

**LAYLA<sup>®</sup>24**

**LAYLA<sup>®</sup> *LapTop***

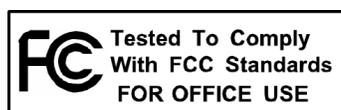
**MONA<sup>™</sup>**

**MONA<sup>™</sup> *LapTop***

**GINA<sup>®</sup>24**

**MIA<sup>™</sup>**

**Owner's Manual Version 3.0 for Windows**



Designed and Manufactured in the U.S. by Echo Corporation

## 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** – (does not apply to Mia) Precautions should be taken so that the grounding capabilities of the unit are not undermined. This equipment is provided with a cord having 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** – (does not apply to Mia) 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

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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.

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**Echo Digital Audio Corporation  
6460 Via Real  
Carpinteria, CA 93013**

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## 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:

- A PCI card wrapped in an anti-static cover **OR** a CardBus Adapter
- An audio interface box (Layla24, Mona, and Gina24 only)
- Four adhesive backed rubber feet (Layla24, Mona, and Gina24 only)
- Four mounting screws with collars (Layla24 and Mona only)
- An interface cable (Gina24, Layla24, and Mona only)

***Please note:** The cable included with the Gina24 system is a shielded audio cable that has been custom manufactured to exacting standards. Use of any other cable, such as a computer printer cable, will substantially reduce the system's overall audio quality.*

- A power cable (Layla24 and Mona only)
- An Esync cable (Gina24 only)
- A CD-ROM containing:
  - Windows 98/Me/2000/XP Drivers
  - Macintosh Drivers
  - .PDF Users Guide
  - Steinberg's Cubasis VST multitrack recording and editing software, Windows and Macintosh versions.
  - Demo versions of digital audio recording, editing, and processing software from a variety of manufacturers
- Quick Start Guide

## System Requirements

In order to use Layla24, Mona, Gina24 or Mia you'll need the following:

- A desktop computer one of the following:
  - A genuine Intel Pentium processor and genuine Intel chipset **OR**
  - An AMD Athlon or Duron processor with one of the following chipsets:
    - Via KX-133
    - Via KT-133
    - Via KT-133a
    - AMD 760/761
- a minimum 128Mb of RAM (256Mb highly recommended)
- A fast, high-capacity IDE or SCSI hard disk drive
- Windows 98/Me/2000/XP
- Peripheral audio equipment, such as a mixer, microphones, studio monitors, musical instruments, etc.

In order to use Layla24 *LapTop* or Mona *LapTop*, you'll need the following:

- A laptop computer with a Type II CardBus Slot
- a minimum 128Mb of RAM (256Mb highly recommended)
- A fast, high-capacity IDE or SCSI hard disk drive
- Windows 98/Me/2000/XP
- Peripheral audio equipment, such as a mixer, microphones, studio monitors, musical instruments, etc.

## **Installation**

Complete installation consists of disabling Windows system sounds, deciding which Echo driver you are going to use, installing the Windows drivers into your system, installing the PCI or CardBus card, connecting the audio interface to the card (Layla24, Mona, and Gina24) and, if necessary, installing a multitrack audio recording/editing application.

### **System Sounds**

We suggest that you turn off your Windows system sounds prior to installing your Echo product. Because most system sounds are sampled at very low sample rates, typically 11 to 22 KHz, each time they are played it will cause the sample rate clock on your Echo product to reset to the slower speed.

To turn off the system sounds, first go to the Windows “Control Panel,” which can be found under “Settings” from the “Start” button. On the control panel you will find an icon titled “Sounds” or “Sounds and Multimedia”. After double clicking on this icon, you will see a window labeled “Schemes” near the bottom of the screen. Click on the small down arrow to the right of the combo-box, and select the “No Sounds” option. Then, click on the Ok button.

### **VxD vs. WDM - Deciding Which Driver to Install**

VxD and WDM are two different driver architectures for Windows. You only have a choice between the two if you are using Windows Me. If you are using Windows 98, you must use the VxD drivers. If you are using Windows 2000 or XP you must use the WDM drivers.

For Windows Me you should use the VxD drivers, unless you fall under one or both of the following two categories:

- 1) You are using an audio program such as Cakewalk’s SONAR, which takes advantage of the WDM architecture to achieve low latencies.
- 2) You are using an audio program that uses an ASIO driver. In general, the WDM ASIO driver provides higher performance than the VxD driver.

For a detailed discussion of the benefits and drawbacks of the WDM architecture, please read Appendix D: WDM-Windows Driver Model FAQ.

## Installing the PCI Card for Desktop Computers

Once you have verified that there are no problems with your system, it is time to install the Echo PCI card into your computer.

1. **Install the drivers.** You should do this **before** inserting the PCI card. Insert the Echo Install CD-ROM into your machine. The Main Install Window should automatically appear; if it does not, simply go to “My Computer,” and double-click on your CD-Rom drive icon. Now, select “Install Windows Drivers,” and follow the on-screen directions.
2. Once you have installed the appropriate driver for your version of Windows, **turn your computer off.**

***IMPORTANT - Unplug your computer and detach all peripherals before proceeding with the following steps.***

3. **Remove your computer’s cover.** This operation differs from computer to computer. Refer to your computer’s manual for a further explanation of this step if necessary.
4. **Select the PCI slot** into which you will install the card. Inside your computer you will likely find two types of expansion slots, ISA and PCI. The PCI slots will be shorter and have a higher connector pin density than the ISA slots. You may use any of the available PCI slots in your computer for the card. Unscrew and remove the bracket covering the expansion slot where you would like to install your Echo product. Put the screw in a safe place, as you will need it later to complete the installation.
5. **Insure that you have fully discharged all static electricity** from your body before handling the card. This can be done through the use of a grounding strap or, more simply, by touching your bare hand to the metal casing of the computer’s power supply. (For this latter method to work, the computer must be plugged in, though not turned on.) After you’ve discharged your static, unplug the computer before proceeding to the next step.
6. Remove the card from its protective anti-static bag. Handle the card carefully by its edges, and **insert it into the selected expansion slot.** Insure that the card’s edge connector (the protruding edge with the gold leads) is seated firmly into the slot. Centering the card over the slot and using a gentle rocking motion while pushing downward into the slot generally works well. Be careful not to force the card into the slot, or bend or twist it while it is being inserted, as this could result in the card being damaged.

7. Now use the screw that you removed earlier, from the protective back plate, to **attach the metal bracket** on the card to the computer's rear panel.
8. **OPTIONAL:** *(If you have Mia, skip this step.)* Esync cables come with Darla24 and Gina24, and can be used to daisy chain several 24-bit Echo cards together. Esync is a proprietary form of super clock that allows you to sync together multiple 24-bit Echo products. Simply attach the Esync cable to the two-prong "Esync Out" connector on the Layla24/Mona/Gina24 PCI card, and attach the other end to the "Esync In" connector on your Darla24 or Gina24 PCI card.
9. **Replace the computer's cover**, and secure it. Reattach its power supply cord and reconnect any peripherals that you may have removed prior to beginning the installation.
10. *(If you have Mia, skip this step.)* Locate the audio interface box (Layla24, Mona, and Gina24), the interface cable, and the power cable. Securely mount the interface into your equipment rack (Layla24 and Mona). If you will not be mounting the box in a rack, remove the backing from the four rubber feet and place one in each bottom corner of the interface unit. Then be sure to locate the rack-mount box in a secure location.

**Plug one end of the cable into the connector** on the Echo PCI card that now protrudes through the back panel of your computer, and secure the cable using the built-in screws located on both sides of the connector. **Attach the other end of the cable** to the rear of the interface and fasten the cable securely with the screws. Plug the power cable firmly into the interface and a power socket, but do not turn on the interface at this time (Layla24 and Mona).

**Caution:** *Never connect the rack-mount interface to the computer while either the interface or your computer is turned on.*

11. You can now **attach external audio devices** to the connectors. Information on attaching external devices to your particular interface may be found in the following sections.
12. Layla24 and Mona only – **Turn on your Layla24 or Mona audio interface box.**
13. **Turn on your computer.** Once Windows starts, it should find new hardware and begin to install the drivers. For most versions of Windows, this process is entirely automatic. For Windows XP, you should select the option "Install the software automatically," and follow the on-screen directions.

**Note:** This version of the WDM driver has not been digitally signed by Microsoft; for Windows 2000 and XP, you will need to confirm that you want to install an unsigned driver.

## Installing the CardBus Adapter for Laptop Computers

Once you have verified that there are no problems with your system, it is time to install Layla24 *LapTop* or Mona *LapTop* into your laptop computer.

1. **Install the drivers.** You should do this **before** inserting the CardBus Adapter. Insert the Echo Install CD-ROM into your machine. The Main Install Window should automatically appear; if it does not, simply go to My Computer and double-click on your CD-Rom drive. Now, select “Install Windows Drivers,” and follow the on-screen directions.
2. Once you have installed the appropriate driver for your version of Windows, **turn your computer off.**
3. Locate the rack-mount interface, the 9-pin interface cable, and the power cable. **Securely mount the interface** into your equipment rack. If you will not be mounting the box in a rack, remove the backing from the four rubber feet and place one in each bottom corner of the interface unit. Then be sure to locate the rack-mount box in a secure location.
4. **Connect the cable.** Before installing the card into the computer, connect the cable between the card and the rack-mount chassis. Make sure the cable is fully inserted into the card and that it “clicks” into place. You may have to wiggle it a bit for this to happen.
5. **Insert the *LapTop* adapter** into your computer. Make sure the computer is turned off. To install the card, insert it into any available CardBus slot. Next, turn on the audio interface box.
6. **Turn on your computer.** Once Windows starts, it should find new hardware and begin to install the drivers. For most versions of Windows, this process is entirely automatic. For Windows XP, you should select the option “Install the software automatically,” and follow the on-screen directions.

**Note:** this version of the WDM driver has not been digitally signed by Microsoft; for Windows 2000 and XP, you will need to confirm that you want to install an unsigned driver.

Once the driver is installed, a small “PCCard” icon will appear in the system “tray” (usually on the lower right-hand side of the screen).

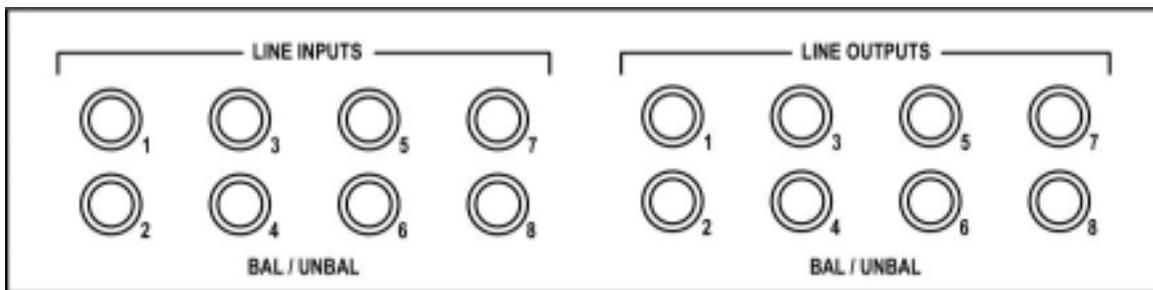
While it is possible to “hot-dock” the card by inserting it with the laptop powered up, it is recommended the card only be inserted or removed with the power off. *If you have to remove the card with your computer turned on, the card **must** first be disabled.* To do this, click on the PCCard icon in the tray, and tell Windows to stop the card. You will get a message when it is safe to remove it. Failure to disable the card before removing it could potentially damage the card or the computer.

## Connecting to Layla24’s Rack-mount Audio Interface

The back panel of Layla24’s rack-mount audio interface contains a wide variety of connections that allow great flexibility in the operation of Layla24. For optimal performance with Layla24, it is critical to use the appropriate cabling and connectors.

### Analog Inputs and Outputs

Layla24 has eight analog inputs and eight analog outputs on the rear panel. The input and output connections can accept unbalanced or balanced signals via ¼” connectors. For more information see “Unbalanced and Balanced Inputs and Outputs” in Appendix E.

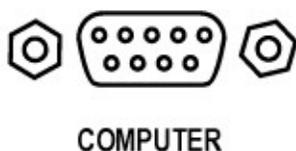


Layla24’s front panel has a ¼” headphone output jack with a corresponding volume knob which monitors analog outputs 1 and 2.



## The Computer Connector

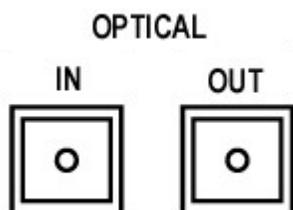
Next to the analog inputs and outputs is a connector labeled **COMPUTER**. This connector is known as a DB-9, and is similar to the serial port on a PC. It is the point at which the audio interface connects to the Layla24 PCI card inside your computer.



A cable was supplied with your Layla24 for this purpose. This custom made cable is manufactured to certain specifications; if you need to replace it, you should only use a replacement cable from Echo.

## ADAT / S/PDIF Optical I/O

Next to the **COMPUTER** connector is a pair of connectors labeled **OPTICAL, IN** and **OUT**. These connectors are used to transmit digital data among digital audio devices via an optical signal.

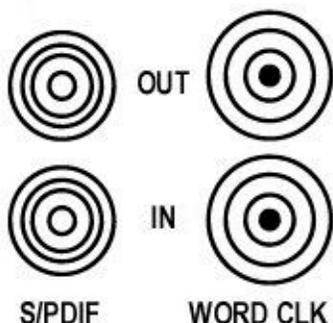


You can use this port for ADAT (8 channels) or for optical S/PDIF (stereo) I/O.

**Note:** Layla24 is only capable of transmitting or receiving one type of digital signal at a time. You must choose either ADAT optical or S/PDIF (optical or RCA); you cannot use both simultaneously. The **Digital Mode Switch** can be found in the Layla24 Console. For more information, refer to the section of the manual that discusses the console.

## S/PDIF

Next to the **ADAT** connectors is a pair of connectors labeled **S/PDIF, IN** and **OUT**. These S/PDIF connectors are used to transmit digital data among digital audio devices via an electrical signal. S/PDIF data can use the full 24-bit sample width used internally on Layla24.



When connecting devices to the S/PDIF jacks on Layla24, the use of standard analog RCA audio cables is not recommended. For reliable S/PDIF operation, 75-ohm coaxial (RG59) video cables are recommended.

## Word Clock

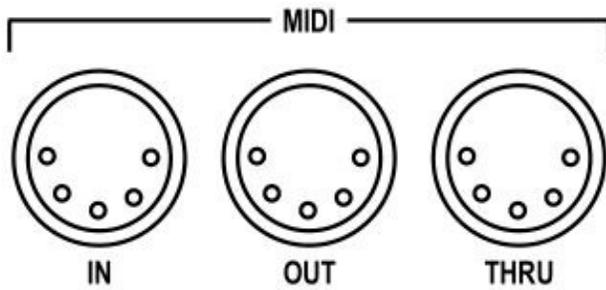
Next to the S/PDIF ports are the **Word Clock** connectors. These connectors allow you to synchronize one Layla24 to another Layla24, Layla, Mona or to other digital audio devices.

