MUTE button is a programmable function button. It lights in a different color for each function, and blinks when activated. The functions are:

1. **Mono (green):** sums the left and right signals being sent to the speaker outputs. This is useful if you need to verify mono compatibility or to check for phase-related issues in your mix.
2. **Dim (orange):** turns down the speaker output by an amount designated in the AudioFuse Control Center software. This is useful if you need to have a conversation in the studio without stopping playback or losing your reference loudness level while mixing.
3. **A/B Speaker (blue):** mutes the Speaker outputs and turns on a pair of B Speaker outputs designated by the user - any analog output pair on the rear panel can be designated for B Speaker use. However, we recommend using 5-6, 7-8 or 9-10, and leaving 3-4 available since outputs 3-4 are mirrored on the front panel and provide other useful functionalities.



NOTE: The A/B Speaker function must be enabled for this feature to work. If the function is not enabled, this button is ignored. You can enable the A/B Speaker function in AFCC's Configuration (gear icon) menu, or on the hardware itself by going to Outputs > Monitor > A/B Configuration.

Next is the **Phones** knob, which provides control of the headphone outputs, provided on both 1/4" and 3.5mm TRS jacks.

4.1.6. Front Panel USB



Monitor and Phones controls and front-panel USB ports

Finally we have two USB-A ports.

The first, labeled **HUB**, works as a powered USB hub for external devices you might need to connect to your computer while you're working: external drives to load and save sample libraries, control surfaces for your DAW, etc.

The second, labeled **MIDI**, is a USB Host port that allows you to plug in a controller that only has a USB connector and no 5-pin DIN MIDI ports. They're getting more common every day, and this jack lets you use them without having to run more cables to your computer or an external USB hub. Unlike the **HUB** connection, this port remains active even when the AudioFuse 16Rig is in standalone mode, giving you MIDI input and output without a computer connected.

Note the small button in the upper right corner with the Arturia logo. This button turns the AudioFuse 16Rig on and off. One push will power the unit on; to power off, hold the button until the display says "Goodbye!" so the unit shuts down in a safe and controlled fashion. You can also tap the button to bring up the AFCC software on the connected computer.



The button will be lit blue when in standalone mode and white if it sees a connected computer.

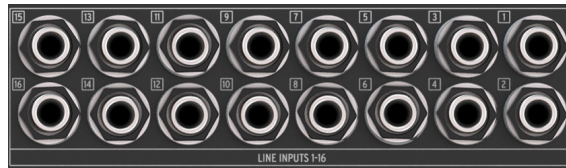
4.2. The Rear Panel



The AudioFuse 16Rig rear panel

The AudioFuse 16Rig's rear panel has a large complement of connectors. Here's a guide, moving from *right to left*.

4.2.1. Line Inputs 1-16

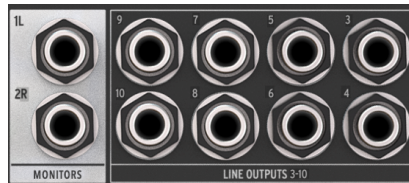


Rear panel audio inputs

At the far right, you'll find 16 TRS balanced analog inputs with world-class A/D converters. These can handle signal levels up to +24 dBu with the Pad switched on, and +12 dBu with the Pad switched off.

Remember that the rear Inputs 1 & 2 are ignored if the front panel inputs are used, unless the **REAR** function is selected. Inputs 3 & 4 work the same way.

4.2.2. Line Outputs 3-10 and Speaker Outputs



Rear panel audio outputs, with Speaker outputs at left

To the left of the Line Inputs are 8 balanced analog line-level outputs. Any of these four pairs of outputs can be sent to another set of speakers and controlled from the front panel if A/B Speaker mode is engaged.

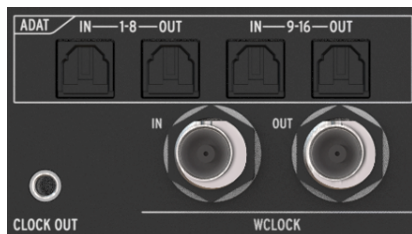


1 These outputs are all DC coupled. That means they are capable of not only playing audio, but running DC control voltages to external modular synthesizers and other analog synth gear. This lets you control your analog synths from your DAW with the right software!

To the left of the Line Outputs and backed in white, the Speaker Outputs are TRS jacks that should be connected to your primary set of monitors. Your DAW will see them as Outputs 1 and 2.

i Note that if you've switched on the A/B Monitoring function, and switch to your alternate or 'B' speakers, these outputs are muted.

4.2.3. Digital Audio I/O



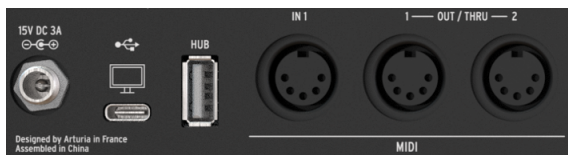
Rear panel ADAT and Word Clock ports

The AudioFuse 16Rig will grow with your setup, offering the ability to send and receive multi-channel digital audio at high resolution. You can use these facilities to connect external preamps (like the AudioFuse 8Pre), or route audio to and from any gear with ADAT optical connections such as standalone A/D and D/A converters.

The four ADAT ports use optical Toslink connectors. They can provide either 16 channels of I/O at 44.1 or 48 kHz, or 8 channels of I/O at 88.2 or 96 kHz.

Underneath are BNC connectors for Word Clock In and Out. They're internally terminated so you don't have to add an external terminator when using them. These jacks assure that all of your digital gear is following the same clock signal, to maintain the highest possible fidelity.

4.2.4. MIDI, USB, and Power



Rear panel MIDI, USB, and power connections

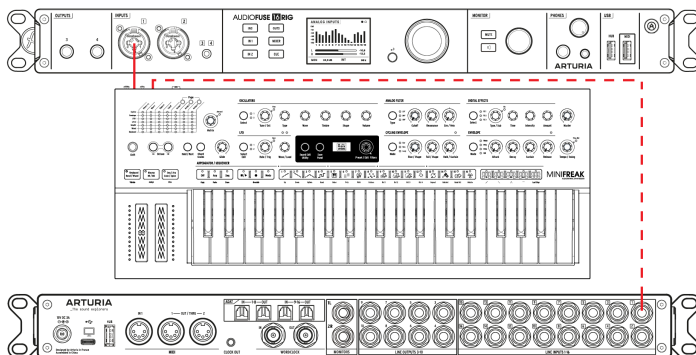
The AudioFuse 16Rig has one MIDI In and a pair of assignable MIDI Out/Thru jacks, all on 5-pin DIN connectors. A 3.5mm jack next to MIDI Out/Thru 2 provides an analog MIDI clock signal for use with certain analog synths, drum machines and modular synthesizers.

To the left of the MIDI jacks are a rear-panel USB-A port that acts as part of the interface's built-in hub, a USB-C port that connects to the host computer for interface control and DAW communication, and a locking connector for the AudioFuse 16Rig's power supply.

5. PUTTING YOUR AUDIOFUSE 16RIG TO WORK

Here are some common ways in which you'll use your AudioFuse 16Rig.

5.1. Recording line level devices

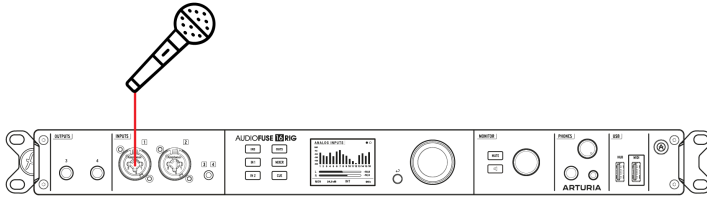


We'll start with the most common use case: hooking up *line level* devices. This includes synthesizers and other keyboards, drum machines, samplers, beat production workstations, etc. These are usually the "hottest" input signals you'll be working with, and shouldn't need much in the way of added gain.

1. Connect your gear to any of the line inputs on the front or rear panel. Remember that if you connect gear to the front panel inputs, the corresponding rear panel inputs will be ignored unless you use the **REAR** function.
2. Press the **INPUTS** button to bring up the Analog Inputs page. Play each instrument, one at a time, and watch the meters.
3. Ideally, the loudest parts of your performance should cause a channel's VU meter to peak between -10 and -6dB.
4. Scroll to select the channel you want to adjust. Watch the numbers under the meters; the currently selected channel will be white. Click the Encoder to select that channel.
5. Set the gain on the channel to get the proper meter reading (the channel meter is on the left of the screen). If your line-level device has a volume control, we recommend you adjust the output level on the device itself, before increasing the gain on the interface. This will get you the cleanest sound; use gain control on the AudioFuse 16Rig only for devices with no level control or very weak output signals.
6. If your sound source is so loud that it overloads the VU meter even at the lowest gain setting, press the Pad button to switch on the 20dB input gain reduction.
7. You can also flip the phase of the input, stereo-link odd/even channel pairs (e.g. 5 and 6 or 9 and 10), and/or force the interface to use the rear panel inputs for Channels 1-4 even if there's something plugged into the front panel inputs.

And there you go! Your instruments are ready to record. Please refer to your software's user guide if you have any questions about how to record.

5.2. Recording a microphone



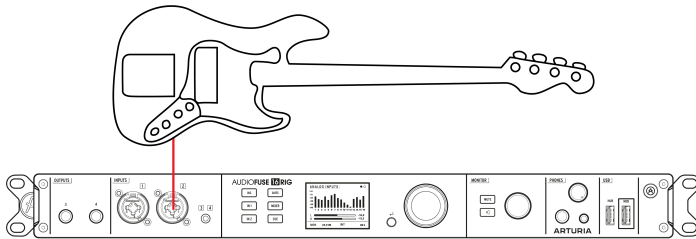
Microphones (especially ribbon mics) generate extremely weak signals, which are referred to as *mic level*. They require the most gain to be heard clearly, so it's a good thing that the mic preamps on Inputs 1 & 2 have a lot of gain and very low noise!

1. Connect your microphone to one of the mic inputs on the front panel.
2. AudioFuse 16Rig will auto-detect that you have connected an XLR cable and that input channel will be given a **48V** button in both the hardware and in AFCC. Switch on 48V phantom power if your mic needs it.
3. Press **IN 1** or **IN 2** to immediately access the appropriate channel. If you've connected an XLR cable, the input channel will now be labeled **FRONT MIC** rather than **ANALOG IN**, and the channel in AFCC will have a microphone icon.
4. If your microphone requires phantom power, press the 48V button to switch on phantom power.
5. Talk or sing into the mic while adjusting the gain. The loudest parts of your performance should cause the channel's VU meter to peak between -10 and -6dB.
6. If your sound source is so loud that it overloads the VU meter even at the lowest gain setting, press the Pad button to switch on the 20dB input gain reduction.
7. If your sound seems weak or "phasey", try flipping the phase with the button labeled **Ø**.

That's it! The microphone input should be available in your music software and ready to record.

i NOTE: If you click and hold the encoder while the interface is showing the display page for a particular channel, the CLIP LED will be cleared. If you do this on the main INPUTS page, all CLIP LEDs will be cleared. This is particularly important if you have the Clipping Reset Time set to "Infinite" in AFCC.

5.3. Recording a Guitar or Bass

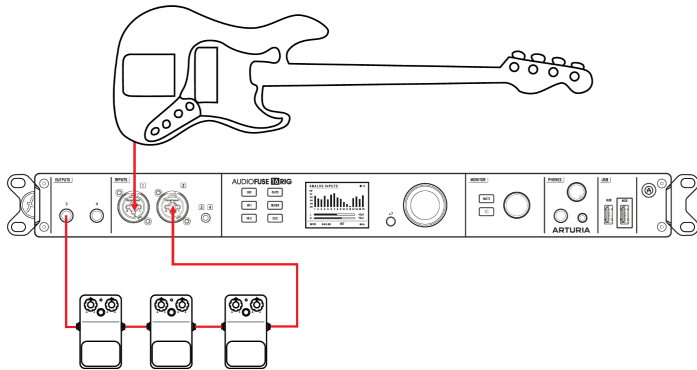


Guitars, basses, and other instruments with magnetic or piezoelectric pickups run at *instrument level*. Signal levels will be hotter than mic level, but usually not as hot as line level. In addition, the pickup needs to see a very high input impedance, or its tone could be compromised.

1. Connect your guitar or other instrument to either of the front panel inputs with a standard 1/4" cable.
2. AudioFuse 16Rig will auto-detect that you have connected a 1/4" cable and that channel in AFCC will add a button labeled **INST**. By default, channels are set to receive a line-level signal; you'll need to click the INST button for instrument-level use.
3. Press **IN 1** or **IN 2** to immediately access the appropriate channel. It will be labeled FRONT LINE rather than ANALOG IN.
4. Scroll to the INST button and click it.
5. Play the instrument while adjusting the gain knob. The loudest parts of your performance should cause the channel's VU meter to peak between -10 and -6dB.
6. If your instrument is so loud that it overloads the VU meter even at the lowest gain setting, press the Pad button to switch on the 20dB input gain reduction.

Your instrument is now set up to feed the clearest, cleanest signal to your software.

5.4. Reamping a Guitar or Bass



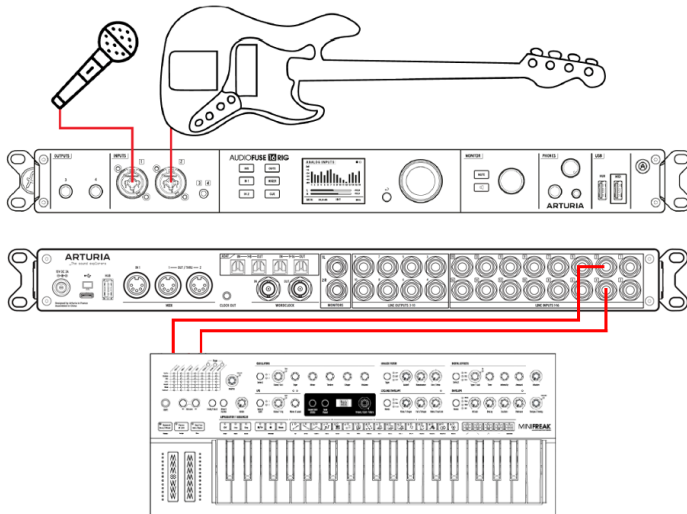
Reamplification or "reamping" is a way to tweak the tone of a recorded track after the fact. The guitar, bass, or other instrument signal is recorded as cleanly as possible, with no effects or amplifier. Then, later on, that signal can be sent back out to a pedalboard or miked amp. The engineer can now tweak the tone all they want – since the performance is already recorded, the guitarist doesn't have to play it over and over.

The trick to reamping is that the signal out of the interface has to convince the pedals or amp that it's coming from a guitar pickup. Therefore it has to present the right signal level and output impedance. Commercial reamplification boxes (sometimes nicknamed "reverse DI boxes") have been around for many years, but the AudioFuse 16Rig has the capability built into the front panel outputs 3 and 4.

