io|14 io|26

Reference Manual

ALESIS

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Introduction

Thank you for purchasing the Alesis IO FireWire audio interface!

You could say Alesis knows a thing or two about recording. Countless artists, engineers, and producers have relied on our digital recorders since the introduction of the original "Blackface" ADAT in 1991. Some publications have even claimed that our ADAT recorders started the "home studio revolution" back in the 1990s. We're proud that our line of affordable tools has made professional-quality recording possible for millions of people around the world.

The IO series FireWire interfaces are the next step in Alesis recording technology. Once you work with the IO|14/26, we're confident you'll appreciate the outstanding sound quality, superior construction, and attention to detail.

For more effective service and product update notices, please register your IO|14 or IO|26 FireWire interface at http://www.alesis.com/.

We're continually delighted by the recordings that have been captured with our products. We hope that your IO|14/26 will be there to inspire and capture your finest performances.

Sincerely,
The People of Alesis

About the IO|14 and IO|26

Our IO|14 and IO|26 audio interfaces are professional-grade tools with everything you need to turn your musical ideas to polished recordings. The two units are virtually identical except that the larger IO|26 has more inputs and outputs than its smaller sibling, the IO|14. The IO|14 and IO|26 feature the following:

- High-speed FireWire (IEEE 1394a) interface for low latency and tons of audio I/O from your computer. The high bandwidth of the FireWire interface allows a single IO|26 to handle 26 inputs and 8 outputs simultaneously (the IO|14 handles 14 inputs and 6 outputs simultaneously).
- Premium 192k analog-to-digital and digital-to-analog converters.
- True 24-bit operation for all digital and analog inputs and outputs.
- High-Definition Microphone Preamplifiers. This new design exhibits superb technical performance and delivers pristine, unclouded sonics. +48v phantom power—required for powering condenser studio microphones—can be applied to any pair of inputs on the unit.
- Switchable guitar inputs for direct recording of guitars and basses.
- Dedicated stereo turntable inputs (IO|26 only).
- Alesis Hardware Direct Monitoring for hassle-free headphone mixes in any recording situation. The included software makes setting up mixes a snap.
- Two headphone outputs optimized for the recording engineer and the artist.
- S/PDIF I/O and ADAT inputs to cover all of your digital connectivity needs..
- Inserts on every analog input for patching additional hardware into your signal path.
- Integrated MIDI I/O on standard 5-pin connectors.
- FireWire bus or AC adapter power.
- Solid construction that's built for many years of heavy use.

Minimum System Requirements

Windows:	Pentium 4 2.4GHz or equivalent (e.g. Centrino) 512Mb RAM Windows XP Service Pack 2 or higher 5400RPM or faster hard drive recommended
Mac:	G4 512Mb RAM OS X 10.4 or higher 5400RPM or faster hard drive recommended

How to Use This Manual

We know this manual will be an integral part of the experience with your IO 14 or IO|26 interface so we've done our best to make it complete, accurate, and helpful for you.

The manual is divided into the following sections describing the various functions and applications of the IO audio interface. While it's a good idea to read through the entire manual once carefully, those having general knowledge about audio interfaces may want to use the table of contents to look up specific topics.

Helpful tips and advice are highlighted in a shaded box like this.

Chapter 1: Hardware Overview describes every section of the IO|14/26's in detail. If you're not sure about the function of a knob, button, connector, or status light, read this section for clarification.

Chapter 2: Installation (Windows) walks you though the installation of the drivers and included software that accompany the IO|14/26. This section covers ASIO, WDM, and MIDI I/O drivers for the PC.

Chapter 3: Installation (Macintosh) discusses installation of the CoreAudio and CoreMIDI drivers for Macintosh computers.

Chapter 4: Getting Started with Cubase LE is designed to help you start recording right away.

Chapter 5: Using the Control Panel shows you how to configure the IO|14/26.

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When something important appears in the manual, an exclamation mark (like the one shown at left) will appear with some explanatory text. This symbol indicates that this information is vital when operating the IO|14 and IO|16 interfaces.

Chapter 6: Hardware Direct Monitoring provides detailed instructions for using the IO1|14/26's built-in digital mixer for low-latency audio monitoring.

Chapter 7: Getting In Deeper: Hardware covers a variety hardware issues such as cabling, using the IO|14/26's insert jacks, and wiring up a home studio.

Chapter 8: Getting In Deeper: Recording discusses various recording methods and techniques and includes a special section for Cakewalk SONAR users.

Chapter 9: Troubleshooting provides various troubleshooting techniques in case of difficulty.

Technical Specifications covers a variety of technical information that technical users will want to know.

And at the end of this manual you'll see a glossary of common terms and a page about the IO|14/26's warranty.

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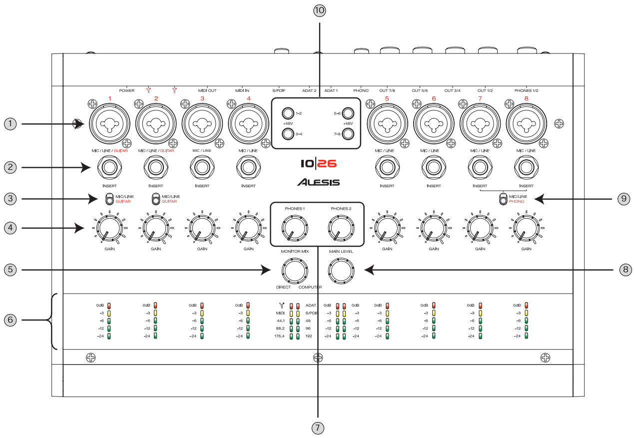
1 Hardware Overview

Section Identification

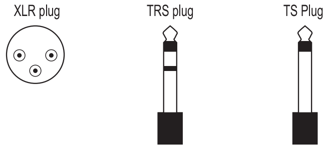
Top Panel

Here's what you'll find on the top of your IO|14/26:



1. **Inputs –** The IO|14/26 features "combo" connectors that allow you to connect either XLR or ¼" cables to the inputs.

   For the IO's inputs, use one of three different cable types:



   Use XLR cables to connect to standard microphones.

   For ¼" connections to balanced gear (like most keyboards and sound modules), use cables with "TRS" plugs. TRS stands for "Tip, Ring, Sleeve."

   For ¼" connections to unbalanced gear (like most electric guitars and basses), use cables with "TS" plugs. TS stands for "Tip, Sleeve." These cables do not have the third wire which TRS cables use to balance the audio signal.

   Choose the right cable for the job. "TRS" cables provide a stronger signal and significantly better noise shielding when used with balanced gear than "TS" cables.

The combo jack inputs
The XLR input "expects" to see a microphone and delivers between 6.8dB and 50dB of gain, depending on the position of the gain knob.

The ¼" input "expects" to see either a line input or an electric guitar or bass, depending on the setting of the Mic/Line/Guitar switch.

In the Line position, the input provides between -15.4dB and 27.8dB of gain. This allows plenty of gain for weak line level sources and also provides the ability to pad down overly powerful line sources.

In the Guitar position, the input provides 6.8dB to 50dB of gain—the same range as the XLR input.

If you're not sure whether an instrument is balanced, your safest bet is to use a balanced ¼" TRS cable.

2. **Inserts** – The IO|14/26 features "insert" jacks on every analog input. These inserts allow you to place additional equipment (like compressors, equalizers, etc.) into your signal path using "insert" cables. Insert cables feature a ¼" TRS connector on one end and two ¼" TS connectors on the other end.. This jack is covered in detail on page 42.

3. **Mic/Line or Guitar Switch (Channels 1-2 only)** – Channels 1 and 2 of your IO|14/26 allow you to switch in a specially designed, high-impedance circuit optimized for recording an electric guitar or bass. If you're recording a microphone or a line-level instrument (keyboard, sampler, DJ mixer, etc.) then set this switch to "Mic/Line." If you're recording a guitar or bass with passive (standard) pickups, set this input to "Guitar."

4. **Gain Knobs** – These knobs let you set the preamplifier gain level. Set the gain with the aid of the meters on the front of the IO14/26. Start with the gain knob turned all the way down (counterclockwise); then slowly turn up the gain until the green LEDs are often illuminated and the yellow LED only illuminates when you play your loudest notes. At this point.

   If the red LED is lighting up (even occasionally), it means the gain is set too high and that you're distorting your signal. Turn the gain knob back down as necessary in order to avoid this distortion.

It's OK to set your gain levels conservatively with the IO|14 and IO|26. These interfaces feature outstanding analog-to-digital converters and preamplifiers, allowing you to capture excellent recordings even if your signals peak at -9dB (or even lower).

Analog and digital distortion are totally unrelated phenomena.

Whereas certain kinds of analog distortion (from guitar amplifiers, stomp boxes, etc.) can sound pleasing, digital distortion sounds awful. If your IO|14/26's meters are going into the red (even every once in a while), it means you're digitally distorting your signal. Turn down the channel gain in these cases.

5. **Monitor Blend Knob** – This knob controls the amount of direct signal (from the analog and digital inputs) that gets blended in to outputs 1/2. This direct signal monitoring bypasses the computer for lag-free listening while tracking.

   The levels and pan positions of each of the direct signals is set using the "Hardware Direct Monitoring" program that ships with the IO|14/26 (see page 35 for more about this program).

   When this knob is turned fully counterclockwise, the Hardware Direct Monitor mix is muted, so you'll only hear the output being returned from your computer's Digital Audio Workstation. When the knob is turned fully clockwise, you'll hear the Hardware Direct Monitoring mix at full volume as well as the signal coming back from your DAW.

6. **Metering Section** – 5-segment "ladder" meters show the precise, digital input signal for each analog channel.

   Status lights indicate Firewire connection to the computer, current sample rate, and ADAT input, S/PDIF input, and MIDI activity.

   Stereo output meters show the levels for outputs 1/2.

7. **Phones 1&2 Volume** – The IO|14/26 has two separate headphone outputs. These knobs let you set the volume for each output.

8. **Main Level** – This knob sets the output level of channels 1/2.

9. **Mic/Line or Phono Switch (IO|26 only)** – Turntables require special "phono" preamps that have higher gain and an "RIAA equalization curve" in order to playback records correctly. If you want to connect a turntable to your IO|26, use the RCA "phono" inputs (on the rear of the unit) and engage this switch.

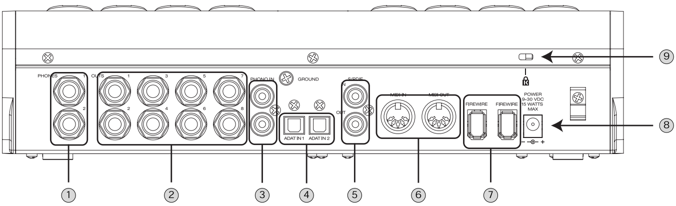
   Note that when you engage this switch, the top panel inputs and gain controls for channels 7-8 are disabled.

10. **Phantom Power** – These switches let you supply +48v "Phantom Power" to condenser microphones that require power. Each button engages/disengages phantom power for a pair of inputs (i.e., channels 1-2, 3-4, 5-6, and 7-8).

Phantom power is only necessary for condenser microphones. Dynamic microphones do not need power to work correctly. Check your microphone's manual to find out if it needs phantom power.

Rear Panel

You'll find the following on the rear of your IO|14/26:



1. **Headphone Outputs** – Connect your headphones to these outputs. The first headphone output always mirrors the analog 1/2 output pair. The second headphone output is assignable in software.

2. **Main outputs (8 outputs on IO|26; 2 outputs on IO|14)** – Use ¼," "TRS" cables to connect these outputs to the balanced inputs of your powered speakers or power amplifier. If your speakers or amplifier only provide unbalanced inputs, use unbalanced ("TS") cables.

   The IO|26 (shown above) has 8 outputs. The IO|14 has 2 outputs.

   If you are using the IO|26 and wish to connect additional speakers, headphone amplifiers or hardware processors, connect them to outputs 3 through 8.

3. **Phono input (IO|26 only)** – Connect your turntable to this input.

   If your turntable has a grounding cable, attach it to the grounding screw to the upper right of the phono inputs. This grounding will minimize humming and buzzing.

4. **ADAT Lightpipe Inputs** – These optical digital inputs can accommodate a wide variety of ADAT-enabled gear. The IO|14 has one ADAT input whereas the IO|26 (shown above) has two ADAT inputs.

   Use ADAT-compatible optical cables to connect to these inputs.