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 are 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 Layla24.

## MIDI

The last set of connections to the Layla24 interface is for MIDI. The MIDI ports can be used for receiving MIDI time code (MTC), or sending MIDI signals from your digital audio/MIDI sequencing software to external sound modules, etc.



MIDI cabling is highly standardized and widely available.

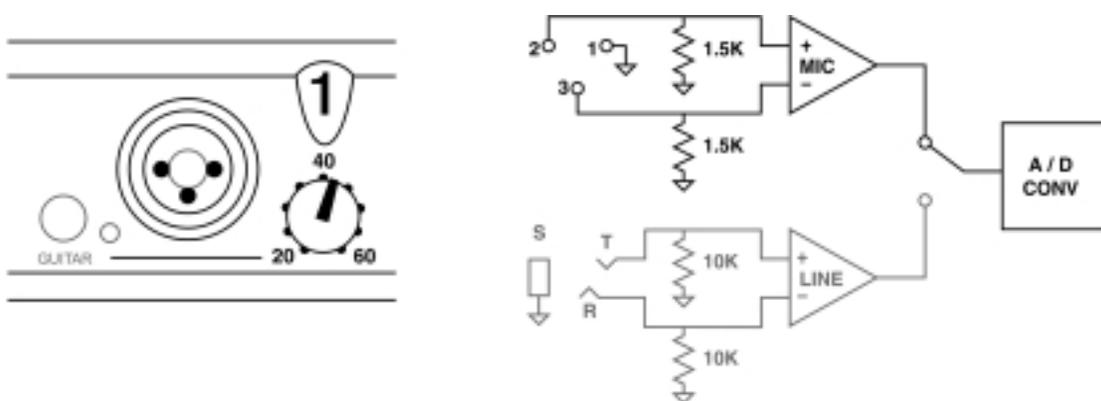
## Connecting to Mona’s Rack-mount Audio Interface

The back panel of Mona’s rack-mount audio interface contains a wide variety of connections that allow great flexibility in the operation of Mona. To achieve the optimal performance with Mona, it is critical that the appropriate cabling and connectors are used.

### Analog Inputs and Outputs

Mona has four analog inputs on the front panel, and six analog outputs on the back panel. The universal input connections can accept balanced or unbalanced signals via ¼” or XLR connectors. Since Mona has built-in preamps, you can just plug your microphone or guitar cable right into any of Mona’s analog inputs. No external preamp or mixer is needed!!

Mona’s “universal” inputs can be thought of as three inputs in one: a microphone input, a line input, and a guitar input.

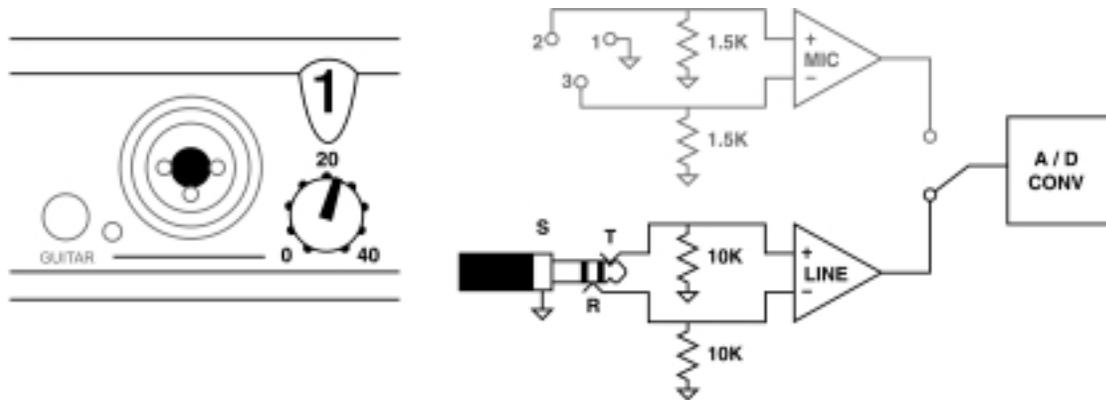


**Mona’s Mic Preamp Input (XLR, Impedance: 1.5K, Trim: 20 – 60dB)**

To use a microphone with Mona, plug the microphone’s XLR cable into any of the analog inputs. Whenever an XLR cable is connected to an input, the microphone preamp is enabled and the gain range of the trim knob is set to range from +20 to +60dB. The input impedance of the mic preamps is 1.5K.

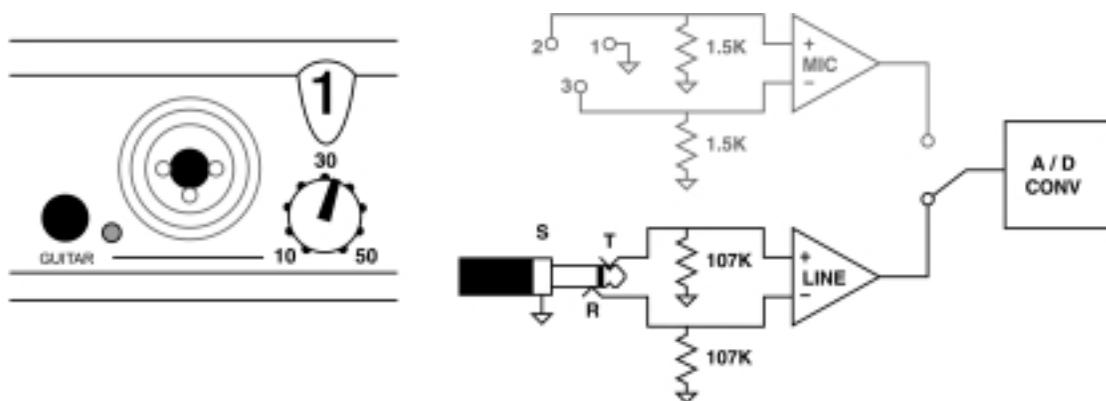
Mona provides 48 volts of phantom power for microphones that require it. To enable phantom power press the button labeled +48V (the LED next to it will light up) and phantom power will be provided to all microphone inputs.

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



**Mona's Line Input (1/4" TRS, Impedance: 10K, Trim: 0 - 40dB)**

When a 1/4" connector is plugged into Mona's universal input jack, the microphone preamp is disconnected and a line input amplifier is inserted in its place. If the **GUITAR** switch is out, the line input provides a balanced connection (for TRS plugs) and the trim knob has a range of 0 to 40dB. The input impedance for line-level signals is 10K. Full-scale input is 22dBu with the trim set to zero.



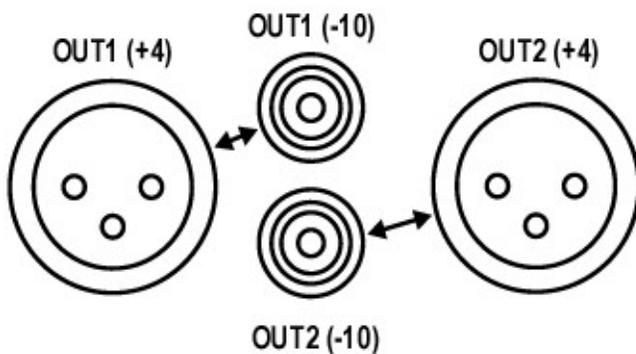
**Mona's Guitar Input (1/4" Plug, Impedance: 107K, Trim: 10 - 50dB)**

An input impedance of 10K is too low to record non-active guitar pickups, so each input also has its own **GUITAR** switch. When the **GUITAR** switch is pushed in (green indicator lit), the input impedance is increased to 107K and the trim knob has a range of 10 to 50dB. The indicator will only light up if there is a 1/4" connector plugged into the jack. This mode will still provide a balanced input when a TRS connector is inserted.

If you want to adjust the level of any input, just turn the knob for that input and watch the corresponding level meter on Mona's front panel. The red bar at the top will light anytime the input signal is greater than -3dB (or within 3dB of clipping). The three yellow bars each represent a range of 3dB and the green bars each represent a range of 6dB.

For optimal audio quality, you should adjust the input trim (knob) so that your loudest recorded signal lights up some of the yellow bars, but not the red. If the signal level ever exceeds 0dB 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! Mona has 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.

Mona’s front panel also has a ¼” headphone output jack with a corresponding volume knob which monitors analog outputs 1 and 2.



Each of Mona’s 6 outputs has one RCA and one XLR connector for your convenience. The RCA jack outputs a –10 dBV (unbalanced) signal, while the XLR jack outputs a +4 dBu (balanced) signal. Both connectors are active at all times, so the output signal will always go out of both jacks.

### The Computer Connector

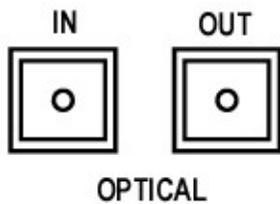
Next to the analog inputs and outputs is a connector labeled **COMPUTER**. This connector is known as a DB-9, and is similar to the serial port on a PC. It is the point at which the audio interface connects to the Mona PCI card inside your computer.



A cable was supplied with your Mona for this purpose. This custom made cable is manufactured to certain specifications; if you need to replace it, you should only use a replacement cable from Echo.

### ADAT/S/PDIF optical I/O

Next to the **COMPUTER** connector is a pair of connectors labeled **OPTICAL, IN** and **OUT**. These connectors are used to transmit digital data among digital audio devices via an optical signal.

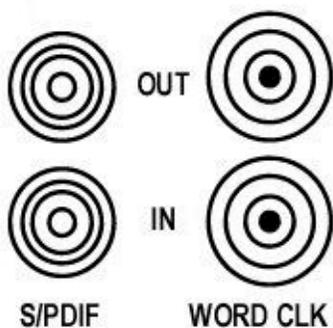


You can use this port for ADAT (8 channels) or for optical S/PDIF (stereo) I/O.

**Note:** Mona is only capable of transmitting or receiving one type of digital signal at a time. You must choose either ADAT optical or S/PDIF (optical or RCA); you cannot use both simultaneously. The **Digital Mode Switch** can be found in the Mona Console. For more information, refer to the console section of this manual.

### S/PDIF

Next to the **ADAT** connectors is a pair of connectors labeled **S/PDIF, IN** and **OUT**. These S/PDIF connectors are used to transmit digital data among digital audio devices via an electrical signal. S/PDIF data can use the full 24-bit sample width used internally on Mona.



When connecting devices to the S/PDIF jacks on Mona, the use of standard analog RCA audio cables is not recommended. For reliable S/PDIF operation, 75ohm coaxial (RG59) video cables are recommended.

## **Word Clock**

Next to the S/PDIF ports are the **Word Clock** connectors. These connectors allow you to synchronize one Mona to another Mona, Layla or Layla24 or to other digital audio devices.

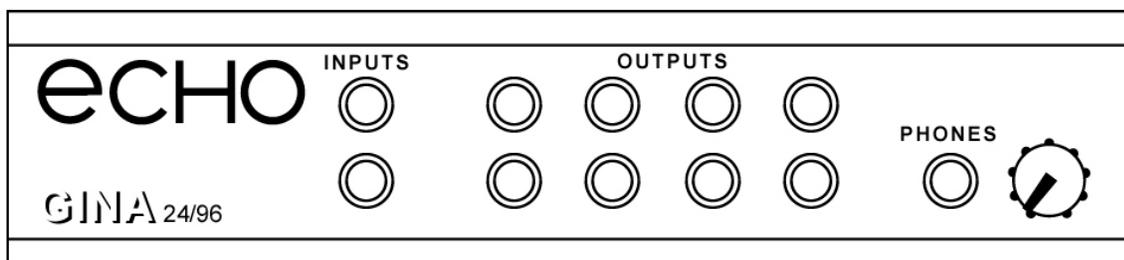
The word clock I/O uses a BNC connector. As with the S/PDIF I/O, a shielded 75-ohm (RG-59) coaxial video cable should be used. BNC connectors are widely used in the electronics industry for both video and computer networking. However, computer networks use 50-ohm (RG-58) coaxial cables and not the 75-ohm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

## Connecting to Gina24’s Audio Interface

The back panel of Gina24’s audio interface contains a wide variety of connections that allow great flexibility in the operation of Gina24. To achieve the optimal performance with Gina24, it is critical that the appropriate cabling and connectors are used.

### Analog Inputs and Outputs

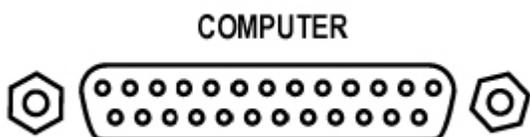
Gina24 has two analog inputs and eight analog outputs on the front panel. The input and output connections can accept balanced or unbalanced jacks via 1/4” connectors. Both the inputs and outputs are set to receive or send a +4 dBu (balanced) signal.



Gina24’s front panel also has a 1/4” headphone output jack with a corresponding volume knob which monitors analog outputs 1 and 2.

### The Computer Connector

On the back panel of the Gina24 audio interface box is a connector labeled **COMPUTER**. It is the point at which the audio interface connects to the Gina24 PCI card inside your computer.

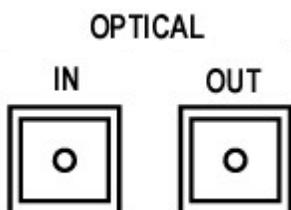


A cable was supplied with your Gina24 for this purpose. This custom made cable is manufactured to certain specifications; if you need to replace it, you should only use a replacement cable from Echo.

**Please Note: The Gina24 cable is NOT interchangeable with Darla24 or Gina (20-bit) cables.**

### ADAT/S/PDIF optical I/O

Next to the **COMPUTER** connector is a pair of connectors labeled **OPTICAL, IN** and **OUT**. These connectors are used to transmit digital data among digital audio devices via an optical signal.

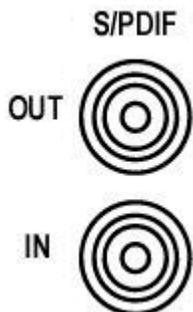


You can use this port for ADAT (8 channels) or for optical S/PDIF (stereo) I/O.

**Note:** Gina24 is only capable of transmitting or receiving one type of digital signal at a time. You must choose either ADAT optical or S/PDIF (optical or RCA); you cannot use both simultaneously. The **Digital Mode Switch** can be found in the Gina24 Console. For more information, please refer to the console section of this manual.

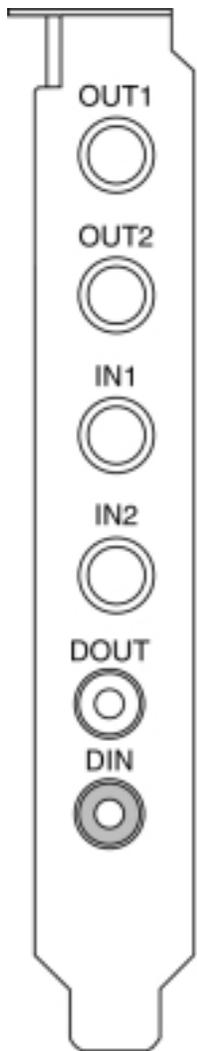
### S/PDIF

Next to the **ADAT** connectors is a pair of connectors labeled **S/PDIF, IN** and **OUT**. These S/PDIF connectors are used to transmit digital data among digital audio devices via an electrical signal. S/PDIF data can use the full 24-bit sample width used internally on Gina24.