1. Record your guitar or bass part clean, as described above.
2. Connect a cable from one of the front panel outputs to your pedalboard or amp.
3. Press the **OUTPUTS** button. Scroll to select Output 3 or 4 and click to access the Analog Out settings pages.
4. Scroll to the third page and use the dropdown menu to select Reamp. A button will appear to let you lift the output ground if you hear hum.

That's all there is to it! You can now assign your DAW to play back your performance through your pedals and amps, and tweak your tone to your heart's content.

5.5. Creating a Mix



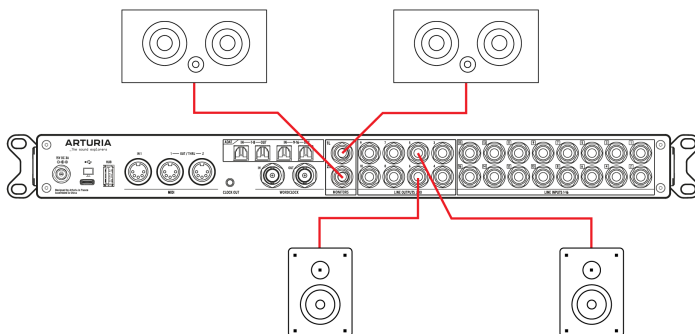
The AudioFuse 16Rig can combine signals with two different mixers: the *Main Mix*, which includes up to four *Aux Sends*, and the *Cue Mix*, which does not. Typically you'd use the Main Mix in the same way you'd use any conventional mixer, while the Cue Mix can feed a headphone amplifier so performers can hear "more me" as they play.

1. Connect and adjust input levels for your line-level sources, microphones, and/or guitars, as described above.
2. Press the **MIXER** button to access the Main Mix or the **CUE** button to access the Cue Mix. (We'll talk about the Main Mix in the following steps, as the Cue Mix is identical but has no Aux Sends.)
3. Scroll to the **MAIN CHANNELS** pages to select which inputs will be present in your mix. The current input will be highlighted in blue; press the encoder to select the input and it will light up solid white. Note that all inputs have the option of being selected as mono channels or linked stereo pairs.
4. In addition to analog sources, there are selectors for ADAT inputs from the rear-panel ADAT inputs and USB playback channels from your recording software. These can also be selected as mono inputs or stereo pairs.
5. Now, return to the **MAIN MIXER** page. You should see all of your added channels, each with a fader level indicator, level meter, and Mute and Solo indicators, as well as meters for the four Aux buses and the Main Mix on the right. If there are more than 8 channels in your mix, subsequent channels will be shown on extra pages after the first one.
6. Scroll to a channel to select it, and click the Encoder to enter its channel settings. Select the parameter you want to adjust, click on it, edit it using the main encoder, and click again to confirm and move on.
7. Repeat this process for all channels, and adjust the stereo level and Aux return levels on the **MAIN MASTER** page.

This method lets you adjust a mix on any or all channels from the front panel of the AudioFuse 16Rig without a computer, and up to eight Main mixes can be saved and loaded as Presets.

i As you can imagine, this process can get pretty tedious. It's much faster and easier to set up and save Main and Cue mixes using the [AudioFuse Control Center \[p.20\]](#) software. Also, channel routings for the various Aux and Mix outputs can only be set within AFCC.

5.6. Using Two Sets of Speakers

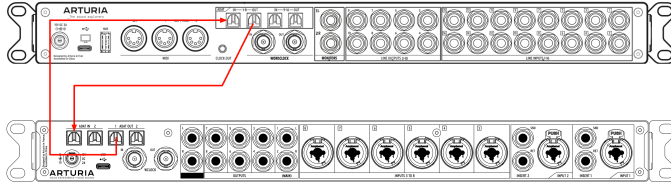


It's quite common for studios to have two sets of monitor speakers set up at any given time. In addition to the main set of monitors, a second set can be used as "check speakers" to hear what the mix sounds like on, say, a car stereo or cheap home entertainment system. Alternatively, a studio might have a very loud and bass-heavy PA set up to impress clients.

Whatever use you might have for a second set of speakers, AudioFuse 16Rig can easily accommodate your needs.

1. Connect your primary speakers to the rear panel Monitor Outs.
2. Select any pair of rear panel outputs to feed your second set of speakers.
3. Press the OUTPUTS button, select MON, and click to enter the Monitor Settings pages. Scroll to the third page, labeled A/B Conf.
4. Here you can activate or deactivate A/B switching as a function using the button with the power symbol. The A/B button switches between the Monitor outs (unlit) and the B Speaker outs (lit). The drop-down menu lets you choose which pair of rear panel outputs will feed the B Speakers.
5. If you wish to make A/B switching a one-button affair, you can set up the front-panel Monitor button to do this. To do this, enter the Settings section from the carousel, and go to the Monitor Settings page. There, set the Monitor Button Mode to "A/B Speaker".

5.7. Connecting to ADAT Digital Audio Devices



The AudioFuse 16Rig can support up to 16 channels of ADAT digital I/O by using the four optical Toslink connectors on its rear panel. Each port carries eight channels of audio in one direction when working at 44.1 or 48k sampling rates. Digitally connected devices can be synchronized through the ADAT connection itself or via the nearby Word Clock BNC ports.

i Note: This section describes how to connect devices using only ADAT cables (i.e., audio and digital sync signals are both transmitted together through the optical cable). For larger, more advanced studios with lots of digital equipment, Word Clock synchronization may be required or preferred. This is covered in the next section.

In the illustration above, we have an AudioFuse 16Rig (top) connected to an AudioFuse 8 Pre (bottom). ADAT optical cables carry digital audio to/from each unit. Synchronization signals are also carried along the cable.

When working with digitally connected audio devices, it's vital that there is only one clock source. Having all devices synchronized to one primary clock prevents clicks, pops, and other audio artifacts that can arise from slightly mismatched clocks.

In this setup, the AudioFuse 16Rig acts as the primary reference clock and so its "Clock Source" parameter is set to "Internal." As a result, AudioFuse 8Pre must have its own clock (the "Sync" parameter on its front panel) set to "ADAT" so that it receives and synchronizes itself to clock information coming from the AudioFuse 16Rig.

To do this:

1. Connect your digital audio devices to each other using the rear panel ADAT ports. The output(s) of the AudioFuse 8Pre connect to the input(s) of the AudioFuse 16Rig and vice versa.

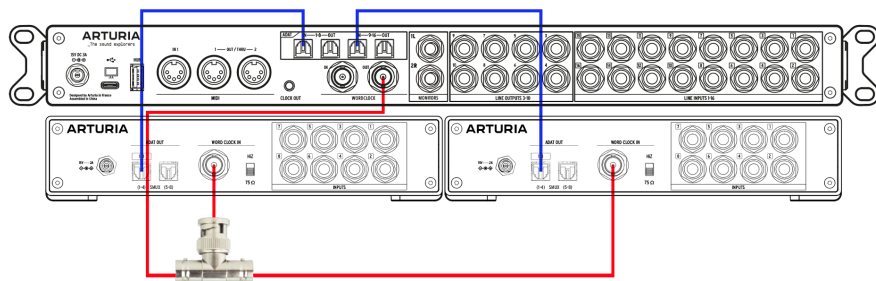
i Note: Each device has two input ports and two output ports. In this example, the second port is only necessary if you are working at higher sampling rates (88.2 or 96k). Since we are working at standard (44.1 or 48k) sampling rates, only the first input and output ports are used.

2. Ensure that the clock source of AudioFuse 16 Rig is set to "Internal" while the clock source of AudioFuse 8Pre (labeled "Sync" on the front panel) is set to "ADAT".

i Note: If AudioFuse 16 Rig is connected to a computer, the Clock Source and Sample Rate parameters in the hardware may be grayed out. If so, you need to change the Clock Source and/or Sample Rate parameters from the AFCC or your DAW.

At this point, the two devices are digitally connected and properly synchronized. In your music software, record-enabling the AudioFuse 16Rig's ADAT inputs 1-8 will let you capture the output of AudioFuse 8Pre and all its pristine preamps.

5.8. Working with multiple ADAT Devices using Word Clock Sync



The previous section described how to connect two digital devices to each other using ADAT optical cables only. In that example, digital synchronization was established through the same optical cable that carries audio signals. However, larger studios with many pieces of digital equipment may need to use Word Clock to sync devices together.

This example is similar to the first example above, except that we now have two AudioFuse X8 IN units connected to the AudioFuse 16Rig through ADAT. This greatly expands the already impressive I/O capabilities of the 16Rig by adding 16 channels of line-level inputs. However, in this setup we can't rely on the ADAT connectors for sync, because the AudioFuse X8 IN has no ADAT inputs and can't receive sync signals that way.

This is not a problem, however, since we can use Word Clock to synchronize the three devices. In fact, Word Clock can be used to sync any number of digital devices with high precision.

As in the previous example, this setup will use optical cables to carry audio signals – this time, from each AudioFuse X8 IN expander to the AudioFuse 16Rig's two ADAT IN ports. Unlike the previous example, we're going to use Word Clock to synchronize all three devices.

In the diagram above, the optical audio cables are shown in blue while the Word Clock cables are shown in red.

Word Clock is carried over special cables using BNC connectors. These cables must have an impedance of 75 ohms (they'll be marked as such when you buy them) – incorrect impedance matching will cause the sync signals to be corrupted. A single cable can be used between two devices with BNC Word Clock inputs or outputs, but to sync multiple devices, BNC cables must be daisy-chained using a T-connector (sometimes just called a 'BNC Tee'). These must also be rated at 75 ohms.

T-connectors can come in various shapes; one style is shown in the diagram above, but a more common style – that really does look a bit like the letter T – is this one:



All that's important is that the T-connector is rated at 75 ohms.

Here's how to set up Word Clock sync in this example:

1. First we make the connections as shown in the diagram above, making sure to have the T-connector attached to the rear panel of the first AudioFuse x8 IN in the chain (the one connected to the AudioFuse 16Rig).

2. The AudioFuse 16Rig will be sending Word Clock signals to the two AudioFuse X8 IN units. Since its clock output will control the other devices, we must set its **Clock Source** parameter to *INTERNAL*.

i Note: If AudioFuse 16Rig is connected to a computer, the Clock Source and Sample Rate parameters in the hardware may be grayed out. If so, you'll need to change the Clock Source and/or Sample Rate parameters from the AudioFuse Control Center software or from your DAW, depending on your computer setup.

3. The first AudioFuse X8 IN will need to accept Word Clock signals without corrupting signals that are going down the cable to the second AudioFuse X8 IN. To do that, set the rear-panel switch to *HiZ*.

4. The second AudioFuse X8 IN is at the end of the chain, so we set its rear-panel switch to $75\ \Omega$ (75 ohm impedance).

5. Finally, we configure both AudioFuse X8 IN units to accept Word Clock sync by pushing the front panel **SYNC** buttons to select *WORD*, as shown here:



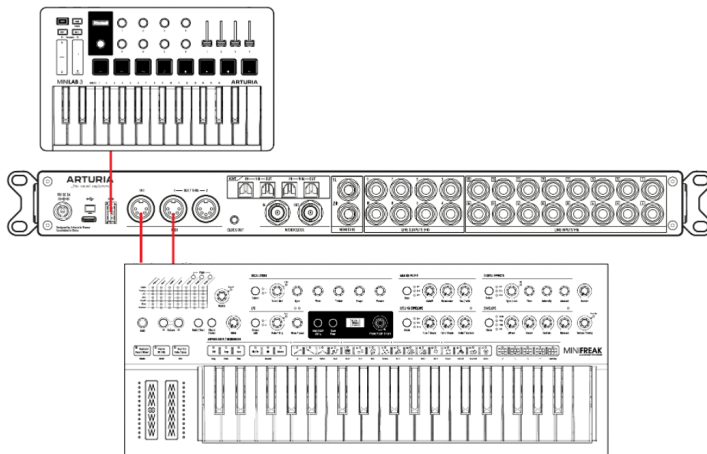
i Note that we must do this last, as the **SYNC** button won't let you select *WORD* if there is no BNC cable connected.

At this point, the three devices have their audio signals connected through ADAT, but are properly synchronized using Word Clock. You're ready to record all of these input signals simply by connecting the AudioFuse 16Rig's ADAT 1-8 and ADAT 9-16 inputs to audio channels in your DAW software.

i If you're trying to set up a system for multiple *output* channels using the AudioFuse X8 OUT, the Word Clock setup is similar. It's possible for the AudioFuse X8 OUT to use ADAT sync, but you can choose *WORD* with the front panel **SYNC** button just as you can on the X8 IN. The rear-panel impedance switches should be set the same way as described above.

i Note that this section only covers the basics of working with Word Clock and there are other (more advanced) ways of synchronizing digital devices to each other. To learn more about this topic, we recommend taking a look at [this article](#).

5.9. Connecting MIDI and USB devices



AudioFuse 16Rig lets you connect both MIDI and USB devices. MIDI inputs and outputs let you connect MIDI synthesizers, drum machines, control surfaces, etc. to the AudioFuse 16Rig to play and record MIDI data. The interface also has multiple USB ports for connecting extra hardware like USB MIDI controllers, software copy protection keys, memory sticks, and more.

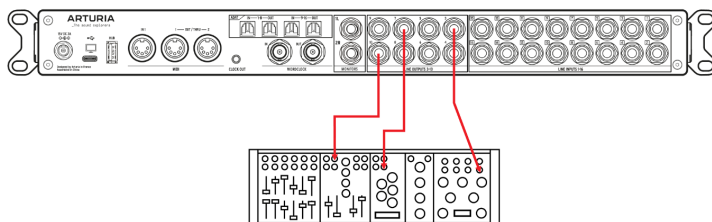
1. Connect any MIDI devices you wish to use to the rear-panel 5-pin DIN **MIDI IN** and dual **MIDI OUT/THRU** ports.
2. Connect the AudioFuse 16Rig to your host computer via the rear panel USB-C port.
3. If you would like to connect multiple USB devices to your computer but you're short on USB-A ports, use the front and rear panel **HUB** connectors to add single USB devices or powered USB hubs. This is especially handy for modern laptops that might have only a few USB-C ports to spare, and might have no USB-A ports at all.

4. The front panel also has a **MIDI** USB-A port. If you connect a USB MIDI device like a keyboard or control surface to this port, it will operate the AudioFuse 16Rig and any connected MIDI gear even when the interface isn't connected to a computer. This can be useful for users that are interested in DAWless jamming or performing live on stage without a laptop. Audio and MIDI always flow through the interface, even if a computer isn't connected. In fact, you can control AudioFuse 16Rig's Main mixer using MIDI messages from any connected controller.

5. Click on the **MIDI** menu to select how the AudioFuse 16Rig will route and respond to MIDI.

If you're going to be recording and playing back MIDI via recording software, consult that software's documentation to learn how.

5.10. Using the AudioFuse 16Rig as a Control Voltage source



AudioFuse 16Rig's outputs 3-10 are DC Coupled. This means these outputs are designed to output not only audio signals but also inaudible "control voltage" signals that are understood by analog synthesizers.

