AUDIOFIRE 8 ™
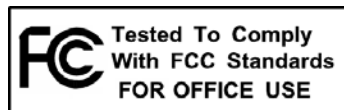
AUDIOFIRE 12 ™

Owner's Manual Version 1.1 for Macintosh

Important Safety Instructions

1. Read Instructions - Be sure to read all of the safety and operating instructions before operating this product.

2. Retain Instructions - The safety instructions and owner's manual should be retained for future reference.

3. Heed Warnings - All warnings on your Echo product and in the Owner's Manual should be followed.

4. Follow Instructions - All operating and use instructions should be followed.

5. Moisture - Water and moisture are detrimental to the continued good health of your Echo product. Do not install or operate your Echo product near sources of water or moisture such as sinks, damp basements, leaky roofs, etc.

6. Heat – Your Echo product should be situated away from sources of heat such as heaters or radiators.

7. Power Sources - This unit should be operated only from the type of power source indicated in this documentation or on your Echo product. If you are unsure about the type of power at your location, contact your local power company.

8. Grounding – Precautions should be taken so that the grounding capabilities of the unit are not undermined. This unit is provided with a cord with an equipment grounding conductor and grounding plug. This plug must be plugged into an outlet that is properly installed and grounded in accordance with all local rules and ordinances. Do not modify the plug provided with the equipment. If the plug will not fit into your outlet, have a proper outlet installed by a qualified electrician.

9. Power Cord Protection – Power supply cords should be routed so that they are unlikely to be walked on or pinched by items placed upon or against them. Pay particular attention to protecting the plugs, outlets, and the point at which the cord exits your Echo product.

10. Servicing - Do not attempt to service this unit yourself, as opening the case will expose you to hazardous voltage or other dangers. All servicing should be referred to qualified service personnel.

11. Damage Requiring Service - Unplug this unit and refer it to a qualified service technician when any of the following occur:

      a) Objects have fallen or liquid has spilled into the unit
      b) The product has been exposed to rain or water
      c) The product does not operate normally or when a marked change in performance is noticed
      d) The product has been dropped or damaged in any way

Registering your product online at **http://www.echoaudio.com/support/register.php** allows us to register key information so that we may handle problems faster and inform you of advance information on upgrades and other news. Thank you in advance for registering. We hope you enjoy your Echo product.

Limited Warranty

Echo Digital Audio Corporation warrants this product, when purchased at an Authorized Echo Dealer in the United States of America, to be free of defects in materials and manufacturing workmanship for a period of one year from the date of original purchase. During the warranty period Echo shall, at its option, either repair or replace any product that proves to be defective upon inspection by Echo. Final determination of warranty coverage lies solely with Echo. Echo reserves the right to update any unit returned for repair, and reserves the right to change or improve the design of the product at any time without notice.

This is your sole warranty. Echo does not authorize any third party, including any dealer or sales representative, to assume any liability on behalf of Echo or to make any warranty for Echo.

Service and repairs of Echo products are to be performed only at the factory (see below) unless otherwise authorized in advance by the Echo Service Department. Unauthorized service, repair or modification will void this warranty.

To obtain factory service:

Contact Echo Digital Audio Corporation at (805) 684-4593, 9AM to 4PM Monday through Friday (Pacific Time). If necessary, you will be given a return authorization number. Products returned without an RA number will be refused. Echo may, at its option, require proof of the original date of purchase in the form of a dated copy of the original authorized dealer's invoice or sales receipt. Pack the product in its original shipping carton and attach a description of the problem along with your name and a phone number where Echo can contact you if necessary. Ship the product insured and freight prepaid to:

Echo Digital Audio Corporation
6309 Carpinteria Ave. #300
Carpinteria, CA 93013

DISCLAIMER AND LIMITATION OF WARRANTY

Echo makes no other warranties, express, implied, or otherwise, regarding Echo products, and specifically disclaims any warranty for merchantability or fitness for a particular purpose. The exclusion of implied warranties is not permitted in some states and the exclusions specified herein may not apply to you. This warranty provides you with specific legal rights. There may be other rights that you have which vary from state to state.

In no event will Echo be liable for any lost profits, or for any consequential, direct or indirect damages, however caused and on any theory of liability, arising from this warranty and sale.

Introduction

Thank you for choosing Echo Digital Audio. We think you'll find your Echo product to be an extremely flexible, high-performance tool for your computer-based hard disk recording system.

What You Should Have Received in the Box

When you opened the box, you should have found the following:

- An audio interface box
- Four mounting screws with collars
- Two removable rack-mount brackets (AudioFire8 only)
- A FireWire cable
- A power cable
- A CD-ROM containing:

  - Echo Console software for Mac OS X
  - Windows XP Drivers
  - PDF Owner's Manual
  - Tracktion multitrack recording, MIDI, editing, and sampling software for Mac OS X & Windows XP

- A Quick Start Guide

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System Requirements

In order to use AudioFire8 or AudioFire12 you'll need the following:

- An Apple brand Macintosh computer (desktop or notebook) with:

  - Mac OS X (10.3.9 or later)
  - An unused FireWire port
  - A G4 or better processor (G5 recommended)
  - A minimum 256Mb of RAM (512Mb highly recommended)
  - A fast, high-capacity IDE, SCSI or FireWire hard disk drive

- Peripheral audio equipment, such as a mixer, microphones, studio monitors, musical instruments, etc.

Installation

To properly install your AudioFire product you will need to: log in as an administrator, check OS X version, run software update, connect the audio interface to a FireWire port and to your other audio hardware, install the Console software, and install Tracktion. The following steps will guide you through installing your AudioFire product.

Log In as Administrator

In order to properly install your AudioFire product you need to make sure that you log in to your Mac with an account that has administrative privileges. To verify that your account has administrative privileges, open "System Preferences" from the Dock or from the Apple menu, select "Show All", and then select "Accounts" in the "System" section. Now, select your account from the list on the left and click on the "Security" tab. Make sure that the checkbox at the bottom labeled "Allow user to administer this computer" is checked. If your account does not have administrative access, then you will need to log out and log in with an account that does.

Check OS X Version

OS X version 10.3.9 (or later) is required use your AudioFire product. If you have a version lower than 10.3.9, your AudioFire product may not install, or may not function properly. To check your OS X version open "About This Mac" from the Apple menu. Verify that you have OS X version 10.3.9 or later. If you do not, then you will need to either purchase an upgrade or download it with Software Update.

Run Software Update

Before you install anything, you should make sure that you have the most recent updates for Mac OS X installed. This is especially important. If Mac OS X is out of date, you may not be able to use your AudioFire product at all. To do this, open "Software Update" from the Apple menu. Download and install all of the updates, including updates for software like iTunes, iDVD, iMovie, QuickTime, GarageBand, etc.

Installing the AudioFire Hardware and Software

Now that you have logged in as an administrator, checked the OS X version, and updated your software, you are ready to install your AudioFire product in your computer.

1. **Install the AudioFire Console.** Insert the AudioFire Install CD-ROM into your machine. Open the "AudioFire Console" folder and double-click the dmg file. Now drag the AudioFire Console icon to your Applications folder, or wherever you want to keep it on your Mac. This is also a good time to copy the manual to your hard drive as well.

2. Once you have installed the Console, locate the AudioFire interface box, the power cable, and the FireWire cable. **Securely mount the interface** into your equipment rack. If you will not be mounting the box in a rack, then be sure to place the interface box in a secure location.

3. Plug the **power cable** firmly into the interface and a power socket, and then turn on the interface by pressing the power button.

4. Plug one end of the **FireWire cable** into the FireWire connector on the rear of the AudioFire interface. Attach the other end of the FireWire cable to an open FireWire port on your computer. (If the connector does not match the port on your computer then you may need to get an adapter.)

5. Your Mac will automatically recognize your new AudioFire product without any fanfare. See the "Software Configuration" section below for more info.

NOTE: You should check the driver downloads area of our website (www.echoaudio.com) for any firmware updates and install them before continuing. The AudioFire hardware must be installed and powered on for the firmware update to load properly.

6. You can now **attach external audio device**s to the connectors on the AudioFire interface box. Information on attaching external devices to the audio interface box may be found in the following section: "Connecting to the Audio Interface".

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Connecting to the Audio Interface

The AudioFire8 and AudioFire12 audio interfaces contain a wide variety of connections that allow great flexibility in operation. For optimal performance, it is critical to use the appropriate cabling and connectors.

Analog Inputs and Outputs – AudioFire8

AudioFire8's front panel has two universal inputs (#s 1 – 2) with meters and trim knobs, a phantom power switch and light (+48v), and a power button with an amber power indicator light (not shown below). There is also a ¼" stereo headphone output jack, with a corresponding volume knob, which monitors analog outputs 1 and 2.