5. **S/PDIF Connectors** – Connect S/PDIF-enabled digital devices (such as the Alesis Masterlink, CD players, DAT machines, MiniDisc Recorders, etc.) to your IO|14/26 using coaxial, RCA-terminated cables.

6. **MIDI Connectors** – Connect your keyboards, sound modules, or other MIDI devices to your IO|14/26 using 5-pin MIDI cables. Remember to chain the OUTs of each device to the INs of other devices.

The IO14/26's outputs are "impedance balanced." This wiring method provides all of the benefits of "fully balanced" wiring when the outputs are connected to balanced gear. Furthermore, impedance balancing, unlike other balancing methods, allows for trouble-free connection to unbalanced devices.

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If you connect an ADAT or S/PDIF input device (or both), you will need to select one of them as the "clock master" in the IO's control panel.

The device you select as the clock master will determine the IO's clock rate and will be responsible for keeping all of the digital signals synchronized. Therefore, if you turn this device off, you will need to select another clok master in the control panel.

7. **FireWire Connectors** – Connect one of these plugs to your computer's FireWire port. You can use the other jack to connect additional FireWire devices (such as hard drives) to your computer. Up to 127 devices can be "daisy-chained" on one FireWire bus.

8. **Power Connector** – Use the supplied AC adapter if your computer does not provide sufficient Firewire bus power to power the IO|14/26 or if you want to preserve battery power.

   Note that many notebook and small form factor computers, like Mac Mini computers, do not provide sufficient power for bus power. For these computers, the external adapter must be used.

9. **Kensington Security Slot** – This locking mechanism allows you to secure your IO|14/26 to a desk or some other heavy object using an optional third-party locking device.

If your Firewire cable comes with one ferrite (a bulge in the cable near the connector), connect the end with the ferrite to the IO|14/26, and connect the end without the ferrite to your computer.

If an AC adapter is attached, the IO|14/26 will use it. FireWire bus power is only used if no AC power is available.

Computers with noisy internal power supplies can send periodic, audible pulsing through the outputs of your IO|14/26. If you hear a low-level pulsing, plug in the AC adapter to bypass the computer's power supply.

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2 Installation (Windows)

Important: Download and install the drivers from http://www.alesis.com--or insert the software CD into your computer's CD drive—BEFORE you plug your IO|14/26 into your computer for the first time.

Install the software/drivers first

Important: Follow these steps BEFORE you plug your IO14/26 into your computer for the first time.

If you have access to the Internet, check http://www.alesis.com for the very latest software updates. The updates posted there are guaranteed to be the most current, best software versions available.

Begin by running the Alesis installer(s). These programs (there may be one or more than one) will install three very important components onto your computer:

- *Drivers.* These are the system components that allow Microsoft Windows to identify and interact with your IO|14/26. You do not interact with the drivers directly, but they must be installed on your computer for the IO|14/26 to work.

- *Control Panel.* The control panel allows you to set sample rates, clock sources, buffer sizes, and other settings.

- *Hardware Direct Monitoring Panel.* This application lets you route the IO|14/26's inputs directly to its outputs for a minimum of latency (delay) when recording.

  The Hardware Direct Monitoring Panel also allows you to change Headphone2 and S/PDIF output assignments.

Windows XP, Service Pack 2 or later is required.

The Windows drivers include the two most popular standards for audio interfacing—WDM (the "Windows Driver Model" built by Microsoft) and ASIO (the "Audio Stream Input/Output" standard used by many audio software applications).

For each of the several installations that occur, click "Continue Anyway" if Windows warns you that the drivers have not passed Microsoft Logo Certification.

Windows Logo Certification warning.
Ignore this warning by clicking "Continue Anyway."

Install Cubase LE (optional)

If you are interested in using Cubase LE as your multitrack Digital Audio Workstation, install it from the CD now.

Connect your IO|14/26

Now, connect your IO|14/26 to your computer using a Firewire cable. Watch for one of the lights on the unit to turn on within a few seconds. If a light does not turn on—or if you are using a notebook computer with a small four-pin Firewire connector—plug in the external AC adapter.

Windows will recognize the IO|14/26 and start the Found New Hardware wizard. The installation process will automatically install these drivers one by one.

When you are prompted whether to install the drivers automatically or search for a specific location, choose to install them automatically.

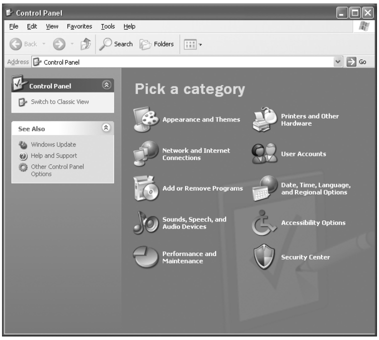
If you are asked whether you want to connect to the Internet to check for the latest driver, choose not to.

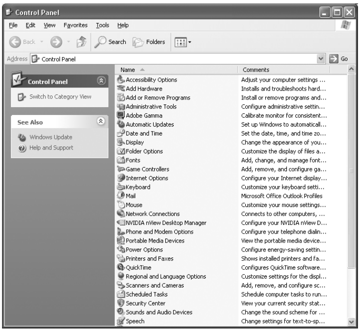
Let the installer continue installing the various sets of drivers until you see a message stating, "Your new hardware is installed and ready to use." Do not cancel any of the installations, as they are all required for proper operation.

Make IO|14/26 the default audio device

To use your IO|14/26 interface as your default Windows sound device, follow these steps:

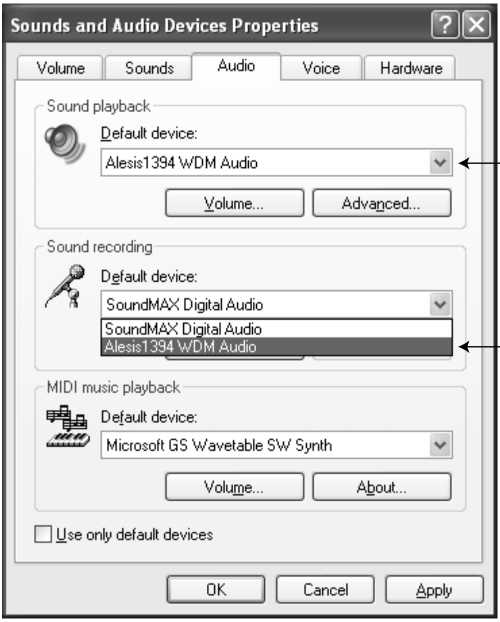
1. From the Windows Start menu, choose "Control Panel." Depending on your Windows preferences, it will appear similar to one of the two pictures below:



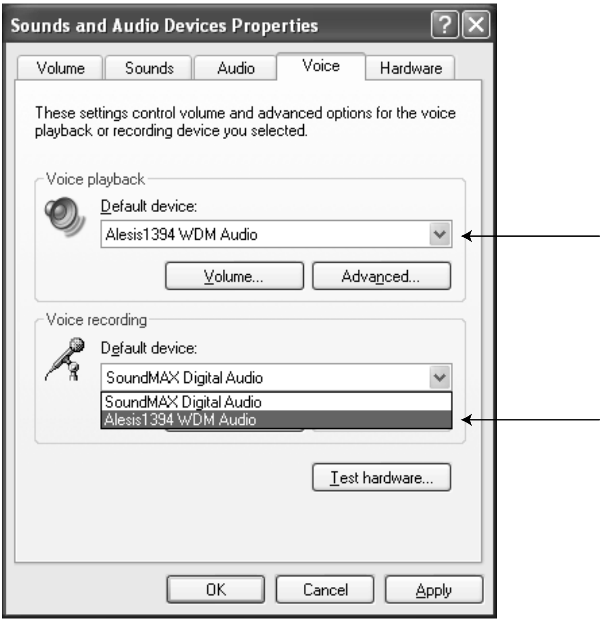


2. Choose "Sounds and Audio Devices".

3. Click the "Audio" tab. Change the default devices for both sound playback and sound recording to your IO interface.



4. Click the "Voice" tab. Change the voice playback and voice recording settings to the IO|14/26.



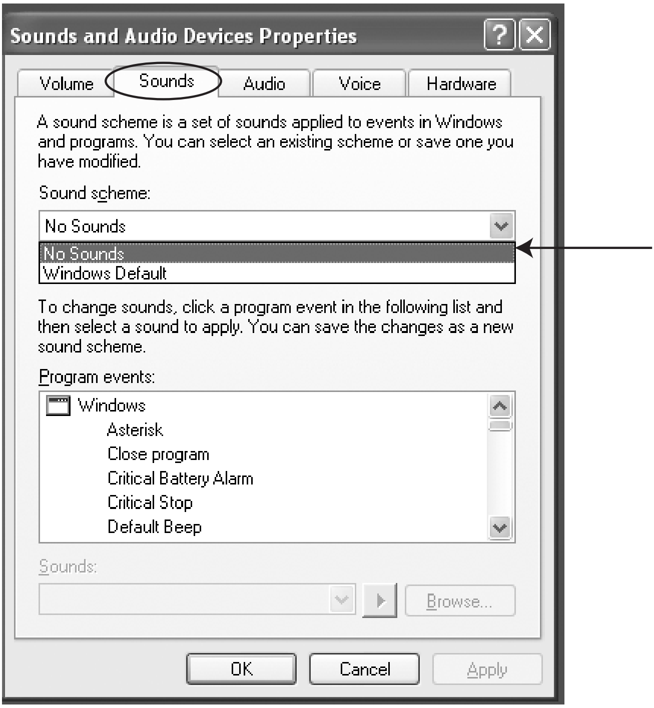
5. Click "Apply" to apply these changes.

Disable Windows System Sounds

Windows System Sounds—the sounds that Windows plays to signal starting up, shutting down, alerts and so forth—can interfere with your audio recording. We **strongly suggest** that you disable these sounds.

1. Click the "Sounds" tab of "Sounds and Audio Devices."

2. Under "Sound Scheme," choose "No sounds."



3. Click "OK" to accept this entry and close the dialog box.

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3 Installation (Macintosh)

Important: Download the drivers from http://www.alesis.com onto a folder on your desktop or insert the software CD into your computer's CD drive BEFORE you plug your IO|14/26 into your computer for the first time.

Install the software/drivers first

Important: Follow these steps BEFORE you plug your IO14/26 into your computer for the first time.

Begin by running the Alesis installer(s). These programs (there may be one or more than one installer) will install three very important components onto your computer:

- *Drivers.* The CoreAudio and CoreMIDI drivers allow your Mac to identify and interact with your IO|14/26.
- *Control Panel.* Access the Control Panel from Audio/MIDI Setup's "Configure Device" button. The control panel allows you to set sample rates, clock sources, buffer sizes, and other settings.
- *Hardware Direct Monitoring Panel.* This application lets you route the IO|14/26's inputs directly to its outputs for a minimum of latency (delay) when recording.
  The Hardware Direct Monitoring Panel also allows you to change Headphone2 and S/PDIF output assignments.

If you have access to the Internet, check http://www.alesis.com for the very latest software updates. The updates posted there are guaranteed to be the most current, best software versions available.

Mac OS X 10.4 or later is required.

Install Cubase LE (optional)

If you are interested in using Cubase LE as your multitrack Digital Audio Workstation, install it from the CD now.

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Your CoreAudio application may describe the IO|14/26 channels simply as "1, 2, 3, 4, etc." Note that, if you are using an IO|14, the S/PDIF channels will not appear at the end of the channel list but rather at an earlier position.

Connect your IO|14/26

Now, connect your IO|14/26 to your computer using a Firewire cable. Watch for one of the lights on the unit to turn on within a few seconds. If a light does not turn on, plug in the external AC adapter.

Important: If you experience problems when using a 4-pin Firewire cable (small connector) to connect the interface to your laptop computer, we recommend that you either switch to higher quality cables or install a six-pin PC card adapter into the notebook.

Make IO|14/26 the active audio device

Open Audio/MIDI Setup and choose your IO|14/26 for both your inputs and outputs.