Using the DC coupled outputs and specific DAW plug-ins such as Ableton Live's CV Tools, you can easily output CV signals from your DAW to control your modular and semi-modular synthesizers, such as Arturia's MiniBrute.

No configuration is needed in the AFCC software to do this, but you may need to adjust the output level and use specific mono cables to connect the AudioFuse 16Rig's outputs to your synths' CV inputs.

6. GETTING AROUND

Now it's time to dive in and learn how to use the AudioFuse 16Rig. The chapters after this one will serve as a step-by-step reference to all of the hardware's functions.

But first, it would help to learn a few basics about how to navigate the front-panel interface controls, and some of the most important parts of the interface's functions and menu structure.



The physical controls on the front panel

6.1. Physical Controls: How do I get from here to there?

6.1.1. The Encoder and Back Button



The **Encoder** controls nearly everything on AudioFuse 16Rig.

Some terminology: You *turn* the encoder to find the page or function you want, *scroll* with it to move through options on a page, and press (*click*) it to turn a function on or off.

The small **Back** button is below and to the left of the Encoder. Press it to cancel a current operation or to step back one level in the menu hierarchy.

6.1.2. Quick Access Buttons



These buttons let you access the most commonly needed functions in your usual workflow. They include:

- **INPUTS (INS):** the Analog Inputs metering screen

- **IN 1:** Controls for Analog Input 1
- **IN 2:** Controls for Analog Input 2
- **OUTPUTS (OUTS):** The Analog Outputs metering screen
- **MIXER:** The Main Mixer screen
- **CUE:** The Cue Mixer screen

Note that some of these screens are only the first of several pages (see below) of functions. In those cases, the button takes you to the first (and most commonly needed) page.



If you turn the Encoder while holding down the **INS**, **OUTS**, **MIXER**, or **CUE** button, the display will immediately jump to the next page in that menu rather than scrolling through all the options on that page. This is a big timesaver!

6.1.3. AudioFuse 16Rig Hardware Shortcuts

Shortcuts	Results
Long Press MAIN Knob	Reset clip indicators
INS / OUTS / MIXER / CUE + MAIN Knob	Quick navigation between subpages of INS / OUTS / MIXER / CUE
Monitor Button + MAIN Knob	Set Monitor Button Function
IN1 / IN2 + MONITOR Knob	Set Input 1 / 2 Preamp Gain
INS / OUTS + MONITOR Knob on a specific Input / Output view	Change the current input / output gain
MIXER / CUE + MONITOR Knob on Mixer / Cue view	Change the Mixer / Cue Master Volume
MIXER / CUE + MONITOR Knob on Mixer / Cue channel view	Change the current channel gain
Double-click Back button	Return to main menu

6.1.4. The Mute and Monitor buttons



MUTE button, Monitor button, and Monitor knob

The **Monitor** knob controls the level coming out of the rear-panel Speaker outs (Or the B Speaker outs, if A/B Speaker monitoring is enabled). It's accompanied by a **MUTE** button that turns red when in use.

Below the **MUTE** button is a programmable **Monitor** button with a speaker icon. It is lit in a different color for each of the functions that it can perform, and blinks bright/dim when activated.

You can assign the button to any of the following three options:

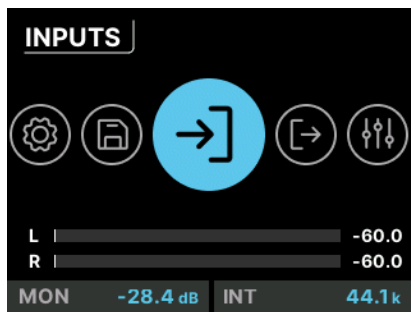
1. **Mono (green)**: sums the signals of Left and Right outputs and sends them to both speakers.
2. **Dim (orange)**: turns down the speaker output by a designated amount that can be set in the AudioFuse Control Center application.
3. **A/B Speaker (blue)**: mutes the Speaker outputs and switches on a pair of rear-panel analog outputs designated by the user - any analog output pair on the rear panel can be designated as the B Speaker output.

 Remember that the A/B Monitoring functionality must be switched on for this third option to work. Otherwise, the button is ignored.

6.2. Screen Layout: what will I see when I get there?

6.2.1. The Carousel

This is the top level of the menu hierarchy. You can recognize it by the large circular icons. If you're not seeing this (because you're on a different screen), just double-click the Back button to bring you back to the Carousel.



The Home screen

Once on the Carousel, you can turn and click the icon for the function *Menu* that you want to access. These are represented on the Home screen by large icons and pop-up labels. They will loop around to the beginning if you keep scrolling - like a carousel (merry-go-round).

The seven menus include:

- **INPUTS**: Metering and controls for the analog and digital inputs.
- **OUTPUTS**: Metering and controls for all analog and digital outputs.

- **MIXER:** Metering and controls for the main Mixer, and to enable/disable channels as needed.
- **CUE:** Metering and controls for the Cue Mixer, and to enable/disable channels as needed.
- **MIDI:** Settings for MIDI input and output ports and routing, as well as a comprehensive MIDI Activity monitor.
- **SETTINGS:** Clock, Monitor, and global preferences, as well as a System Info page to show basic information about your particular AudioFuse 16Rig's hardware.
- **PRESETS:** Controls to save and load presets. There are 8 slots available.

6.2.1.1. Information Displayed on the Carousel

There are certain bits of information that are always visible at the bottom of the Carousel screen:

- Meters for the Main outputs, with levels shown in dB
- Level setting for the Monitors (set with the Monitor knob)
- Clock source and sample rate

6.2.2. Pages

Many menus have multiple *pages*. These are indicated by small circles at the top right of the screen.



Page icons

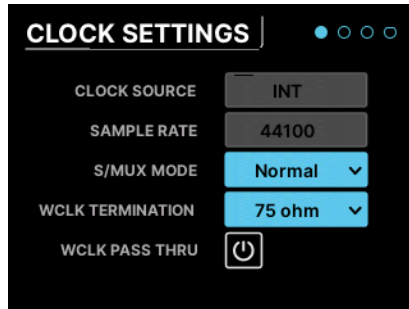
The number of circles naturally indicates the number of pages in that section, and the page you're currently on is highlighted. In most cases, if you scroll past the last page, you will be taken to the first page.

6.2.3. Scroll, Select, Activate

Once you're on the page you want, use the Encoder to scroll to the function you want to work with. Then click to change it.

6.3. "Why is this function greyed out?"

Sometimes you'll run into a function that you can't select. This isn't a bug! The interface is smart enough to know when a feature can't be used, and will tell you that by making it inaccessible.



Parameters on a typical screen page

The most obvious example is Inputs 1 & 2. If nothing is plugged into one of the front panel inputs, none of the functions specific to that input will be available. For example, if you connect an XLR cable to the front panel input, a 48V switch will appear. If there is no mic connected, that button will not be shown since there is no need for it.

6.4. Don't forget the software!

Remember that all of these functions and more are available in the AudioFuse Control Center software, in a neat graphical interface with instant visual feedback. While we recommend that you read the software's user manual, you will find that many of the features there are self-explanatory once you've learned your way around the hardware.

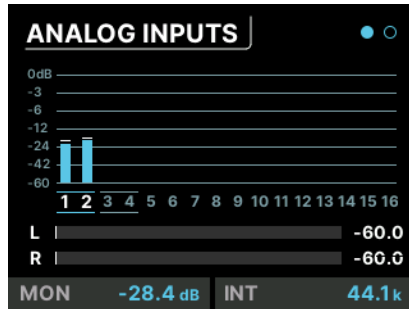
7. MENUS AND FUNCTIONS

The AudioFuse 16Rig's Carousel screen gives you access to seven different sets of functions. In this chapter, we'll go through them one by one, and show which functions are available on each page.

7.1. INPUTS

7.1.1. ANALOG INPUTS / ADAT INPUTS

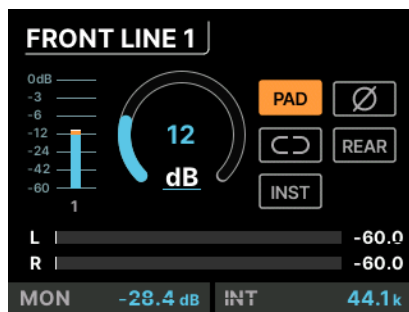
When you click on the **INPUTS** icon or press the front panel **INPUTS** button, you're taken to the **ANALOG INPUTS** page:



Scrolling from left to right highlights each channel. When you get to the end, the view shifts to the next page, which is **ADAT INPUTS**.

For all analog inputs, you can click on the number of the channel you want to edit (highlighted in white). The second page displays incoming ADAT levels, and is there for visual feedback only (it does not have any editable parameters).

7.1.2. ANALOG IN / FRONT LINE / FRONT INST / FRONT MIC



The screen can appear in 4 ways depending on what's plugged in. This one is for the FRONT LINE.

Each input channel page has the following features:

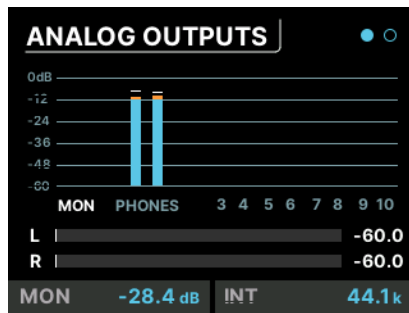
- Level meter
- Gain knob
- Buttons to activate or deactivate
 - Pad
 - Phase Invert (Ø)
 - Stereo Link (chain link icon)

Depending on the channel you're looking at, there may also be other options offered:

- Front panel XLR channels also have a button for 48V Phantom Power
- Front panel 1/4" channels also have a button for Instrument Level (**INST**). This sets the jack's input impedance to 1 million Ohms, which assures that a guitar or bass plugged into the front jack will not have the tone of its magnetic pickups adversely affected by the interface. If you're recording a guitar, bass, or electric piano into one of these jacks, the **INST** setting will assure you the most transparent tone.
- Inputs 1-4 have a **REAR** button, which overrides the front panel input in favor of the rear-panel one. This lets you keep different inputs plugged in on the front and back, and easily switch between them as needed.

7.2. OUTPUTS

When you click on the **OUTPUTS** icon or press the front-panel **OUTPUTS** button, you're taken to the **ANALOG OUTPUTS** page:



Scrolling from left to right highlights each output channel:

- **MON** indicates the Main Monitor output
- **PHONES** indicates the Phones output
- Numbers 3-10 indicate those rear panel outputs

Click on the name or number of the channel you want to edit (highlighted in white).



If a pair of outputs has been designated as the B Speaker pair, there will be a box around those two channels, and clicking on them will bring up the **A/B CONF** page (see below).

7.2.1. MONITOR

Clicking on **MON** accesses the three **MONITOR** pages:

- The first page has a Gain knob (which is set with the front-panel **MONITOR** knob) and buttons for the Mute, Dim, Mono, and A/B functions. Note that A/B will be greyed out if the **A/B** function hasn't been switched on, which is done on the third page as explained below.
- The second page has indicators showing which signals are being sent to OUT 1 and OUT 2.
- The third page is the **A/B CONF** (configuration) page. This page has a button to turn A/B capability on or off, a button to switch between A and B if the function is activated, and a drop-down menu to select the output pair that will feed the B speakers. There's also a Trim knob for adjusting the B Speaker level to match that of the Main Monitors.

7.2.2. PHONES

Clicking on **PHONES** accesses the three **PHONES** pages:

- The first page has a level meter and Gain knob for the Phones output signal. This onscreen parameter can be set with the front-panel **PHONES** knob.
- The second page has indicators showing which signals are being sent to the headphones (OUT 11-12).
- The third page has a drop menu that lets you adjust the impedance of the headphone output to match whatever headphones you happen to be using. Setting this appropriately means you can use low, standard, and high impedance headphones with ease.

7.2.3. ANALOG OUT 3 & 4

Analog Outputs 3 and 4 offer more capabilities than the other analog outputs. They are available not only from the rear panel, but also from two jacks on the front panel that can be set to several different functions that you'll find handy when working with external gear.

Clicking on Output 3 or 4 accesses three **ANALOG OUT** pages, all of which have output metering:

- The first page has a Gain knob and a Stereo Link button
- The second page has an indicator showing which signal is being sent to that channel's **OUT** jack.

- The third page has a drop-down menu that lets you select what the front-panel **OUTPUT** jack will transmit. Options are:
 - **Line:** A conventional line-level output.
 - **Reamp:** Sends the chosen source out through the front-panel output at the appropriate level and impedance to feed a pedalboard or amplifier for reamplification.
 - **DirectThru:** Creates a true zero-latency connection from the front-panel **INPUT** jacks 1 and 2 to the **OUTPUT** jacks 3 and 4. This is important if you want to record a clean guitar signal to your computer while also playing through your pedals and amp; this connection taps into your cable run without introducing latency that can affect your playing.
 - **AltPhones:** Converts each output on the front panel into its own stereo headphone jack. These outputs must share the same source (i.e., they must mirror each other and you can not have separate things playing from output 3 and 4), and their output level is set using the Gain knob on page 1. However, the AltPhones do not have to mirror the main headphone outputs of AudioFuse 16 Rig. You can set their source in the Routing Matrix.

7.2.4. ANALOG OUT 5-10

Clicking on any other channel number accesses the two ANALOG OUT pages, both of which have output metering:

- The first page has a Gain knob and a Stereo Link button
- The second page has an indicator showing which signal is being sent to that channel's **OUT** jack.

7.3. MIXER

Clicking on the **MIXER** icon, or pressing the front panel **MIXER** button, takes you to the **MAIN MIXER** page.



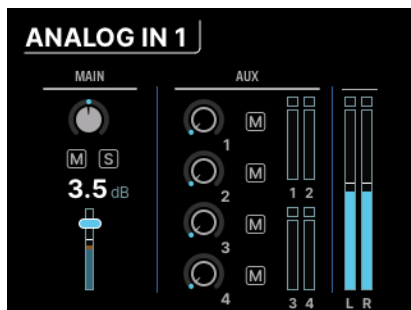
Each channel in the mix has its own indicators for level, mute, and solo. There are also meters for the Aux sends and L/R Main mix. Analog channels come first, then ADAT channels, then USB channels.

If there are more channels in the mix than there is room on the screen, scrolling to the right will move you to the next screen and more channels.

Note that the only clickable items in this view are the channel selectors. If you want to change something like volume, mute or solo, click on any channel number to select it for adjustment.

i If the mixer appears empty, it's because no channels have been added yet. You can add channels by going to the **MAIN CHANNELS** selectors (pages 3 and 4 in the **MIXER** section) and adding whichever channels you would like to have available in the mixer.

7.3.1. MIXER CHANNEL SETTINGS



These screens are labeled by the input type and channel numbers, e.g. ANALOG 3-4 or USB 5-6.

On the left, you'll find the usual channel fader controls: pan, mute, solo, and fader level. If two channels have been stereo linked, there will be two pan controls for individual adjustment.