AudioFire8 has six analog line inputs (#s 3 – 8) and eight analog line outputs on the rear panel. These analog inputs (#s 3 – 8) operate with an input impedance of 10Kohms. The input and output connections can be used with balanced or unbalanced cables via the ¼" connectors. By default, the inputs and outputs are set to receive or send a +4dBu signal.

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You can switch between +4dBu and –10dBV for each individual input and output via the console software (see section below). There are also two channel inserts on the rear panel where you can connect a serial effects device (compressor, EQ, etc.) for the universal inputs (#s 1 – 2). The channel inserts are wired: tip = send, ring = return, and sleeve = ground.

Universal Inputs – AudioFire8

The universal input connections can accept balanced or unbalanced signals via ¼" or XLR connectors. Since AudioFire8 has built-in preamps, you can just plug your microphone or guitar cable right into the universal input. No external preamp, direct box, or mixer is needed!!

To use a microphone, plug the microphone's XLR cable into any of the universal inputs. Whenever an XLR cable is connected to a universal input, the microphone preamp is enabled and the gain range of the trim knob is set to range from +10 to +59dB. The input impedance of each microphone preamp is 1.5Kohms. Full-scale input is 9dBu with the trim set to minimum gain.

AudioFire8 provides 48 volts of phantom power for microphones that require it. To enable phantom power, press the button labeled +48v (the LED next to it will light up) and phantom power will be provided to all microphone inputs.

> **Note:** Phantom power will not be applied to any line level equipment or guitars plugged in using cables with ¼" connectors.

To record a guitar or line level signal, just plug a cable with a ¼" connector into one of the universal inputs. Whenever a ¼" cable is connected to a universal input, the guitar/line preamp is enabled and the gain range of the trim knob is set to range from 0 to +45 dB. The guitar/line preamp can support both guitar & line level signals. The input impedance of each guitar/line preamp is 102Kohms. Full-scale input is 18dBu with the trim set to minimum gain.

If you want to adjust the input level of a universal input, just turn the trim knob and watch the corresponding level meter. The red light at the top will light anytime the input signal is greater than –6.5dBFS (or within 6.5dB of clipping). The yellow

light represents a signal greater than –12.5dBFS, and the green light represents a signal greater than –36.5dBFS.

For optimal audio quality, you should adjust the input trim knob so that your loudest recorded signal lights up the yellow light, but not the red. When the red meter light shines extra bright – you've clipped! If the signal level ever exceeds 0dBFS the signal will be "clipped" and you will hear a "pop" or "tick" in the recording. This is a *very bad thing*, and clipping should be avoided at all costs! There is enough headroom so that you can be conservative in this area, and there is no need to push the input levels right up to the edge of clipping.

Analog Inputs and Outputs – AudioFire12

AudioFire12's front panel has meters for the twelve analog inputs and outputs, indicator lights for sample rate and clock synchronization, and a power button with an amber power indicator light (not shown below).
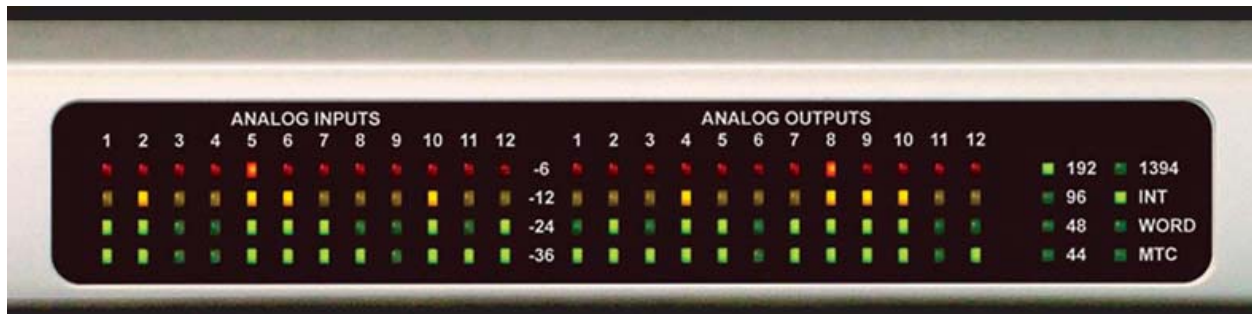
AudioFire12 has twelve analog line inputs and twelve analog line outputs on the rear panel. The analog inputs operate with an input impedance of 10Kohms. The input and output connections can be used with balanced or unbalanced cables via the ¼" connectors.

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By default, the inputs and outputs are set to receive or send a +4dBu signal. You can switch between +4dBu and –10dBV for each individual input and output via the console software (see "Console" section below).

The FireWire Connector

On the rear panel of each AudioFire interface box is a pair of connectors with the FireWire symbol above them. These six-pin IEEE 1394A connectors operate at 400 Mbps and are used to connect the audio interface to your computer, and to other FireWire devices. These connectors do not pass power from the computer to other FireWire devices, so other devices will require external power to function.

A FireWire cable was supplied with your AudioFire product.

S/PDIF (AudioFire8 only)

Next to the **FireWire** connectors on the back of AudioFire8 is a pair of connectors labeled **S/PDIF**, **IN** and **OUT**. These S/PDIF connectors are used to transmit digital audio data between digital audio devices via an electrical signal.
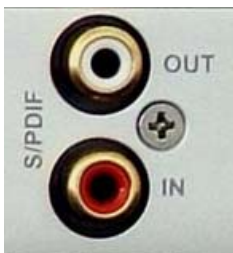
When connecting devices to the S/PDIF jacks, the use of standard analog RCA audio cables is not recommended. For reliable S/PDIF operation, 75-ohm coaxial (RG59) video cables are recommended.

MIDI

Both AudioFire8 and AudioFire12 are equipped with MIDI I/O connectors. The MIDI ports can be used for receiving MIDI time code (MTC), or sending and receiving MIDI signals between your digital audio/MIDI sequencing software and external sound modules, keyboards, MIDI controllers, etc.
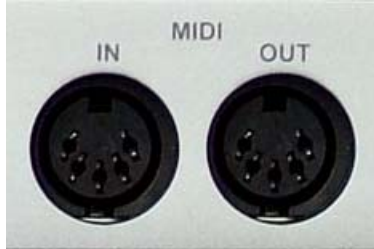
MIDI cabling is highly standardized and widely available.

Word Clock

Next to the MIDI ports are the **Word Clock** connectors. These connectors allow you to synchronize your Echo product to any other device equipped with a Word Clock connector.

The Word Clock I/O uses a BNC connector. As with the S/PDIF I/O, a shielded 75-ohm (RG-59) coaxial video cable should be used. BNC connectors have been widely used in the electronics industry for both video and computer networking. However, computer networks use 50-ohm (RG-58) coaxial cables and not the 75-ohm (RG-59) cable used by video. For reliable word clock operation, use only video grade cables with your Echo product.

Software Configuration

Now that you have installed your AudioFire product, you can verify that everything is working properly.

Open "System Preferences" from the Dock or Apple menu, select "Show All", and select the "Sound" panel. Under the "Output" tab, you should see your AudioFire product listed as a sound output device. If you want to use your AudioFire product with iTunes, iDVD, iMovie, QuickTime, etc. you will need to select it as the sound output device.