When connecting devices to the S/PDIF jacks on Gina24, the use of standard analog RCA audio cables is not recommended. For reliable S/PDIF operation, 75ohm coaxial (RG59) video cables are recommended.

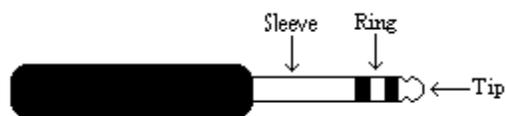
## Connecting to Mia



### Analog Inputs and Outputs

Mia has two analog outputs (OUT1 and OUT2) and two analog inputs (IN1 and IN2) on the back. The input and output connections can accept balanced or unbalanced jacks via ¼” connectors. Both the inputs and outputs are set to receive or send a +4 dBu (balanced) signal.

For optimum performance it is recommended that you use balanced signals when connecting to Mia’s analog connectors. Mia uses TRS connectors (tip, ring sleeve) for connecting balanced 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). Mia will also accommodate the two conductor unbalanced style connector.

### S/PDIF Digital Inputs and Outputs

Under the analog connectors is a pair of connectors labeled **DOUT** and **DIN**. These S/PDIF connectors are used to transmit digital data among digital audio devices via an electrical signal. S/PDIF data can use the full 24-bit sample width used internally on Mia.

When connecting devices to the S/PDIF jacks on Mia, the use of standard analog RCA audio cables is not recommended. For reliable S/PDIF operation, 75ohm coaxial (RG59) video cables are recommended.

## Using Audio Software – Gina24, Layla24, and Mona

To audio software, the Echo hardware appears as a group of stereo wave devices. You can view the list of these devices either from within your audio software or from the Windows control panel. The items you see in the list will depend on what driver you installed. The examples below are for Layla24; Mona and Gina will look similar; the only differences being in the number of input and output devices. (Obviously, the names will also be different.)

For playback devices, you should see:

### Win98/WinMe (VxD)

Layla24 1/2 Analog Playback  
Layla24 3/4 Analog Playback  
Layla24 5/6 Analog Playback  
Layla24 7/8 Analog Playback  
Layla24 S/PDIF Playback  
Layla24 1/2 ADAT Playback  
Layla24 3/4 ADAT Playback  
Layla24 5/6 ADAT Playback  
Layla24 7/8 ADAT Playback

### WinMe/2000/WinXP (WDM)

Layla24 1/2 Analog  
Layla24 3/4 Analog  
Layla24 5/6 Analog  
Layla24 7/8 Analog  
Layla24 1/2 Digital  
Layla24 3/4 Digital  
Layla24 5/6 Digital  
Layla24 7/8 Digital

And for recording:

### Win98/WinMe (VxD)

Layla24 1/2 Analog Record  
Layla24 3/4 Analog Record  
Layla24 5/6 Analog Record  
Layla24 7/8 Analog Record  
Layla24 S/PDIF Record  
Layla24 1/2 ADAT Record  
Layla24 3/4 ADAT Record  
Layla24 5/6 ADAT Record  
Layla24 7/8 ADAT Record

### Win2000/WinXP (WDM)

Layla24 1/2 Analog  
Layla24 3/4 Analog  
Layla24 5/6 Analog  
Layla24 7/8 Analog  
Layla24 1/2 Digital  
Layla24 3/4 Digital  
Layla24 5/6 Digital  
Layla24 7/8 Digital

Please refer to the documentation for your audio software to find out how to assign inputs and outputs.

If you are using the VxD drivers, you cannot use the S/PDIF and ADAT devices at the same time; please refer to the VxD console section of the manual to find out how to set the hardware to either S/PDIF or ADAT mode.

The WDM drivers work a little differently, depending on the current digital mode. Gina24, Layla24, and Mona have a **digital mode switch** that lets you set how the hardware transmits and receives digital audio signals. The digital mode switch can be set to S/PDIF RCA, S/PDIF optical, or ADAT optical.

The following is a table showing what the WDM wave devices do for each digital mode:

Device	S/PDIF RCA	S/PDIF optical	ADAT optical
Layla24 1/2 Digital	S/PDIF	S/PDIF	ADAT 1/2
Layla24 3/4 Digital	not used	not used	ADAT 3/4
Layla24 5/6 Digital	not used	not used	ADAT 5/6
Layla24 7/8 Digital	not used	not used	ADAT 7/8

In other words, if you want to play or record using S/PDIF, set the hardware to either S/PDIF RCA or S/PDIF optical and configure your software to use “Layla24 1/2 Digital.” Since S/PDIF is stereo (two channels), digital channels 3 – 8 are not used.

If you want to play or record using ADAT optical, set the hardware to ADAT optical mode and configure your software to use all of the digital channels.

Please refer to the Console section of the manual for more information on digital modes.

## Using Audio Software – Mia

To audio software, the Mia hardware appears as a group of stereo wave devices. You can view the list of these devices either from within your audio software or from the Windows control panel. The items you see in the list will depend on what driver you installed.

Win98/WinMe (VxD)

Mia 1/2 Virtual Out  
Mia 3/4 Virtual Out  
Mia 5/6 Virtual Out  
Mia 7/8 Virtual Out

WinMe/2000/WinXP (WDM)

Mia 1/2  
Mia 3/4  
Mia 5/6  
Mia 7/8

And for recording:

Win98/WinMe (VxD)

Mia 1/2 Analog Record  
Mia S/PDIF Record

Win2000/WinXP (WDM)

Mia 1/2  
Mia 3/4

You'll find these same input and output options available when using your audio record/editing software, with the devices assignable on a per-track basis. See your software owner's manual for details on how to make these device assignments.

If you are using the WDM drivers, the listed output devices correspond to Mia's virtual outputs; routing is done in the console. Also, "Mia 1/2" corresponds to Mia's analog inputs, and "Mia 3/4" corresponds to the S/PDIF inputs.

This completes the installation of the hardware and software.

## **Mia’s Virtual Outputs**

Most cards with a pair of analog inputs and outputs will appear to software as having just that, a single pair of inputs and outputs. When one application appropriates the card for its use, all other software is “locked-out” from using the card at the same time.

While there is rarely any need to record to multiple applications simultaneously, there are numerous instances where this is desirable for outputs. A good example would be playing back digital audio while at the same time using a software synthesizer for MIDI. Another example would be using multi-track editing software. Before multiple tracks can be played back on a stereo device, they must be mixed down within the software to the two output tracks. This takes up extra CPU cycles that might otherwise be dedicated to software effects or some other task.

Mia gets around this by combining the use of “virtual” outputs with its “multi-client” drivers. Mia appears to software as if it has eight separate outputs. These are mixed together with Mia’s on-board DSP to produce the actual or “physical” outputs that connect to external equipment without any CPU intervention. Any virtual output can be sent to any actual output along with any combination of other virtual outputs. The levels and routing of the virtual outputs is fully controllable with Mia’s Console, which is described later in this manual.

An example of this would be using Reality from Seer Systems with Cakewalk’s Pro Audio. Reality could be set up to play back through Mia’s outputs 7 and 8. Cakewalk’s Pro Audio, which combines MIDI sequencing with digital playback, could then play back its MIDI output through Reality while using Mia’s outputs 1 through 6 for six separate tracks of digital output.

This gives Mia most of the advantages of products with eight separate analog outputs, without the added expense of the extra converters or having to use an external mixer to mix the outputs down to a final pair of stereo outputs.

Virtual outputs are not available on Gina24, Layla24, or Mona.

## Console3 (WDM)

This section of the manual applies only if you installed the WDM drivers.

A “virtual control surface” application called Console3 is included with the WDM driver. The Console allows you to control the audio I/O and clocking functions of your Echo 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, select the digital mode, and adjust input monitoring.

If you’re curious, the name “Console3” means that this is Echo’s third-generation design for a software console.

### Running the Console

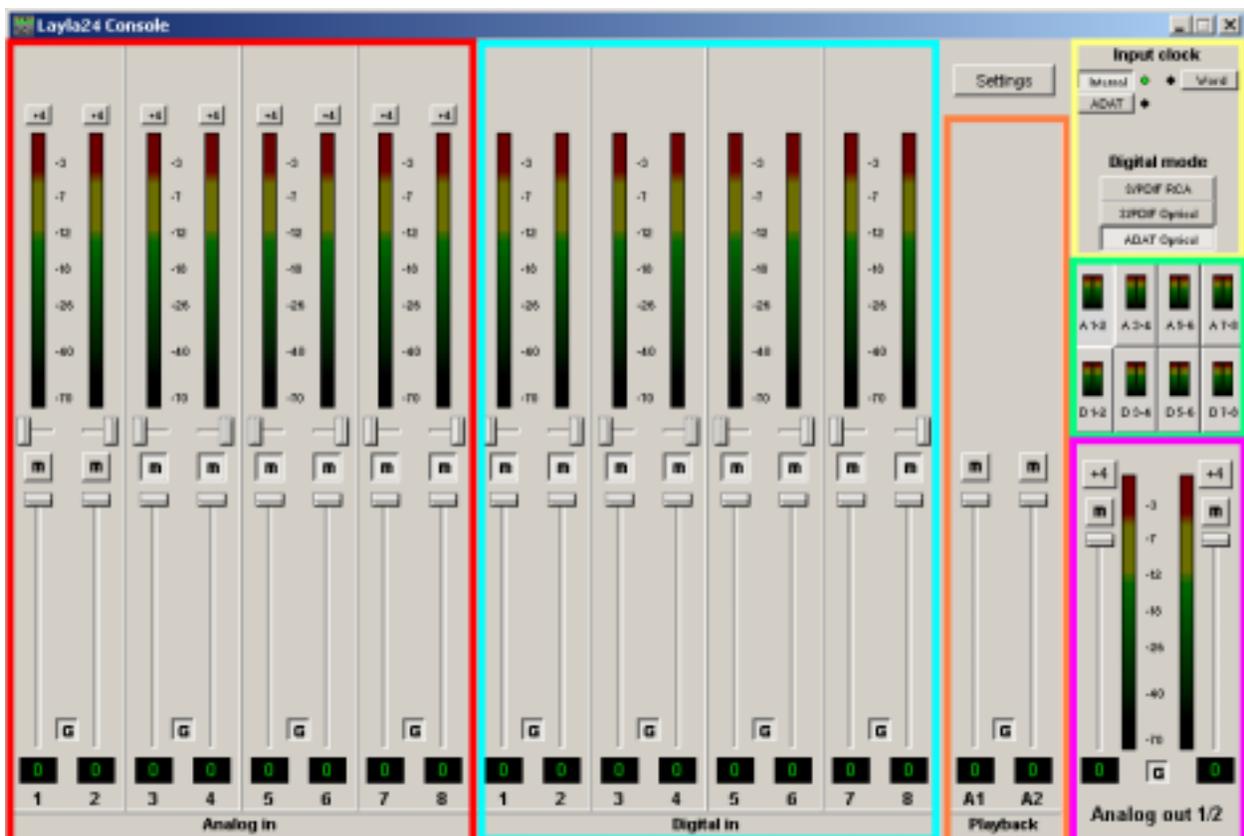
Once you have run the installer, you will find the console under your Start menu under the group “Echo Digital Audio.” Here, you have two options for running the console: “Show all Console3 windows” and “Start the tray icon.”

If you select “Start the tray icon,” a red “Echo” icon will be placed on your system tray. This is called the launcher. Double-clicking the launcher icon will display a console window for each Echo card in your system. If you only want to open a console window for one of your Echo cards, right-clicking the launcher icon will display a menu that lists the cards in your system. Select the card corresponding to the console you want to display.

If you select “Remove from tray,” the icon will vanish. All of your open console windows will close as well.

### 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 is a stereo pair of outputs on your 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 may be mixed to an output bus.



Screenshot of Layla24 Console3 main window - other Echo hardware will work similarly.

## Bus select

The bus select panel is the area outlined in green. Eight different output busses are shown here: four analog busses in the top row and four digital busses on the bottom row. Each bus has its own button marked with either an A or D (Analog or Digital) and a channel number pair. The button in the “down” position is the currently active bus. All monitor controls remain in effect even when not displayed by maintaining a level setting for each monitor path it controls. Clicking on an output bus button simply selects the settings that are displayed. In this case Analog 1-2 (A 1-2) is the active bus. This means all input monitor parameters such as volume, pan, and mute apply to the audio coming out Analog outputs 1 and 2 of Layla24. To select an output bus, click on one of the buttons. Alternatively, you can use the function keys F1 through F8.

## Master output bus control

The master bus control is the area outlined in purple. At the bottom is the label “Analog out 1-2.” This indicates you are currently controlling everything mixed to analog outputs 1 and 2. These channels correspond to the currently selected button on the bus select panel.

The two buttons labeled “+4” are the *nominal level* select buttons. Since they both read “+4,” this indicates these outputs are sending a +4 dBu (professional level) signal. To send a –10 dBV (consumer level) signal, click the button. This feature allows you to connect either professional or consumer gear to each output. Mona does not include “nominal level buttons” here. This is because the Mona’s RCA connectors only output a +4 dBu (professional level) signal, and the XLR connectors only output a –10 dBV (consumer level) signal.

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. Holding down the Ctrl key and clicking on a fader will set that fader to 0 dB, and clicking on the readout below the fader will allow you to enter a numerical value for the fader setting.

The *gang button*, marked with a “G,” sets the gang mode. If you are in gang mode, the mute buttons and nominal level buttons will work together; for example, clicking the mute on one channel activates the mute button on the other channel. Also, ganging the faders ties them together so they will maintain their relative placement with regard to each other. Holding down the Shift key 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 outlined in red and have the label “Analog in” at the bottom. There is one strip, or *input monitor*, corresponding to each of Layla24’s eight physical analog inputs. The controls affect the audio going to the currently selected output bus; in this case it is outputs Analog 1 and 2.

The topmost control is a button labeled “+4”; this button allows you to select the nominal level for this analog input. Since the button currently reads “+4,” it is ready to accept a +4 dB signal. Click the button to change this input to –10 mode. This button will affect the level of what you record as well as the monitor level. Mona does not include the “nominal level” buttons here either; this is because Mona’s input level is controlled on its audio interface box.

Below the “nominal level” button is the peak meter for this channel, labeled in decibels. Under the peak meter is the monitor pan slider; this pans the input channel between the two channels of the output bus. Holding down the Ctrl key and clicking on a pan slider will set the slider to the center position. Note that this does not affect your recording. Below the pan slider is the monitor mute. This controls whether or not you want this input channel to be mixed directly out to the currently selected output bus. Again, this does not affect recording. This is followed by the monitor fader, and it controls the input monitor level. Like the

pan and mute controls, this does not affect your record level. Also, clicking on the readout below the fader will allow you to numerically enter a fader setting.