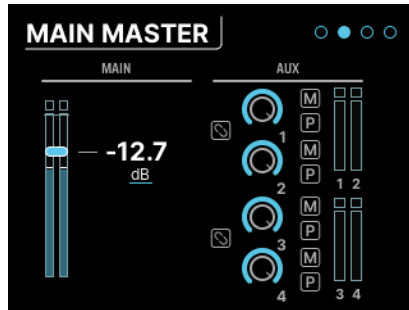
On the right is the Aux section, where the send amount and muting for the four Aux sends can be adjusted. Finally there are meters for the Auxes and Main mix.

Press the Back button to return to the MAIN MIXER page.

i Remember that if you hold the **MIXER** button while turning the encoder while on a particular mixer channel's page, you'll jump to the next channel's corresponding page. This lets you quickly scroll through channels to check or adjust parameters like pan position or Aux send levels. When you're holding down the **MIXER** button, the **MONITOR** knob directly controls the channel level. That means you can rapidly jump from channel to channel and tweak levels, just by holding down the **MIXER** button and going back and forth between the Encoder and the **MONITOR** knob!

7.3.2. MAIN MASTER

Scroll past the last Main Mixer channel selector to access the second page of settings – the MAIN MASTER page.

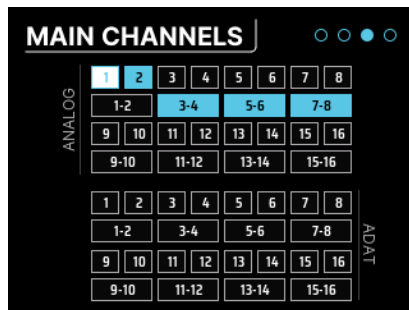


This page shows the Main Master fader on the left, and the Master Aux controls on the right. Each Aux has its own level and mute controls, plus a switch to set the Aux to be pre-fader.

Aux 1 & 2 and/or Aux 3 & 4 can be linked to become a stereo Aux if desired, using the two Stereo Link buttons on this screen.

7.3.3. MAIN CHANNELS

The last two pages of the MIXER menu are the MAIN CHANNELS pages. Here, you can scroll through and select/deselect any channel(s) you wish to have in the Main mix.

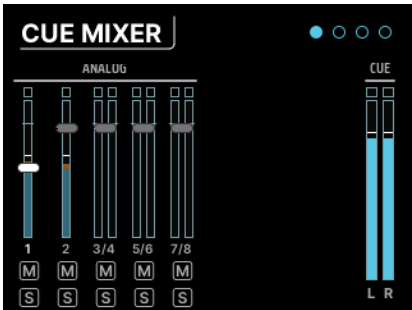


As you can see, each channel is available to select on its own, or you can select a stereo pair of channels instead.

- The first of these two pages is where you select from up to 16 Analog and/or 16 ADAT channels.
- The second page is where you select from up to 32 USB "return" channels. These are channels that come from your computer (for example your DAW's various outputs).

7.4. CUE

Clicking on the **CUE** icon takes you to the **CUE MIXER** pages. The Cue mix is a separate mixer that is often used to give performers a custom headphone mix, perhaps isolating certain elements of the Main mix to be turned up while tracking.



However, it's not just for headphones – you can use this mixer just like any other mixer. For example, if you're working with lots of modules or synthesizers, you can create a stereo submix and then route that mix out of the analog or ADAT outputs for further processing.

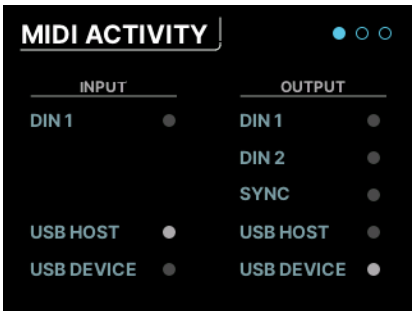
The controls and functions of the Cue mix are identical to those of the Main mix, with the following exceptions:

1. There are no Aux buses.
2. The Cue mix is not always active. It must be switched on in order to work; this is done either by clicking the CUE button in the Cue Mix tab of the AFCC software or by double-clicking the front panel **CUE** button, which lights up red while the Cue mix is active. Double-click again to turn it off.

7.5. MIDI

Clicking on the **MIDI** icon in the carousel accesses the **MIDI** menu, which has 3 pages:

7.5.1. MIDI ACTIVITY



This page has a set of MIDI activity indicators, that light up when MIDI signals are received from the appropriate inputs or sent to the corresponding outputs.

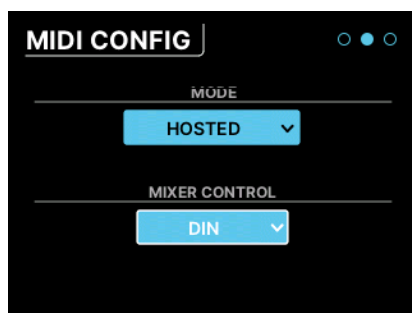
INPUT sources include:

- **DIN1**: the rear panel 5-pin DIN **MIDI IN**
- **USB HOST**: the front panel USB port labeled **MIDI**
- **USB DEVICE**: MIDI messages coming from your music software and/or computer (via USB)

OUTPUT sources include:

- **DIN1** and **DIN2**: the two rear panel 5-pin DIN **MIDI OUT/THRU**
- **SYNC**: the rear panel **CLOCK OUT** minijack
- **USB HOST**: the front panel USB port labeled **MIDI**
- **USB DEVICE**: MIDI messages going to your music software and/or computer (via USB)

7.5.2. MIDI CONFIG



The **MODE** control is a drop-down menu that lets you set how the AudioFuse 16Rig responds to MIDI.

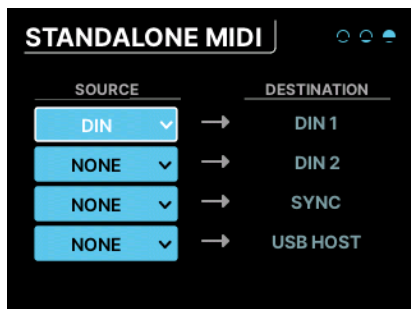
- **STANDALONE**: The AudioFuse 16Rig passes MIDI from sources to destinations without the need for a computer. These routings are set in the **STANDALONE MIDI** page described below.
- **HOSTED**: The interface connects the attached MIDI gear to a host computer. The music software on the host computer can then decide where the MIDI signals are routed.

AudioFuse 16Rig's Main mixer is MIDI controllable. The **MIXER CONTROL** drop-down determines which MIDI inputs port will provide MIDI control messages to the AudioFuse 16Rig's mixer. This lets you quickly manipulate mixes using a control surface rather than using the front panel controls or the AFCC software.

Choices include: **NONE**, **DIN**, **USB HOST**, **USB DEVICE**

 If you're sure you don't want to use MIDI to control the mixer, set **MIXER CONTROL** to **NONE** to avoid accidentally changing mixer settings with your connected MIDI devices.

7.5.3. STANDALONE MIDI



When in standalone mode, the AudioFuse 16Rig needs to know how to route MIDI signals between its various ports. This menu lets you set that up.

There are four fixed destinations: **DIN1**, **DIN2**, **SYNC**, **USB HOST**.

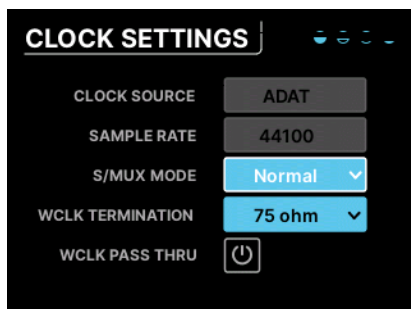
Each can be connected to one of the following sources: **NONE**, **DIN**, **USB HOST**, **USB+DIN**

 What happens if you send two or more sources to one destination? The AudioFuse 16Rig will *merge* their MIDI signals, acting like a sort of MIDI mixer.

7.6. SETTINGS

Clicking on the **SETTINGS** icon accesses four pages of system settings and information displays.

7.6.1. CLOCK SETTINGS



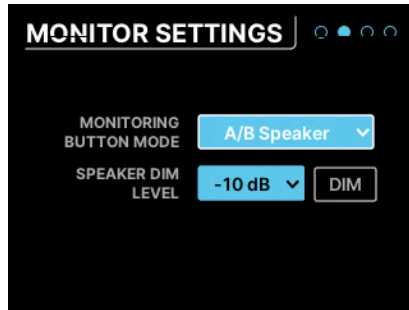
When connecting multiple digital hardware devices in the same rig, solid *clock sync* is vital. If two devices' clocks don't line up and run in sync, audio artifacts will be introduced in the form of clicks, pops and various kinds of noise (known as *jitter*). That's why AudioFuse 16Rig is equally capable of providing tight clocking to other devices, or following the clock generated by another device.

AudioFuse 16Rig can operate on its own internal clock, or use an input clock signal from the ADAT inputs or the Word Clock input. The clock source and sample rate are displayed at the bottom of most screens (and shown here on the **CLOCK SETTINGS** page), and they can be set on the interface itself or using the [AudioFuse Control Center \[p.20\]](#) software.

Parameters on this page include:

- **Clock Source:** This can be set to Internal (**INT**) Clock, **ADAT** Clock, or **WORD** Clock. It defaults to **INT** and cannot be changed if no other digital device is connected to the interface.
- **Sample Rate:** This can be set to 44100, 48000, 88200, 96000, 176400, or 192000 Hz. If the Clock Source is set to **INT** and the AudioFuse 16Rig is connected to a host computer, then the sample rate will be set by the host and cannot be changed within the interface itself.
 - Sample rates above 48000 Hz will be available with limited capabilities.
 - At 88200 or 96000 Hz, the interface offers 16 analog inputs, 12 analog outputs, and 8 ADAT inputs and outputs.
 - At 176400 or 192000 Hz, the interface offers 8 analog inputs and 10 analog outputs, with no digital inputs or outputs.
- **S/MUX Mode:** When set to Normal, the interface automatically detects the incoming ADAT clock and sets its sampling rate accordingly. This is the default and should work well in most cases. However, if you select Force, the interface will interpret incoming data as 88.2/96 kHz even if it "sees" 44.1/48 kHz. This is necessary because some devices' ADAT outputs don't have the right metadata embedded in their signals, so the sample rate can be misinterpreted. The Force function prevents this.
- **Word Clock Termination:** The BNC cables that transmit word clock data are very sensitive to changes in impedance. Usually when a connection terminates, it wants to see a 75 ohm impedance. Under some circumstances, though, a very high impedance is needed. This control lets you choose either 75 ohm or Hi-Z termination.
- **Word Clock Pass Thru:** If your AudioFuse 16Rig is part of a larger word clock network and is synchronized to an external clock, it can pass word clock data through the interface to another device in a daisy chain.

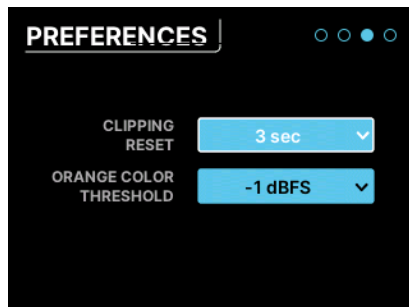
7.6.2. MONITOR SETTINGS



This page contains several speaker-related settings:

- **Monitoring Button Mode:** This dropdown menu selects what the Monitoring button on the front panel does. The chosen function causes the button to light up in a particular color, so you can tell at a glance what it does. The button will blink when activated. Settings include:
 - **Mono (green):** sums the signals of Left and Right outputs and sends them to both speakers. This is useful when checking mono compatibility or when checking for phase-related issues.
 - **Dim (orange):** turns down the speaker output by a designated amount (see below). This lets you take a moment to chat with someone else in the studio with the sound lowered, then return it to its previous level precisely.
 - **A/B Speaker (blue):** mutes the main Speaker outputs and switches on a pair of outputs designated by the user – any analog output pair on the rear panel can be designated as the B Speakers, and trim controls assure that levels match the A Speakers (i.e. the Monitor outputs). Note that this button does not do anything if the A/B function isn't activated.
- **Speaker Dim Level:** This drop-down sets the amount by which the Dim button lowers the speaker level. You can choose either -10dB or -20dB.
- **DIM button:** Engages and disengages dimming without leaving this page. It's a handy way to test which amount of dimming you prefer.

7.6.3. PREFERENCES



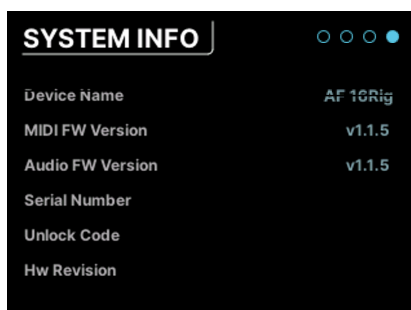
Here you can configure AudioFuse 16Rig to suit your preferences.

Clipping Reset: This drop down sets the amount of time it takes for clipping indicators to automatically clear. This is a global setting that affects all meters in the hardware and also AFCC. Options include 1.5 seconds, 3 seconds, or Infinite.

 Note: If you set the Clipping Reset time to "Infinite" the clip indicators will not clear automatically. In this case you will need to clear the meters manually by clicking and holding the encoder for 3 seconds.

Orange Color Threshold: This drop down sets the point at which the VU meter color changes from green to orange. This is a global setting that affects all meters in both the hardware and also in AFCC.

7.6.4. SYSTEM INFO



This "About" screen lists some of your interface's operational details, such as:

- MIDI Firmware Version
- Audio Firmware Version
- Serial Number
- Hardware Revision

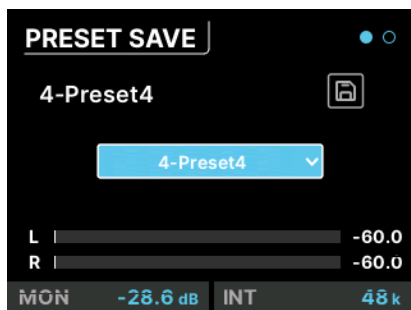
7.7. PRESETS

One of the fun things you can do with the AudioFuse 16Rig is to use it in standalone mode, with no computer attached to it. It's a very handy mixer with lots of routing flexibility, and it doesn't need a host program to be useful in that regard.

However, it does require a place to save all of its routing and mix settings when the computer is not connected. That's one of the many uses for **Presets**.

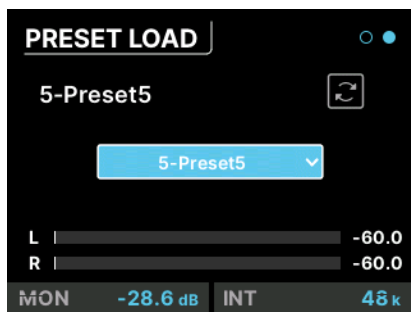
Presets can also be very useful even when the AudioFuse 16Rig isn't in standalone mode; if you have different workflows for recording music, doing voiceovers, reamping instruments, and so forth, you can speed up your work by saving each configuration as a Preset.

7.7.1. PRESET SAVE



This page has a Save button and a drop-down menu for choosing which of the eight available preset slots you want to save to. The current preset is displayed at top left; if there is a star after the name, it means the preset has been altered from the saved version.