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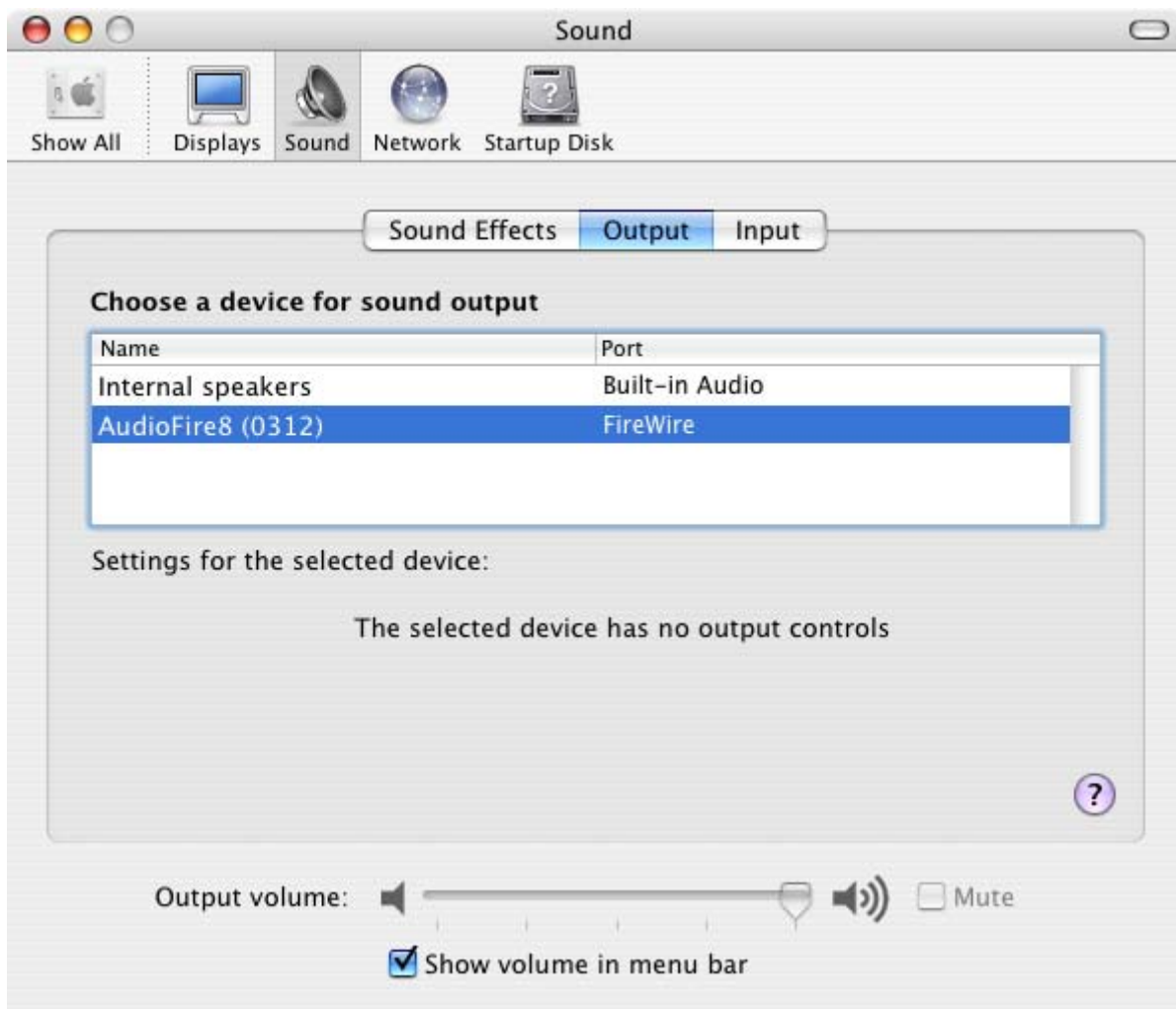
You can also check out your hardware with the "Audio MIDI Setup" application. Go to your Applications folder on your hard disk and from there to the Utilities folder. Open "Audio MIDI Setup."

In addition to listing all the audio devices in the system, Audio MIDI Setup lets you examine the various audio formats supported by the hardware in your Mac.

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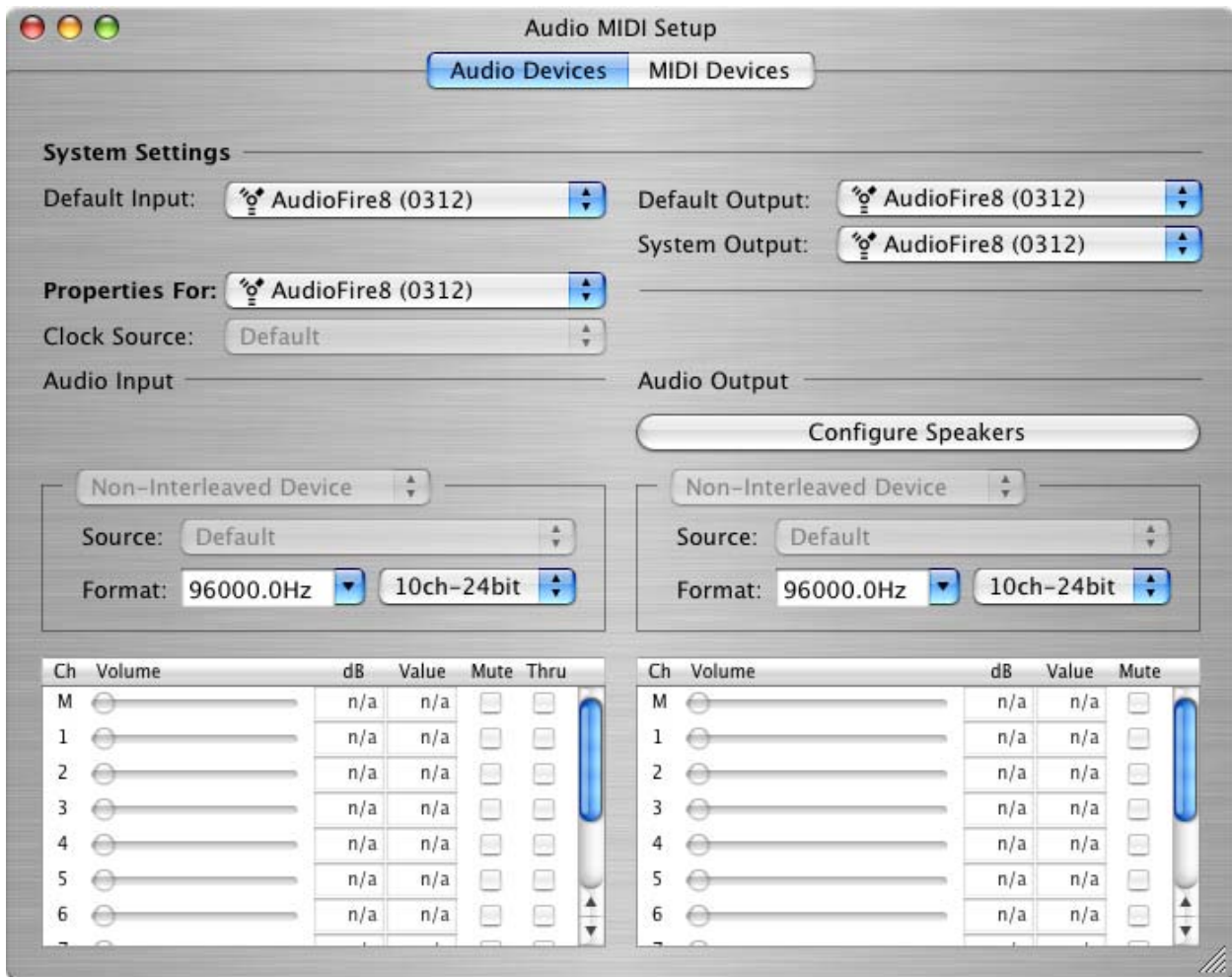
By clicking on the "MIDI Devices" tab, you will display the available CoreMIDI devices. You should see your AudioFire8 or AudioFire12 hardware shown as an icon here.

Next, you'll want to install Tracktion and learn about the AudioFire Console software. Both are covered in the following sections of this manual.

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Installing Tracktion

Both AudioFire8 and AudioFire12 come bundled with the full version of Tracktion, audio multitrack recording, MIDI, editing, and sampling software.

Install Tracktion from the AudioFire CDROM

To install the software:

1. Insert the AudioFire CDROM into your CD-ROM drive. A CDROM icon will appear on your desktop called "AudioFire". Open it, and then open the "Tracktion" folder.

2. Drag the Tracktion icon to the "Applications" folder.

Tracktion comes with an HTML "Quick-start Guide" and help files that you can open from the "Help" button inside Tracktion. So, you'll be able to get up and running with a minimum of hassles (hopefully none at all!). You can also access the Tracktion "Quick-start Guide" and help files with the F12 key on your keyboard.

Authorize your copy of Tracktion

Your copy of Tracktion will run in demo mode until you authorize it. Your AudioFire CDROM came with a sticker that has an authorization code and a website address printed on it. Go to that website and follow the instructions to authorize your copy of Tracktion.

NOTE: **<u>DO NOT LOSE YOUR CODE!!</u>** Echo Digital Audio Corporation will not be responsible for lost Tracktion authorization codes.

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Launching Tracktion and Assigning Inputs & Outputs

Now it's time to launch the actual program and set a few options:

- Locate and double-click the Tracktion icon in the "Applications" folder or wherever you copied it to earlier. You may also want to put it in the Dock for easy access.

When Tracktion first runs, it takes you to the Projects page. Move the mouse pointer over various objects on the screen and you will notice that Tracktion uses pop-up help to tell you about the interface.

Note that whenever you click on something you will see any available details, controls or settings for it in the information windows at the bottom middle of the screen. You can learn a lot just by clicking on things and reading the pop-up help.

You can also access a quick start guide and a more detailed help document by clicking on the "help!" button in the bottom left corner and selecting "show the Tracktion help pages…" or by pressing the F12 key on your keyboard. The first thing you'll want to do is set up Tracktion to play and record with your AudioFire product.