The analog inputs also have gang buttons, and their operation is the same as the gang button for the master outputs.

## Digital Inputs

The controls for the digital inputs are outlined in light blue and have the label “digital in” at the bottom. Their operation is identical to the analog input controls except the digital input controls don’t include a nominal level button.

For Layla24, Gina24 and Mona, whether you are monitoring the S/PDIF or the ADAT inputs depends upon the digital mode you have selected. If you are in either S/PDIF mode, the S/PDIF inputs will come in digital inputs 1 and 2. If you are in ADAT mode, the console expands to display the eight ADAT digital inputs.

## Playback Strips

The controls for the level of audio being played by an application are outlined in orange and have the label “playback” at the bottom. Notice you cannot pan or otherwise redirect here; if you are playing out of analog 1 and 2 from your application, it will only be sent to that same bus. This differs with Mia; the following section explains the difference.

## Virtual Outputs

The controls for the level of audio being played by an application are outlined in orange and have the label “Virtual out” at the bottom. Most cards with a pair of analog inputs and outputs will appear to software as having just that, a single pair of inputs and outputs. When one application appropriates the card for its use, all other software is “locked-out” from using the card at the same time.

While there is rarely any need to record to multiple applications simultaneously, there are numerous instances where this is desirable for outputs. A good example would be playing back digital audio while at the same time using a software synthesizer for MIDI. Another example would be using multi-track editing software. Before multiple tracks can be played back on a stereo device, they must be mixed down within the software to the two output tracks. This takes up extra CPU cycles that might otherwise be dedicated to software effects or some other task.

Mia gets around this by combining the use of “virtual” outputs with its “multi-client” drivers. Mia appears to software as if it has eight separate outputs. These are mixed together with Mia’s on-board DSP to produce the actual or “physical” outputs that connect to external equipment without any CPU intervention. Any virtual output can be sent to any actual output along with any combination of other virtual outputs. The levels and routing of the virtual outputs is fully controllable with Mia’s Console.

An example of this would be using Reality from Seer Systems with Cakewalk’s Pro Audio. Reality could be set up to play back through Mia’s outputs 7 and 8. Cakewalk’s Pro Audio, which combines MIDI sequencing with digital playback, could then play back its MIDI output through Reality while using Mia’s outputs 1 through 6 for six separate tracks of digital output.

This gives Mia most of the advantages of products with eight separate analog outputs, without the added expense of the extra converters or having to use an external mixer to mix the outputs down to a final pair of stereo outputs.

## 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 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.

## Clock and digital mode settings

The area within the yellow outline on the upper right lets you select your sync source and digital mode. If a given clock is detected, the indicator next to the clock button will turn green. You will only be allowed to select clocks that have been detected. Also, you may only select the digital mode if there aren’t any applications actively playing or recording.

**Note:** Sample rates above 48kHz will not be available while ADAT mode is selected. This is because ADAT does not support sample rates higher than 48kHz.

## Synchronizing Multiple Devices

Layla24, Mona, Gina24 and Mia are designed to work alongside other audio equipment. If you are planning on using your card with other audio equipment, please note the following:

The Windows drivers included in this package support multiple Echo cards within the same system. In addition, your new Echo hardware will operate alongside other Echo products. Echo hardware can also 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) in your system 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 or that will be affected by clock drift.

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

**Word Clock** – This is a synchronization signal that connects to the BNC connector labeled **Word Clock** on Layla24/Mona's back panel. This synchronization clock runs at the 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. Layla24/Mona 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, Layla24/Mona can only sync to Word 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.

**S/PDIF** – 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, whether via optical cables or RCA cables, Echo products will utilize the synchronization clock that is embedded in the S/PDIF while it decodes the bitstream. Your Echo digital audio system can only sync to S/PDIF clock if the master device is set to a sample rate between 30kHz and 50kHz. 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.

**ADAT** – (Layla24/Mona/Gina24) The Alesis ADAT optical interface can send and receive up to 8 simultaneous channels of digital audio data over fiber optic

cables. Unlike wire cables, fiber optic cables do not add noise or interference. Layla24/Mona/Gina24 can send out (or “master”) ADAT clock as well slave to it. If you want the interface to slave to an ADAT clock, just set the input clock to the **ADAT** setting. Layla24/Mona/Gina24 can only sync to ADAT clock if the master device is set to a sample rate between 40kHz and 50kHz. Otherwise you will get noise and/or loss of sync.

**Note:** To preserve the high audio quality of Layla24/Mona/Gina24’s converters it is best to use the **Internal** input clock setting instead of synchronizing to an external ADAT clock, which may add unwanted jitter to the signal.

**Esync** – Esync is a proprietary form of super clock that is used to synchronize two or more Echo 24-bit products like Layla24, Mona, Gina24, and Darla24. While Gina24 and Darla24 have both input and output connectors on their PCI cards, Layla24 and Mona only have the Esync output connectors, and Mia has no Esync connectors. Layla24/Mona/Gina24 is always outputting Esync 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 simple. Suppose that Layla24 is the only audio device used in your system. Since you have no other devices to synchronize with, simply select **Internal** for Layla24's input clock. Layla24 will then use its own clock to control its operation.

Now a little more complicated set-up: You have two Layla24s connected. Simply set Layla24 #1 to **Internal** for its input clock. Now connect the first Layla24 to the second one via a BNC cable running from Word Clock Out on Layla24 #1 to Word Clock In on Layla24 #2. Now select **Word** for Layla24 #2's input sync. The second Layla24 will slave to the first one and your Layla24s will now operate in unison.

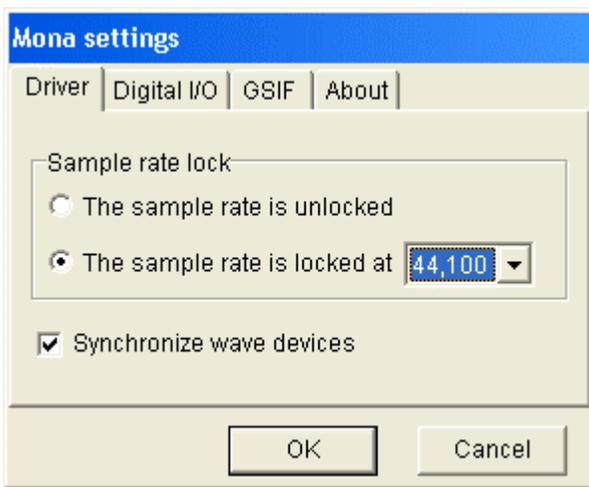
No matter how many devices you are synchronizing, the concept is essentially the same. You are merely "daisy-chaining" devices together using compatible clocks. One device will operate as the source of the master clock, with each successive device using that clock to sync.

## Console Settings

Clicking the Settings button above the playback strips shows a dialog box for altering console settings. The Settings dialog gives you four different tabs to choose from: “Driver,” “Digital I/O,” “GSIF,” and “About.”

### Driver

Here you can do two things: lock the sample rate and tell the driver whether or not to synchronize wave devices.



*This is the settings window for Mona. This window is the same for all Echo24 products.*

### Locking the sample rate

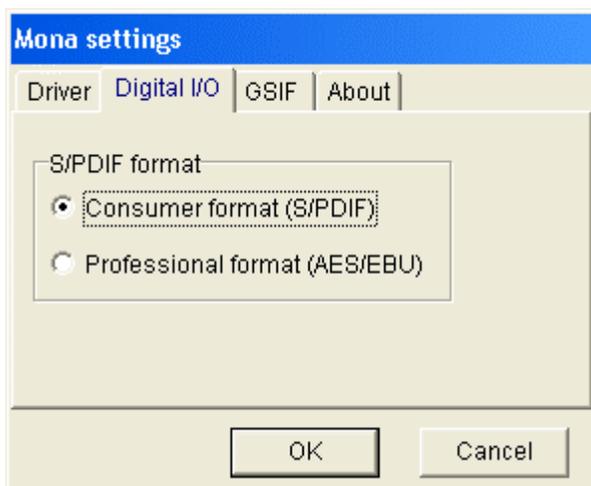
Locking the sample rate can be very important, since Windows will frequently try to set the hardware to the highest rate it supports. Then, Windows sample rate converts between that rate and the rate at which you are playing or recording. For instance, you may be trying to play at 44.1 kHz, but Windows sets the hardware to 96 kHz and converts between the two. This is undesirable because you may be trying to do a digital transfer at a specific rate. In addition, sample rate conversion will reduce your audio quality and consume a great deal of CPU power. Locking the sample rate forces Windows to only use the hardware at the locked rate. In general, we recommend you set the sample rate first in your audio application. Then, lock that same rate within the console. Locking the sample rate is not really necessary for applications that do not use the legacy wave/MME interface (such as SONAR, Cubase, etc.).

## Synchronize wave devices

This is mainly for developers who are using tools like Visual Basic or Delphi and want the inputs and outputs to be totally independent. Unless you have a good reason to turn it off, you should leave this switch on. Also, if you are running multiple applications on one card and you can't get it to work, try un-checking this box.

## Digital I/O

This lets you set the format of your S/PDIF outputs.



Your Echo card 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.

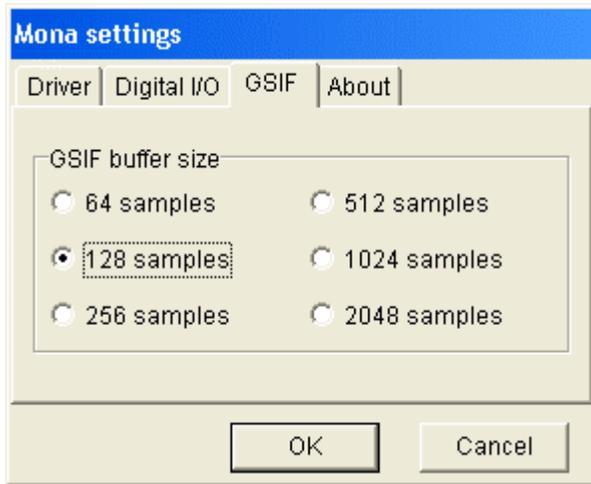
The S/PDIF output defaults to the Professional format. If you are recording from your Echo hardware into a consumer DAT deck, CD recorder, or Mini Disc recorder, the device may not be able to recognize the signal until you switch the output to Consumer.

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.

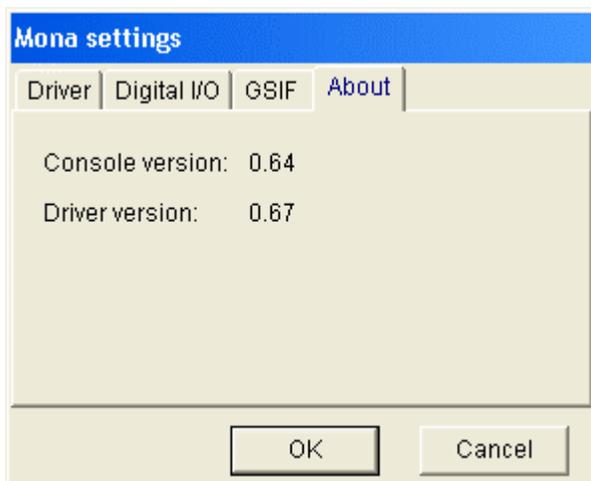
## GSIF

This window allows you to set the latency of GigaStudio in terms of samples. Select the value here that provides the best tradeoff between performance and reliability.



## About

This displays the console version and driver version numbers. We feel that this is the single most exciting feature on the new console.



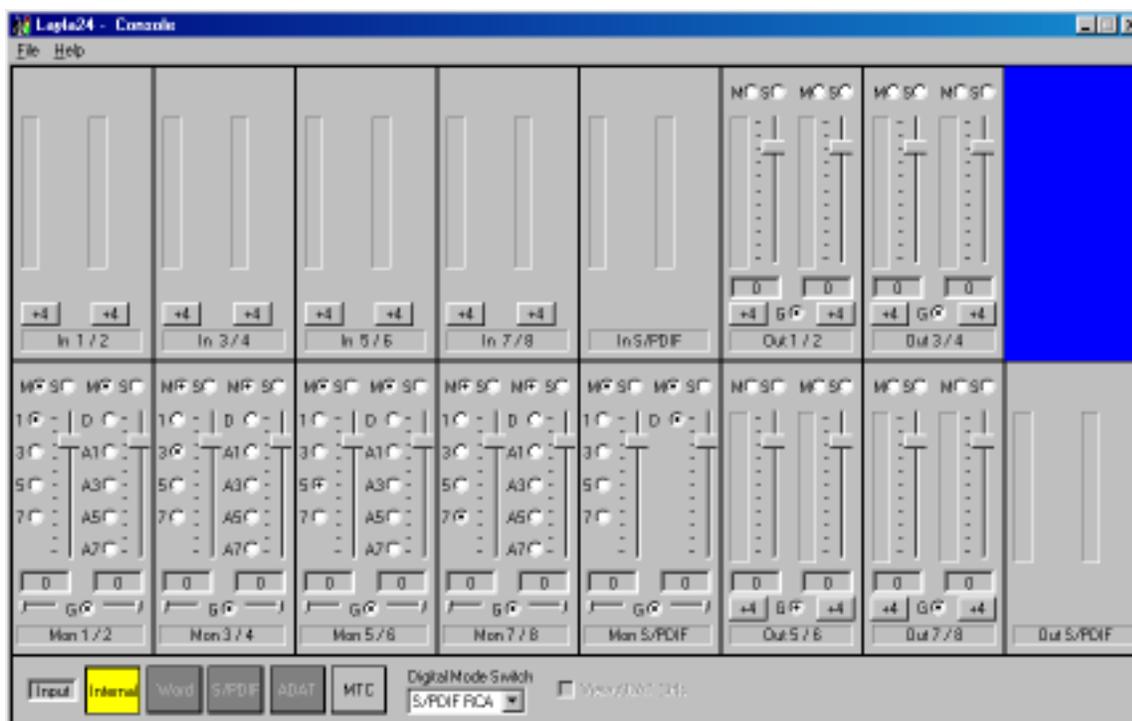
## VxD Console – Gina24, Layla24, and Mona

This section of the manual only applies if you installed the VxD drivers and are using Gina24, Layla24, or Mona.

A “virtual control surface” application called the Console is included with the VxD driver. Every time you start Windows you will see it flicker across the screen as it loads. Then it will minimize itself to the taskbar. The Console allows you to control the audio I/O and clocking functions of your Echo product, and it brings these controls to a single easy-to-use location. From the console you can mix the virtual outputs, control the output levels, select clocks, and adjust input monitoring.

**Note:** While this portion of the manual primarily refers to Layla24, the concepts will be similar for Gina24 and Mona. Mia’s console will be explained in the following section.