7.7.2. PRESET LOAD

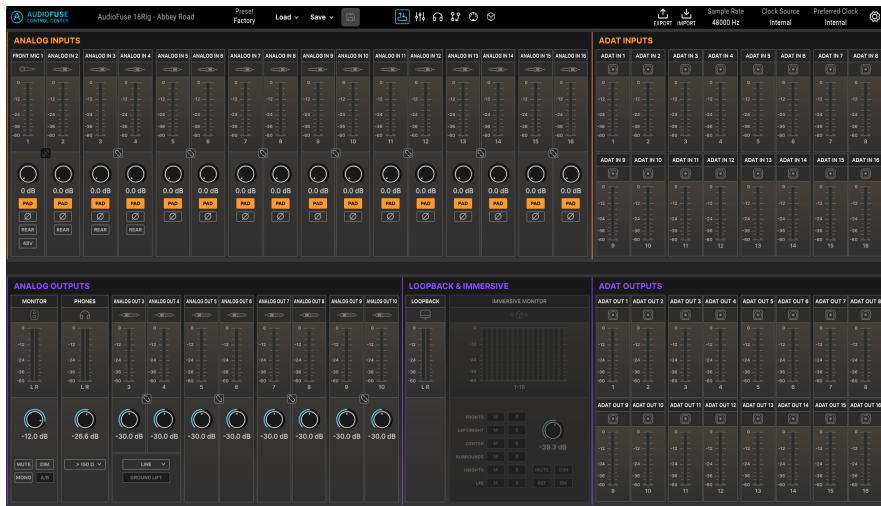


This page is identical to the PRESET SAVE page, but has a Load button instead. Notice that the Load preset actually has nine presets. This is because there is a preset called 'O-Factory' that loads AudioFuse 16Rig's factory defaults and can not be overwritten.



Note that Presets can be renamed in AFCC, so they're easier to differentiate than "Preset-1" or "Preset-8".

8. AUDIOFUSE CONTROL CENTER



While nearly all of the AudioFuse 16Rig's functions can be accessed from the front panel controls, you may find it more convenient to control your interface from a computer. Like all AudioFuse interfaces, the AudioFuse 16Rig can be set up and controlled from the AudioFuse Control Center (AFCC) software.

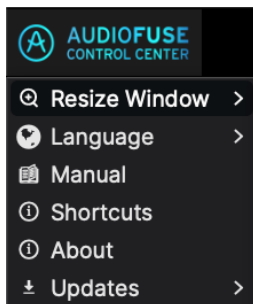
AFCC makes everything easier, from monitoring levels to configuring channels to setting up MIDI controls and mixing. It can control multiple AudioFuse interfaces connected to the same computer; in this chapter, we'll cover its functions and layout for the AudioFuse 16Rig.

8.1. The Upper ToolBar



The **Upper ToolBar** gives you quick access to general controls for AFCC.

8.1.1. AFCC Menu



At the far left of the Upper Toolbar is the **AFCC Menu**. Click on the logo to get a drop-down menu with the following three choices:

- **Resize Window:** Lets you choose an optimal interface size for your computer's display.
- **Manual:** opens this manual.
- **Shortcuts:** Opens a window with all AFCC and AudioFuse 16Rig shortcuts conveniently located in one place. Memorizing these will really help you speed up your workflow in the longrun.
- **About:** opens a window with the current software version and a list of credits. (Now you know whom to thank for this cool program!)
- **Updates:** Enable (On) or disable (Off) the automatic updates check for this software and the device firmware

Next to the AFCC Menu logo is the type of interface connected (in this case, "AudioFuse 16Rig") and the name of your interface, which you can set in the [Settings menu \[p.55\]](#).

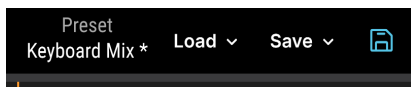
8.1.2. Screen Selectors



The five **Screen Selectors** in the center of the Upper Toolbar let you select the desired screen (view) of the plug-in interface. We will cover these in detail below; they are:

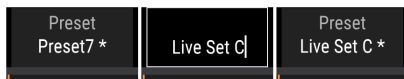
- **Overview**
- **Mixer**
- **Cue**
- **Routing**
- **MIDI**
- **Immersive**

8.1.3. Preset Controls



Next are the **Preset Controls**. These include the following functions:

8.1.3.1. Preset Name [and Rename]



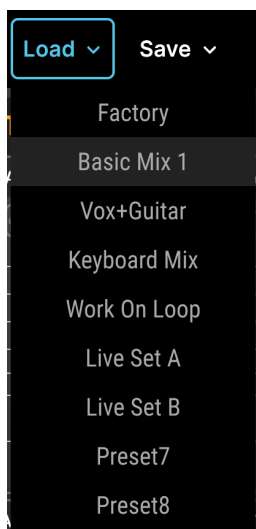
To rename a preset (left), click inside the field and retype (center), then press Enter or click outside the field to save the name (right).

The name of the currently loaded preset is displayed first. If there's a star (*) after the name, it indicates that the current settings in AFCC don't match what's saved in the preset - in other words, there are unsaved edits. These will be lost if you don't save the preset before loading another one.

Click on the preset name to change it to something you'll quickly recognize, like "Synth Mix" or "Voiceover". Preset names can be up to 12 characters long.

i NOTE: Renaming a preset and saving an edited preset are two separate operations - you can save a preset without renaming it, and you can rename a preset without saving any edits you've made (if any). In the left image above, the star in "Preset7 *" indicates that the original Preset7 has been edited. Even with the new name saved, the star is still there to show that the preset itself has unsaved edits. These will be lost if you load a different preset before saving this one.

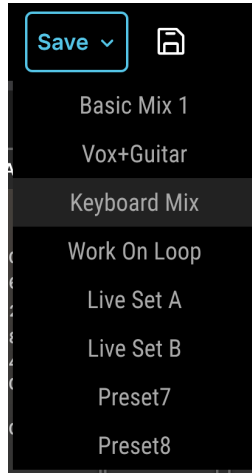
8.1.3.2. Preset Load



The **Load** icon reveals a drop-down menu of the eight presets plus the Factory preset, a default you can always use if you want to start with a clean slate.

i The Factory preset can't be edited and saved, or overwritten. If you start with the Factory preset and edit it to fit your needs, you must save the result into one of the eight preset slots.

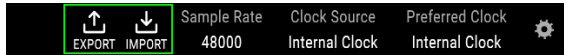
8.1.3.3. Preset Save and Quick Save



*Note the Quick Save
diskette icon next to the
Save button.*

The **Save** icon opens a drop-down menu indicating where an edited preset may be saved. Next to it is the floppy-diskette **Quick Save** icon, which you click to save an edited preset into the same slot.

8.1.4. Import & Export Configuration Settings



The **EXPORT** and **IMPORT** buttons in the Menu bar let you save the current state of your AudioFuse 16Rig to a file, then recall it later on the same unit or on another 16Rig. Configuration files use the .afp extension.

8.1.4.1. What is saved

An export captures every digitally controlled setting of the device, along with its firmware and software options:

- Input: Mic inputs 1-2 (gain, +48V phantom power, pad, instrument Hi-Z mode, phase) ; Line inputs 3-18: pad ; Inputs 1-4: front or rear connector selection ; Pair link on all analogue inputs.
- Main and Cue mixers: input selection, pan, solo, mute and level.
- Immersive mode settings.
- Analogue and digital output routing sources, loopback source and headphone source.
- Main monitoring: mute, mono, DIM, level and alternative speaker sets.
- MIDI control mappings: external control surface mappings and MIDI CC assignments.
- All settings from the Advanced Settings panel.
- Clocking: sample rate, clock source and termination.

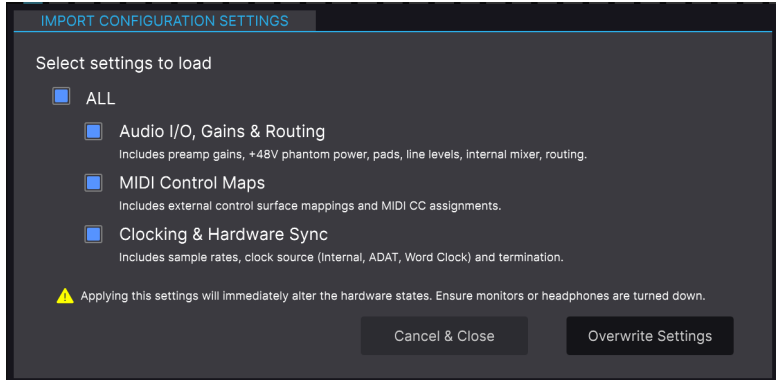
Headphone volume is under analogue control and cannot be recalled by the AFCC, so it is not saved. DAW settings are not included.

8.1.4.2. Exporting your settings

1. Click **EXPORT** in the Menu bar.
2. Choose a name and a location in the save dialog, then confirm.

8.1.4.3. Importing settings

1. Click **IMPORT** in the Menu bar, then select an .afp file.
2. The AFCC checks the file before anything is applied. The file must come from an AudioFuse 16Rig, and its software and firmware versions must match the ones currently in use. If the check fails, the import is aborted and a dialog details the reason.
3. Once the file passes, the Import Configuration Settings window opens.



The import is scoped, so you decide which part of the configuration is recalled. This matters because the three groups change at very different rates:

- Audio I/O, gains and routing belong to the session: they follow the instruments, mics and monitoring setup you are working with that day.
 - Clocking and hardware sync belong to the studio itself. Word Clock, ADAT and sample rate are usually wired once, shared with the rest of your rack, and stay put from one session to the next.
 - MIDI control maps belong to your control surface: they depend on what is connected to the 16Rig, not on the material you are tracking.
4. Click **Overwrite Settings** to apply your selection, or **Cancel & Close** to leave the device unchanged.



⚠: Export your current configuration first if you want to be able to return to it.

Loading a colleague's routing setup should not force your converters onto their clock source, or remap your controller to a surface you do not own. Leave **Clocking & Hardware Sync** and **MIDI Control Maps** unticked and both stay as they are, while everything else is recalled.

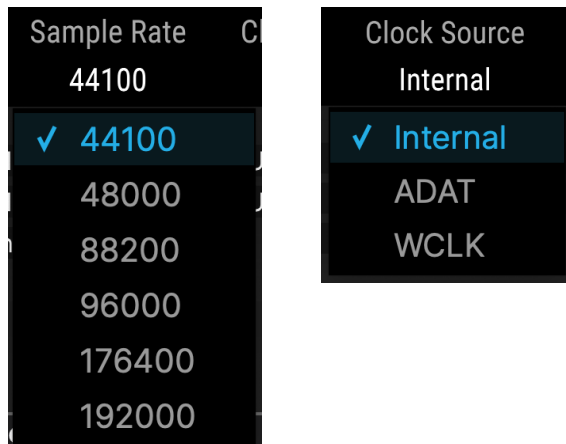
8.1.4.4. Configuration files and presets

The 16Rig stores up to eight presets in its internal memory, recallable from the front panel in standalone mode. Configuration files work alongside that memory rather than with it.

An export always writes the current device state, whether or not that state still matches the preset you last recalled. It does not write the contents of the eight preset slots. In the same way, an import restores a device state. It does not write to a preset slot, so save the imported state to a slot yourself if you want to keep it.

i To back up your full preset memory, recall each preset in turn and export its state, one file per slot. To restore a slot, import the matching file, then save it to the slot you want.

8.1.5. Sample Rate and Clock Source



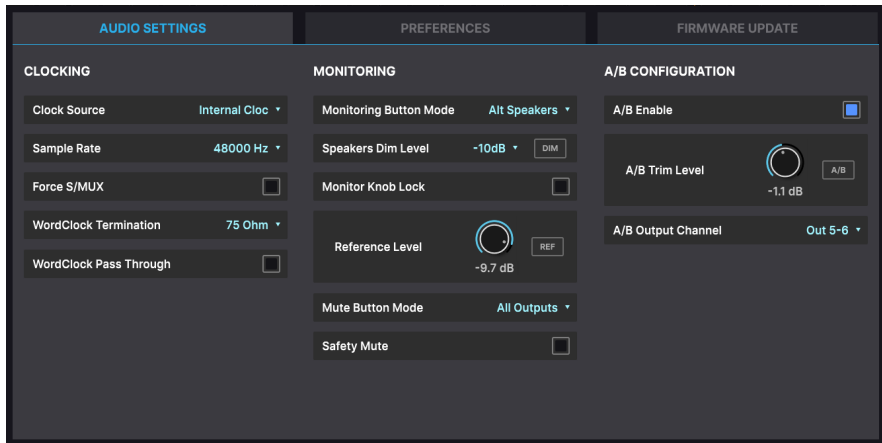
These drop-down menus let you select the Sample Rate, from 44.1 to 192 kHz, and Clock Source, whether Internal, ADAT, or Word Clock.

i NOTE: Selecting a sample rate higher than 48 kHz will limit the capabilities of the AudioFuse 16Rig. See the section on [Clock Settings \[p.53\]](#) for more information.

8.2. Global Settings

The last icon in the Upper Toolbar, the gear, opens a pop-up **Settings** menu with three tabs.

8.2.1. Audio Settings



These settings cover clocking, monitoring, and configuration of the A/B Speakers. They include:

- **Clock Source**
- **Sample Rate**
- **Force S/MUX**
- **WordClock Termination**
- **WordClock Pass Through**
- **Monitoring Button Mode**
- **Speakers Dim Level** and **DIM** button
- **A/B Enable**
- **A/B Trim Level**
- **A/B Output Channel**

These features are identical to, and offer the same options as, their corresponding hardware settings as described in the hardware [CLOCK SETTINGS](#) [p.53] and [MONITOR SETTINGS](#) [p.55] pages under the **SETTINGS** menu, as well as the **A/B CONF** settings in the **MONITOR** [p.47] menu.

8.2.2. Preferences

The screenshot shows the 'PREFERENCES' tab selected in the top navigation bar. The interface is divided into two columns. The left column contains 'METERING' and 'HTTP API' sections. The 'METERING' section has two settings: 'Clipping Reset' set to '3 sec' and 'Orange Color Threshold' set to '-12 dBFS'. The 'HTTP API' section has 'Server' checked and 'Port' set to '64274'. The right column contains the 'DEVICE' section with 'Device Name' set to 'Abbey Road', 'Serial Number' set to '8850400063010537', 'After Power Cut' set to 'Auto-Restart', and 'A Button Mode' set to 'Open / Show'.

SECTION	SETTING	VALUE
METERING	Clipping Reset	3 sec
	Orange Color Threshold	-12 dBFS
HTTP API	Server	☒
	Port	64274
DEVICE	Device Name	Abbey Road
	Serial Number	8850400063010537
	After Power Cut	Auto-Restart
	A Button Mode	Open / Show

You can adjust these settings to your personal preference. These settings affect both AFCC and the AudioFuse 16Rig hardware. For more informations about the HTTP API, please refer to the chapter [Advanced remote control \[p.82\]](#) of this manual.