Audio Settings

1. Click the "Settings" tab at the top of the screen.
2. Select "Audio Devices" on the left.
3. Where it says "Wave device:" click on the drop down menu and select your AudioFire product.
4. Now you should be able to see a list of the available audio devices for your AudioFire product. Go ahead and enable the inputs and outputs you want to use with Tracktion by clicking on the red X and waiting for it turn into a green checkmark.
5. Make sure one of the outputs of your AudioFire product is the "default wave output", and that it is hooked up to some speakers or headphones. You can check this setting on the right side of the "Settings – Audio devices" screen of Tracktion.
6. Now you're ready to play some music. Click on the "Projects" tab at the top of the screen and select one of the sample projects that come with Tracktion.
7. You will see a listing of all the files for that project in the large area to the right. Double-click on the file with the same name as the project. This will open the project.
8. Now click on the play button in the bottom right section of the screen and you'll hear the sample project with your new AudioFire product.
9. Enjoy the music!!

There are more detailed instructions for Tracktion in the HTML "Quick-start Guide" and help files, which are available by hitting the F12 key while in Tracktion. You can also find hints and tips in the Tracktion user's forums available through the Tracktion webpage here:

http://www.mackie.com/products/tracktion/tracktion.html

NOTE: Tracktion version 1 does not support sample rates above 96kHz.

This completes the installation of the hardware and software for AudioFire8 and AudioFire12.

Console

If you followed the instructions in Step 1 of the "Installing the AudioFire Hardware and Software" section of this manual, then you installed a "virtual control surface" application called the AudioFire Console. The console allows you to control the audio I/O and clocking functions of your AudioFire product, and it brings these controls to a single easy-to-use location. From the console you can control the output levels, select synchronization clocks, and adjust input monitoring. The examples in this section will be for AudioFire8. The controls for AudioFire12 are similar.

You can open the AudioFire console by going to the "Applications" folder (or wherever you chose to put it during installation) and double-clicking on the icon. Alternately, you can drag the icon to the Dock for easier access and just open it from there.

Console Window

The basic metaphor for the console interface is a digital mixing board, and it works in terms of ***output busses***. An output bus represents a pair of outputs on your AudioFire hardware. For example, analog outputs 1 and 2 are an output bus. The console window allows you to select an output bus and control what is mixed and sent to that output bus. Audio played through the selected channels by an application, any analog input, and any digital input (AudioFire8 only) may be mixed to an output bus.

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The AudioFire8 Console Window: Analog Out 1-2 Tab selected.

Bus Select Tabs

The output bus is selected by clicking on one of the tabs at the top of the Console window. The above example shows the Console window for AudioFire8. Five different output busses are shown here: four analog busses and one digital bus. Each output bus has its own tab, marked either "ANALOG" or "DIGITAL", and a channel number pair. When you select a tab its title will turn white, and the other tab titles will be gray. In this case "ANALOG OUT 1-2" is the currently selected output bus. Clicking on an output bus tab simply selects the **settings that are displayed**. This means all input monitor parameters such as volume, pan, and mute **only apply** to the audio coming out **analog outputs 1 and 2** of AudioFire8. To select, view and change the input monitoring and playback settings for another set of outputs you must select a different output bus by clicking on one of the other

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output bus tabs. AudioFire12 does not have S/PDIF, so it does not have a digital output bus; it has 6 analog output bus tabs instead.

Master Output Bus Control

The master bus control is the area on the far right of the Console window. At the top are the titles "ANALOG", "OUT 1", and "OUT 2." This indicates you are currently controlling everything mixed to analog outputs 1 and 2. These channels correspond to the currently selected output bus tab.

Beneath the titles are two green label fields that can be changed by clicking in them and typing. Below the green label fields are mute buttons, level meters and faders. The *mute buttons* and *faders* affect **everything being mixed to this bus**. Thus, pulling the fader down will make both the input monitors and playback quieter. The level meters display the amount of signal being sent out of the selected outputs.

The *gang button*, marked with a "G," sets the gang mode. If you are in gang mode, the mute buttons will work together for the left and right channels; for example, clicking the mute on one channel activates the mute button on the other channel. Also, ganging ties the faders together so they will maintain their relative placement with regard to each other. Holding down the Shift key temporarily reverses the state of the gang button; that is, if the gang button is down, holding down shift puts you in non-ganged mode and vice versa.

Analog Inputs

The controls for the analog inputs are on the left side of the Console and have the titles "ANALOG" and "IN 1" through "IN 8" at the top. There is one strip, or *input channel*, corresponding to each of AudioFire8's eight physical analog inputs. The input level meters remain viewable at all times, whereas the input monitor controls (pan, solo, mute, faders and gang) that you see will change depending on which output bus is currently selected.

The input monitor controls are a little complicated. Basically, there is *a different set of input monitor controls for each output bus*, but *only one set is viewable and changeable at a time*. In the example above, the only set of input monitor controls that is viewable and changeable is the one for the output bus called "ANALOG OUT 1-2". To see and change a different set of input monitor controls you will need to select a different output bus (see the "Bus Select" section above).

The level meters do not change when a different output bus is selected. They always reflect the signal level being received on the inputs. In the example 2 pages earlier you can see that the signal level on the ANALOG OUT 1-2 meters is less than that on the input meters. This is because the ANALOG IN 1-2 faders have been adjusted to –19, thereby decreasing the amount of the input signal that is being sent to analog outputs 1-2. If we were to select the ANALOG OUT 3-4 tab the Console would appear as in the example below.

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Note that since the ANALOG OUT 3-4 tab is selected you are no longer looking at the input monitor and output controls for analog outputs 1 and 2. You are now looking at the controls for analog outputs 3 and 4. You can still see the input signal on the input level meters, but the output level for analog outputs 1 and 2 is no longer visible since you have changed to the ANALOG OUT 3-4 bus.

At the top of each input channel, below the title, is a green label that can be changed by clicking in it and typing. The titles, labels, and level meters are the only things in this section that do not change when different output bus tabs are selected.

Moving down the input channel strip you find where the input monitor controls begin, starting with the input monitor pan knob. This knob changes how the input channel is monitored between the two channels of the currently selected output bus. Double-clicking on a pan knob will set it to the center position. Below each pan knob is a non-selectable green field that tells you the current setting of the pan knob. Note that adjusting the pan knob will not affect your recording, as the input monitor controls only affect how the input signal is sent to (or monitored on) the outputs.

Below the pan knob are the monitor solo and mute buttons. Activating the solo button will make sure that the selected output bus only monitors the inputs that are soloed. Activating the mute button on an input channel will prevent it from being monitored by the currently selected output bus. Again, this does not affect recording. These are followed by the monitor fader, which controls the input monitor level. Like the pan, solo, and mute controls, this does not affect your recording level. The faders also have gang buttons, and their operation is the same as the gang button for the master output bus controls.

Digital Inputs (AudioFire8 only)

The controls for the digital inputs can found to the right of the analog input controls and have the title "DIGITAL", "IN 1", and "IN 2" at the top. Their operation is identical to the analog inputs.

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Playback Controls

The playback controls are to the left of the master output controls and are similarly titled, except instead of "OUT" they are titled "PLAY". The playback controls affect the level of audio being played by an application out the currently selected output bus. These controls operate the same as the input monitor controls, except there are no pan knobs, solo buttons, or level meters.

Adjusting Playback Volume

Output volume adjustments are made in the digital domain. When you lower a volume fader below 0dB, you are actually decreasing the number of available bits, thereby taking away from the potential dynamic range of the system. Therefore, we suggest that whenever possible you leave the output faders (Playback & Output Bus) set to 0dB, and perform any necessary attenuation on your external mixer. When the playback volume controls are set to 0dB and the input signal approaches the maximum pre-clipping level, you can achieve the full 24-bit dynamic range of the system.

Console Settings

Clicking the Settings button above the playback strips opens a window for altering console settings. The Settings window gives you five different tabs to choose from: "Input," "Output," "Clock," "Digital," and "About."