The Console software was automatically installed at the same time that you installed the VxD driver. If installation was successful, you should see a red capital letter **L**, **M**, **G** for Layla24, Mona, and Gina. The letter/s will appear in your Windows taskbar (typically in the bottom right-hand corner of your screen). Double-clicking on the letter will activate the console program. It should look like this:

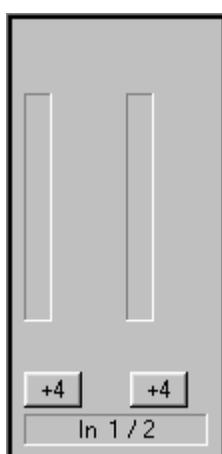


The console functions are grouped into three areas: inputs (in the upper left corner), monitors (directly below the inputs that are being monitored), and outputs (on the right hand side). The controls for a particular function/stereo pair are then

further grouped into a box that contains selection buttons, faders, and other controls and displays as determined by the function.

## Console Controls

Let's take a look at the control surface. The input control area is located in the upper left portion of the console surface. For each input device pair you will find a pair of input meters. Each input also has a nominal level switch that switches between professional (+4 dBu) and consumer (–10 dBV) levels. This way you can connect either professional or consumer gear to each input. Make sure to set the nominal level switches to match the equipment that you have connected to Layla24's inputs. Note that these switches are not present in Mona's console because the input level is adjusted on the audio interface box.

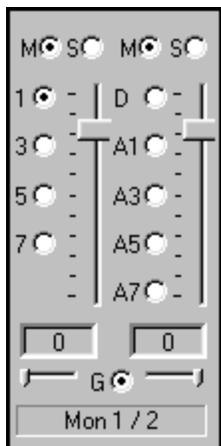


## Monitor Controls

Below the input channels are the input monitor controls. The input monitor controls allow you to monitor the input signal via any of the available outputs on your Echo hardware. Each input channel pair has a corresponding monitor control pair directly below it on the console. At the top of each monitor pair are radio buttons that allow you to selectively mute (M) or solo (S) each channel. In the middle you will find a pair of faders that will allow you to set the levels of the monitor signals. At the bottom of the control area there is a radio button that allows you to “gang” (G) the faders. Ganging the faders ties them together so that they will maintain their relative placement with regard to each other. Below each fader is its corresponding pan control. In the default pan positions (see example below) the signal from input 1(left) will be monitored by output 1(left), and the signal from input 2(right) will be monitored by output 2(right). Moving a pan control will determine how much of the input signal is sent to each output channel

in the selected output pair. Both pan controls and faders can be reset to default by pressing the **Ctrl** button and left clicking on the control.

Instead of level meters, each monitor control has a series of numbered buttons. These buttons allow you to select which output channel pair controls are displayed, so you can adjust them.



The monitors are one of the most powerful functions of the console. When you are recording, these controls allow you to set the level at which each input signal will be monitored at each of the outputs, independent of the main output level control.

When setting a monitor level, you must first select the input that will be monitored, and then select the output for which you will set the monitor level. Selection of the input is really just a matter of using the correct monitor control box. A monitor setting will always affect the input pair that is directly above it. To set the monitor for inputs 1/2, for example, simply go to the monitor control directly below inputs 1/2 and click on one of the buttons found in the center of the control.

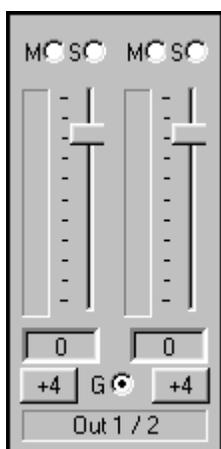
Let's say you want to set the level at which input pair 1/2 is monitored for each of the four stereo analog output pairs on Layla24. You would first find the monitor control that corresponds to inputs 1/2 (the control on the far left, directly below the input control for channels 1/2). Clicking on the button within this box labeled "1" causes the control box to display the currently selected levels at which inputs 1/2 are being monitored at outputs 1/2. Clicking on the button labeled "3" would cause the control box to display the currently selected levels at which inputs 1/2 were being monitored at outputs 3/4. You can move through the remaining buttons, setting the level at which inputs 1/2 are monitored at each of the four output pairs, as well as the S/PDIF "D" and ADAT "A#" outputs. The console program remembers any settings that you make, and all settings for all outputs are maintained independently.

**Note:** You cannot monitor the S/PDIF inputs through the ADAT outputs and vice versa (Layla24, Mona, Gina24). This is because you can only use one digital mode (S/PDIF or ADAT) at any given time. For more information, please see the section “**Digital Mode Switch**” later in this manual.

Remember that all of the monitor controls remain in effect even when they’re not displayed. The degree of attenuation (or muting) of each level is set by the monitor controls. The console program constantly maintains a level setting for each of the monitor paths it controls. Clicking on an output selection button simply selects the settings that are displayed.

## Output Controls

The right hand side of the console is dedicated to the controls for analog output channels one through eight (an output meter pair is included for the two S/PDIF output channels, but these levels are not adjustable). The output controls mirror those found on the monitor controls (Gang, Mute, Solo, & faders) with a few exceptions. Instead of pan controls you have nominal level switches that switch between professional (+4 dBu) and consumer (–10 dBV) levels. This allows you to connect either professional or consumer gear to each output. You will also find level meters in the middle area instead of the numbered buttons. Just like the monitor controls, the output faders can be reset to default by pressing the **Ctrl** button and left clicking on the fader.

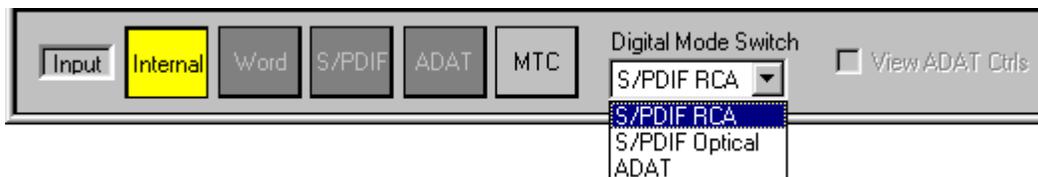


## Adjusting Monitor and Playback Levels

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 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.

## Setting Clock Sources and Destinations

At the very bottom of the Console there are buttons that allow you to select the synchronization clocks. The console program will detect which input clocking options are available, and automatically disable those that are unavailable. Depending on what external devices you have connected, you may have as many as four out of five options available (Layla24/Mona).



*Input clocks, Digital Mode Switch & View ADAT Ctrl's are at the bottom of the Console.*

If you are unsure which clock synchronization selections are appropriate for your installation, be sure to read the next section, which deals with the types of supported clocks.

## Synchronizing Multiple Devices

Layla24/Mona/Gina24/Mia is designed to work alongside other audio equipment. If you are planning on using your Echo hardware with other audio equipment, please note the following:

The Windows drivers included in this package support multiple Echo cards within the same system. In addition, your new Echo hardware will operate alongside other Echo products. Echo hardware can also 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) in your system 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 or that will be affected by clock drift.

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

**Word Clock** – This is a synchronization signal that connects to the BNC connector labeled **Word Clock** on Layla24/Mona's back panel. This synchronization clock runs at the 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. Layla24/Mona 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, Layla24/Mona can only sync to Word 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.

**S/PDIF** – 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, whether via optical cables or RCA cables, Echo products will utilize the synchronization clock that is embedded in the S/PDIF while it decodes the bitstream. Your Echo digital audio system can only sync to S/PDIF clock if the master device is set to a sample rate between 30kHz and 50kHz. 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.

**ADAT** – (Layla24/Mona/Gina24) The Alesis ADAT optical interface can send and receive up to 8 simultaneous channels of digital audio data over fiber optic cables. Unlike wire cables, fiber optic cables do not add noise or interference. Layla24/Mona/Gina24 can send out (or “master”) ADAT clock as well slave to it. If you want the interface to slave to an ADAT clock, just set the input clock to the **ADAT** setting. Layla24/Mona/Gina24 can only sync to ADAT clock if the master device is set to a sample rate between 40kHz and 50kHz. Otherwise you will get noise and/or loss of sync.

**Note:** To preserve the high audio quality of Layla24/Mona/Gina24's converters it is best to use the **Internal** input clock setting instead of synchronizing to an external ADAT clock, which may add unwanted jitter to the signal.

**Esync** – Esync is a proprietary form of super clock that is used to synchronize two or more Echo 24-bit products like Layla24, Mona, Gina24, and Darla24. While Gina24 and Darla24 have both input and output connectors on their PCI cards,

Layla24 and Mona only have the Esync output connectors, and Mia has no Esync connectors. Layla24/Mona/Gina24 is always outputting Esync clock. Now, let's take a look at some example configurations and how you might set them up from a synchronization standpoint.

Let's start simple. Suppose that Layla24 is the only audio device used in your system. Since you have no other devices to synchronize with, simply select **Internal** for Layla24's input clock. Layla24 will then use its own clock to control its operation.

Now a little more complicated set-up: You have two Layla24s connected. Simply set Layla24 #1 to **Internal** for its input clock. Now connect the first Layla24 to the second one via a BNC cable running from Word Clock Out on Layla24 #1 to Word Clock In on Layla24 #2. Now select **Word** for Layla24 #2's input sync. The second Layla24 will slave to the first one and your Layla24s 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.

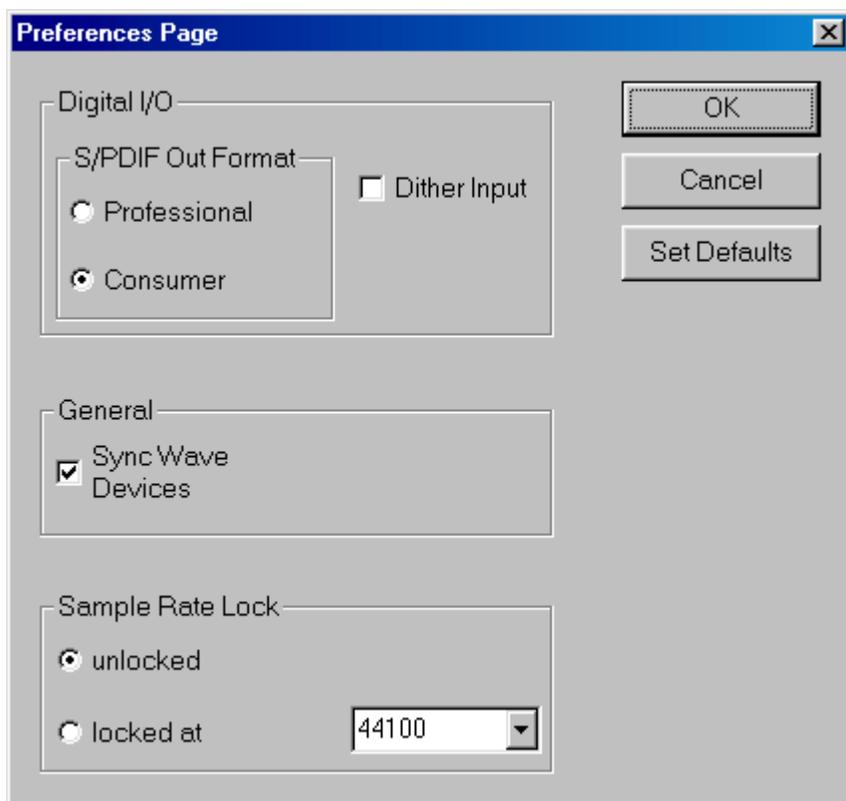
## The File Menu

At the top left of the Console window you will find the **File** menu. By selecting the **File** menu, several configuration options become available to you. The first menu option is “**Preferences.**” The **Preferences** option brings up the Console “**Preferences Page**” window. We’ll get to that window shortly, but first let’s go on to the rest of the **File** menu options.

The second option under the **File** menu is “**Hide**” which closes the Console window. You can reopen the Console by clicking on the appropriate colored letter in the bottom left of the Windows taskbar.

The third option under the **File** menu is “**Exit**” which closes the current instance of the Console. This will not only close the Console window, but also shut down the Console application and remove it from the system tray. Don’t worry, if you want to bring the Console back, you can do it by disabling, then re-enabling the **Show console on taskbar** checkbox in the **Mia by Echo Properties** window (see below under “Additional Configuration Settings”). It will also come back when you restart Windows.

## The Preferences Page



The Preferences Page is accessible through the Console File menu.

### Digital I/O - Selecting the S/PDIF Output Format

Your Echo hardware 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 **Professional** format. If you are recording from your Echo hardware into a consumer DAT deck, the deck may not be able to recognize the signal until you switch Mia’s output to **Consumer**.

We have provided a software switch in the driver that allows you to select which format the Echo hardware transmits. In the window that appears you’ll see a pair of check boxes in an area labeled **S/PDIF Out Format**: one box is labeled **Consumer** and the other **Professional**. Select the appropriate format for your DAT (if you don’t know which one to use and are having difficulties, simply try the one that is not currently checked). No Echo hardware transmits the SCMS bit, regardless of which format is selected.

## Digital I/O - Dither Input

In the top right is a checkbox labeled **Dither Input**. This checkbox allows you to enable or disable dithering on the digital inputs. It is off by default. Most of the time you will want to leave this setting disabled. The only time you will want to use dithering is when you are receiving a 24 bit signal over the digital inputs, but you want to record it as a 16 bit wave file. In this case, dithering will improve the audio quality.

## General - Sync Wave Devices

This is mainly for developers who are using tools like Visual Basic or Delphi and want the inputs and outputs to be totally independent. Unless you have a good reason to turn it off, you should leave this switch on.

If you have a program that is telling you that our hardware isn't full duplex, chances are you can fix this by either disabling this switch or enabling **Multi-client audio** (see below).

## Sample Rate Lock

These controls allow you to enable or disable the **Sample Rate Lock** feature, as well as set the sample rate you want to lock to. While the sample rate is locked, all sample rate change requests from applications will be ignored and the hardware will remain at that sample rate no matter what.