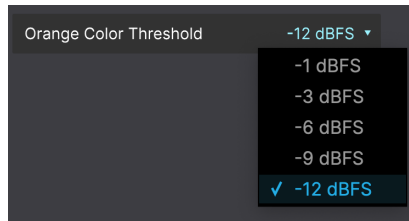
8.2.2.1. Clipping Reset

The screenshot shows the 'MIXER' section with a dropdown menu open for the 'Clipping Reset' setting. The dropdown menu lists three options: '1.5 sec', '3 sec' (which is selected and marked with a checkmark), and 'Infinite'.

OPTION
1.5 sec
✓ 3 sec
Infinite

This controls how long a level meter will indicate clipping before it resets itself. Options include 1.5 seconds, 3 seconds, and Infinite (clip indicators remain lit until manually reset, as explained [here \[p.29\]](#)).

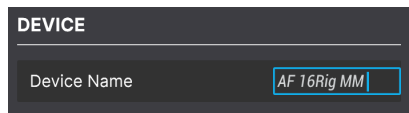
8.2.2.2. Orange Color Threshold



The AudioFuse 16Rig's meters show green for lower-level signals and red for clipping. In between, there's a level range indicated in orange. Some users like a lot of warning before they get close to clipping, while others want less.

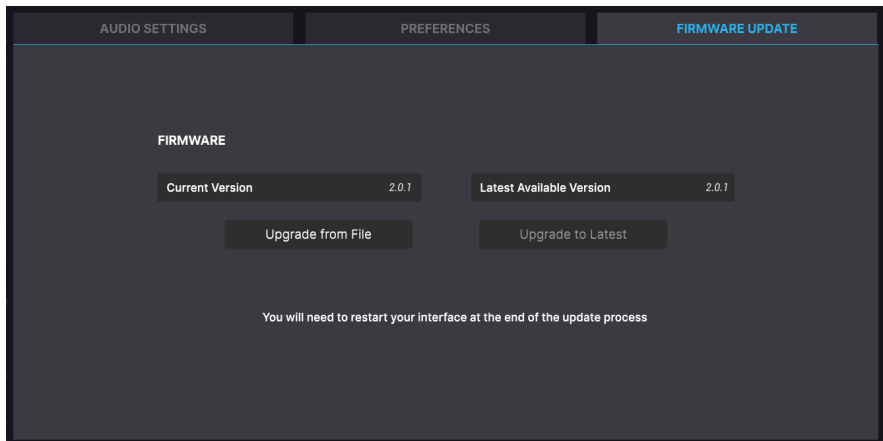
This drop-down menu lets you choose the level at which the meter displays turn orange, from a very conservative -12 dBFS to a riding-the-edge -1 dBFS. Note that your selection here is represented in both AFCC and the hardware.

8.2.2.3. Device Name



You can name your AudioFuse 16Rig. This name will be displayed in the Upper Toolbar, and is useful you have more than one AudioFuse interface connected to your computer.

8.2.3. Firmware Update



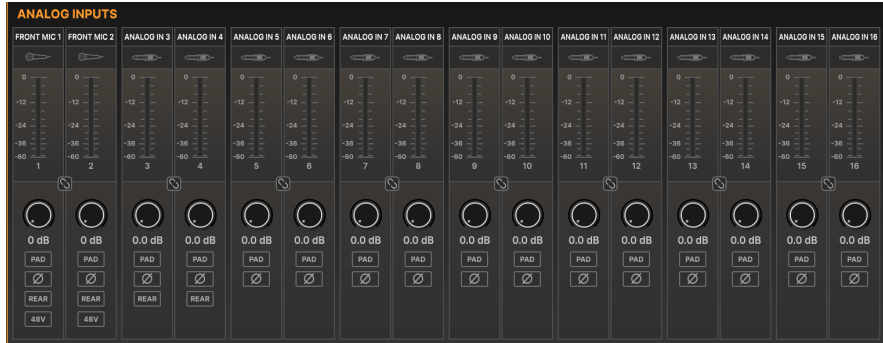
Sometimes Arturia will release new firmware for AudioFuse 16Rig, whether to fix bugs or add new features. Use this tab to check your current firmware version vs. what's available online at Arturia's website, download the latest firmware, and install it.

8.3. Overview Screen



Clicking the first of the Screen Selector icons opens an overview of the AudioFuse 16Rig's inputs and outputs, with level meters and various controls.

8.3.1. Analog Inputs



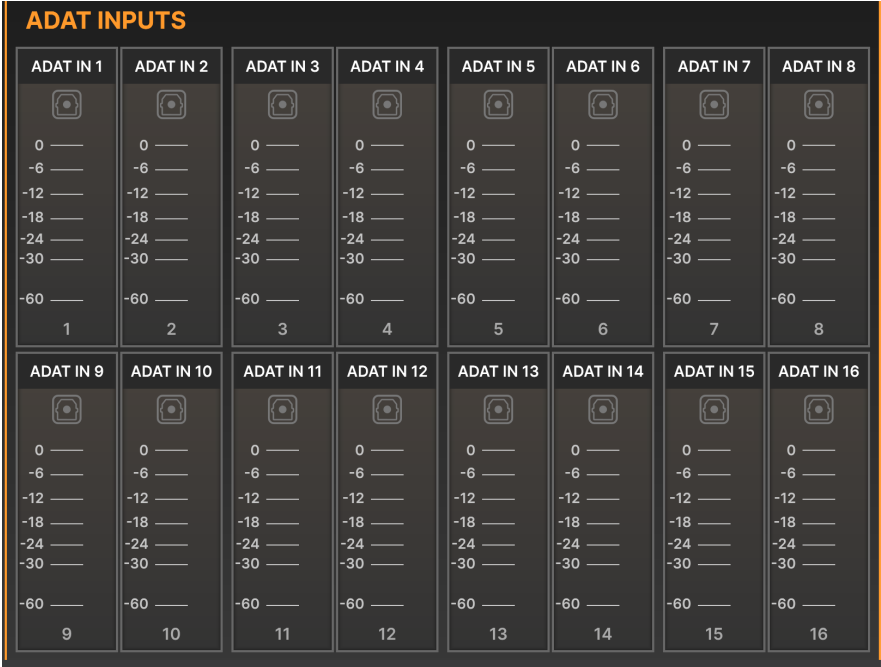
The upper left section of the Overview screen contains meters and controls for the Analog Inputs. These duplicate the features of the hardware, but allow for all inputs to be seen at once. Each input has a level meter, stereo **Link** button shared with the adjacent input, gain control, and buttons for **Pad** and **Phase**.

In addition, the first four inputs can be switched to the **Rear** jacks on the hardware. This lets you leave cables plugged into the Input 1 and 2 XLR/TRS jacks and the Input 3 and 4 minijack while sending other signals to their corresponding rear-panel inputs. Remember, if the **Rear** button is not pressed, the corresponding rear inputs will be ignored.

Inputs 1 and 2 will also display a **48V** button for phantom power if an XLR cable is plugged into the front panel, or an **INST** button for high-impedance instrument-level signals if a 1/4" cable is plugged into the front panel.

i Note that some of these function buttons (48V and INST, for example) will only appear if appropriate front-panel connections are made. If the appropriate cable is not connected to the front panel, the button(s) will not be shown.

8.3.2. ADAT Inputs



The upper right portion of the Overview screen has 16 level meters for the ADAT digital inputs. There are no controls associated with these meters.

8.3.3. Analog Outputs



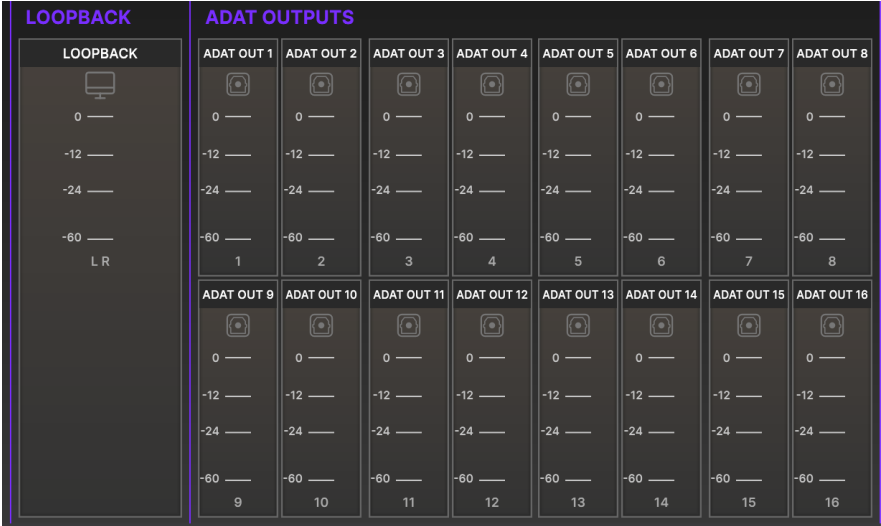
The lower left portion of the Overview screen has controls and level meters for all of the Analog Outputs. Each output channel has metering, a stereo **Link** button to connect adjacent mono channels, and an output level control.

In addition, the **MONITOR** output pair has controls for **MUTE**, **DIM**, **MONO**, and **A/B** functions, corresponding to those on the hardware. If A/B Speaker capability is not enabled, the A/B button is greyed out.

The **PHONES** section has an indicator showing the position of the Phones knob on the front panel. Note that this on-screen control is for display purposes only. To change the level, use the physical knob on the front panel. This section also has a drop down menu letting you adjust the output level of the headphone amp to better match the impedance of your headphones, letting you easily use both low and high impedance headphones with AudioFuse 16Rig.

Analog Outputs 3 and 4 on the front panel have a drop-down menu for selecting their function: **LINE**, **REAMP**, **DIRECT THRU**, and **ALT PHONES**. In addition, a **GROUND LIFT** switch is available to eliminate possible ground loop hum when connecting the AudioFuse 16Rig outputs to guitar amps and other gear.

8.3.4. Loopback and ADAT Outputs



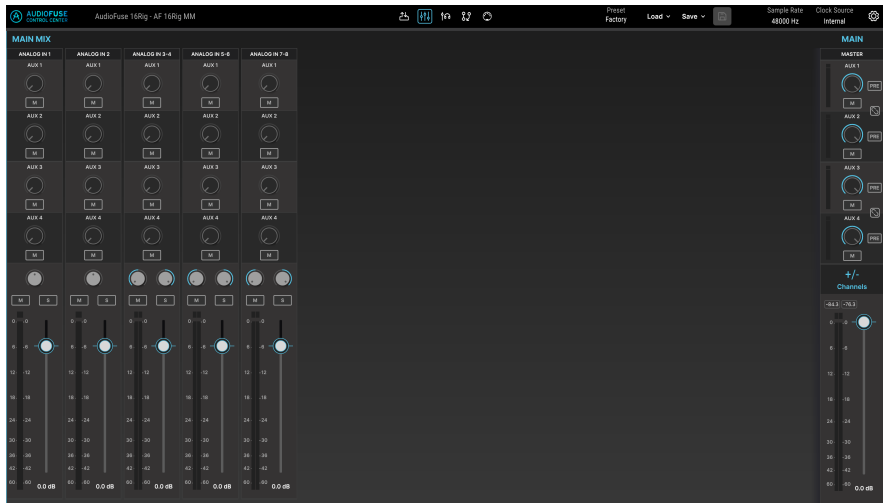
The lower right portion of the Overview screen provides output level metering for the stereo Loopback signal and for all 16 ADAT digital outputs.

8.4. Main Mixer Screen



This screen brings up AudioFuse 16Rig's Main mixer, and provides functions that let you build and set it far more quickly and easily than with the user interface of the AudioFuse 16Rig itself.

When you open this screen for the first time, it will look like this:



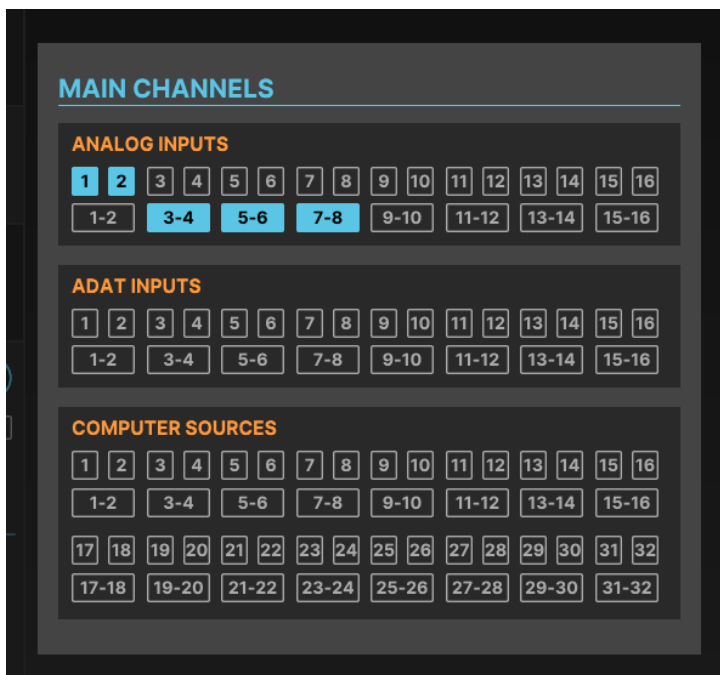
The Factory preset populates the Main mixer with a set of eight analog input channels as shown here.

The controls for the **MAIN** section are present on the right. They include:

- **AUX 1-4:** These are controls for the four Aux buses. Each has a level meter at left, level control, and **M** (Mute) and **PRE** (Pre-fader) buttons. In addition, Aux 1 and 2 can be linked to form one stereo bus, and Aux 3 and 4 can do the same.
- **+/- Channels:** Click here to add channels to the mix, as described below.
- The Master output level meters and fader are here. Numerical displays above the meters show the highest level seen so far, in dBFS.

8.4.1. Adding Channels

If the starting configuration isn't precisely what you need, the Main mixer screen allows you to build a custom mix, adding the inputs you need and leaving out the ones you don't. This is done with the **Main Channels** pop-up window.



Click on the +/- **Channels** button to open the Main Channels window. All available inputs are displayed, both in mono and linked stereo. Active channels are lit while inactive channels are not. Click on an inactive channel to add it to the mixer. Click again to deselect a channel and remove it from the mixer.

Once you've selected the channels you want, click anywhere outside the Main Channels pop-up to dismiss it. Your populated Main mixer screen might look something like this:



Each channel has a send level and **M** (Mute) button for each of the Aux buses, a Pan control, **M** (Mute) and **S** (Solo) buttons, a level fader, and a meter with clip indicator.