## Inputs: Nominal Levels

The "Input" tab allows you to change the *nominal level* settings for the analog inputs. By default, the inputs are set to receive a +4dBu signal. You can switch between +4dBu (professional level) and –10dBV (consumer level) for each analog input by selecting the appropriate radio button. This feature allows you to connect either professional or consumer gear to each input.
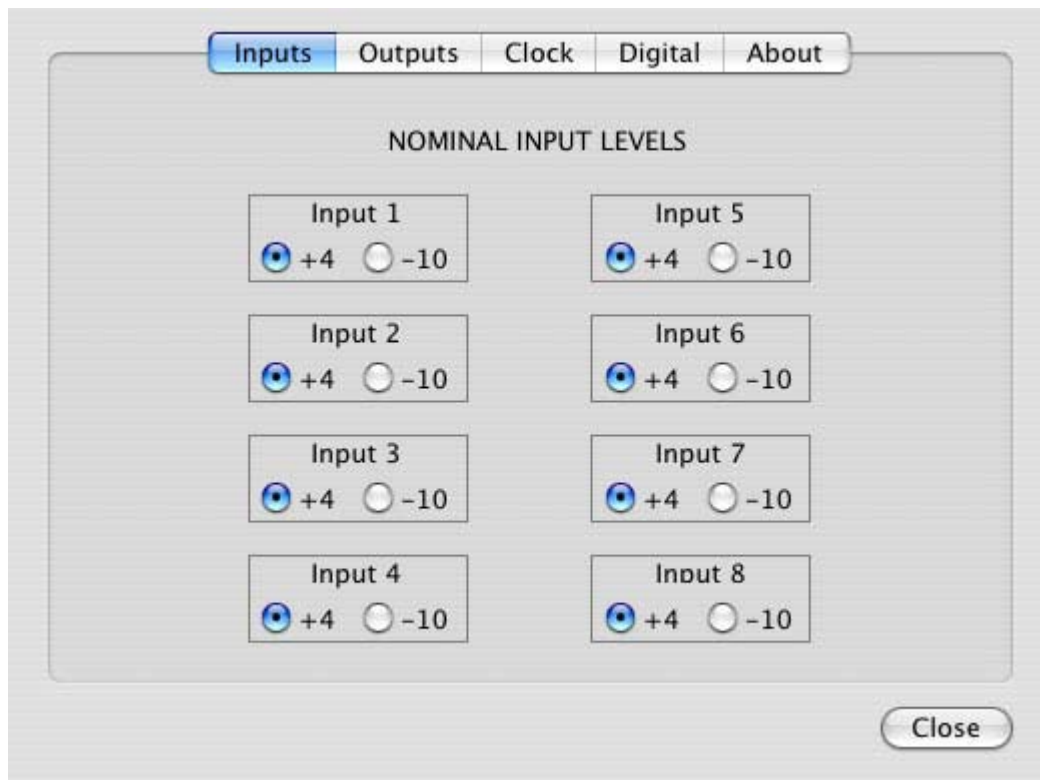
## Outputs: Nominal Levels

The "Output" tab allows you to change the *nominal level* settings for the analog outputs. By default, the outputs are set to send a +4dBu signal. You can switch between +4dBu (professional level) and –10dBV (consumer level) for each analog output by selecting the appropriate radio button. This feature allows you to connect either professional or consumer gear to each output.
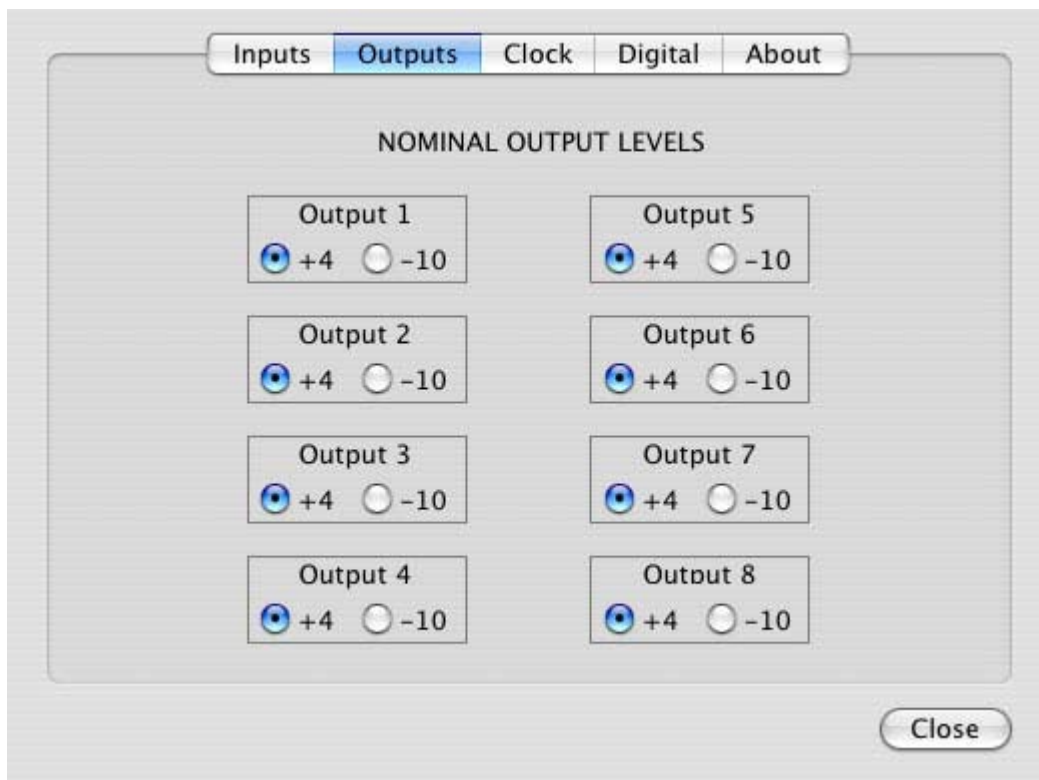
Clock Settings

The "Clock" tab lets you select your clock rate and the input clock source. The clock rate, also known as sample rate, affects recording and playback. For more information please refer to Appendix B: An Introduction to Digital Recording.

The input clock setting is used to synchronize your AudioFire hardware to another piece of audio hardware. If a given synchronization source (input clock) is detected, the indicator next to the input clock button will turn green. You will only be allowed to select input clocks that have been detected. For more information please refer to "Synchronizing Multiple Devices" later in this manual.
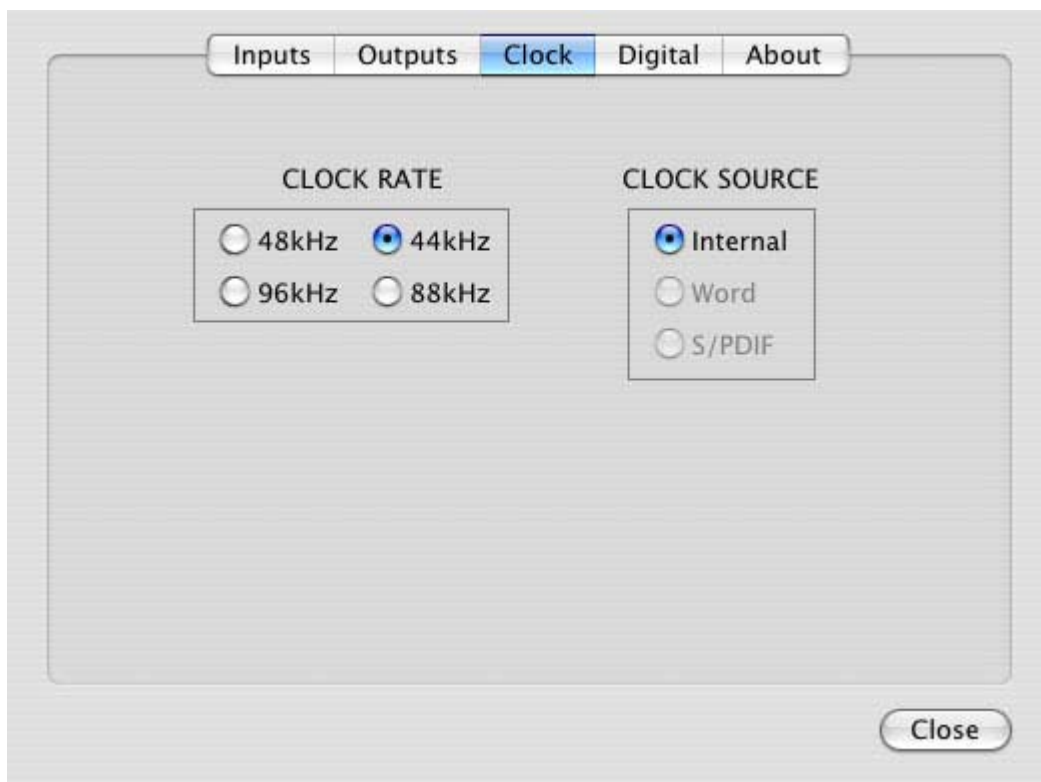
Digital: S/PDIF Settings