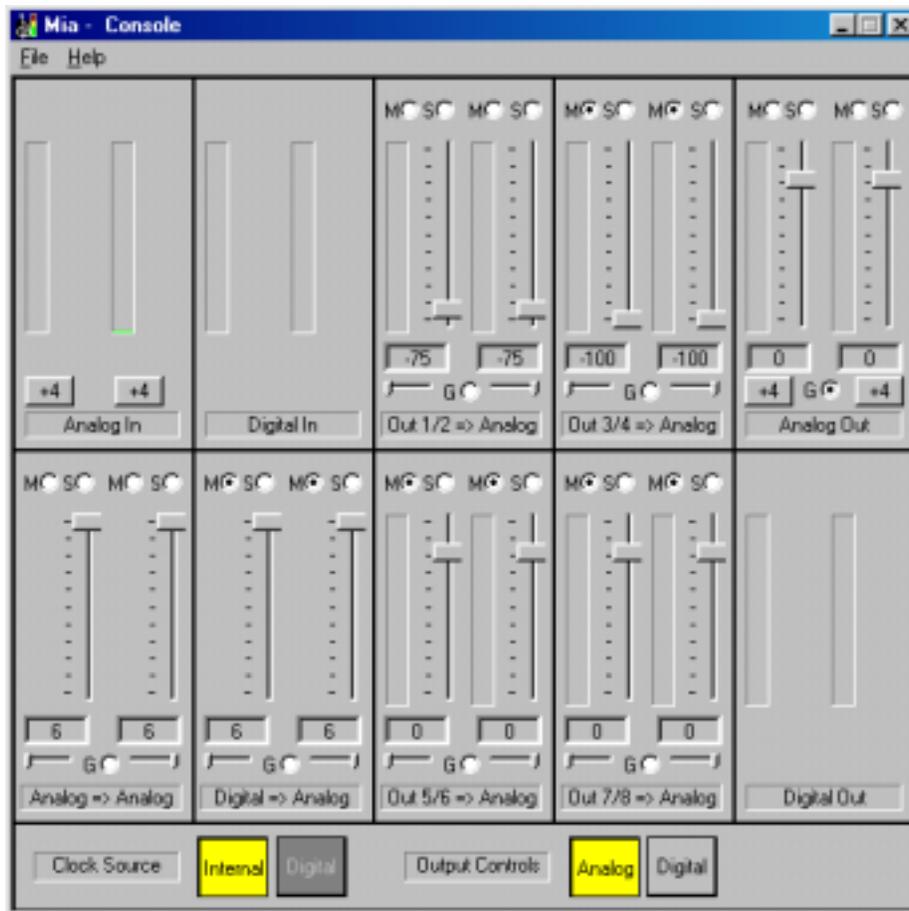
This feature is most useful in **Multi-client audio** mode (see below). Suppose you have locked the sample rate to 44100. If you are recording a project at 44100 and another sound tries to play at, say, 22050, the sample rate won't change. Sample rate locking only applies while your Echo hardware is set to Internal clock. This setting defaults to **unlocked**.

## VxD Console - Mia

This section only applies if you installed the VxD drivers and are using Mia.

A “virtual control surface” application called the Console is included with the VxD driver. Every time you start Windows you will see it flicker across the screen as it loads. Then it will minimize itself to the taskbar. The Console allows you to control the audio I/O and clocking functions of Mia, and it brings these controls to a single easy-to-use location. From the console you can mix the virtual outputs, control the output levels, select clocks, and adjust input monitoring.

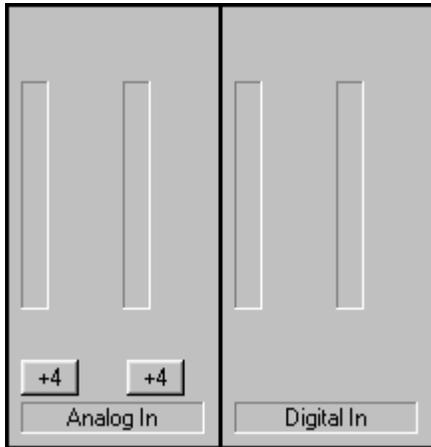
The Console software was automatically installed at the same time that you installed the Mia driver for Windows 98/Me. If installation was successful, you should see a blue capital letter **M** in your Windows 98/Me taskbar (typically in the bottom right-hand corner of your screen). Double-clicking on the “**M**” will activate the console program. It should look like this:



The console functions are grouped into three areas: inputs and monitors (on the left), virtual output mixing controls (in the middle), and outputs (on the right hand side). The controls for a particular function/stereo pair are then further grouped

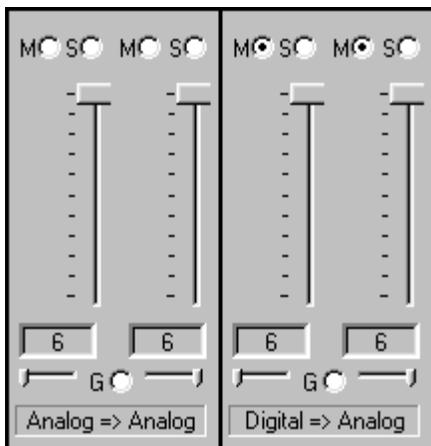
into a box that contains selection buttons, faders, and other controls and displays as determined by the function.

## Input Controls



Let's take a look at the control surface. The input control area is located in the upper left portion of the console surface. For each input device pair you will find a pair of input meters. In addition, the analog inputs have controls that allow you to select between +4 dBu (professional) input levels or -10 dBV (consumer) input levels. Clicking on the button will toggle back and forth between the two levels.

## Monitor Controls



Below the input channels are the input monitor controls. These allow you to monitor the input signal via any of the outputs on your Mia. Each input channel pair has a corresponding monitor control pair directly below it on the console. At the top of each monitor pair are radio buttons that allow you to selectively mute (M) or solo (S) each channel. In the middle you will find a pair of faders that will allow you to set the levels of the monitor signals. At the bottom of the control area there is a radio button that allows you to “gang” (G) the

faders. Ganging the faders ties them together so that they will maintain their relative placement with regard to each other. Below each fader is its corresponding pan control. In the default pan positions (see example below) the signal from input 1(left) will be monitored by output 1(left), and the signal from input 2(right) will be monitored by output 2(right). Moving a pan control will determine how much of the input signal is sent to each output channel in the selected output pair. Both pan controls and faders can be reset to default by pressing the **Ctrl** button and left clicking on the control.

Notice that the monitor controls are labeled at bottom indicating which input is being monitored and which output they are going to. In the pictures above, controls for the analog outputs are shown. There is a duplicate set of controls for

monitoring through the digital outputs that are hidden from view. Clicking on the large “Output Controls” button labeled “Digital” at the bottom of the console will display the monitor controls for monitoring through the digital outputs. These controls are labeled “Analog => Digital” and “Digital => Digital”.

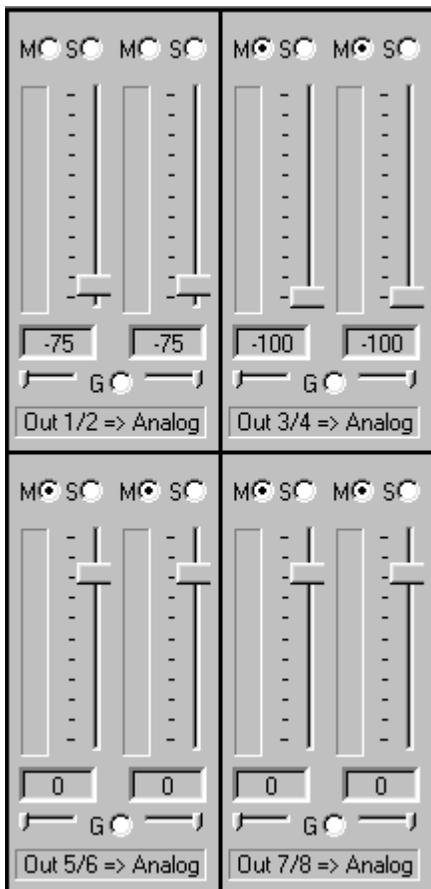
The monitors are one of the most powerful functions of the console. When you are recording, these controls allow you to set the level at which each input signal will be monitored at each of the outputs, independent of the main output level control.

When setting a monitor level, you must first select whether you are adjusting the monitors for the analog or digital outputs by clicking on the appropriate “Output Controls” button on the bottom of the console. Then adjust the input that will be monitored. Selection of the input is really just a matter of using the correct monitor control box. A monitor setting will always affect the input pair that is directly above it. To set the monitor for the analog inputs, for example, simply go to the monitor control directly below the analog inputs and click on one of the buttons found in the center of the control.

Let’s say you want to set the level at which the analog inputs are monitored through the S/PDIF outputs on Mia. You would first click on the “Digital” output control button at the bottom of the console to select the digital controls. Then you would adjust the monitoring levels by using the sliders and buttons within the monitor control labeled “Analog => Digital). The console program remembers any settings that you make, and all settings for all outputs are maintained independently.

Remember that all of the monitor controls remain in effect even when they’re not displayed. The degree of attenuation (or muting) of each level is set by the monitor controls. The console program constantly maintains a level setting for each of the monitor paths it controls. Clicking on either the “Analog” or “Digital” output control button simply selects the settings that are displayed.

## Virtual Output Controls

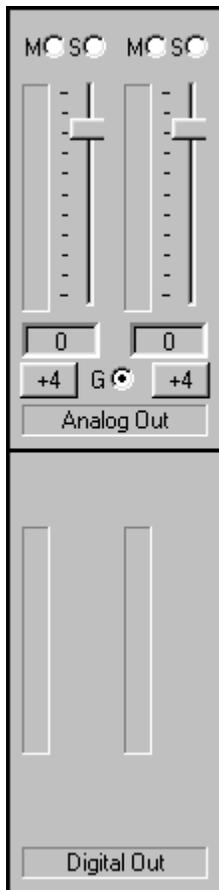


The middle of the console is dedicated to the controls for virtual output channels one through eight. The output controls look like the monitor controls (Gang, Mute, Solo, & faders), except that there are also meters that show the levels for each of the virtual outputs. Just like the monitor controls, the output faders can be reset to default by pressing the **Ctrl** button and left clicking on the fader.

The controls for mixing virtual outputs 1/2 and 3/4 to the analog outputs are shown above. Just like the monitor controls, there are duplicate controls for mixing to the digital outputs that are not shown. To view these controls, select the “Digital” output controls button at the bottom of the console.

All virtual output sliders default to 0 dB when the console is first installed. Since multiple outputs are being added together, you may need to lower these levels to avoid clipping at the analog or digital output. Also, when the console is first installed virtual outputs 1/2 are routed to the analog outputs and virtual outputs 3/4 are routed to the digital outputs.

## Output Controls



The right hand side of the console is dedicated to the controls for the “physical” analog and digital outputs. An output meter pair is included for the two S/PDIF output channels, but these levels are not adjustable. The analog output controls are similar to those found on the virtual output controls (Gang, Mute, Solo, & faders), except that there are no pan controls and in their place there are buttons that allow you to select between +4 dBu (professional) input levels or –10dBV (consumer) input levels. Clicking on the button will toggle back and forth between the two levels. Just like the monitor controls, the output faders can be reset to default by pressing the **Ctrl** button and left clicking on the fader.

## Adjusting Monitor and Playback Levels

Mia’s output volume adjustments are made in the digital domain. When you lower a volume slider, you are actually decreasing the number of available bits, thereby taking away from the potential dynamic range of the system. To avoid this, we suggest that whenever possible you leave the output sliders set to their maximum positions, and perform any necessary attenuation on your external mixer. When the Mia playback volume controls are set to maximum and the input signal approaches the maximum pre-clipping level, you can achieve the full 24-bit dynamic range of the system.

## Setting Clock Sources and Output Controls

At the very bottom of the Console there are buttons that allow you to select the clock that is used by Mia. You may choose between Mia’s own internal or “on-board” clock or you may synchronize to an external digital input. The console program will detect whether a digital input clock is available, and automatically disable the “Digital” Clock Source button if a digital clock is unavailable.

There are also buttons for selecting controls that affect either the analog outputs or the digital outputs. When “Analog” is selected, the monitors and virtual output controls for the analog outputs are displayed within the Console. When “Digital” is selected, the controls for the digital outputs are shown.



*Clock Source and Output Control buttons are at the bottom of the Console.*

## Synchronizing Multiple Devices

Mia is designed to work alongside other audio equipment. If you are planning on using Mia with other audio equipment, please note the following:

The Mia Windows drivers included in this package support multiple Mias within the same system. In addition, your new Mia will operate alongside other Echo products. Mia can also 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) in your system must support synchronization with Mia’s digital input or outputs. 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 or that will be affected by clock drift.

Mia can slave to or generate synchronization signals with its digital (S/PDIF) interface. The S/PDIF (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 Mia will utilize the synchronization clock that is embedded in the S/PDIF while it decodes the bit stream.

**Note:** When recording from a S/PDIF port, you must select **Digital** 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 **Digital** as your input clock.

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

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

Now a little more complicated set-up: You have two Mias connected. Simply set Mia #1 to "Internal" for its input clock. Now connect the first Mia to the second one via an S/PDIF cable running from DOUT on Mia #1 to DIN on Mia #2. Now select Digital for Mia #2's clock source. Your Mias 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.

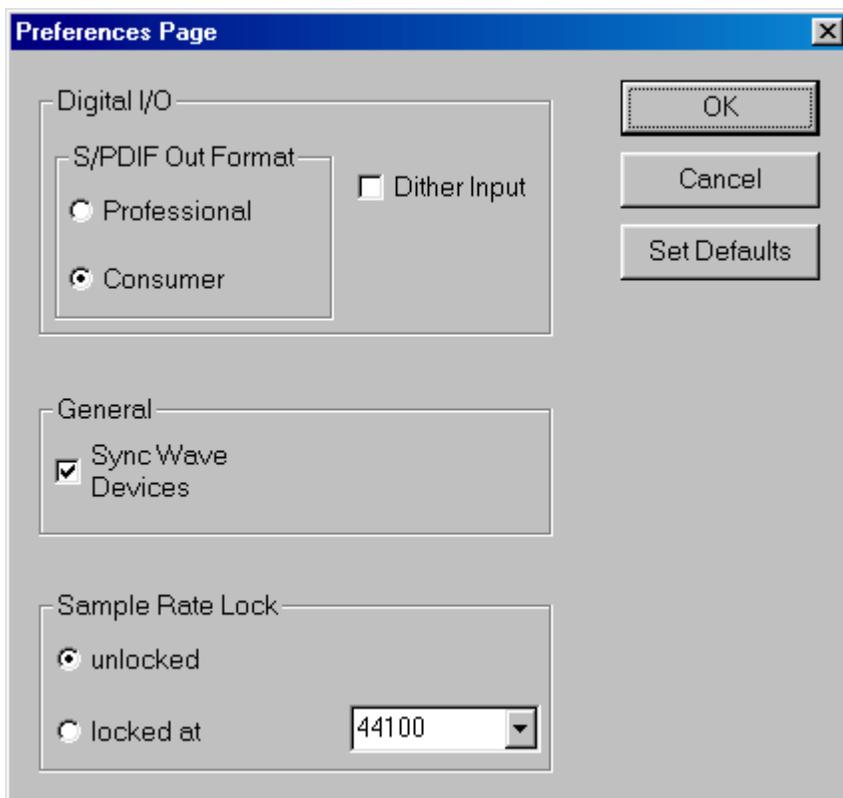
## The File Menu

At the top left of the Console window you will find the **File** menu. By selecting the **File** menu, several configuration options become available to you. The first menu option is “**Preferences.**” The **Preferences** option brings up the Console “**Preferences Page**” window. We’ll get to that window shortly, but first let’s go on to the rest of the **File** menu options.

The second option under the **File** menu is “**Hide**” which closes the Console window. You can reopen the Console by clicking on the blue “**M**” in the bottom left of the Windows taskbar.

The third option under the **File** menu is “**Exit**” which closes the current instance of the Console. This will not only close the Console window, but also shut down the Console application and remove it from the system tray. Don’t worry, if you want to bring the Console back, you can do it by disabling, then re-enabling the **Show console on taskbar** checkbox in the **Mia by Echo Properties** window (see below under “Additional Configuration Settings”). It will also come back when you restart Windows.

## The Preferences Page



*The Preferences Page is accessible through the Console File menu.*

## Digital I/O - Selecting the S/PDIF Output Format

Mia 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.