4 Getting Started with Cubase LE

Introducing Cubase LE

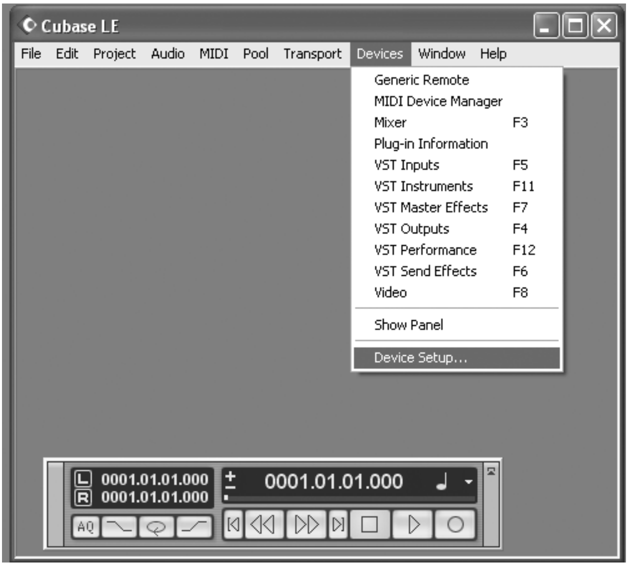
Your IO|14/26 ships with Cubase LE, a powerful audio and MIDI Digital Audio Workstation.

The following instructions are designed to get you set up and recording audio with Cubase LE quickly. For more information on using this software, consult the documentation available in Cubase's Help menu.

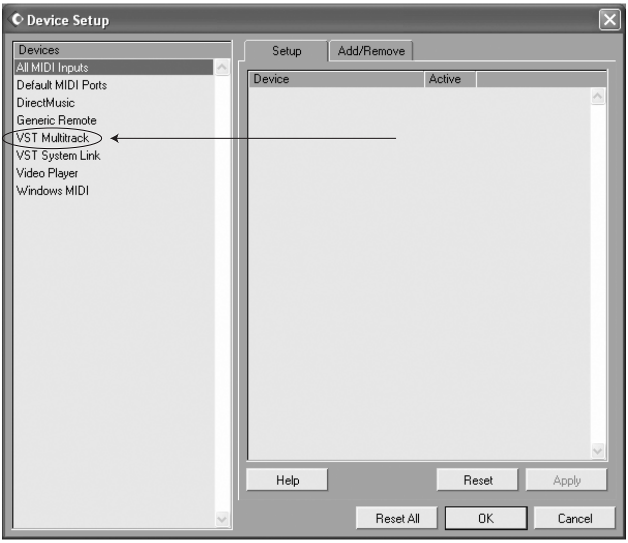
Windows only: selecting the IO|14/26 as your audio and MIDI device.

For Mac users, the audio device used by Cubase is the same one that you select in Audio/MIDI setup.

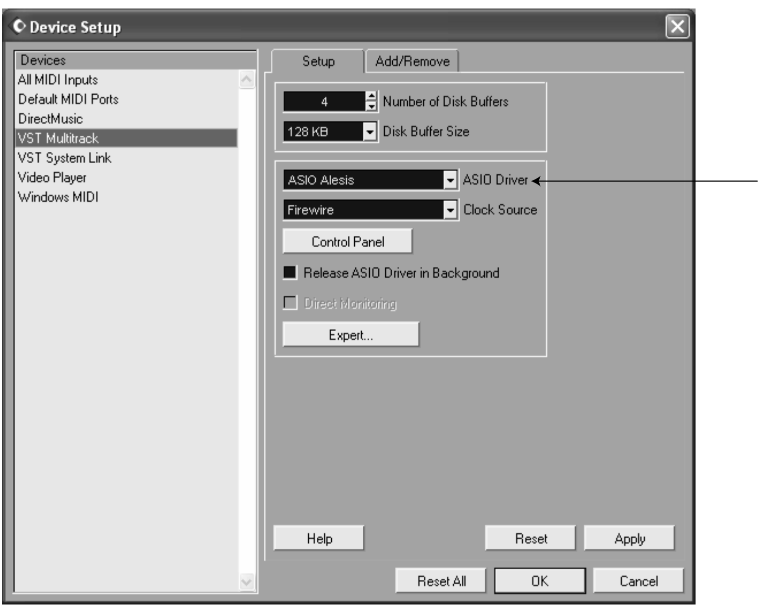
1. Choose the menu option "Devices" | "Device Setup…."



2. Click the "VST Multitrack" option.



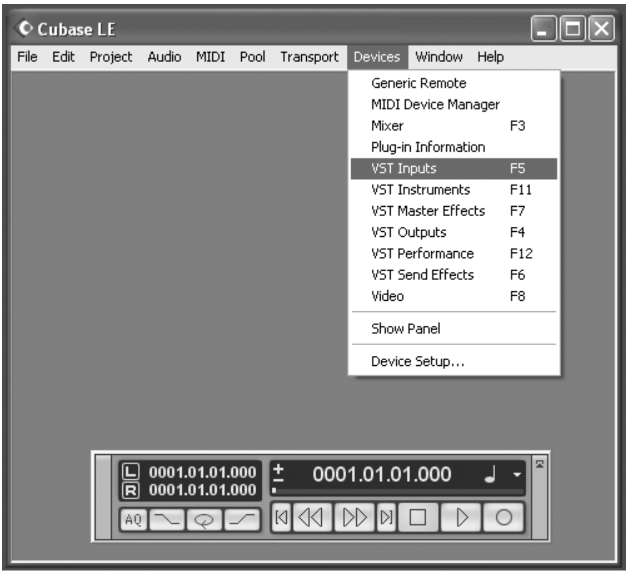
3. From within the ASIO Driver drop-down box, choose the IO|14/26. Press Apply to accept the change.



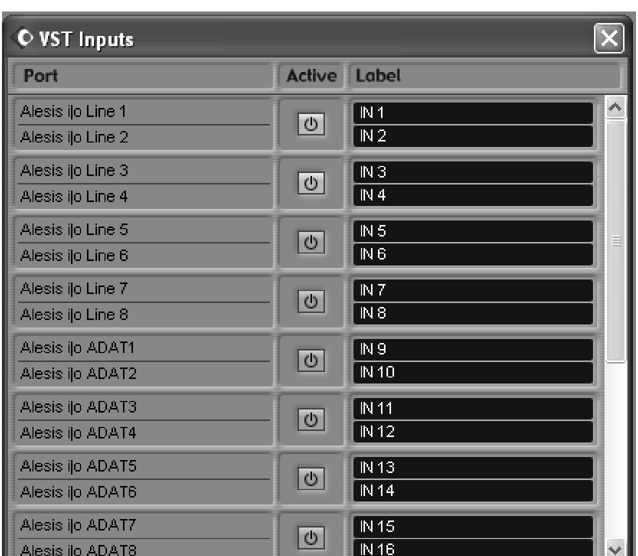
4. To enable MIDI, click on the “Default MIDI Ports” option on the left-hand side and select IO|14/26’s MIDI ports for input and output. Again, click “Apply” to accept the changes. Then press “OK” to exit this screen.

5. Now that the IO|14/26 is selected as the audio device, individual channels must be activated for use. Again, return to the “Devices” menu, and select “VST inputs.”

Different versions of Cubase function similarly, but not exactly, to the examples shown here.

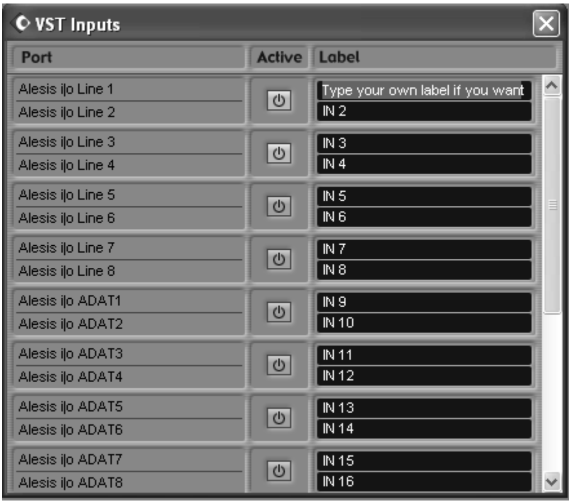


6. Click the activation buttons for channel pairs that you want to use.



Some versions of Cubase, including Cubase LE, do not allow all of the physical inputs to be used simultaneously.

7. In Cubase, you can rename channels by clicking in the "Label" area. This is useful if, for instance, your lead vocals are always on channel 1, your bass drum is always on channel 2, etc.

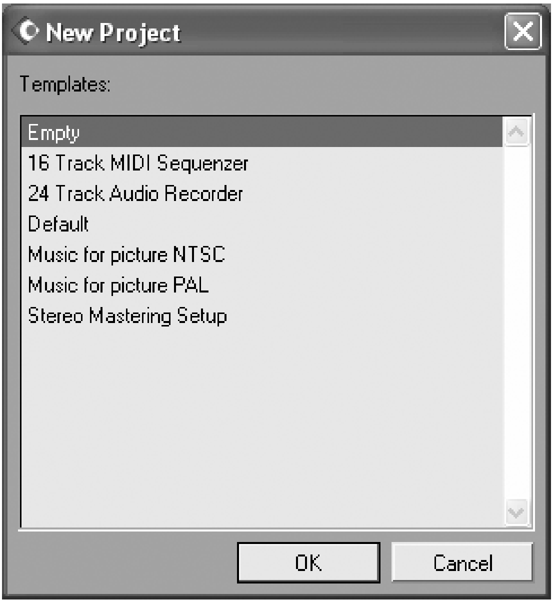


Creating a new file

Now, you're ready to create an audio project.

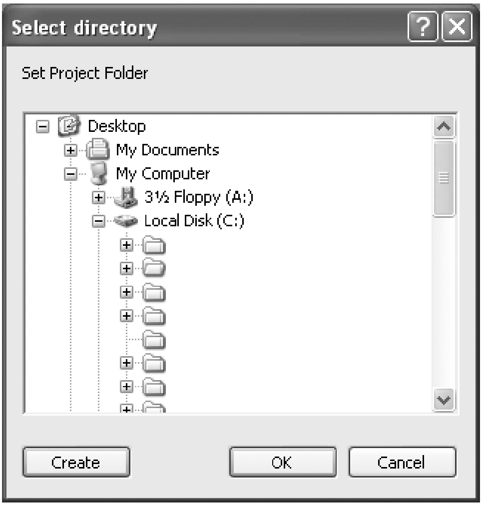
1. Choose "File" | "New Project…"

2. You can begin with a template or an empty file. For now, begin with an empty file.

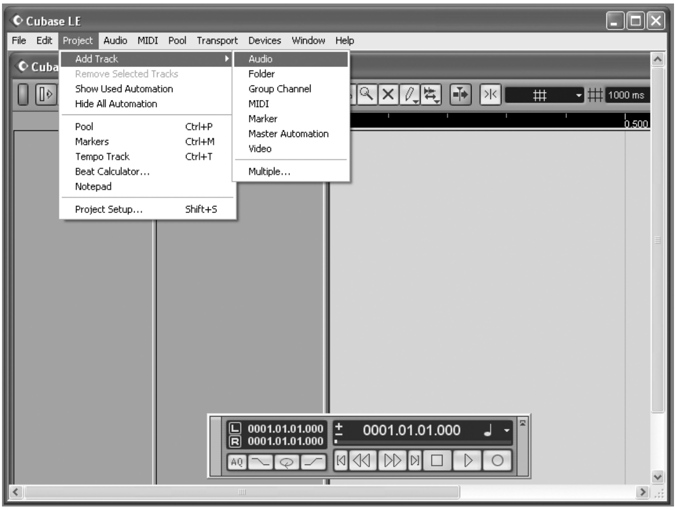


3. Cubase needs to know where to place audio. Choose a directory here.

An excellent scheme for storing your projects is to create a directory called "audio projects." Then, within that folder, create a new folder for each song you work on. Cubase will store your song file and all associated audio files in that same folder.



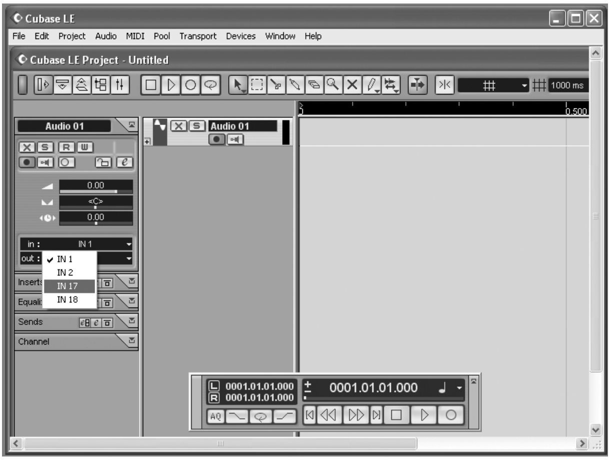
4. Now, you have a blank project. Add an audio track for recording by choosing "Project" | "Add Track" | "Audio."