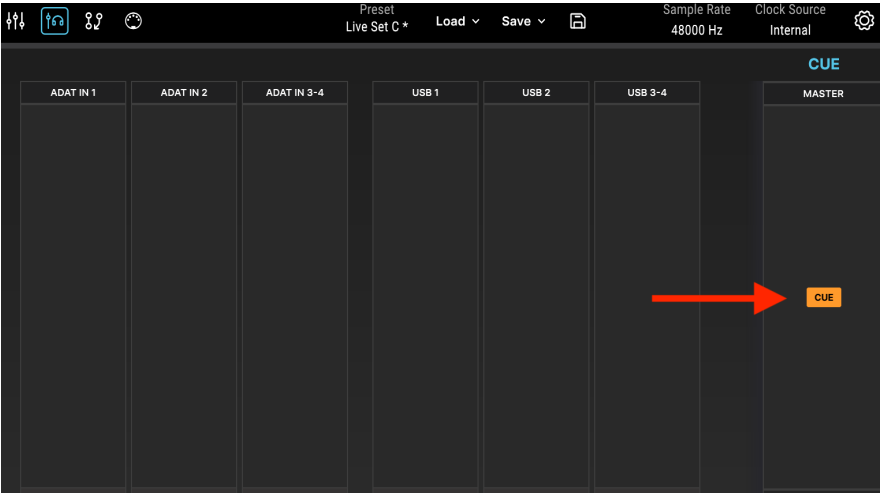


Remember that the Main Mixer's output needs to be routed to a pair of analog or digital outputs for you to hear it. If you see activity in the mixer's level meters but you cannot hear the output, check your connections in the [Routing Matrix \[p.74\]](#) and make sure the output of the mixer is routed to where you want it to go.

8.5. Cue Mixer Screen



The Cue mixer screen is identical to the Main mixer screen in features and operation, aside from the lack of Aux buses. Clicking the +/- **Channels** button brings up a **Cue Channels** pop-up window that works the same way as in the Main mixer.



There is one other important difference when it comes to the Cue Mixer. There is a **CUE** button above the Master fader, which turns the Cue mix on or off. This corresponds to the **CUE** button on the front panel of the AudioFuse 16Rig itself.

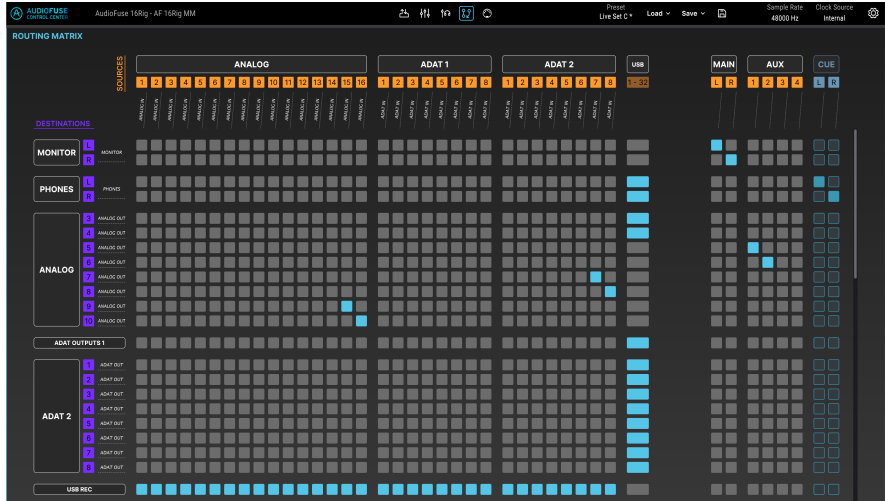


If you cannot hear the output of the Cue mixer, please make sure the **CUE** button is switched on and the output of the Cue mixer is routed to your desired analog or digital outputs. This can be done through the AudioFuse 16Rig's Routing matrix.

8.6. Routing Matrix Screen



With so many inputs and outputs, the AudioFuse 16Rig provides many choices for routing signals. The Routing Matrix screen lets you make sense of the many possible routing options and see at a glance where signals are going.



The Routing screen shows a matrix of **SOURCES** (columns listed from left to right and numbered in gold) and **DESTINATIONS** (rows listed from top to bottom and numbered in purple). To make a connection, simply click the point where the source and destination intersect in the matrix. Connections are shown with blue squares.

When working in the router, you will often need to make lots of "diagonal" connections, which can be tedious to do one-by-one (often over 8 or even 16 connections at a time). To make life easier, we've added a click-drag function that draws a line and makes diagonal connections quickly and easily for you.

i TIP: Don't think of the sources and destinations as "inputs" and "outputs"! You can get tripped up easily that way. After all, when you send the Main mix to the Monitor jacks, you're sending an "output" to another "output", which can quickly become confusing as you connect more and more things. Instead, any time you're working in the Routing Matrix, just think of things as being sources and destinations. It's a simple trick but hopefully it keeps things clear in your mind.

8.6.1. Sources

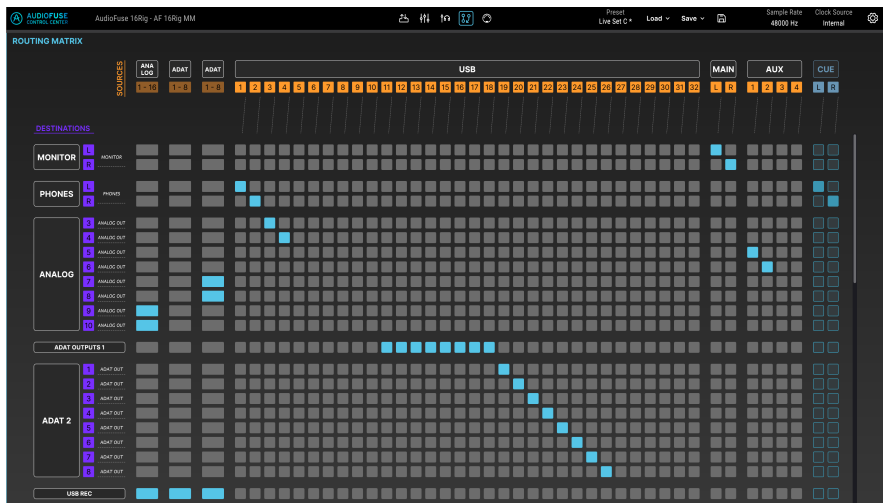
From left to right, the available Sources are:

- **ANALOG:** 16 input channels
- **ADAT 1:** 8 input channels
- **ADAT 2:** 8 input channels
- **USB:** 32 channels of audio playing back from your DAW
- **MAIN:** the stereo output of the Main mixer
- **AUX:** the four Aux outputs of the Main mixer
- **CUE:** the stereo output of the Cue mixer

Because there are so many sources, it is impossible to show them all at once on the screen. As a result, half of the sources are minimized (or "collapsed") while the other half is being shown, and you switch between them by clicking on the name of any minimized group of sources.

In the screenshot above, the **ANALOG**, **ADAT 1**, and **ADAT 2** sources are displayed, and the **USB** sources have been collapsed into a single column. If a connection exists between *any* of the USB sources and a particular destination, that row will have a blue rectangle.

If you click on any of the black boxes with group labels (e.g. ANALOG or ADAT 2), the screen will switch to the following view:



Here, the **USB** sources are displayed individually while the other three groups of sources have been collapsed into single columns.

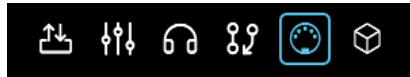
8.6.2. Destinations

In the two screenshots above, you can also see the available groups of Destinations:

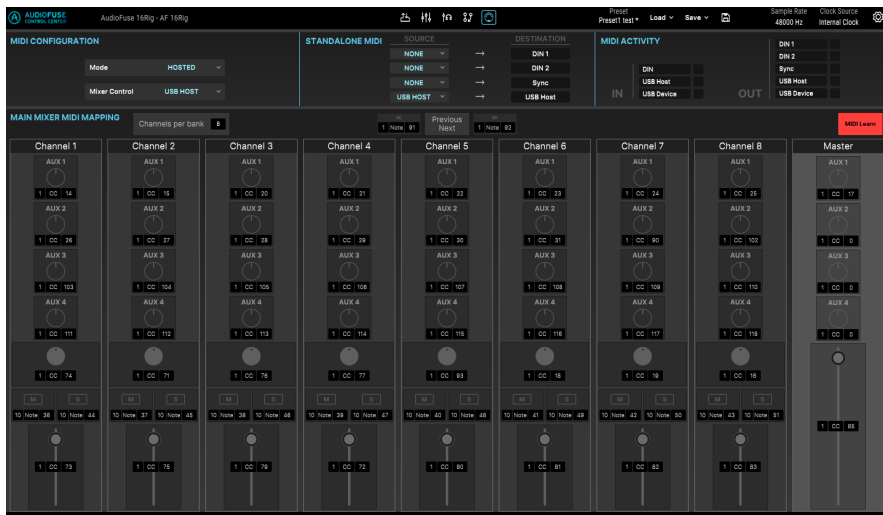
- **MONITOR:** the rear-panel **MONITOR** outputs
- **PHONES:** the front-panel **PHONES** outputs
- **ANALOG:** the 8 **LINE OUTPUTS**, numbered 3 through 10
- **ADAT 1:** 8 channels at **ADAT OUT 1**
- **ADAT 2:** 8 channels at **ADAT OUT 2**
- **USB:** 32 channels going to your DAW to be recorded

Unlike the sources, which switch back and forth between two possible horizontal views, any group of destinations can be collapsed into a single row or left fully expanded. If too many rows are open for your computer screen to show them all, you can scroll through them vertically with your mouse or scroll wheel in the usual way.

8.7. MIDI Screen

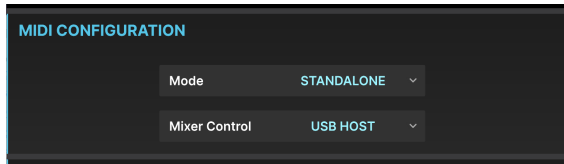


This screen allows you to monitor and configure various MIDI-related parameters. Crucially, this includes the ability to map the various onscreen controls of the Main mixer to your MIDI controller's physical controls (knobs, sliders, etc.) so that you can control your mix via MIDI. You'll probably find it easier to mix with an external control surface than with your computer mouse!

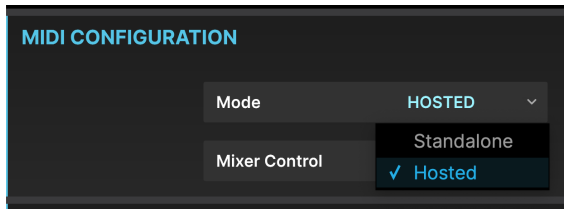


Across the top of the MIDI Screen are several important MIDI-related control sets:

8.7.1. MIDI Configuration

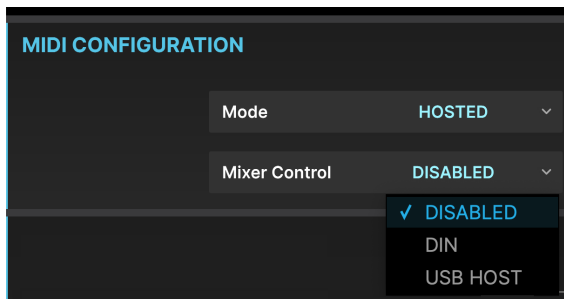


The **MIDI CONFIGURATION** section has two controls with drop-down selection menus.



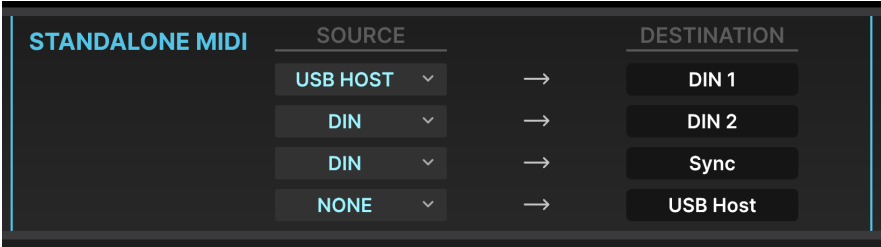
The **Mode** menu lets you choose whether the AudioFuse 16Rig is operating in **Standalone** mode or **Hosted** mode.

- In Standalone mode, MIDI message routing is controlled by the AudioFuse 16Rig's [Standalone MIDI \[p.53\]](#) settings.
- In Hosted mode, the connected computer handles MIDI routing between inputs and outputs.



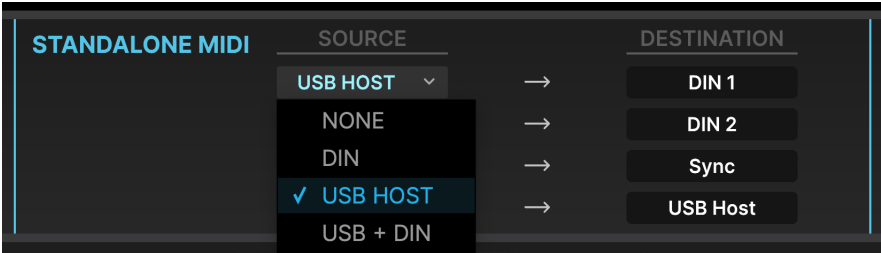
The **Mixer Control** menu chooses between whether the Main mixer is controlled by MIDI messages coming from the rear-panel DIN port, the USB HOST port, or whether MIDI mixer control is entirely disabled. If you're not planning to control the Main mixer via MIDI, we recommend leaving this disabled to prevent accidentally changing your mixer settings.

8.7.2. Standalone MIDI



The **STANDALONE MIDI** controls at the top center of the screen let you specify the routings of MIDI Inputs to MIDI Outputs among the AudioFuse 16Rig's various ports when it's in Standalone mode.

There are four fixed destinations: **DIN 1**, **DIN 2**, **Sync**, and **USB HOST**.



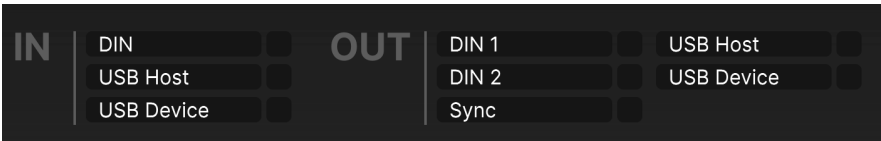
The following options are available for sources to route to these destinations:

- **NONE**
- **DIN**
- **USB HOST**
- **USB + DIN** (data from both ports is merged)



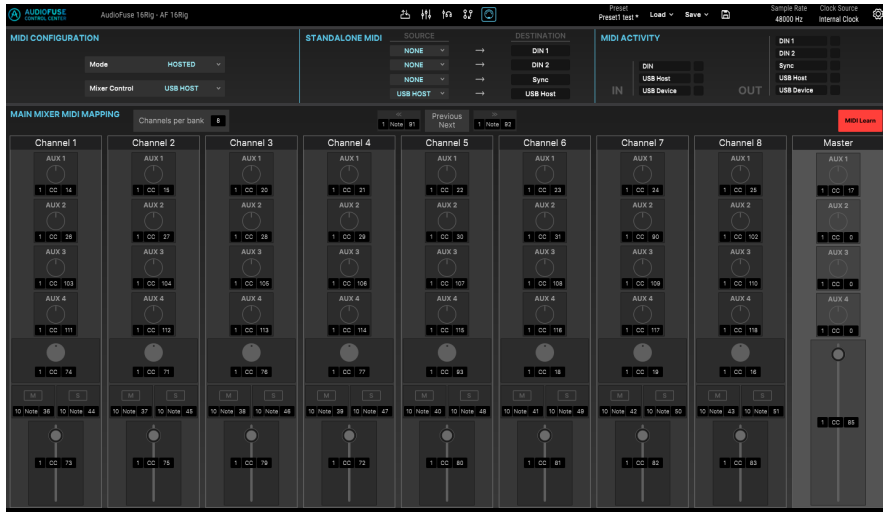
Note that the **USB + DIN** option is not available as a source for the **Sync** destination, to avoid conflicting MIDI timing messages.