Mia’s S/PDIF output defaults to the **Professional** format. If you are recording from Mia into a consumer DAT deck, the deck may not be able to recognize the signal until you switch Mia’s output to **Consumer**.

We have provided a software switch in the Mia driver that allows you to select which format Mia transmits. In the window that appears you’ll see a pair of check boxes in an area labeled **S/PDIF Out Format**: one box is labeled **Consumer** and the other **Professional**. Select the appropriate format for your DAT (if you don’t know which one to use and are having difficulties, simply try the one that is not currently checked). Mia never transmits the SCMS bit, regardless of which format is selected.

## Digital I/O - Dither Input

In the top right is a checkbox labeled **Dither Input**. This checkbox allows you to enable or disable dithering on the digital inputs. It is off by default. Most of the time you will want to leave this setting disabled. The only time you will want to use dithering is when you are receiving a 24 bit signal over the digital inputs, but you want to record it as a 16 bit wave file. In this case, dithering will improve the audio quality.

## General - Sync Wave Devices

This is mainly for developers who are using tools like Visual Basic or Delphi and want the inputs and outputs to be totally independent. Unless you have a good reason to turn it off, you should leave this switch on.

If you have a program that is telling you that our hardware isn't full duplex, chances are you can fix this by either disabling this switch or enabling **Multi-client audio** (see below).

## Sample Rate Lock

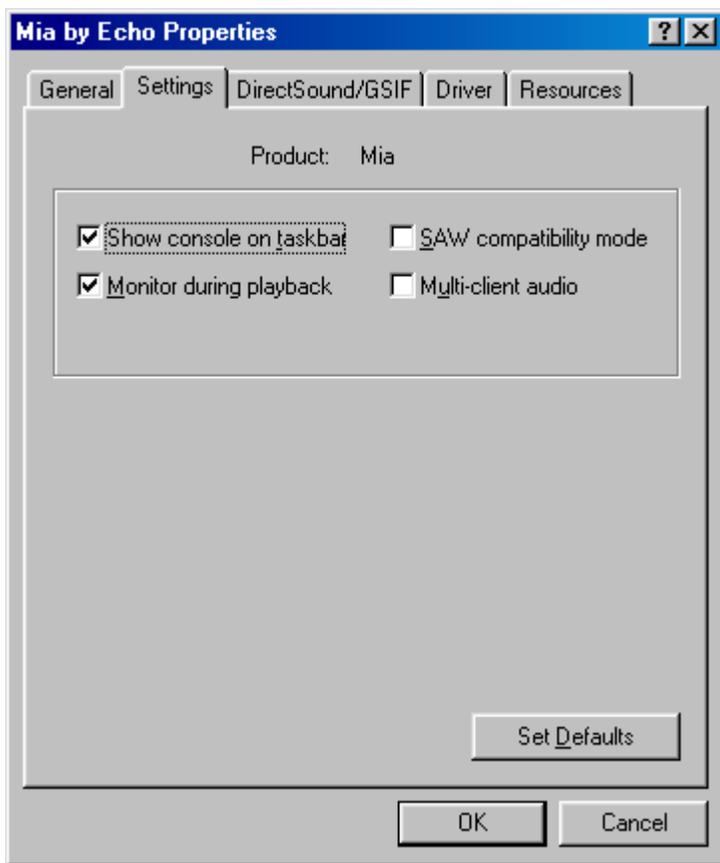
These controls allow you to enable or disable the **Sample Rate Lock** feature, as well as set the sample rate you want to lock to. While the sample rate is locked, all sample rate change requests from applications will be ignored and the hardware will remain at that sample rate no matter what.

This feature is most useful in **Multi-client audio** mode (see below). Suppose you have locked the sample rate to 44100. If you are recording a project at 44100 and another sound tries to play at, say, 22050, the sample rate won't change. Sample rate locking only applies while Mia is set to Internal clock. This setting defaults to **unlocked**.

## Additional Configuration Settings (VxD Drivers)

In addition to those settings that can be made using the Console program, there are a number of other configuration options available only to users of Windows 98/Me. These are found in the Windows Control Panel. To access these controls, begin by pressing the Windows **Start** button. Select **Settings**, then **Control Panel**. In the Control Panel window, locate the icon for **System**, and double-click on it. Click on the tab titled **Device Manager**. Now locate the line titled **Sound, video and game controllers**. Finally, locate the line titled **Layla24/Mona/Gina24/Mia by Echo**, and double-click. This will bring up the **Echo Properties** window, which we saw briefly in the “Installing the VxD Software Drivers” section of this manual.

We have previously seen the “General” screen, so let’s move on to the “Settings” screen by clicking on the tab labeled **Settings**.



*Additional settings can be found in the “Mia by Echo Properties” page. This window is similar for all types of Echo hardware.*

## **Show console on taskbar**

The first selection in the **General** section of the **Settings** screen allows you to determine whether the Console program will occupy a space on the Windows taskbar (for example, Mia’s blue “**M**” in the bottom right-hand corner of the Windows desktop).



Checking this option enables this feature, while unchecking will disable it.

## **Monitor during playback**

In the default mode, your Echo product allows you to monitor your input signal whether you are recording, playing back, or simply idle. In some situations, such as when performing punch-in recording, it may be desirable to have the input signal muted until Record is enabled. In the **Echo Properties** window that appears you’ll see a checkbox labeled **Monitor during playback**. Uncheck the box to disable input monitoring during playback; recheck it to enable input monitoring during playback. The Monitoring controls are found in the **Console** (see above).

## **SAW compatibility mode**

When using Mona, Gina24, or Mia with earlier versions of the Software Audio Workshop (SAW) program from Innovative Quality Software, there are some special considerations that must be accommodated by the driver. SAW expects an equal number of inputs and outputs on. For SAW compatibility, the driver creates several non-functional mirrors of input channels 1 and 2 (making an equal number of ins and outs). To insure the hardware will operate properly with earlier versions of SAW, check the box labeled **SAW Compatibility Mode**.

## Multi-client audio

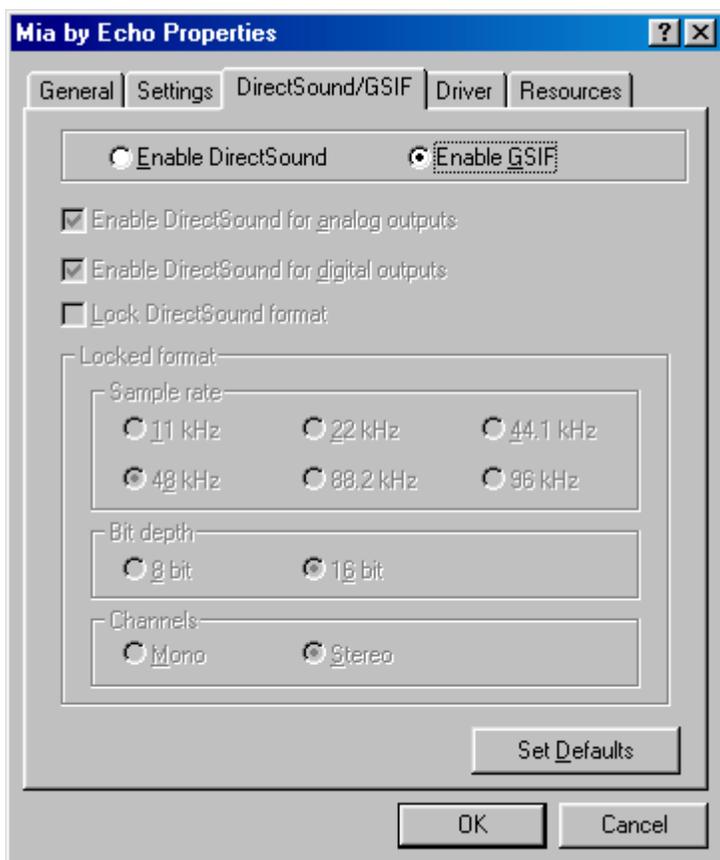
Multi-client audio lets you use different inputs and outputs on your Echo audio system with different audio programs at the same time. Otherwise, you would have to use a different audio device with each audio application.

For example, suppose you own both Cubase VST (multitrack audio) and Reality (software synthesizer). You can now set Cubase VST to play out through outputs 1 through 6, and set Reality to play out through outputs 7 and 8.

The default setting for Multi-client audio is disabled. Although this feature is very useful, it can also be difficult to use properly. We highly recommend that you read the appendix entitled “**Multi-client audio FAQ**,” found later in this manual, before using this feature.

## DirectSound/GSIF Settings

Click on the **DirectSound/GSIF** tab. The window will change to look like this:



The two radio buttons on the top toggle between the **DirectSound** and **GSIF** modes, which are mutually exclusive. While **DirectSound** is enabled, you will not be able to use your Echo product with the Gigasampler or Gigastudio software samplers. Conversely, when **GSIF** is enabled, you will not be able to use the Echo DirectSound driver.

The first checkbox is called “**Enable DirectSound for analog outputs.**” This allows you to enable or disable the DirectSound driver for the analog outputs only.

The second checkbox is called “**Enable DirectSound for digital outputs.**” This allows you to enable or disable the DirectSound driver for the digital outputs only.

**DirectSound** can only support a limited number of outputs, so if you have several audio devices with a large number of outputs you may not be able to find the output you are looking for. In this situation, you can disable some of the **DirectSound** outputs, and then when you restart you should be able to find the **DirectSound** output that was missing before.

The third checkbox is labeled “**Lock DirectSound format.**” This allows you to force the DirectSound driver to playback files at a certain sample rate and bit depth. It also allows you to lock it to stereo or mono.

This setting only applies if you are using software that uses multiple DirectSound outputs simultaneously, such as the applications offered by Stage Research. For this reason, it defaults as disabled.

**Note:** This is not the same as the “sample rate lock” feature found in the Console.

## Cubasis VST

### Installing Cubasis

Included with your Echo 24 system is a special version of Steinberg’s Cubasis VST, a complete audio and MIDI recording studio. Here you can record and arrange your musical ideas, create a mix, add effects and finally produce a finished file ready for burning to CD-R. To install the software:

1. Insert the Echo 24 Install CD-ROM
2. The Main Install Window should automatically pop up. If it doesn’t, go to My Computer and double-click on the CD-ROM icon.
3. From the Main Install Window, select “Install Cubasis.” Then, simply follow the Install Wizard.

Cubasis Setup will copy all of the necessary files to your hard drive. It will also add a program group called “Steinberg Cubasis VST” to the “Program” menu available from the “Start” button menu.

Cubasis VST comes with very thorough “Operation Manual” in .PDF format, so you’ll be able to get up and running with a minimum of hassles (hopefully none at all!). You can access the Cubasis VST “Operation Manual” from the same program group in the “Start” menu.

### Launching Cubasis VST and Assigning Inputs & Outputs:

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

1. Locate the Cubasis VST program on the Start menu or on the desktop.
2. Select the program from the menu (or double click on the icon) to launch the program. Cubasis VST will start and display an Arrange window.

### Audio Settings

1. Pull down the Audio menu and select “System...” The Audio System Setup dialog will then appear.

2. Pull down the ASIO Device pop-up menu, and select your ASIO driver; if you installed the VxD drivers, select “ASIO Echo Layla24/Mona/Gina24/Mia,” or if you installed the WDM drivers, select “ASIO Echo WDM.” In simple terms, the ASIO Device is the mechanism that allows audio data to be transferred between Cubasis VST and your Echo product.
3. Close the System dialog by clicking OK.

## Choosing audio playback and record channels

Pull down the Audio menu and select “VST Channel Mixer.” In the bottom right-hand corner of the VST Channel Mixer window, you can select the pair of stereo outputs on your Echo hardware where you want to send your audio. Click on the “pull down” menu below the Master faders, and select the stereo pair of outputs that you have connected to your audio monitors, speakers or external mixing board. For example, if you have your speakers connected to outputs 1&2 on your Echo hardware, choose your outputs 1&2 within this “pull down” menu.

**Note:** By default, Cubasis VST will activate your first set of stereo inputs on your hardware. Also, please keep in mind that Cubasis VST is a “stereo input and stereo output program.”; that is, you can only use a single pair of outputs and a single pair of inputs at any given time.

There are more detailed instructions for Cubasis VST in the “Operation Manual” and “Steinberg on the Web.” Both sources can be found in the Start menu’s “Programs” list.

## Contacting Echo Customer Service

If you experience any trouble with your Echo hardware please go to the support area of our website at [www.echoaudio.com](http://www.echoaudio.com), and check out the tutorials & troubleshooting 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 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 using the Appendices in the owner's manual and/or Cubasis VST on-screen help.

Thank you for buying an Echo product!

## Contacting Steinberg Customer Service

Note: All Cubasis VST support issues are handled solely by Steinberg.

**Please do not call Echo for technical support regarding Cubasis VST.**

You can reach Steinberg by selecting "Steinberg on the Web" from the pull down **Help** menu within Cubasis VST. This will automatically launch your Web browser, and take you to the Steinberg website. Simply click on the "Support" link, and then "Contact."

## Appendix A: General Troubleshooting Guide

**Note:** For the latest information about software compatibility, please visit our website, [echoaudio.com](http://echoaudio.com).

**Problem:** After installing your Echo product, one or more of your peripheral devices no longer functions properly.

**Solution:** During the installation of your Echo product under Win98/Me it is possible that an interrupt conflict was created. Please see **Appendix B: Resolving Interrupt Conflicts** for more information on this topic.

**Problem:** You installed your Echo product and restarted Windows. Now Windows indicates that it has found new hardware and wants you to install a driver for it. This hardware was working before you installed your Echo product, and the driver for this hardware was already installed.

**Solution:** It is possible that during the installation process you moved an existing card from one slot to another. Windows keeps track of installed devices, not only by peripheral type, but also by physical location. In moving a card from one slot to another, Windows will no longer recognize the card as one for which a driver is installed. You must re-install the driver for the card that was moved.

**Problem:** You installed multiple Echo cards, and you get several error messages telling you that Windows is unable to open the Mixer Library.

**Solution:** You can ignore these error messages and click “OK”. Now restart Windows. After you restart, everything should work normally. The error messages only appear when you install an additional Echo card. This will be fixed in a future driver release.

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

**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 your Echo card, 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 your Echo hardware transmits. To access this switch go to the **Console**. Select **Preferences** under the **File** menu. In the window that appears you'll see a pair of check boxes in an area labeled **S/PDIF Out Format**; one box 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:** Your Echo card never transmits the SCMS bit, regardless of which mode is selected.

**Problem:** When the Windows start-up sound plays, it plays at an altered pitch.

**Solution:** When your Echo card is set to sync with an external device, it will playback at the rate generated by that device. If the startup sound you are using was sampled at 8kHz, but you are synchronized with a device running at 44.1kHz, the startup sound will play back at this faster rate. You have four choices - ignore the altered pitch, switch to internal clock, use the sample rate lock feature, or use a different device for the startup sound playback. We suggest that you never use your Echo card for playback of any of Windows' system sounds.