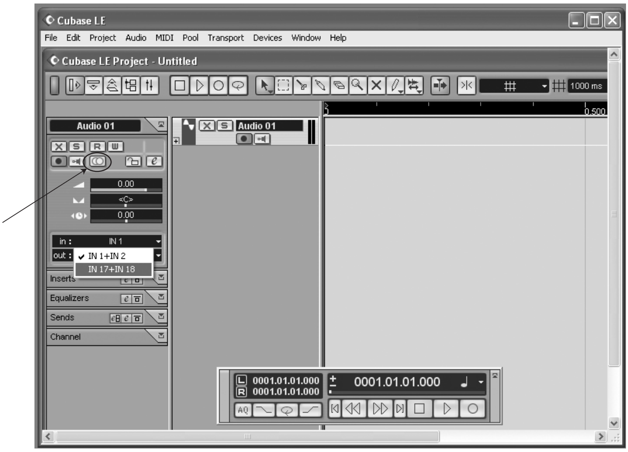
5. Be sure that the "inspector"—a strip on the left-hand side of Cubase that shows all sorts of information about the selected track—is active.

   If your view is similar to that shown below, the Inspector is active. If you do not see all of the information on the left-hand side, the Inspector is not active. To activate it, press the "show Inspector" button towards the upper left of the screen (just below the "Edit" menu in the following picture).

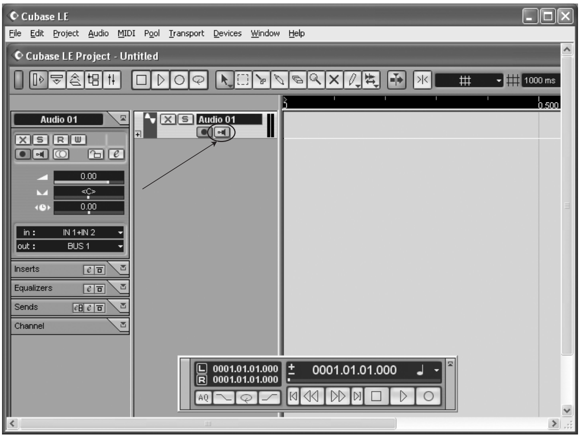
Later, you can hide the Inspector if you want to save space on your screen.



6. Choose an input for your track by selecting it from the "in" area on the left. To record stereo on the track, click the button highlighted below.



7. If you want to monitor your audio with Cubase's effects (distortion, reverb, etc.), press the direct monitoring button next to the Record Enable button.



Using Cubase's direct monitoring requires the audio to make a round-trip through the computer, which causes a small but noticeable delay as the digital audio is processed. To avoid any echo effects, open the Alesis Hardware Direct Monitoring panel and mute the corresponding input. This way, you will only hear the signal with effects without also hearing the pure signal output from the IO.

8. Add additional audio tracks as needed. Record-arm each one and press the RECORD button to begin recording.

For additional information, consult Cubase's documentation.

5 Using the Control Panel

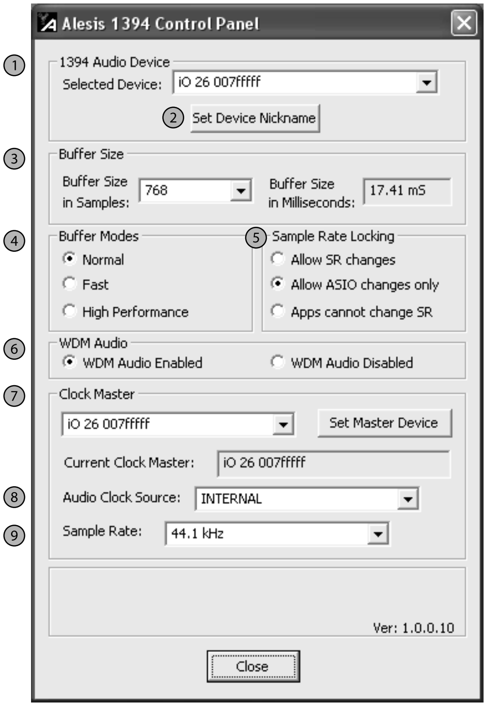
Working with the IO14/26 control panel

Accessing the control panel

In Windows, open the control panel from the shortcut on your desktop, the Programs menu, or from within your audio application.

On the Mac, access this panel from the “configure device” button of Audio/MIDI Setup.

To open the control panel from within Cubase in Windows, select “Devices” | “Device Setup....” | “VST MultiTrack.” Then, click the “Control Panel” button.



1. Select the Alesis IO.

Be sure that your IO14/26 is shown here. If it is not shown, then your computer does not “see” it. Check the IO’s connections to your computer.

2. Create a nickname for your IO|14/26 (optional)

You can change the name of the IO|14/26 as it’s seen by the recording program.

Once this is done, choose “Reset All” in Cubase (or the equivalent command in other programs) to update the display of the device and channel names.

Creating a nickname for your device is entirely optional.

3. Adjust latency by changing the buffer size

"Latency" refers to the amount of time it takes for audio to get into and out of the computer. In the best of all possible worlds, there would be no such thing as latency—we would hear audio the moment it was created. However, computers have limited processing power, and they can "choke"—cutting off recording or crashing programs—if they are asked to handle too much data all at once.

To minimize this risk, audio can be stored in a buffer for a certain amount of time. This buffering helps smooth out the stream of data that the computer needs to handle. In the end, all of the audio is sorted out and played correctly, but with a delay.

To counteract the delays you hear when you are monitoring incoming audio through the computer and your latency settings are high, turn off your DAW's input monitoring feature. Use the included Alesis Hardware Direct Monitoring application instead.

Here are the basic considerations to consider when adjusting buffer sizes:

Lower buffer size = less latency but higher risk of audio problems

Higher buffer size = more latency but lower risk of audio problems

Very high buffer size = possible system instability

For most systems, there is a "sweet spot" where latency is not too high and system performance is stable. Experiment with raising or lowering buffer sizes to hit this sweet spot.

Many people use two latency settings—a lower one when recording tracks and a higher one when mixing.

As you begin adding plug-in EQ, compression, and so forth to your project, your computer will need to work harder. Consider increasing your buffer size at this time.

4. Choose a Buffer Mode

Three different modes allow you to customize the way that your computer organizes and uses buffers.

- If you use your computer for all sorts of applications besides music (Word Processing, email, etc.), choose "Norma"l Buffer Mode.

- If your computer is relatively new and is used almost exclusively for music applications and has very few other processes running, choose "Fast" mode.

- Choose "High Performance" mode if you have an extremely well specified computer, use few background tasks, and are not shuttling very large amounts of data through the Firewire bus.

Experiment with each of these settings in your first few weeks using the IO14/26. Depending upon your computer configuration and the types of audio projects you do, you may find that one setting clearly outperforms the others.

5. Specify how sample rates can change

Since Windows (and various Windows applications) have a nasty tendency to try and take control over your audio sample rate—often without notice—this section allows you to ignore those sample change events.

- If you do not mind your sample rate changing freely, choose "Allow SR Changes."
- To allow only ASIO applications (like Cubase) to change the sample rate, select "Allow ASIO changes only."
- You can lock the sample rate—such that it can only be changed using this control panel—by selecting "Apps cannot change SR." This is the safest of all these options.

Whichever setting you choose, the sample rate can always be changed from within this control panel. Note, however, that if you have an audio application open, you can cause conflicts with it if you try to set the sample rate differently here compared to the setting in the open audio application.

6. Enable or Disable WDM audio (Windows only)

If you are using ASIO applications exclusively and do not need access to Windows system sounds, Windows Media Player or other media players, consider unchecking this box. Doing so helps ensure that unwanted audio from other applications does not intrude on your intended audio output.

7. Set the clock master

If you are using multiple Firewire audio devices at once, chain one to the next, and designate the one at the beginning of the chain as the clock master.

8 Specify the audio clock source

In digital recording systems, all digital input devices must be synchronized to the same clock. For IO|14/26, this means that you need to take care to set the clock source properly when you use either the S/PDIF or the ADAT inputs.

1. INTERNAL. Use this setting if:
   a. You **are not** connecting any external S/PDIF or ADAT input devices;

   b. You **are** connecting external S/PDIF or ADAT input devices, but you are running a cable from the IO|14/26's S/PDIF output into one of these devices. This is a rare case.

If you're using the IO|14/26 by itself, you don't need to worry about clocking. Just make sure that the "Clock Source" parameter in the IO|14/26's control panel is set to "Internal."

If you're connecting your IO|14/26 to other equipment through the ADAT or S/PDIF connections, keep in mind the following rule of digital audio:

<u>When connecting devices digitally, all of your devices must be locked to ONE clock.</u>

2. S/PDIF or ADAT. Use one of these settings if you have attached either a S/PDIF or ADAT device.

   If you have attached more than one of these devices, synchronize their clocks using dedicated cables; then choose one of the as the clock master.

   Note that ADAT2 on the IO|26 cannot be the clock master. If you are only using one ADAT input on the IO|26, use the first input. If you are using both ADAT1 and ADAT2 with two different hardware devices, be sure to clock one device to the other.

3. FIREWIRE. Use this setting if you have another Firewire audio devices connected to the IO14/26 and you want that device's clock to drive the IO14/26's clock. No additional cables are necessary—the IO14/26 will read the clock signal coming from the other device over the Firewire cable.

Your IO|14/26 includes a state-of-the-art clock recovery circuit that generally eliminates clocking-induced glitch artifacts even when clock sources are incorrectly assigned. Nevertheless, do take care to assign the clock source correctly. Otherwise, sample accuracy is likely to suffer.

When you use two or three digital input sources, lock the devices together. Most commonly, you will use the devices' BNC Word Clock connectors for this purpose.

Start with one device and cable its Word Clock Output to the Word Clock Input of the second device. If you have a third device, connect the second device's Word Clock Output to the third device's Word Clock Input.

For each device, if there is a "master/slave" setting, be sure to set it appropriately. (Only the first device in the chain should be the "master.")

There are other ways to lock your clocks together, such as using a dedicated clock distribution hardware device. However, the method outlined here is perfectly fine, as well as being the most simple and economical.

9. Set the sample rate

Set the sample rate here. Note that this setting must be made even if you are using an external device (ADAT, S/PDIF or Firewire) as your clock master.

Some audio programs require that you change the sample rate under their Project Setup or similar menus as well. For instance, in Cubase, be sure that the sample rate selected here matches that under the "Project" | "Project Setup…" menu.

To ensure S/PDIF lock, set the clock source and sample rate in the Alesis control panel <u>before</u> connecting, powering on, or changing the sample rate of your S/PDIF device.

The Alesis Hardware Direct Monitoring and Routing Application

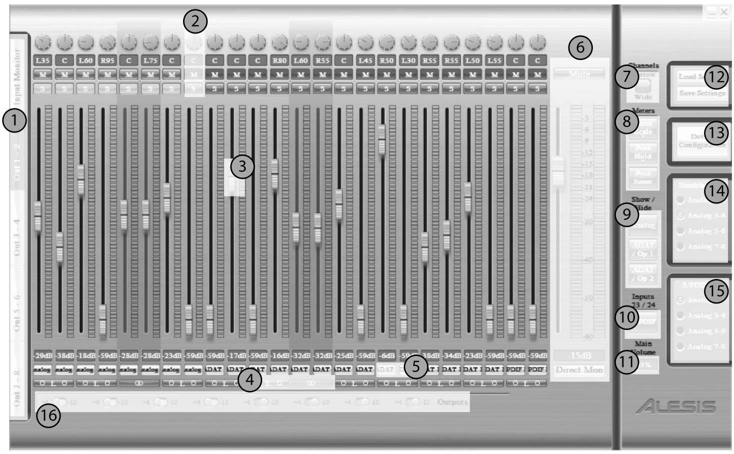
A powerful digital mixer is built into your io. This digital mixer allows you to route your analog and digital inputs directly to the outputs, completely bypassing the computer. This hardware-based routing allows musicians to monitor their performances with no perceptible delay.

There may be times when you find this functionality to be unnecessary. For instance, when you are recording simpler projects that put less strain on your computer, you can generally choose lower buffer settings in the Alesis IO control panel. This way, you can use the input monitoring feature of your DAW recorder, and the latency (delay) experienced by your performers will be very slight.