The "Digital" tab lets you set the format of the S/PDIF outputs for AudioFire8.
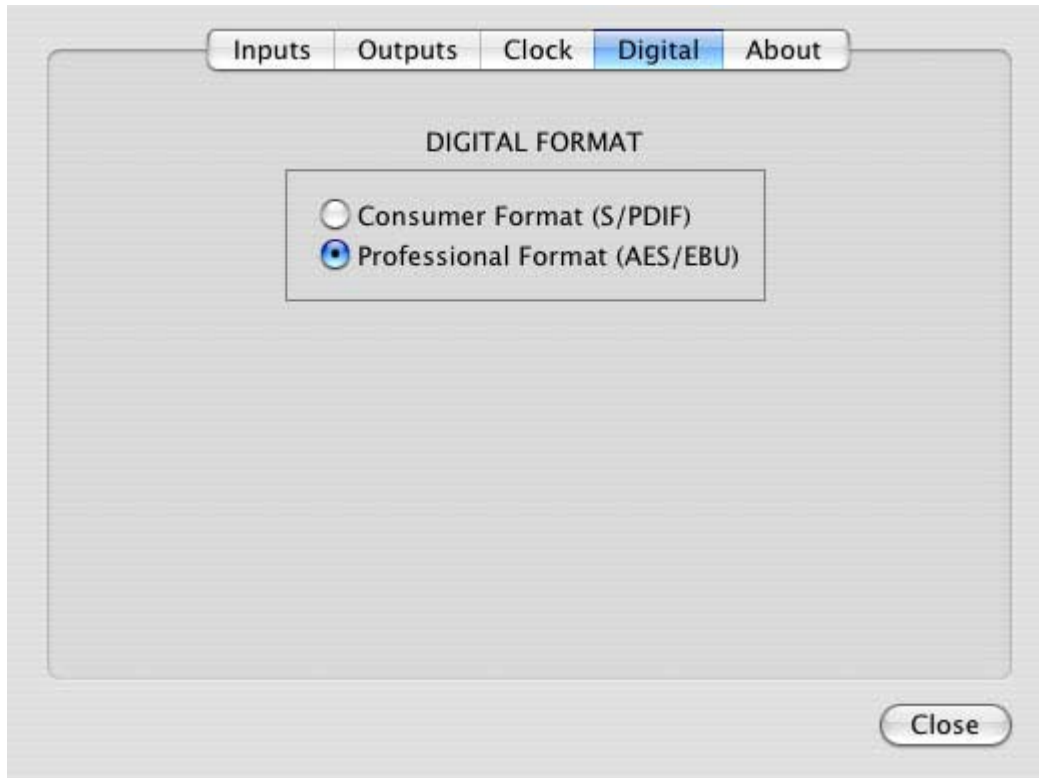
AudioFire8 can transmit digital information in either of two formats: "professional" or "consumer." The primary difference between the two is in the implementation of the SCMS copy protection bit, which, in the Consumer format, prevents the user from making digital copies of a digital copy.

The S/PDIF output defaults to the Consumer format. If you are recording from your AudioFire8 into a professional DAT deck, CD recorder, or Mini Disc recorder, the device may not be able to recognize the signal until you switch the output to Consumer.

Select the appropriate format for your DAT, CD recorder, or Mini Disc recorder (if you don't know which one to use and are having difficulties, simply try the one that is not currently checked).

> **Note:** No Echo hardware transmits the SCMS bit, regardless of which format is selected.

About

The "About" tab displays the console version and driver version numbers. We feel that this is the single most exciting feature of the console ☺

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Console Sessions

Console settings can be saved as a console session to be reloaded later. To save a console session, click the "Save" button in the main console window. To load a previously saved session, press the Load button in the console.

By default all sessions are saved to your user directory. Sessions are not cross compatible between different types of hardware; that is, an AudioFire8 session cannot be used with an AudioFire12 session. All console settings are saved, including:

- Volume faders and pan knobs
- Solo, mute, and gang buttons

The various options available in the "Settings" window are not saved as part of the session. To return the console settings to their default state, simply load the session titled "default."

Synchronizing Multiple Devices

AudioFire8 and AudioFire12 are designed to work alongside other audio equipment. If you are planning on using AudioFire8 or AudioFire12 with other audio equipment, please note the following:

Echo hardware can peacefully coexist with audio equipment from other manufacturers, but be aware that operating *alongside* another product is not the same as operating *with* it. In order for accurate synchronization to occur, the other audio product(s) must support a synchronization mode that is compatible with your particular Echo hardware. Without such synchronization, the individual pieces of equipment will act independently of each other. This scenario may be fine for some musical applications; however, it is not appropriate for situations where sample-accurate synchronization is required.

Let's take a brief look at the various synchronization types.

Word Clock – The **Word Clock** connectors can be found on the back panel. When your AudioFire product is set to "Internal" this synchronization clock runs at the AudioFire product's selected sample rate. Think of it as a kind of electronic metronome, which clicks back and forth at the digital sample rate. It is one of the most widely used forms of synchronization in digital audio. Your AudioFire product is always generating Word Clock on its Word Clock output BNC connector. Although it can generate Word Clock at any sample rate it is set to, your AudioFire product can only sync to Word Clock if the master device is set to a sample rate between 30kHz and 100kHz (200kHz for AudioFire12). Otherwise you will get noise and/or loss of sync.

> **Note:** When synchronizing your AudioFire product to an external device via S/PDIF or Word Clock, if the sample rate is changed on the external device then you may have to reset the input clock setting on your AudioFire product. To do this, merely click on the input clock button again (S/PDIF or Word).

S/PDIF (AudioFire8 only) – The Sony/Phillips Digital Interchange Format is a serial bit-stream that has a clock signal embedded in the data stream. When recording from an S/PDIF source, AudioFire8 will utilize the synchronization clock that is embedded in the S/PDIF while it decodes the bit-stream. AudioFire8 can only sync to S/PDIF clock if the master device is set to a sample rate between 30kHz and 100kHz. Otherwise you will get noise and/or loss of sync.

> **Note:** When recording from a S/PDIF port, you must select **S/PDIF** as the input clock. For greater flexibility, this is not done automatically. If you find that your S/PDIF recordings contain pops or skips, be sure that you have selected **S/PDIF** as your input clock.

Now let's take a look at some sample configurations and how you might set them up from a synchronization standpoint.

Let's start with a simple example. Suppose that AudioFire8 is the only audio device used in your studio. Since you have no other devices to synchronize with, simply select **Internal** for AudioFire8's input clock. AudioFire8 will then use its own clock to control its operation.

Now a little more complicated set-up: You want to have an AudioFire8 synchronized to another device via word clock. Simply set the AudioFire8 to **Internal** for its input clock. Now connect the AudioFire8 to the other device via a BNC cable running from Word Clock Out on the AudioFire8 to Word Clock In on other device. Now select **Word** for the other device's input sync. The other device will slave to your AudioFire8 and they will now operate in unison.

No matter how many devices you are synchronizing, the concept is essentially the same. You are merely "daisy-chaining" devices together using compatible clocks. One device will operate as the source of the master clock, with each successive device using that clock to sync.

Contacting Echo Customer Service

If you experience any trouble with your Echo product please go to the support area of our website at www.echoaudio.com, and check out the FAQ's we have there. If you can't find a solution to your problem there, please fill out the provided technical support email form. This form will be sent to our technical support staff, and they will respond to you quickly. **Please fill out the form completely.** The best way to get the help you need is by giving us plenty of detailed information about your computer system, your audio software and hardware, and the problem you are having.

We do ask you to please read through this manual and the support area of our website before contacting us. Also, you may find an answer to your problem in the Appendices of this document – starting on the next page.

Thank you for buying an Echo product!

Contacting Mackie Customer Service

> Note: All Tracktion support issues are handled solely by Mackie. **Please do not call Echo for technical support regarding Tracktion.**

You can find helpful information about Tracktion and links to user and support forums on the Internet at:

http://www.mackie.com/products/tracktion/tracktion.html

If you are still having problems with Tracktion you can contact Mackie Technical Support here:

http://www.mackie.com/support/index.html

Appendix A: General Troubleshooting Guide

Problem: You can't get your AudioFire product to install properly under Mac OS 8, OS 9, or OS X (10.3.8 or lower).

Solution: There is no driver support for AudioFire8 or AudioFire12 under Mac OS 8 or 9. There is only driver support for Mac OS X (10.3.9 or later).

Problem: You've installed your AudioFire product, but it isn't listed in the System Preferences or the Audio MIDI Setup application.

Solution: Your installation of OS X may be out of date. Try running Software Update; you'll find it in the System Preferences. You need version 10.3.9 or later.

Problem: You can't use more than one Echo product at a time with your audio software.

Solution: This is a limitation of your audio software. Currently, Apple's CoreAudio spec does allow multiple hardware devices to be used simultaneously by an audio software program, but most audio software programs currently do not support this feature.

Problem: When you connect a bus powered FireWire device to your AudioFire product it does not receive power.

Solution: Your AudioFire product does not pass power from the computer on its FireWire connector. Use a FireWire device with an external power adapter.

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Problem: You are unable to get your DAT recorder to recognize the S/PDIF output from AudioFire8.