8.7.3. MIDI Activity



The top right corner of the screen has a set of **MIDI ACTIVITY** monitors, so you can see if MIDI data is flowing to/from the various inputs and outputs.

8.7.4. Main Mixer MIDI Mapping



The **MAIN MIXER MIDI MAPPING** screen is laid out as a "bank" of eight channels, with the following controls available for MIDI control on each channel:

- Aux 1 through Aux 4 send levels
- Channel Mute and Solo status
- Channel level

If you're working with a larger mix, the MIDI configuration works in blocks (or "banks") of 8 channels, and there are MIDI commands to let you move between blocks, jumping to the previous or next block. The currently selected block in the Main Mixer has channel labels that appear in blue text while all other channel labels appear in white. For example, if you have a 24-channel mix, moving from one block to the next will give you control of channels 1 through 8, then 9 through 16, then 17 through 24. As you move from block to block, you'll notice the channel labels change from white to blue depending on what set of controls is under MIDI control.

In addition, the following MIDI controls are available in the Master section:

- Aux 1 through Aux 4 return levels
- Master level

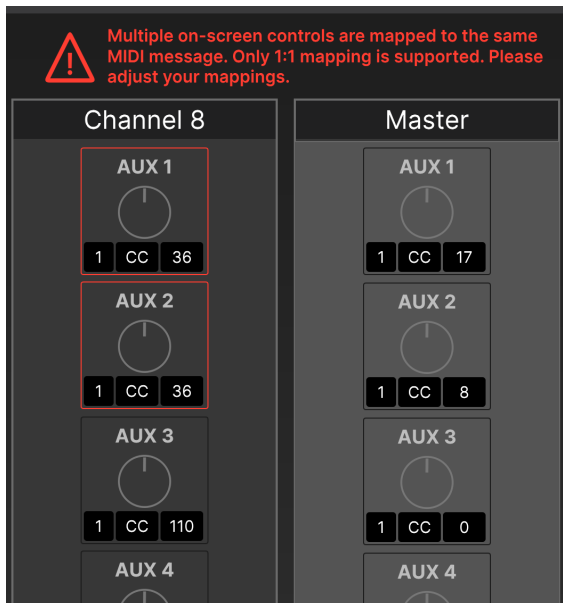
8.7.4.1. Setting MIDI control messages manually



Under each point of control, you can click on the three black boxes to set the following parameters:

- MIDI Channel (1 to 16)
- Type of MIDI Message (Off, MIDI Control Change (CC), or MIDI Note)
- CC or Note number

i Note that each on-screen control must have its own unique channel+CC/Note+value assignments for MIDI mapping to work properly. If two or more controls share the same assignments, you'll get an error message showing the controls that are in conflict, and telling you to change the values for one or the other, as shown below:



8.7.4.2. Adjusting Channels per Bank

Some desktop control surfaces have fewer than 8 channel controls (sliders, knobs, etc.) In fact, many of the smallest controllers only have one channel's worth of controls. To accommodate smaller controllers, AFCC's MIDI screen has a "Channels per bank" drop-down menu that lets you set the number of mixer channels that can be in a bank. So, for example, if your controller only has one channel's worth of controls, set this to 1 and you're all set.

8.7.4.3. MIDI Learn

To make MIDI programming easier, we've added a MIDI Learn function (if you don't see this on your screen, make sure you have updated to the latest firmware and AFCC!)

To use the MIDI Learn function:

1. Click the MIDI Learn button at the top right (it turns red when MIDI Learn mode is engaged)
2. Click any on-screen control you'd like to re-map
3. Push, turn or move a physical control on your MIDI Controller
4. Continue steps 2-3 until you have mapped all desired controls
5. Click the MIDI Learn button again to switch off MIDI Learn mode

The Main Mixer is now re-mapped to your hardware controller.

8.8. Advanced remote control

As a first for Arturia, the introduction of the Open HTTP API enables new ways for remotely control selected Arturia devices.

This API is intended for advanced users who want to streamline their daily workflow.



Arturia provides the API and its documentation, but cannot provide support for custom applications.

8.8.1. The Open HTTP API

The Arturia AudioFuse 16rig features an Open HTTP API enabling full remote control and automation of most of the parameters like Gain, Mute, and Solo, and other settings. This API has been designed for professional, broadcast, and high-level content production environments.

The system includes automated port management within the AudioFuse Control Center. You can find more information on the technical documentation.

The API documentation and the associated code samples can be downloaded from Arturia's website here: <https://link.arturia.com/fuse-api-doc>

8.8.1.1. The Arturia Fuse Stream Deck plugin

Arturia now offers support for the Elgato series of Stream Deck devices. The Arturia Fuse Stream Deck plugin lets you control key AudioFuse functions directly from an Elgato Stream Deck, eliminating the need to open the AudioFuse Control Center. These allow you to control compatible Arturia devices from Stream Deck buttons. Here's a list of supported functions:

Scope	Functions	AudioFuse Studio Support	AudioFuse 16rig Support
Clock	Source/Sampling rate	x	x
Monitoring	Volume, Mute, DIM, Mono, A/B Speaker-Set	x	x
Monitoring	Reference level recall		x
Monitoring	Speaker Solo/Mute, Surround & Immersive settings		x
Monitoring	Listening Source	x	
Monitoring	Phone Source & Mono	x	
Preset	Recall		x

In short, if you use a Stream Deck, you can perform common AudioFuse functions with customizable buttons instead of opening the AudioFuse Control Center.

Stream Deck buttons update in real time to reflect the current status of your interface. The plugin is compatible with AudioFuse 16Rig, and requires the latest release of the AudioFuse Control Center on Windows or macOS.

You can download the Fuse plugin from here: <https://link.arturia.com/fuse-stream-deck>

8.9. Hover Tips



One last thing to point out, if you didn't notice it in the very first screenshot: hovering your mouse over any control will display a detailed explanation of what the control does, shown in the Hover Tips display at the bottom of the plug-in window. After all, everybody needs a little extra help now and then!

9. SPECIFICATIONS

9.1. System Requirements

- 4 GB RAM
- Intel i5 or later (including Apple Silicon)
- 1 GB free hard disk space

9.2. Audio Specifications

Line Inputs	Measured Values
Input impedance	20k Ω (symmetric), 10k Ω (asymmetric)
Maximum input level	Rear: +24dBu, Front +22dBu
Pad mode	Rear: -12dB, Front: -10dB
Frequency response 20Hz to 20kHz	+/-0.06dB typical
Dynamic range	118dB (A-weighted)
THD+N @ 1kHz	-113dB (A-weighted)

Instrument Inputs	Measured Values
Input impedance	1.1M Ω asymmetric
Maximum input level	+20dBu
Gain range	55dB
Pad mode	-10dB
Frequency response 20Hz to 20kHz	+/-0.1dB typical
Dynamic range	114dB (A-weighted)
THD+N @ 1kHz	-105dB (A-weighted)

Microphone Preamps	Measured Values
Input Impedance	2,7k Ω
Maximum input level	+22dBu
Gain range	55dB
Pad mode	-20dB
Equivalent Input Noise (EIN)	-129.5dBu (A-weighted)
Frequency response 20Hz to 20kHz	+/-0.08dB
Dynamic range	117dB (A-weighted)
THD+N @ 1kHz	-110dB (A-weighted)

Speaker Outputs	Measured values
Output impedance	240Ω balanced, 120Ω unbalanced
Maximum output level	+20dBu
Frequency response 20Hz to 20kHz	+/- 0.02dB
Dynamic range	126dB (A-weighted)
THD+N @ 1kHz	-114dB (A-weighted)

Line Outputs (DC coupled)	Measured values
Output impedance	240Ω balanced, 120Ω unbalanced
Maximum output level	+24dBu
Frequency response 20Hz to 20kHz	+/- 0.04dB typical
Dynamic range	120dB (A-weighted)
THD+N @ 1kHz	-106dB (A-weighted)
DC voltage amplitude	+/- 8.7V (asymmetrical)

Phones Outputs	Measured values
Output impedance	5Ω
Maximum output level	+14.4dBu
Dynamic range	109dB (A-weighted)
Power @ 33Ω @1%THD	271mW @-78dB THD
THD+N @ 1kHz	-103dB (A-Weighted)



Supported Frequency Rates: 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz

9.3. Latency Performances

9.3.1. Analog to Analog roundtrip latency

Buffer size	Measured values
192kHz - 32 samples buffer size	4.3ms
96kHz - 32 samples buffer size	4.3ms
96kHz - 16 samples buffer size	3.9ms
48kHz - 32 samples buffer size	4.8ms
48kHz - 16 samples buffer size	4.4ms
48kHz - 8 samples buffer size	4.0ms



Note: Latency performances have been measured on a Windows PC using ASIO driver.

9.3.2. USB loopback roundtrip latency

Buffer size	Measured values
192kHz - 32 samples buffer size	3.6ms
96kHz - 32 samples buffer size	4.5ms
96kHz - 16 samples buffer size	3.5ms
48kHz - 32 samples buffer size	4.5ms
48kHz - 16 samples buffer size	4.6ms
48kHz - 8 samples buffer size	4.0ms



Note: Latency performances have been measured on a Windows PC using ASIO driver.

10. APPENDICES

10.1. USB Audio mapping

AudioFuse 16Rig displays differing numbers of input and output channels in your DAW depending on your selected sampling rate.

Sampling Rate	Inputs	Outputs
44.1 / 48 kHz	34	34
88.2 / 96 kHz	18	18
176.4 / 192 kHz	10	10

The tables below contain detailed information about inputs and outputs at each sampling rate.

 The given mapping is the default mapping for Factory presets. You can easily reassign and remap everything in the AudioFuse Control Center using the [Matrix \[p.74\]](#).

10.1.1. Audio Mapping at 44.1 / 48 kHz sampling rates:

Channel	Computer Input (REC)	Computer Output (Playback)
1	Front/Rear In 1	Speakers/Headphones Left
2	Front/Rear In 2	Speakers/Headphones Right
3	Front/Rear In 3	Front/Rear Out 3
4	Front/Rear In 4	Front/Rear Out 4
5	Rear In 5	Rear Out 5
6	Rear In 6	Rear Out 6
7	Rear In 7	Rear Out 7
8	Rear In 8	Rear Out 8
9	Rear In 9	Rear Out 9
10	Rear In 10	Rear Out 10
11	Rear In 11	ADAT Out 1
12	Rear In 12	ADAT Out 2
13	Rear In 13	ADAT Out 3
14	Rear In 14	ADAT Out 4
15	Rear In 15	ADAT Out 5
16	Rear In 16	ADAT Out 6

Channel	Computer Input (REC)	Computer Output (Playback)
17	ADAT In 1	ADAT Out 7
18	ADAT In 2	ADAT Out 8
19	ADAT In 3	ADAT Out 9
20	ADAT In 4	ADAT Out 10
21	ADAT In 5	ADAT Out 11
22	ADAT In 6	ADAT Out 12
23	ADAT In 7	ADAT Out 13
24	ADAT In 8	ADAT Out 14
25	ADAT In 9	ADAT Out 15
26	ADAT In 10	ADAT Out 16
27	ADAT In 11	USB Playback
28	ADAT In 12	USB Playback
29	ADAT In 13	USB Playback
30	ADAT In 14	USB Playback
31	ADAT In 15	USB Playback
32	ADAT In 16	USB Playback
33	Loopback In (Left)	Loopback Out (Left)
34	Loopback In (Right)	Loopback Out (Right)

10.1.2. Audio Mapping at 88.2 / 96 kHz sampling rates:

Channel	Computer Input (REC)	Computer Output (Playback)
1	Front/Rear In 1	Speakers/Headphones Left
2	Front/Rear In 2	Speakers/Headphones Right
3	Front/Rear In 3	Front/Rear Out 3
4	Front/Rear In 4	Front/Rear Out 4
5	Rear In 5	Rear Out 5
6	Rear In 6	Rear Out 6
7	Rear In 7	Rear Out 7
8	Rear In 8	Rear Out 8
9	Rear In 9	Rear Out 9
10	Rear In 10	Rear Out 10

Channel	Computer Input (REC)	Computer Output (Playback)
11	Rear In 11	ADAT Out 1
12	Rear In 12	ADAT Out 2
13	Rear In 13	ADAT Out 3
14	Rear In 14	ADAT Out 4
15	Rear In 15	ADAT Out 5
16	Rear In 16	ADAT Out 6
17	ADAT In 1	ADAT Out 7
18	ADAT In 2	ADAT Out 8

10.1.3. Audio Mapping at 176.4 / 192 kHz sampling rates:

Channel	Computer Input (REC)	Computer Output (Playback)
1	Front/Rear In 1	Speakers/Headphones Left
2	Front/Rear In 2	Speakers/Headphones Right
3	Front/Rear In 3	Front/Rear Out 3
4	Front/Rear In 4	Front/Rear Out 4
5	Rear In 5	Rear Out 5
6	Rear In 6	Rear Out 6
7	Rear In 7	Rear Out 7
8	Rear In 8	Rear Out 8
9	Free	Rear Out 9
10	Free	Rear Out 10

11. DECLARATION OF CONFORMITY

11.1. FCC

WARNING: DO NOT MODIFY THE UNIT!

Any modifications or other changes to this unit not approved by the party responsible for compliance could void the user's authority to operate this equipment.

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) This device must accept any interference received, including interference that may cause undesired operation.

Responsible Party in USA: Zedra, 185 Alewife Brook Parkway, #210, Cambridge, MA 02138, United States T: +1 857 285 5953

Trade Name: ARTURIA, Model Number: AudioFuse 16Rig

Note: 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 a residential installation. 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. However, 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 into 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.

11.2. CANADA

This class B digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe B est conforme à la norme NMB-003 du Canada

11.3. CE

This device has been tested and found to comply with the limits of the European Council Directive on the approximation of the laws of the member states relating to Electromagnetic Compatibility according to 2014/30/EU, and Low Voltage Directive 2014/35/EU.

11.4. ROHS

This device has been produced with lead free solder and fulfills the requirements of the ROHS directive 2011/65/EU.

11.5. WEEE

This symbol indicates that the electrical and electronic equipment should not be disposed of as general household waste at its end-of-life. Instead, the products should be handed over to the applicable collection points for the recycling of electrical and electronic equipment for proper treatment, recovery, and recycling in accordance with your national legislation and the Directive 2012/19/EU (WEEE – Directive on Waste Electrical and Electronic Equipment). For more information about collection points and recycling of these products, please contact your local municipal office, your household waste disposal service, or the shop where you purchased the product.

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