However, when you add more tracks and plug-ins, you will need to increase your buffer sizes in order for your system to keep operating smoothly. At this point, it makes sense to mute the Input Monitoring on your DAW and use the IO14/26's digital mixer.

Using the Hardware Direct Monitoring (HDM) panel

Once you have connected all your recording gear and established a computer connection (as shown by the light on the io), open the Alesis Hardware Direct Monitoring and Routing panel by clicking on the shortcut on your desktop.



1. Output Monitor Tabs

Use these tabs to select the output pairs to which you will blend in the incoming signal. Each tab's HDM mix is completely separate from the others.

Setting up direct monitor mixes for single musicians and for multiple musicians
Most commonly, you'll have a stereo set of speakers attached to outputs 1/2. This is the output pair to use for your Digital Audio Workstation's stereo outputs.

If you are recording a single musician, connect your inputs, and then select the tab for outputs 1/2. Now, you can easily blend in the musician's direct audio with the music coming back from the Digital Audio Workstation.

When you are recording multiple musicians—each of whom desires a different monitor mix—use these tabs to create a separate "monitor mix" for each output pair.

Experiment with creating multiple mixes—doing so puts no strain on your computer.

Re-naming the tabs
You can re-name each of these tabs by double-clicking on them and entering your new name. (Names like "Singer" and "Drummer" can be a lot more helpful than "Out 3-4.") Note that the names you type will not alter the physical routing taking place in the IO14/26.

At 176.4k and 192k sample rates, hardware direct monitor mixes can only be heard on outputs 1-4. Audio will play back from the computer as normal on all the channels, but the analog and digital inputs cannot be monitored on outputs 5-8.

2. Pan/Mute/Solo

Set the left/right pan position of the audio here. Also, you can solo and mute individual channels.

Using the *Shift* key, click the pan knob to return it to center. Also, hold down *Shift* while clicking to mute or solo (or un-mute or un-solo) all of the inputs at one time.

If you hear the input signal twice—with a short delay between each instance—then you are most likely monitoring through both the HDM panel and also your DAW software. Either turn off input monitoring on your DAW or mute the corresponding input on the HDM panel.

3. Volume

Set the volume using this slider.

4. Stereo Channel Linking and Unlinking

Use this button to link or unlink the controls for stereo channel pairs.

5. Channel Names

Just like the output tabs, the incoming audio channels can be re-named. (Words like "vox" and "snare" probably have more value than "Analog1" and "ADAT12.") The names you provide carry over to all of the tabs.

6. Master HDM Mix Fader

This is the master fader for the Direct Monitor mix for this output pair. Bringing this fader up or down affects all the channels being monitored using HDM but does not affect the audio coming back from the computer.

Once you have set your balances for multiple instruments using the individual slider, this control allows you to bring your whole mix into balance with the audio from your Digital Audio Workstation.

For outputs 1 and 2, the MIX BLEND knob on the front panel of the hardware can also be used to control this slider.

7. Narrow/Wide View Switch

You can switch between "Narrow" and "Wide" views of the HDM panel. Note that the Input Monitor tab is always in Narrow mode.

8. Metering Options

Choose from a number of different metering options here.

The metering options do not change how audio is actually recorded. They only change how the audio is metered on screen.

Meter Scale
Under "Meter Scale," choose "High," "Medium," or "Safe." These options change the thresholds at which the meters change to yellow and then to red. *WHATEVER SETTING YOU CHOOSE, THE ACTUAL AUDIO BEING RECORDED IS UNTOUCHED.* Rather, these modes can help keep you from recording too "hot," resulting in digital "overs."

Peak Hold time
Set how long the highest peak of the incoming audio should be displayed.

For highly unpredictable audio (live concerts, maniac drummers, etc.), consider using "Safe" metering and trying to keep your audio in the yellow area of the meter scale. Then, if an unpredictable audio event occurs, your audio is more likely to be recorded cleanly.

For predictable dynamic sources, like distorted electric guitar or sources where you've inserted a hardware limiter in the recording path, don't be afraid to push your meters up into the red zone. (Red in this panel means "caution," not "uh-oh!")

Peak Reset
Click this button at any time to reset the peak levels of the meters.

9. Bank Hide/Show buttons

In "Wide" mode, not all channels are visible at once. If you are not using some of your channels, choose to hide them.

Note that these buttons are grayed out in "Narrow" mode.

10. ADAT-S/PDIF Toggle

The HDM mixer can mix 24 inputs. At 44.1kHz and 48kHz, the io|26 offers 26 inputs, 2 more than the total. Therefore, at these sample rates, choose to monitor either ADAT 15/16 or S/PDIF.

Your choice here will not affect the number of channels seen by your DAW. Your DAW will always see all IO|14/26 channels. This choice only affects the audio that you monitor through this HDM application.

If you still need to monitor those last two channels, you can always do so through your Digital Audio Workstation software, though some increased latency will result.

11. Main Level

The position of the MAIN LEVEL(outs 1/2) knob is indicated here for convenience (Windows only).

Numbers 12 through 15: Load/Save and Audio Routing Settings

12. Save/Recall Settings

Your HDM settings will only be recalled if you save them. Save them as "default.hdm" and they will be recalled every time that the panel is opened. Save setups that you want to recall for specific sessions under some other name.

13. Device Configuration (Control Panel shortcut)

Click "Device Configuration" to go immediately to the IO14/26 Control Panel, where you can set the sample rate, clock master, device nickname, etc.

14. Headphones 2 Assignment

The first set of headphones always follows the Out 1/2 mix. The second pair of headphones is freely assignable to follow any of the analog output pairs.

15. S/PDIF output assignment

The S/PDIF digital output, like the second headphones output, can be assigned to follow any output pair.

For the IO|26, these output pairs all correspond to physical, line-level outs. They all exist in hardware. While the IO|14 only has one stereo pair of line-level outputs, the IO|14 reports these additional output pairs to your Digital Audio Workstation. These additional pairs allow you to make headphone 2 and S/PDIF assignments to more than just the built-in stereo output.

This "virtual" routing providesIO|14 owners with much of the power of its big brother, the IO|26.

The S/PDIF output mirrors the volume of the analog 1/2 outputs.

Changes made to the main volume encoder on the front of the unit, and changes made to the +4/-10 output level switches on the Hardware Direct Monitoring application, will change the level of the S/PDIF output.

7 Getting In Deeper: Hardware

Cabling 101

Cables are a crucial (and often overlooked) part of a studio. Many beginners run into problems because they use inappropriate or poor quality cabling to connect their gear and their recordings suffer as a result. Don't let this happen to you! Use the following guidelines to maximize your sound quality:

1. **Use balanced cabling wherever possible** – The IO|14 and IO|26 are fully balanced recording devices that will give you the best sound quality when using "balanced" cabling wherever possible. Technically speaking, balanced cables carry your signal over three conductors (known as "hot" "cold" and "ground) as opposed to "unbalanced" cables which only have two conductors (known as "hot" and "ground"). Because of this design difference, balanced cables pick up much less radio frequency (RF) and electromagnetic (EM) noise. Musically speaking, this means you'll have much less unwanted humming, hissing, and buzzing noises in your recordings.

2. **Minimize Cable Length** – As your cable length increases, so does your signal's susceptibility to unwanted noise. Try to minimize your cable runs as much as possible to preserve sound quality. Don't sweat over minor differences in length (i.e., using 20' of cable when you only need 15'), but definitely don't use a 100 ft. cable if all you need is 10 ft.!

3. **Use High Quality Cables** – Not all cables are the same! Try to use well constructed, high-quality cables whenever possible. Two cables may look the same on the outside, but a high quality cable will have better shielding and soldering on its connectors. This means a good cable will perform better and last much longer than a cheap one.

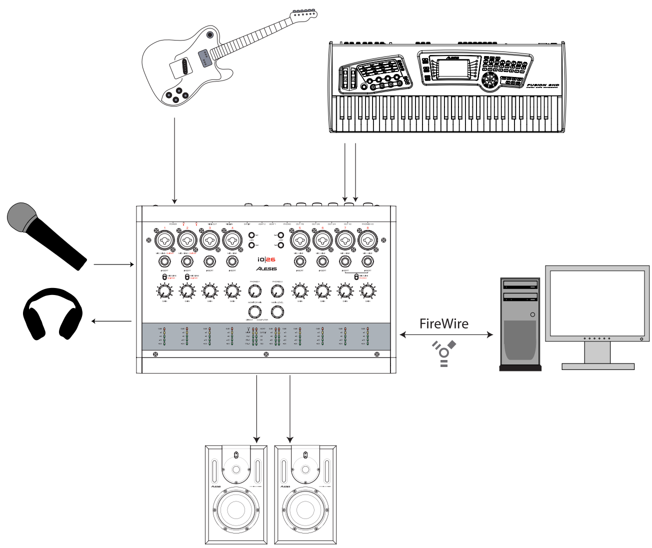
4. **Keep 'em separated!** – Try to keep your audio cables and power cables separate from each other. Power cables tend to emit lots of electromagnetic noise and if you bunch all of your cables together to make them look tidy, you're increasing the noise that is picked up by your audio cables. Group your cables in two bunches if possible (i.e., an "audio" group and an "other" group) and keep these groups separate. Even a few inches between the two groups will substantially cut down on noise. If your power and audio cables need to cross paths, have them do so at 90 degree angles to minimize contact.

Common Hookup Scenarios

Singer / Songwriter

The following setup will be commonly used by singer/songwriters working in their home studios. It allows for the artist to hook up guitars, keyboards, and microphones to the IO|14/26 and to monitor him/herself on headphones. The guitar, keyboard, or microphone shown below can easily be swapped with other equipment (i.e., samplers, CD players, more microphones, etc.) to match the artist's needs. All of the arrows indicate analog ¼" or XLR cabling unless otherwise noted.

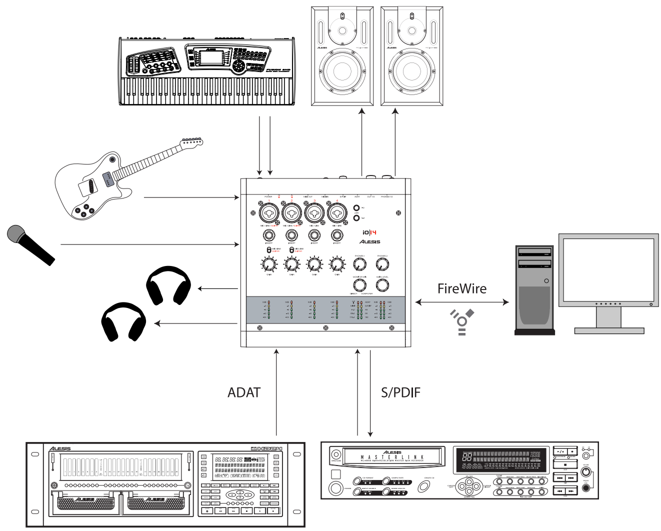
Note that the computer can be replaced with a laptop making the recording rig completely portable.



Full Studio Setup

The following setup makes full use of the IO|14's audio inputs and outputs. All of the arrows indicate analog ¼" or XLR cabling unless otherwise noted.

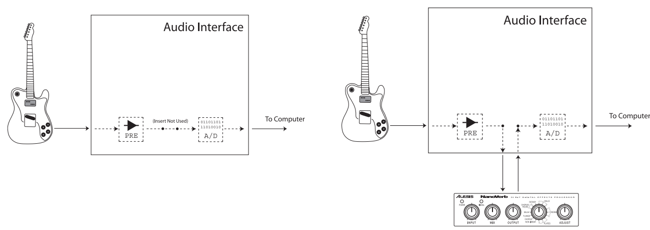
An IO|26 would allow you to connect four additional analog inputs as well as 8 additional digital inputs via a second ADAT port.



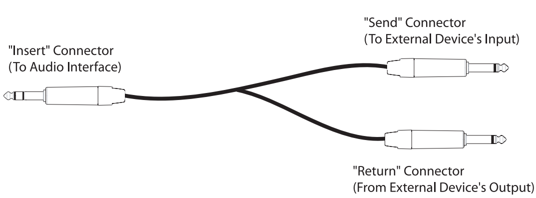
Using the Insert Jacks

Sometimes, you may want to add additional gear into your signal path before your analog-to-digital converter digitizes your signal and sends it to the computer. For example, many bass players like to compress their instrument with an analog compressor before recording into the computer. Inserts help you do this by letting you tap into your signal *after* the IO|14/26's preamplifier but *before* the A/D converter. The two diagrams below demonstrate where the inserts fit into your signal path.