Solution: Digital information is transmitted in either of two modes, "professional" or "consumer." The professional mode is usually implemented in devices that are likely to be used in professional recording environments, whereas the consumer mode is commonly implemented on equipment designed for home use in the consumer market. The primary difference between the two modes is in the implementation of the SCMS copy protection bit, which, in the consumer format, prevents the user from making digital copies of a digital copy. In most professional equipment, this copy-protection bit can be turned off or on according to the user's needs. In consumer products, the SCMS bit is always enabled.

Unfortunately there is no way for the transmitting device to automatically detect which format the receiving device is able to accept. If you have a DAT deck that is not able to read the S/PDIF output from AudioFire8, chances are it is transmitting in the mode that the deck is not equipped to handle.

We have provided a software switch in the driver that allows you to select which mode AudioFire8 transmits. To access this switch go to the **Console**. Click on the **Settings** button and select the **Digital** tab. In the window that appears you'll see a pair of radio buttons in an area labeled **S/PDIF Format**; one radio button is labeled **Consumer** and the other **Professional**. Select the appropriate format for your DAT (if you don't know which one to use, simply select the one that is not currently checked). Now click the **OK** button and again try recording to your DAT.

> **Note:** AudioFire8 never transmits the SCMS bit; regardless of which mode is selected.

Problem: When you play an audio file, it plays at an altered pitch.

Solution: When your AudioFire product is set to synchronize with an external device, it will play back at the rate generated by that device. If the sound you are playing was sampled at 44.1kHz (for example), but you are synchronized with a device running at 96kHz, the sound will play back at this faster rate. You have four choices - ignore the altered pitch, switch to your AudioFire product's internal clock, change the sample rate of the external device, or use a different device for the sound playback.

Problem: Your AudioFire product doesn't seem to recognize the synchronization clock to which it is connected.

Solution: Although it may seem obvious, the first thing to check is that there is a physical connection between the device generating the clock and your AudioFire product. Just because multiple devices are connected to the same computer doesn't mean they are synchronized. Next, be sure that you have selected the desired input clock source in the Console for your AudioFire product.

Problem: You're syncing to an external device and suddenly you've lost sync.

Solution: Whenever your AudioFire product is syncing to an external device and you change the sample rate in that device, you may need to reset the input clock setting in the AudioFire Console software. Just click on the appropriate input clock button to reset it.

Problem: Recordings made using AudioFire8's S/PDIF port contain occasional pops or skips.

Solution: When recording with AudioFire8's S/PDIF port, you must manually select the S/PDIF clock as the input clock. This can be done using the **Console**.

Problem: The sound cuts in and out, or the left or right channel doesn't work.

Solution: You may have a bad audio cable. Try using a different cable in the problem channel.

Problem: There is no sound unless I pull the audio plug out a little.

Solution: Not all ¼" plugs are made to the exact same dimensions. Try a plug/cable from a different manufacturer.

Appendix B: An Introduction to Digital Recording

Converting Sound into Numbers

In a digital recording system, sound is represented as a series of numbers, with each number representing the voltage, or amplitude, of a sound wave at a particular moment in time. The numbers are generated by an *analog-to-digital converter*, or ADC, which converts the signal from an analog audio source (such as a guitar or a microphone) connected to its input into numbers. The ADC reads the input signal several thousand times a second, and outputs a number based on the input that is read. This number is called a *sample.* The number of samples taken per second is called the *sample rate*.

On playback, the process happens in reverse: The series of numbers is played back through a *digital-to-analog converter*, or DAC, which converts the numbers back into an analog signal. This signal can then be sent to an amplifier and speakers for listening.

In computers, *binary numbers* are used to store the values that make up the samples. Only two characters, 1 and 0, are used. The value of a character depends on its place in the number, just as in the familiar decimal system. Here are a few binary/decimal equivalents:

BINARY	DECIMAL
0000000000000000	0
0000000000000001	1
0000000000000011	3
0000000000000111	7
0000000000001111	15
0000000000011111	31
1111111111111111	65,535

Figure A. Binary numbers and their decimal equivalents

Each digit in the number is called a *bit.* The binary numbers expressed in Figure A are sixteen bits long, and have a maximum value of 65,535. The more bits that are used to store the sampled value, the more closely it will represent the source signal. In a 16-bit system, there are 65,535 possible combinations of zeroes and ones; so 65,535 different voltages can be digitally represented. (see Figure A above).
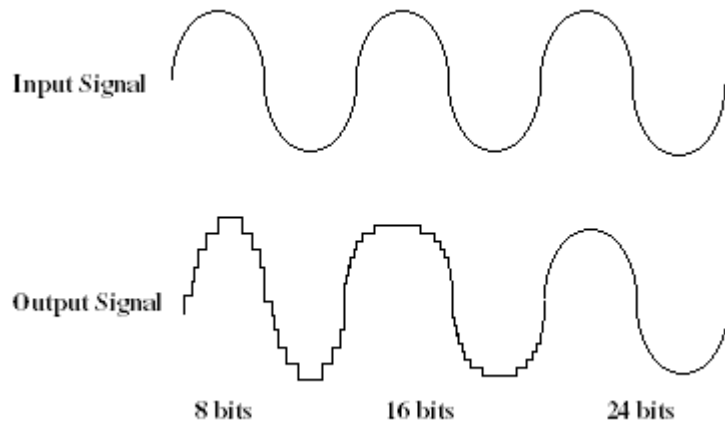
Input Signal

Output Signal

8 bits   16 bits   24 bits

Figure B. The more bits there are available, the more accurate the representation of the signal and the greater the dynamic range.

Your Echo product's analog inputs use 24-bit ADCs, which means that the incoming signal can be represented by any of over 16 million possible values. The output DACs are also 24-bit; again, over 16 million values are possible. The S/PDIF inputs and outputs also support signals with up to 24-bit resolution. Your Echo product processes signals internally with 24-bit resolution to insure that there is no degradation to the audio signal as it is processed through the system.

The number of bits available also determines the potential dynamic range of the device. Moving a binary number one space to the left will multiply the value by two, so each additional bit doubles the number of possible values that may be represented. Each doubling of the number of values provides 6dB of additional dynamic range (see decibel section below). So, a 24-bit system can theoretically provide 144dB of dynamic range (6dB times 24 bits = 144dB) versus a 16-bit system with a maximum dynamic range of only 96dB.

Also important to the quality of a digital recording is the frequency with which the samples are stored, called the sample rate. In order for a waveform to be faithfully digitized, it must be sampled at a minimum of twice the highest frequency to be stored. Failure to sample frequently enough results in a kind of distortion called aliasing. (If you like technical issues, do some research on The Nyquist Theorem, which explains why this happens). In addition to aliasing, sampling too slowly will result in reduced high frequency reproduction. Your Echo product allows you to sample sound at up to 96,000 times per second (192,000 with AudioFire12).

Once the waveform has been transformed into digital bits, it must be stored. When sampling in stereo at 96kHz using a 24-bit word size, the system has to accommodate 4,608,000 bits per second. In the past, storing this vast amount of data was problematic. Today, computer-based digital recording systems record the data directly to the computer's hard disk. Today's hard disks are capable of storing large amounts of data, though the performance of hard drives can vary substantially. The speed and size of your hard drive will be a major determining factor in how many tracks of audio you will be able to simultaneously record and playback.

Decibels

Audio signal levels are generally expressed in units called "decibels" which are abbreviated as "dB". This is a "logarithmic" scale where each doubling of signal level is represented by an increase of 6dB. Therefore a signal of 6dB is twice as big as a 0dB signal and a signal of 12dB is four times as big as a 0dB signal.

Since digital audio signals are represented by binary data, each bit of audio information represents 6dB. A 16-bit number can represent a total range of 96dB and a 24-bit number can represent a total range of 144dB (6 times the number of bits). It's much easier to say that one signal is 72dB less than another instead of saying it is 1/4096 the size of the other one. It also more accurately represents the way we hear sounds, since the smaller signal in the above example will still be audible and not appear to be only 1/4096 as loud when we listen to it.

Just as there are different types of degrees used to represent temperature (Fahrenheit, Celsius, etc), there are different types of decibels used to represent the

level of analog audio signals. The most common are dBu and dBV decibels. Both of these represent voltage levels and still double for every increase of 6dB. It is only the reference point, or 0dB level that is different. A 0dBV signal has a voltage level of 1.0 volts. A 0dBu signal has a voltage level of .775 volts. Since .775 is approximately 2dB less than 1.0, converting dBV levels into dBu levels is as simple as subtracting 2dB (2.21 to be exact).