In the diagram to the left, no insert is used and your signal passes straight from the preamp to the A/D converter. In the diagram on the right, an effects processor is inserted into the signal path just after the preamp (before the A/D converter).



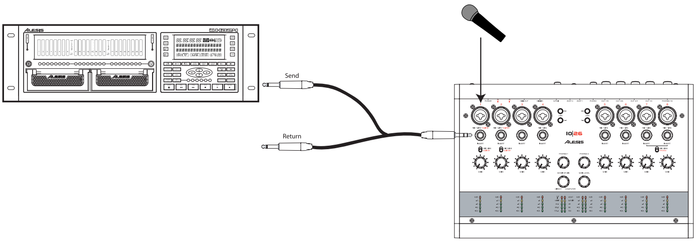
Note that you need a special "insert cable" to utilize this connection. These cables have a ¼" TRS connector on one end and two ¼" TS connectors on the other end. The TRS side goes into the "insert" jack on the IO|14/26. The two TS connectors are usually labeled "send" and "return." The "send" plug goes to your external device's input whereas the "return" plug connects to the device's output.



Additional Uses of Insert Jacks

Inserts can be used in two additional ways that you may find useful. They are as follows:

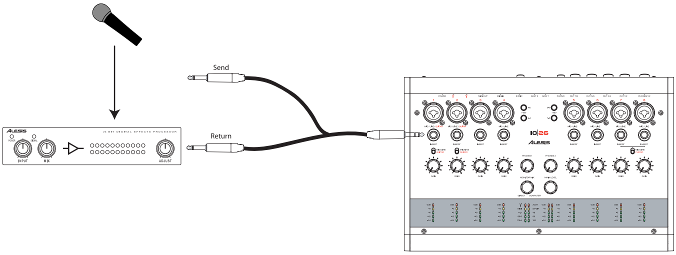
1. **Using the IO|14/26 as a preamplifier** – In certain situations, you may want to use the IO|14/26's preamps by themselves. For example, if you're recording a performance onto an external hardware recorder (such as an Alesis HD24) but you need more preamplifiers, you can use an insert's "send" cable to send your preamplified signal to the recorder and simply not use the "return" connection on the insert cable.



2. **Bypassing the IO|14/26's preamplifier** – The IO|14/26 has excellent sounding preamplifiers built in, but if you have a megabucks or vintage preamp that you want to use instead, experiment with connecting it to the insert jack. The insert jack allows you to use your external preamp while completely bypassing the IO|14/26's built in preamplifier.

   To bypass the IO|14/26's internal preamplifier, connect the "return" of the insert cable to your external device. Leave the "send" of the insert cable disconnected.

   Doing this will unbalance the external preamplifier's signal. In most cases this is not a problem, but check with your preamplifier's manufacturer for any noise, level, or grounding concerns before connecting your equipment in this way.



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Watch Your Levels While Recording

A basic principle in digital recording is that you want to capture the loudest signal you can, but at the same time you never, ever want to exceed the maximum digital threshold. If you do, you'll introduce the nasty distortion known as "digital clipping" into your recording.

Your Alesis IO provides a number of methods to help you avoid digital clipping:

1. **24-bit recording** - Operating at 24 bits, the IO offers 256 times more resolution than that of 16-bit compact disks.

   One result of that increased resolution is that you have far more headroom to work within. You can easily record such that your loudest peaks for an unpredictable player are no greater than -12dBfs to -18dBfs. ("dBfs" stands for "decibels at full scale". In digital, once you reach full scale, you're using all of the available resolution. Attempting to record at greater than 0dBfs results in digital clipping.)

2. **Dedicated hardware metering of digital signal** - The meters on the front panel of the IO show your audio signal as it is seen by the digital converters. In other words, they're totally accurate, unlike analog meters which only approximate the digital signal strength.

3. **Dedicated input monitoring panel** - In the HDM application, there is a special tab for Input Monitoring. This tab provides an uncluttered view of the strength of your incoming signals. (Note that the tab is always in "narrow" view to provide a quick overview of all the channels.)

4. **Pre-fader level monitoring** - Again in the HDM application, all metering is always "pre-fader." That means that the metering shown to you is independent of how you set the volume sliders. This allows you to see your actual signal strength even as you're creating various custom mixes for the people you are recording.

If you haven't experienced digital clipping before, spend a moment to learn what it sounds like. Plug a microphone or line level source into your IO. In the HDM panel, turn up the fader for that channel in an appropriate output tab. Be sure that the "Direct Mon" slider towards the right is turned up as well. You want to hear your audio.

Watch the HDM panel metering, and turn up the IO's preamp until you're registering signals near the top of the scale. Then, go just a bit further. Notice the harsh, buzzing kind of distortion that appears. This distortion is digital clipping. It can ruin your recordings and should be avoided at all costs (unless you want it as a special, unmusical effect).

Base Sample Rates: 44.1/88.2/176.4kHz versus 48/96/192kHz

There are two "base" sampling rate standards in the world of professional audio—44,100 samples per second (44.1kHz) and 48,000 samples per second (48kHz). Broadly speaking, audio CD's operate at 44.1kHz, while film and television operate at 48kHz.

High definition sample rates—including 88.2kHz, 96kHz, 176.4kHz, and 192kHz—are all simply multiples (doublings and quadruplings) of the 44.1kHz and 48kHz standards. The IO14/26 offers all of these sample rates.

If you're unclear on what base sample rate to use, consider this guideline:

- If your recordings are slated for release on CD, MP3, Casette, Vinyl, etc. set your sampling rate to 44.1, 88.2, or 176.4k for best results.

- If your project is slated for DVD, film, or television, set your sampling rate to 48, 96, or 192k for best results.

If you're working on a commercial project and you're not sure what rate to use, ask your technical supervisor before you begin working.

Changing base sample rates mid-stream in projects is fairly straightforward in most DAW programs. However, doing so may cause very slight but still audible degradation of your audio.

High Resolution Recording

The Upside of High Definition recording

Recording at high definitions (i.e., anything at 88.2k or above) means you're capturing sound at well beyond the range of human hearing. Doing so has three sonic advantages:

1. **At the Hardware Level:** All analog-to-digital converters need to heavily filter your signal's highest frequencies to prevent a nasty form of distortion called "aliasing" from taking place. Only sounds above the sampling limit are removed, but the filter itself causes unwanted phase-shifts that some critical listeners can hear (mind you… these are quite subtle changes in your audio)

   When you're recording in HD, the anti-aliasing filter is much more gradual and set at a very high frequency (well over the upper limit of human hearing). This all but eliminates any of the phase distortion that you may have heard.

2. **At the Software Level:** Since software plugins such as equalizers and compressors have more sample data to work with, carefully programmed plugins can sound better.

3. **Archiving:** If you're capturing a special recording that may have some historical value in the future, it makes sense to capture it with the highest level of technical accuracy.

   Alesis has designed your IO14/26 to sound excellent at all sample rates. Decide for yourself if the benefits and tradeoffs are worth it for a particular session.

The Downsides of High Definition Recording

1. **Disk Usage –** Recording at high sample rates consumes hard disk space much more quickly than standard rates. The following table describes sampling rate vs. disk usage for a 60-second snippet of monophonic (1 track) audio:

Length	Bit Depth	Sampling Rate	Disk Usage
60 Seconds	24 bit	44.1k	7.9 MB
60 Seconds	24 bit	48k	8.6 MB
60 Seconds	24 bit	88.2k	15.9 MB
60 Seconds	24 bit	96k	17.3 MB
60 Seconds	24 bit	176.4k	31.8 MB
60 Seconds	24 bit	192k	34.6 MB

   You can see how this may become a problem on large musical projects. For example, whereas a 5-minute song with 16 channels of 24-bit audio would require up to 635MB to record at 44.1k, the same song would need approximately 2.54 gigabytes of storage if you're recording at 176.4k!

   Please note:
   At quad sample rates (176.4k and 192k), only inputs 1-4 of the IO|26 are operational.

   We recommend that you take a look at your available hard disk resources when deciding which sampling rate to use. You can use the following formula to estimate the total disk space required for a song:

   Song length (in seconds) X Number of Channels X Sampling Rate X 3

   So… Our hypothetical 5-minute song would be calculated in the following way:

   300 sec x 16 channels x 44,100 x 3 = 635,040,000 byes (about 635 MB)

   That same song recorded at 176.4k would be:

   300 sec x 16 channels x 176,400 x 3 = 2,540,160,000 bytes (or 2.54 GB)

Geek talk: Why do we multiply by 3?

We have to multiply our disk usage by 3 because we assume you'll be recording at 24 bit. One "byte" of disk space contains 8 "bits" of information. Thus, if you are recording at 24-bit resolution, you'll need 3 bytes to contain all of that sample data (since 3 x 8 = 24). This is why your disk usage must be multiplied by 3.

If you were recording at 16 bit (which we do not recommend due to the reduction in sound quality) you would only multiply your total by 2 since a 16 bit sample only needs 2 bytes to describe the sample (2 x 8 = 16).

If all of this sounds like Martian to you—don't worry. You don't really need to know this stuff and there's no quiz at the end of the manual!

2. **Processor Usage –** A second drawback of recording in high definition is that you'll use substantially higher amounts of CPU resources. This is because your computer processor has to deal with twice as many samples operating at 88.2k than it does when processing at 44.1k and four times as many samples when recording at 176.4k. The following hypothetical scenario should clarify:

Sample Rate	Maximum Plugins you can run on your computer
44.1/48k	40
88.2/96k	20
176.4/192k	10

If you generally don't use many audio tracks and plugins, this won't affect you very much. If you use tons of audio tracks and plugins, this may tilt you in favor of recording at a lower sampling rate (or get a faster computer if you insist in recording in HD).

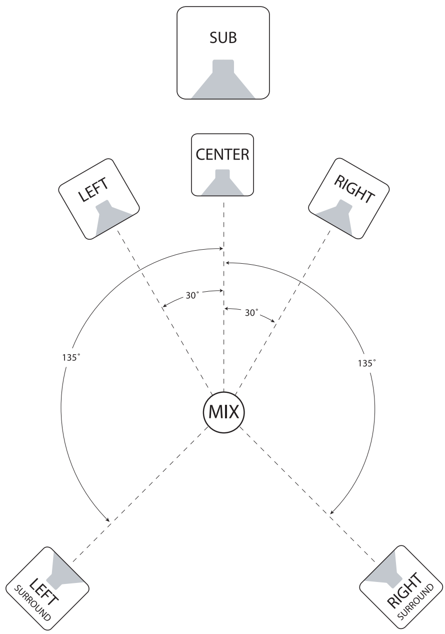
3. **Fewer ADAT optical inputs –** If you're using an external analog-to-digital converter to add more inputs to your IO14/26, you can add 8 (IO|14) or 16 (IO|26) channels at 44.1 or 48k, but you can only add 4 channels of 88.2k or 96k audio.

The ADAT2 port on the IO|26 is only operational at 44.1k and 48k sample rates.

Surround Sound (IO|26 only)

The IO|26's 8 analog outputs make it perfect for multichannel surround-sound applications (such as sound for film, TV, or DVD). If your software supports surround mixing, simply hook up your speakers to the IO|26 and refer to the software's documentation on how to set up a surround mixing environment.

Note that there are several common surround formats including 4-channel (a.k.a. "quad"), 6-channel ("5.1"), and 8-channel ("7.1"). There are also many non-standard and "custom" mixing scenarios and every situation has its own rules and requirements on speaker type, placement, and other factors. It would be beyond the scope of this manual to cover each type of surround configuration, but the following "5.1" mixing scenario is commonly used and will work for most applications. The diagram below illustrates the setup:



The "Mix" circle represents the "sweet spot" where the engineer sits. The left and right speakers are positioned 30° off center. The rear two channels are positioned 135° off of the center speaker.

The distance from the mixing engineer to each speaker should be identical (or as close to it as possible).

Note that subwoofer placement depends on the size and shape of the mixing room. The sub should be placed in a location that provides the most linear frequency response. This generally requires some trial-and-error before the optimal subwoofer position is located.

Again, if you're working on a commercial project, make sure to speak to the supervisor and iron out the details before jumping in! This may save you from time-consuming conversion or remixing in the future.

More on Surround Mixing

The Grammy organization has excellent (and free) resources available on the web for people interested in surround mixing. Please see http://www.grammy.org/pe_wing/guidelines/for more about this.

Using the IO14/26 with Sonar and other WDM applications

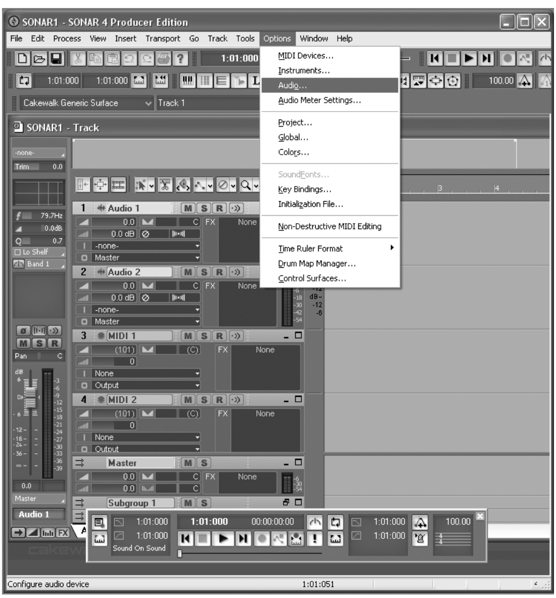
WDM or ASIO? Experiment with both protocols

An increasing number of Windows audio applications—including Sonar—offer a choice of either ASIO or WDM operation. It's worth experimenting with the two different modes of operation. ASIO will often (but not always) prove to be the superior choice.

For applications that only work using WDM, the following instructions should prove helpful.

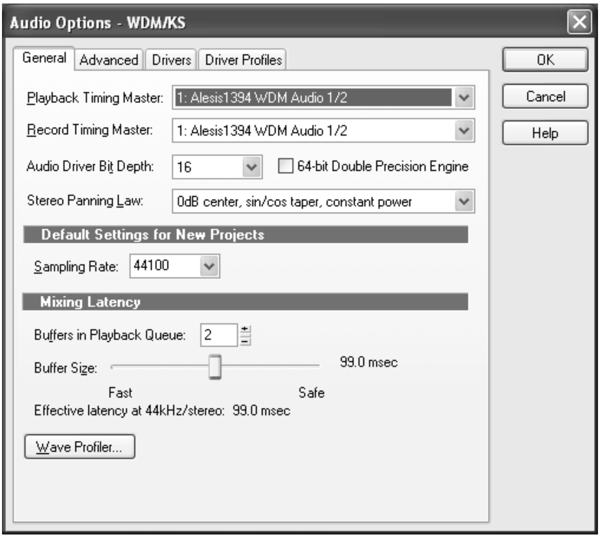
Choosing the IO14/26 as your audio device

1. Choose the menu "Options" | "Audio…."



Unlike ASIO, WDM allows different audio devices to be used at the same time. However, doing so can cause synchronization problems. Therefore, we suggest that you use the IO14/26 as your sole computer-connected audio input/output device when using WDM.

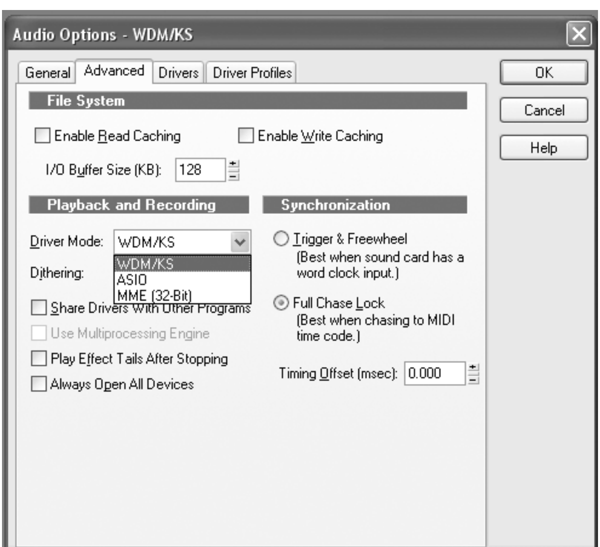
2. On the "General" tab, select any available IO14/26 channels as the Playback and Record timing masters.



This discussion uses Cakewalk's Sonar software, but the principles provided here apply to all WDM-based recording software.

For applications like Sonar that support both WDM and ASIO modes, you may want to experiment with each to see if either mode offers greater stability. Alesis generally recommends using ASIO mode when possible.

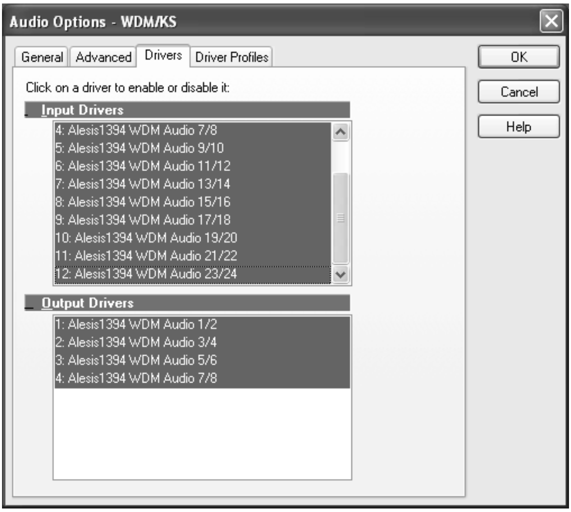
3. Click the "Advanced" tab. For WDM operation, be sure that the "Driver Mode" is set to "WDM/KS." (If you change this setting, you will need to exit and then restart Sonar.)



A limitation of the WDM architecture is that, each time you change between "low" (44.1/48), "medium" (88/2/96), and "high" (176.4/192) sample rates, the WDM profiler needs to be run again.

Therefore, when changing amngst these sample rates, first close SONAR,then change the sample rate on the Alesis control panel. Finally, re-open SONAR and allow its WDM profiler to run again.

Move to the "Drivers" tab. Click on each input pair and also on the output pair to make them available to Sonar.



If you're having problems operating the IO|14 or IO|26, this troubleshooting index may help you resolve your issues.

Symptoms	Cause	Solution
No sound from the IO|14/26.	No power.	Plug in power adapter or FireWire cable. If using FireWire bus power, be sure that your computer's FireWire port can provide power to the IO|14/26.
	Main Output level set too low.	Raise the MAIN LEVEL knob.
	Speakers (or amplifier) is turned off or down.	Turn speakers (or amplifiers) on or up.
	Headphone level is too low.	Turn up the PHONES knob for your headphone output.

	Cables not hooked up properly.	Check outputs to make sure cables are plugged in correctly (and securely).
	Bad cable(s).	Check all cables; substitute cables with known good ones.
Audio signal is distorted.	Channel input gain is too high.	Turn down the input gain knob for the distorting channel (use the LED meters to locate channels that are distorting).
	Output levels are too high.	Turn down your audio software's output level.
	-10/+4 input switch set incorrectly.	Be sure that the "-10/+4" input switch for each input is set correctly in the IO|14/26's control panel
	Sound source is too loud.	Turn down the output level of the sound source. If you are using a microphone with a volume "pad," engage the pad switch.

Audio signal carries an unwanted hum.	Not using balanced cables.	Make sure you are using balanced (XLR or 1/4" TRS) cables wherever possible.
	Improper grounding	Insert a "ground lift switch" into the offending cable's signal path.
	Damaged cable	Replace cable with a known good cable.
Microphone level is too low.	Phantom power is not turned on.	Turn on phantom power using the "+48V" switch on top panel of IO|14/26.
	Microphone is damaged.	Test the microphone on other audio devices. If you detect damage, contact the manufacturer or dealer.
No or low sound from a channel input.	Gain set too low.	Increase the gain for that channel.
	Instrument volume is too low.	Turn up the instrument's volume knob.
	Instrument volume is too low.	Turn up the instrument's volume knob.
Turntable input is too low in level and/or lacks bass.	"Mic/Line" and "Phono" switch is set incorrectly (this applies only to the IO|26)	Hook up your turntable to outputs 7-8 (on your IO|26) and set the "Mic/Line or Phono" switch to "Phono."

LED input meters not working.	Input level too low	Turn up the GAIN knob or the output level of your instrument.
LED output meters not working.	Output level too low (or muted)	Un-mute and/or turn up the output level of your audio software
	Audio is not routed to the correct outputs in your audio software.	Make sure your audio software is routing your signal to the correct outputs.
Computer does not see the IO|14/26.	FireWire connection must be established between your computer and the IO|14/26.	Unplug the FireWire and power cables from your IO|14/26. Reconnect the FireWire and power cables. If this does not work, leave the IO|14/26 on and power-cycle your computer. Do not use "restart" but instead actually turn off the computer and then turn it on again.
Computer sees the IO|14/26, but no sound is received and/or transmitted.	IO|14/26 is not set as primary sound device.	Windows: in the computer's Control Panel, go to the Sounds/Multimedia area. In the Audio section, set the default sound recording and playback devices to the IO|14/26. Mac: perform a similar operation under Audio/MIDI Setup.
FireWire audio has crackling or glitches, or audio plays/records at incorrect pitch.	Buffer size is set too small.	Increase the buffer size in the IO|14/26 control panel.
	Computer configuration may be incompatible with FireWire audio.	Certain FireWire chipsets have design limitations or IRQ assignment restrictions that must be resolved before audio can work correctly on them. See your FireWire chipset documentation for further information if required.
No power.	Defective power supply (or FireWire cable).	Replace power supply (only use Alesis-recommended AC output power supply).
	FireWire cable does not carry power.	If you are powering your IO|14/26 through the FireWire

		cable, you need to make sure you're using a 6-pin FireWire cable that can carry power. The mini, 4-pin connector found on most Windows notebook computers is not capable of providing bus power.
	Computer provides insufficient power for bus power operation.	Use the supplied AC adapter.

Computer or audio application does not see the IO|14 or IO|26 interface

Basic troubleshooting

First, check that the IO|14/26 is seen by the computer: Under "Sounds and Audio Devices" (Windows) or "Audio MIDI Setup" (Mac), look for your IO|14 or IO|26 to be listed as an available device.

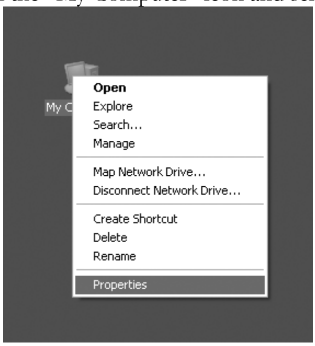
If your IO|14/26 is not shown, check that the Firewire cable is properly connected. Turn your IO|14/26 off, wait a few seconds, and then power it up again. Repeat this process until the IO|14 or IO|26 is found.

Advanced troubleshooting under Windows

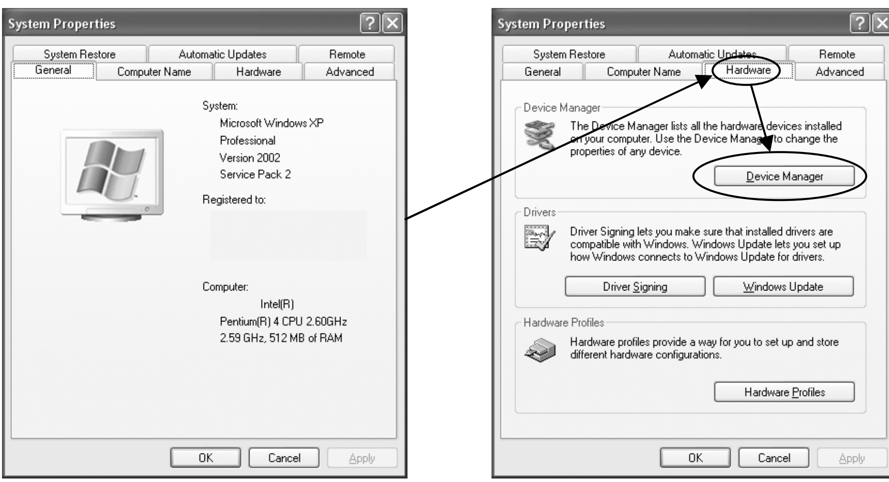
The best way to see that your IO|14/26 is connected and operating properly is through the Windows Device Manager. This is a powerful Windows component that requires some navigation but yields very detailed information.