Signals are also occasionally represented with units of dBm. This is an older unit that measures power instead of voltage levels with 0dBm representing 1 milliwatt. Earlier tube-based audio equipment used standardized input and output impedances of 600 ohms, so a 0dBm signal was produced with a voltage of .775 volts. Since most of today's equipment uses impedances other than 600 ohms, it is more useful to represent signals by voltages rather than power and the dBu unit was introduced. A signal level of 0dBu is identical to a level of 0dBm.

Digital signals, after they are recorded, no longer directly represent any physical quantity such as voltage or power and 0dB is generally used to represent a "full-scale" or maximum signal level. All other signal levels are lower and are expressed as negative decibels. Most meters on digital equipment have 0dB at the top and range downward from there. A signal that is 30dB below full scale would simply be referred to as a –30dB signal.

Nominal Signal Levels and Headroom

Today's equipment is generally referred to as +4dBu equipment (professional) or –10dBV equipment (consumer). These levels are the typical or "nominal" signal levels you can expect to see with professional (studio) equipment such as mixers or with consumer equipment such as home stereos and CD players. A +4dBu signal has a voltage level of 1.23 volts and a –10dBV signal has a voltage level of .316 volts.

The above nominal levels represent typical or average levels that are often exceeded when recording loud signals such as drum beats. The difference between the nominal level and the loudest signal that can be recorded without clipping is called "headroom". Your Echo product provides approximately 14dB of headroom allowing an 18dBu signal to be recorded.

Unbalanced and Balanced Inputs and Outputs

An unbalanced signal, commonly used for guitars and consumer electronics, contains two components, a ground signal and a "hot" or active signal. The ground is the barrel of a ¼" connector and the shell of an "RCA" style connector.

A balanced signal contains two active signals instead of one in addition to the ground. These are referred to as the "plus" and "minus" signals. A balanced input amplifier amplifies the difference between these two signals. Any extraneous noise picked up from power lines or other sources will appear equally on both the plus and minus inputs. This is called "common mode" noise since it is common to both signals and the input amplifier will subtract the noise on the minus input from the noise on the plus input.

If the input amplifier is perfectly balanced and the noise on both plus and minus is precisely equal, the noise will completely cancel out. In the real world this is not the case and some of the common mode noise will still make it through, although at a much reduced level. How well an input amplifier rejects this common mode noise is called the "common mode rejection ratio" (abbreviated as CMRR) and is expressed in dB.

Balanced outputs typically drive the plus and minus components of a balanced signal in one of two ways. A "differentially" balanced output signal is one where the minus output is the mirror image of the plus output. That is, if the "plus" output is at +1 volt, then the "minus" output is at −1 volt.

An "impedance" balanced output provides most of the benefits of a differentially balanced output at a lower cost. In this case only the plus component actually carries the signal. The minus component is tied to ground through a resistor so that the output impedance matches that of the plus component. Since the impedances are matched, any noise added to the minus component will still be approximately the same as that picked up by the plus component and will be subtracted out by the input amplifier. Your Echo product uses this type of output.

Balanced signals connect with either XLR connectors or TRS (tip, ring, sleeve) connectors. Your Echo product uses XLR connectors for microphones and TRS connectors for connecting balanced (and unbalanced) line level signals.
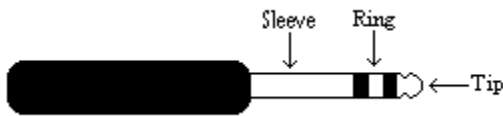
The three sections of a TRS connector are used to transmit the three components of a balanced signal (T = plus, R = minus, S = ground).

Dynamic Range

Dynamic range represents the difference between the maximum signal that can be recorded and the "noise floor", or level of noise with no signal present. A system with a high dynamic range will be quieter than one with a lower dynamic range. Dynamic range is a very important specification, and your echo product uses converters that have very high dynamic range.

Theoretically, a 24-bit system has a dynamic range of 144dB and a 16-bit system has a dynamic range of 96dB. Two questions immediately come to mind:

1) Why does my Echo product only have a dynamic range of 110 to 114dB?
2) For mastering 16-bit CDs with a dynamic range of 96dB, isn't anything more than 96dB just overkill?

First, today's analog-to-digital converters typically produce a full-scale input voltage with an input of +7dBu. If they were to have 144dB of dynamic range, they would have to be capable of resolving signals as small as –137dBu (7dBu – 144dBu) or approximately 10 nano-volts. That's 10 *one-billionths* of a volt! Transistors and resistors produce noise in this range just by having electrons moving around due to heat. Even if the converters could be perfectly designed to read these levels, the low noise requirements of the surrounding circuitry such as power supplies and amplifiers would be so stringent that they would either be impossible or too expensive to build.

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In answering the second question, consider the fact that music is often compressed or amplified after it is recorded, and that some headroom is necessary when recording to avoid clipping. The only way that 96dB would be adequate is if all music were recorded so that the peaks were just under full-scale and no compression or amplification was going to be applied after recording. Any time recorded music is amplified, so is the noise at the low end. Your Echo product has enough dynamic range to allow sufficient headroom and post-processing to be applied while still keeping the noise either off the recording completely or down as far as possible.

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Appendix C: Specifications

<u>AudioFire8 - Audio Performance</u>

Analog Inputs (x2 balanced universal XLR/TRS & x6 balanced TRS):

- Universal Inputs (x2 balanced XLR/TRS):
  - Frequency Response: 10Hz to 20kHz, ±0.2dB
  - Microphone Input Level (balanced XLR):
    - EIN: -129dBu
    - Dynamic Range: 111dB A-weighted
    - THD+n: <0.004% A-weighted
    - Input Impedance: 1.5Kohms
    - Gain Adjustment: +10 to +59dB
    - Maximum Input Level at Minimum Gain: +9dBu
  - Guitar/Line Input Level (balanced TRS):
    - Dynamic Range: 110dB A-weighted
    - THD+n: <0.001% A-weighted
    - Input Impedance: 102Kohms
    - Gain Adjustment: 0 to +45dB
    - Maximum Input Level at Minimum Gain: +18dBu
- Normal Line Inputs (x6 balanced TRS):
  - Frequency Response: 20Hz to 20kHz, ±0.1dB
  - Dynamic Range: 113dB A-weighted
  - THD+n: <0.002% A-weighted
  - Linearity: ±0.01dB, 0dBFS to 100dBFS
  - Nominal Input Level: +4dBu or –10dBV
  - Input Impedance: 10Kohms
  - Maximum Input Level: +18dBu

Analog Outputs (x8 balanced TRS):

- Frequency Response: 10Hz to 20kHz, ±0.1dB
- Dynamic Range: 114dB A-weighted
- THD+n: <0.002% A-weighted
- Nominal Output Level: +4dBu or –10dBV
- Maximum Output Level: +17.2dBu

AudioFire8 - Hardware

- Host Interface: 1394a (FireWire)
- Two high quality mic preamps with phantom power (+48v)
- Eight balanced ¼" analog inputs with precision 24-bit 128x oversampling analog-to-digital converters
- Eight balanced ¼" analog outputs with high performance 24-bit 128x oversampling digital-to-analog converters
- S/PDIF digital I/O with up to 24-bit resolution
- Trim control, level meter, and channel inserts for analog inputs 1 and 2
- Headphone output with volume control
- On-board 32-bit / 1.6 gigaflop DSP
- 24-bit data resolution maintained throughout entire signal path
- Multiple sample rates from 32kHz to 96kHz
- Variable sample rate feature lets you select any rate between 32kHz – 96kHz
- MIDI In/Out
- Word clock I/O

AudioFire12 - Audio Performance

Analog Inputs (x12 balanced TRS):

- Frequency Response: 20Hz to 20kHz, ±0.1dB
- Dynamic Range: 113dB A-weighted
- THD+n: <0.002% A-weighted
- Nominal Input Level: +4dBu or –10dBV
- Input Impedance: 10Kohms
- Maximum Input Level: +17.5dBu

Analog Outputs (x12 balanced TRS):

- Frequency Response: 10Hz to 20kHz, ±0.1dB
- Dynamic Range: 114dB A-weighted
- THD+n: <0.002% A-weighted
- Nominal Output Level: +4dBu or –10dBV
- Maximum Output Level: +17.5dBu

AudioFire12 - Hardware

- Host Interface: 1394a (FireWire)
- Twelve balanced ¼" analog inputs with precision 24-bit 128x oversampling analog-to-digital converters
- Twelve balanced ¼" analog outputs with high performance 24-bit 128x oversampling digital-to-analog converters
- Level meters on front panel for analog inputs and outputs, and sync
- On-board 32-bit / 1.6 gigaflop DSP
- 24-bit data resolution maintained throughout entire signal path
- Multiple sample rates from 32kHz to 192kHz
- Variable sample rate feature lets you select any rate between 32kHz – 192kHz
- MIDI In/Out
- Word clock I/O

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