1. Start by finding the "My Computer" icon either on your desktop or from the Start menu.

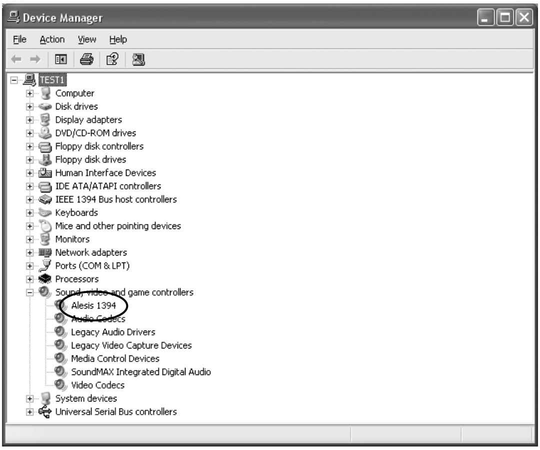
2. Right-click the "My Computer" icon and select "Properties."



3. The active tab at the top of this window is now the "General" tab. Click the "Hardware" tab. Then, click the "Device Manager" button.



4. Finally, expand the "Sound, video and game controllers" section. An "Alesis 1394" entry should appear.



No connection
If "Alesis 1394" does not appear, Windows does not see the interface as being connected to the computer.

1. Check your Firewire cable and try powering the mixer off and on again.

2. Expand the "IEEE 1394 Bus host controllers" section. Check that your Firewire card is listed as properly working.

3. As a last resort, power your computer off and on.

When restarting your computer, turn it off all the way. Pull out the AC power cord, keep it out for a few seconds, and then plug it back in again before you restart your computer.

This ensures that your computer's motherboard and PCI cards (including your FireWire card) are properly reinitialized.

Faulty connection
If "Alesis 1394" appears, but there is a yellow question mark or exclamation point next to it, the drivers are improperly installed. Right-click on the "Alesis 1394" listing and choose "Uninstall…", and uninstall the device. The next time you connect the IO14/26 to your computer, you will need to re-install the drivers again.

Audio playback or recording is at the wrong speed

Adjust the buffers in the ASIO control panel or through your WDM application's audio setup options.

It is impossible to say which buffer settings are perfect for each system, but generally buffer sizes around the middle or low-middle of the available range provide the best results.

Audio playback or recording stutters or drops out

Again, work with the buffer settings in the ASIO control panel or through your WDM application's audio setup options in order to find the most reliable setting.

Many audio applications include advanced setting dialog boxes where pre-fetch and other parameters can be set. Explore those options.

Help on the web
Most major recording programs are supported by lively user forums. The odds are that, if you're having problems, someone who posts at these forums has already experienced them and found solutions.

Don't be shy about visiting these forums and posting your questions.

Audio echoes during recording

This is probably happening because your "delayed" signal (the signal that passes through your computer's audio application back out to your speakers) is mixing with your "hardware direct monitoring" mix.

To resolve this problem, either disable your audio software's "input monitor" option or, if you want to monitor through your software, mute the appropriate input channel(s) on the Alesis Hardware Direct Monitoring application. For more information on the HDM application, see page 35.

Problems with notebook computer audio recording

Notebook computer recording problems (hum, dropouts, etc.) can generally be diagnosed and solved in one of the following ways:

1. The power from your notebook computer may be marginally insufficient to guarantee stable performance. Use the included AC adapter whenever you connect your IO|14/26 to a notebook computer or small form factor computer (such as a Mac Mini computer or a computer built around a microATX motherboard).
2. The notebook computer may be introducing a ground loop problem. To test for this, unplug your notebook's power adapter and run on battery power. If this solves your problem, consider using a three-to-two prong "ground lifter" between your computer and your AC power source. Be sure to observe all necessary safety precautions.
3. If your computer has a mini Firewire jack, be suspicious of your four-to-six pin Firewire cable. Many poorly constructed cables cause problems. Try using a respected, brand-name cable.
4. Again, if your computer has a mini Firewire jack, consider using a six-pin Firewire PCMCIA card or ExpressCard to bypass the built-in firewire connection altogether.

Technical Specifications

Analog Inputs

Sample rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz
Frequency response:	+/- 0.05dB from 20Hz to 22kHz
Dynamic range:	112dB, A-weighted
Signal-to-noise ratio:	112 dB, A-weighted, minimum gain
THD+N:	0.001 -0.0015% @ 1kHz/0dBFS
Crosstalk:	-110dB @ 1 kHz
Preamplifier THD+N:	<0.0007 at 20dB gain
Preamplifier slew rate:	15 volts/microsecond
Microphone gain range:	+6.8dB to +50dB
Microphone impedance:	1.2kOhm
Line gain range:	-15.4dB to 27.8dB
Line impedance:	16kOhm
Guitar gain range:	+6.8dB to +50dB
Guitar impedance:	1 MegaOhm

Turntable Input (IO|26 only):

Sample rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz
Frequency response:	+/-1dB from published RIAA curve
Dynamic Range:	95dB, A-weighted
Signal-to-noise ratio:	95dB, A-weighted
THD+N:	<0.05%
Crosstalk:	-100dB @ 1 kHz
Impedance:	47Kohm

Line-Level Outputs

Sample rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz
Level:	-10dBv/+4dBu nominal (switchable), +19dBu maximum
Frequency response:	+/- 0.075 from 20Hz to 22kHz
Dynamic range:	108dB, A-weighted
Signal-to-noise ratio:	108dB, A-weighted
THD+N:	0.0011%-0.0017% @ 1KHz/18-19dBU
Crosstalk:	-105dB @ 1 kHz

Output impedance:	220 Ohm

Headphone Outputs

Frequency response:	+/-0.1 dB, 22Hz to 22kHz
Power (@ 32 Ohms):	50mW
Gain range:	20dB
THD+N:	<0.05 %
Signal-to-noise ratio:	100 dB, A-weighted
Output impedance:	32 Ohm
Load impedance range:	32 to 600 Ohms

S/PDIF and ADAT digital inputs

Bit depth:	24 bit
Sample rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz

Dimensions **(W x D x H)**

(Without packaging or power adapter)

IO|14:	7.9375" x 8.0" x 2.875" / 202mm x 203mm x 73mm, 3.5lbs / 1.6kg
IO|26:	12.1875" x 8.0" x 2.875" / 310mm x 203mm x 73mm, 5.0lbs / 2.3kg

Glossary

Here are the definitions to some terms you'll probably encounter while using your IO|14/26 FireWire Audio Interface.

Term	Definition
ASIO	ASIO is an acronym for "Audio Stream Input/Output." It is an audio protocol developed by Steinberg and used by many software manufacturers to communicate with audio hardware.
balance	A control that lets you adjust the ratio of one signal to another. For example, you can adjust the balance between your zero-latency mix and your computer's output using the MONITOR BLEND knob.
bus	The electrical component that carries signals from multiple sources to a single destination such as an amplifier.
channel	A path through which an audio signal flows.
clipping	The distortion that takes place when your signal is too loud for channel's circuitry to handle. When viewed on an oscilloscope, this distortion makes your signal's peaks and troughs look like they have been cut (or "clipped") off.
codec	Compression/decompression algorithm. Different CODECs are used by different digital audio devices and file formats.
condenser microphone	A type of high-quality microphone that produces a weak signal, usually requiring an external power source like the ones provided by your IO|14/26's "phantom" power switch.
DAW	Digital audio workstation. DAWs can be either standalone, like the Akai DPS series, or software-based.
dB (decibel)	A common unit of measure for audio.
dry	Term used to describe an audio signal free of effects. The opposite of "wet."
dynamic microphone	A common type of microphone that does not require external power. Dynamic microphones are generally cheaper and more durable than condenser microphones but don't reproduce sound as well.
effects processor	A unit whose purpose is to provide effects for audio signals. Some common effects include reverb, chorus, flange and delay. Effects processors come in many shapes and sizes, from small pedals up to 19" rackmount units.
EQ (equalizer)	The part of your mixer (or other device) that manipulates an audio signal by lowering the level of some frequencies and/or increasing the levels of others. EQ is used to fine-tune a signal's highs and lows.

fader	A device that allows you to control the level of an audio signal by sliding the fader up and down a straight path. Each input on the IO|14/26 has its own fader in the Zero Latency Monitoring application.
Firewire	A standard for connecting external devices to a computer. Also called IEEE 1394a
gain	The measure of extra amplification applied to an audio signal. Each analog input on your IO|14/26 has its own gain knob, which can be used for boosting mic and line signals.
IEEE 1394a	Another name for Firewire.
insert	A special "access point" found in mixers and some audio interfaces that allows you to place an audio device (such as a compressor) in the signal path. Inserts are placed after the preamplifier and before the remaining electronics in your signal path. They require special "insert" cables.
latency	The time it takes for audio to travel from the IO|14/26, through the computer's recording program, and out again to your speakers (or heaphones). Latency is measured in either samples or milliseconds.
level	The amount of power driving an audio signal. The most common names given to levels of varying voltage are, from lowest to highest, "microphone" level, "instrument" level and "line" level.
master section	The section of a mixer where the main mix is controlled.
mic preamp	An amplifier that boosts a microphone or instrument-level signal up to line level.
mixer	A device whose purpose is to combine and output a number of audio signals, allowing various types of signal manipulation.
mono (monaural)	Refers to an audio signal that has only one channel. The opposite of stereo.
pan	A control that lets you position a mono signal within the stereo spectrum by altering the level of the signal being sent to the left channel as opposed to the right.
phantom power	A way of providing power to condenser microphones. Also known as +48volt power.
return	A line input whose function is to carry back to the mixer an audio signal that has been sent from the mixer. Returns are used in conjunction with "insert" points.

sample rate	Digital audio is chopped up into tiny time slices. The sample rate is the number of time slices captured in one second. 44.1kHz—44,100 samples per second—is the standard used for Compact Disc audio. 48kHz—48,000 samples per second, is commonly used for film and video.
send	A line output whose function is to send a signal from the mixer to an external device, usually an effects processor. Like returns, sends are used in conjunction with insert cables.
stereo	Refers to an audio signal that has two channels.
unity gain	Refers to the setting of an audio channel at which the signal leaves the channel at the same level at which it entered. Unity gain is marked by a 0 on the MultiMix's faders.
WDM	The Windows Driver Model. This is the default standard by which Microsoft Windows communicates with audio devices.
wet	An audio signal that has had effects or other manipulations applied. The opposite of "dry."

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Alesis Limited Warranty

ALESIS CORPORATION ("ALESIS") warrants this product to be free of defects in material and workmanship for a period of one (1) year for parts and for a period of one (1) year for labor from the date of original retail purchase. This warranty is enforceable only by the original retail purchaser and cannot be transferred or assigned. For the most effective service, the purchaser should register the purchase on the ALESIS website at http://www.alesis.com/support/warranty.htm.

During the warranty period ALESIS shall, at its sole and absolute option, either repair or replace free of charge any product that proves to be defective on inspection by ALESIS or its authorized service representative. In all cases disputes concerning this warranty shall be resolved as prescribed by law.

To obtain warranty service, the purchaser must first call or write ALESIS at the address and telephone number available on the Alesis Website to obtain a Return Authorization Number and instructions concerning where to return the unit for service. All inquiries must be accompanied by a description of the problem. All authorized returns must be sent to ALESIS or an authorized ALESIS repair facility postage prepaid, insured and properly packaged. Proof of purchase must be presented in the form of a bill of sale, canceled check or some other positive proof that the product is within the warranty period. ALESIS reserves the right to update any unit returned for repair. ALESIS reserves the right to change or improve design of the product at any time without prior notice.

This warranty does not cover claims for damage due to abuse, neglect, alteration or attempted repair by unauthorized personnel, and is limited to failures arising during normal use that are due to defects in material or workmanship in the product.

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THIS CONTRACT SHALL BE GOVERNED BY THE INTERNAL LAWS OF THE STATE OF CALIFORNIA WITHOUT REFERENCE TO CONFLICTS OF LAWS. This warranty gives you specific legal rights, and you may also have other rights required by law which vary from state to state.

This warranty only applies to products sold to purchasers in the United States of America or Canada. The terms of this warranty and any obligations of Alesis under this warranty shall apply only within the country of sale. Without limiting the foregoing, repairs under this warranty shall be made only by a duly authorized Alesis service representative in the country of sale. For warranty information in all other countries please refer to your local distributor.

For more effective service and product update notices, please register your IO|14/26 online at http://www.alesis.com.

Alesis Contact Information

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IO|14/IO|26 FireWire Audio Interface Reference Manual
Revision C by Leo Der Stepanian and Fred Morgenstern

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