**Problem:** Your Echo card 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 Echo hardware. 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.

**Problem:** Recordings made using the S/PDIF port contain occasional pops or skips.

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

**Problem:** On Windows startup you hear a high pitched squealing sound or your S/PDIF signal has gone crazy.

**Solution:** Reset the sample rate. Windows always sets the sample rate to 11kHz at startup. Because S/PDIF doesn't support this rate, you will have problems syncing until you reset your Echo hardware's sample rate to one within spec (like 44.1 or 48kHz). You can do this by playing a wave file of the appropriate sample rate through Echo card's outputs. Then you will be able to sync properly

**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 1/4" plugs are made to the exact same dimensions. Try a plug/cable from a different manufacturer.

## **Appendix B: Resolving IRQ Conflicts - Windows 98/Me**

We estimate that 95% of Echo installations will go without a hitch—that Plug-and-Play will properly assign resources to your card without any conflicts. The other 5% of you may experience interrupt problems (mostly caused by non-Plug-and-Play ISA cards or PCI cards that don't like to share). These problems can manifest themselves by causing your new Echo card to not function properly; in some instances, devices will fail that were previously working just fine.

If you are having problems with your computer after installing a new Echo card (or if you are unable to successfully install the card at all), please read the sections below describing what interrupts are, how they are assigned, and how to work around interrupt conflicts. Hopefully, we'll be able to help you solve your problem without too much trouble.

### **What is an Interrupt?**

A computer has two ways of telling when a device, such as an audio card, is ready to exchange data with a program. One way is to have the computer periodically ask or “poll” the device to see if it's ready to transfer new data. Since it is important to do the transfer as soon as it is ready, the program must poll the card quite often, which wastes the computer's resources.

The other method is to have the card “interrupt” the program when it is ready to transfer data. It does this by sending a signal over one of the many interrupt wires connected to the slot into which it is plugged. X86 and Pentium PC's have 16 possible interrupts (0 to 15), five of which are reserved by the motherboard for the keyboard, system timer, etc.

### **ISA and PCI Interrupts**

There are two types of expansion connectors found on today's computers. The older style connectors are called ISA (for Industry Standard Architecture) connectors. ISA connectors have 11 of the 16 PC interrupts wired to them. Some ISA cards (also called “legacy” cards) have you select which interrupt is used by configuring jumpers on the card. Newer Plug-and-Play ISA cards have their interrupts selected by either the computer's BIOS (the built-in program that starts the computer) or by Windows 98/Me.

The newer PCI (Peripheral Connect Interface) bus transfers data faster than ISA and was designed to support Plug-and-Play from the start. PCI connectors have only four interrupts wired to them (A, B, C, and D). However, these interrupts can be shared by more than one PCI connector slot and each connector's interrupt can

be assigned or “steered” to one of the 11 available PC interrupts by either the BIOS or Windows 98/Me. As far as the user is concerned, PCI interrupts use the same numbers and are assigned the same way as Plug-and-Play ISA interrupts. Note, however, that a PCI card can share an interrupt with another PCI card but not with an ISA card.

## **Plug-and-Play and Interrupts**

When your computer first starts up, the BIOS will check each connector slot to see if a Plug-and-Play card is installed. Each Plug-and-Play card has built into it a list of what resources it requires including how many interrupts it needs and which ones it can use. The BIOS will then configure each card and, if needed, assign an appropriate interrupt. It does this for both ISA and PCI Plug-and-Play cards. Windows 98/Me will also check for and configure Plug-and-Play cards, possibly changing some of the settings to suit its needs.

## **Legacy Cards and Interrupt Conflicts**

Problems can arise when the system contains older “legacy” ISA cards with their jumper-configured (or possibly hard-wired) interrupts. The computer’s BIOS and Windows 98/Me have no automatic way to detect these cards and determine which interrupts, if any, are being used. For example, you may have a MIDI card installed with its jumpers configured to use IRQ 9. Since the BIOS cannot detect the legacy card it will consider IRQ 9 to be a free interrupt. Consequently, it may assign IRQ 9 to your Echo card. The next time you try and use your MIDI card or play back audio via your Echo card, neither device will function properly.

Or, maybe you have a SCSI card that had previously been assigned to IRQ 10. The BIOS may now decide to assign IRQ 10 to your Echo card, and move the SCSI card to IRQ 9. An interrupt conflict now exists between the SCSI card and the older MIDI card.

## **PCI Cards and Interrupt Conflicts**

Some plug & play PCI cards can also cause interrupt conflicts when they try to share an IRQ with another PCI card. Many PCI cards can share an IRQ with no problems at all, but some just don’t like to share. For example, you may have a network card installed, and Windows has it configured to use IRQ 10. Since that is the only card assigned to IRQ 10, Windows assumes that it is OK to let it share with the next PCI card that comes along. Consequently, it may assign your Echo card to share IRQ 10. This network card happens to be unfriendly and doesn’t like to share, so the next time you try to use it or your Echo card, neither device will function properly.

## Resolving Interrupt Conflicts

Although there is no way to automatically detect legacy interrupts, Windows 98/Me and some BIOS's allow you to reserve specific interrupts for legacy use. Once an interrupt is reserved it will not be assigned by Plug-and-Play to another card. Or, if you have a PCI card that doesn't like to share you may be able to reassign it to a different IRQ.

The first thing to do if you think you have an interrupt conflict is to determine which interrupt and which card is causing the problem. To do this, right-click on the **My Computer** icon on your desktop and select **Properties**, or go to **Start – Settings – Control Panel** and double-click the **System** icon. In the window that appears, select the **Device Manager** tab. Now, highlight **Computer**, and click the **Properties** button. A new window will appear that lists all of the interrupts (IRQs) that Windows 98/Me recognizes are in use and the cards or resources to which they are assigned. Write down the interrupts that are listed and the devices that are assigned to them. Pay particular notice to the interrupt that your Echo card is assigned to.

Once you know which interrupt your Echo card is trying to use, you need to locate which card is potentially causing the conflict. To do this you may need to dig out the manuals for whatever ISA cards are installed and examine the jumper settings to determine which interrupts, if any, are used. Hopefully, at this point you will find an offending card that was not listed by **Device Manager**, but is using an interrupt that Windows has assigned to another device. If you don't have any ISA cards, or your legacy ISA card is not causing the conflict, you should take note of which PCI card is assigned to the same IRQ as your Echo card.

If you have lost your manuals and are having trouble figuring out which card may be causing a problem, you can try the brute force approach. Remove all cards from your system and then re-install them one at a time until the system quits working. The card that makes the computer stop working correctly will be the one that needs to be dealt with.

Once the offending card has been identified you have a few different options. One option is to simply remove the card from your system. However, you've likely got the card in there for a reason, so unless it's an old sound card you don't care about keeping, you'll probably want to use an approach to remedying the problem that allows the card to remain in the system.

If the offending card is a legacy ISA card, another option is to change the IRQ on the card to an IRQ that isn't currently being used as reported by the Reporter program. While this may fix the problem at hand, the next time you add another new Plug-and-Play card you may wind up with another conflict. If you choose this option it is still a good idea to reserve the IRQ as described below.

If the offending card is a PCI card, you may want to skip to the section on “Reconfiguring an IRQ within Windows 98/Me - PCI” a few pages down.

## Reserving an IRQ within the BIOS – Legacy ISA

Other than removing the offending legacy card, the next most effective method is to reserve the IRQ for legacy use within your BIOS. To do this you will need to enter the BIOS configuration screen when your computer first starts up. This is usually done by pressing the **Delete** key or **F1** key right after the memory test during the boot-up routine. Once you're in the BIOS control panel, check to see if you are given the option of manually configuring your interrupts. Some BIOS's, such as those from Award, allow you to choose between "Plug-and-Play" and "legacy ISA" for each interrupt. If you select "legacy ISA" for the interrupt that is being used by your legacy card, the BIOS will know not to assign that interrupt to a Plug-and-Play card.

Unfortunately, some BIOS's don't support this capability. If your BIOS is one of those, don't worry: You still have another option.

## Reserving an IRQ within Windows 98/Me – Legacy ISA

Windows 98/Me also allows you to reserve interrupts for legacy use. In most cases this works just as well as reserving the IRQ in the BIOS. However, you may still have problems if you boot up an older version of DOS from a floppy where Windows 98/Me can't do its Plug-and-Play magic. To reserve an IRQ within Windows 98/Me do the following:

1. Click on the **Start** button and select **Control Panel** under the **Settings** option.
2. Double click on the icon labeled **System**.
3. Select the tab labeled **Device Manager** and click the **Properties** button.
4. Select the tab labeled **Reserve Resources** and make sure that **Interrupt Request** is selected.
5. Click on the **Add** button and enter the IRQ that is being used by your legacy card.
6. Windows may prompt you that the IRQ is currently in use by another device. Click on **Details** to see which device Windows currently thinks is using this IRQ. It should be your Echo card or whatever device is currently conflicting with your legacy card. If that is the case, go ahead and reserve the IRQ.
7. When Windows asks, reboot your machine. When Windows restarts your Plug-and-Play cards will not be assigned to the reserved IRQ.

Hopefully, your system will now be functioning properly. If not, please refer to the "Contacting Customer Service" section of this manual. We'll do our best to help you solve your problem.

## Reconfiguring an IRQ within Windows 98/Me - PCI

Now that you have identified the offending card and the IRQ in contention, you will want to try to reconfigure its IRQ setting. To do this, you must open the Device Manager. Select **Start – Settings – Control Panel – System**, and click the **Device Manager** tab. Find the entry for the PCI card that is causing the problem, highlight it, and click the **Properties** button. In the properties window that appears, select the **Resources** tab. Make sure that “Use automatic settings” is unchecked, highlight “Interrupt Request,” and click the **Change Setting** button. Some cards will allow you to change the IRQ setting at this point, while others will give you a message saying, “This resource setting cannot be modified.” If you are allowed to change the IRQ, a window will appear that lets you choose from several different values. Scroll through the different values until you find one that says, “No devices are conflicting.” Choose that one and keep selecting **OK** until Device Manager is closed, then restart Windows.

If you are unable to reconfigure the IRQ and you have a free PCI slot, you may want to try moving your Echo card or the offending card to a different slot. Make sure your computer is off first, and follow the safety instructions in this manual for hardware installation. Sometimes this can help, because Windows will assign a different IRQ depending on which slot the PCI card is in.

Hopefully, your system will now be functioning properly. If not, please refer to the “Contacting Customer Service” section of this manual. We’ll do our best to help you solve your problem.

## Appendix C: Multi-client audio FAQ

### Q: What is “Multi-client audio”?

Multi-client audio is a feature that lets you use different inputs and outputs on your Echo card with different applications at the same time. This way, you can use several different audio applications with just one Echo card. Otherwise, you would have to use a different audio device for each audio application.

For example, suppose you own both Cubase VST (multitrack audio) and Reality (software synthesizer). You can now set Cubase VST to play out through outputs 1 through 6, and set Reality to play out through outputs 7 and 8. In previous Echo card drivers, you had to choose between one software program and the other.

Note that all of the various interfaces are multi-client. You could have an ASIO application, a wave application, and a DirectSound application all running at the same time. It's very flexible.

If you want to use multi-client audio, you need to enable it in the main driver properties page. See the section entitled “Additional Configuration Settings” in this manual.

**Note:** Multiclient audio is always enabled with the WDM driver.

### Q: So what's the catch?

The catch is this: all the apps using the Echo card at the same time must be set at the same sample rate. This is a limitation of the hardware.

Say that you didn't follow this rule in the previous example. Say that you initially had Cubase VST and Reality set up for 44.1 kHz- so far, so good. Then you load a song into Cubase that is set up for 48 kHz. Cubase will change the sample rate of the hardware. Unfortunately, this means that Reality is now also coming out at 48 kHz, so all the sounds played out of Reality will sound higher in pitch.

If you're planning to use multi-client audio, the best thing you can do is to sit down and plan it out. Make a chart of each input and output on the Echo card and write down what application is going to use what input or output. Go through all the applications you plan to use this way and set them all up for the same sample rate and set up their inputs and outputs correctly.

You should also turn off your system sounds. Most system sounds tend to be at 11 kHz or 22 kHz. Imagine that you're recording the best guitar solo of your life and someone clicks on something on the screen. The 22 kHz system sound will play

out of an unused output and drop the sample rate of the whole system to 22, ruining your take.

If you really want system sounds, go buy a \$30 cheapo sound card and play them through that. Set the preferred device in the Multimedia control panel to the cheapo sound card and check "Use preferred devices only."

**Q: What are the rules that I should follow if I want to use multiple audio applications at the same time?**

- Set all your applications to the same sample rate. (You can also use the "sample rate lock" feature in the Console to help with this.)

-Turn off your system sounds.

- Figure out ahead of time which inputs and outputs each application is going to use. Inputs and outputs cannot be shared between applications.

**Q: What I really want to do is use a software synthesizer or sampler like Gigasampler, Reality, or Retro AS/1. What's the best way to use this with an audio and MIDI sequencer like Cubase VST or Cakewalk?**

The best thing you can do is run the synthesizer/sampler application first, then run the sequencer. Make sure that each application is set to use a different set of inputs and outputs.

With the VxD ASIO drivers, you need to be careful with assigning inputs and outputs to get Cubase VST to work in multi-client mode. With WDM, the ASIO driver hides channels from Cubase that are already in use, which greatly simplifies getting multi-client mode to work.

**Q: Can I run a Direct Sound program, an ASIO program, and a wave program all at the same time?**

Sure. If you don't understand the question, don't worry about it.

**Q: What's the downside to all this?**

The main downside is that your computer may very well have a hard time keeping up! If you fire up three different audio apps at once, your computer is probably going to struggle unless you have a really hot machine. The key here is memory, more memory, and a fast CPU.

## Appendix D: WDM-Windows Driver Model FAQ

### Q: What is a WDM driver and why should I care?

WDM stands for “Windows Driver Model.” Microsoft realized they were asking hardware manufacturers to write a lot of different drivers: support for Windows NT, Windows 95, support for the wave API, support for DirectSound, etc. To encourage good driver development, they came up with the idea of WDM : you can just write one driver that can run on all the different Windows platforms that support all the different audio APIs.

You should care about WDM and our hardware if you meet any of the following criteria:

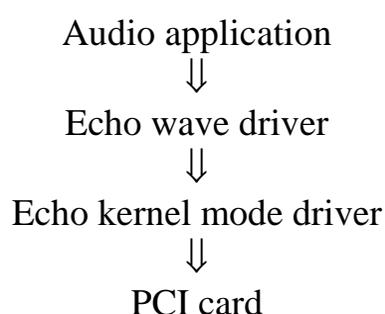
- You are running Windows 2000 or Windows XP.
- You are running SONAR or some other audio application that talks directly to WDM drivers.

If you are running Windows 95, 98, 98SE, or Me and are using the VxD drivers, there probably isn’t a compelling reason to switch to WDM (unless you really, really like the new console, or unless you have an audio application that takes advantage of WDM).

### Q: How is this driver different from the previously released Windows 2000 beta drivers?

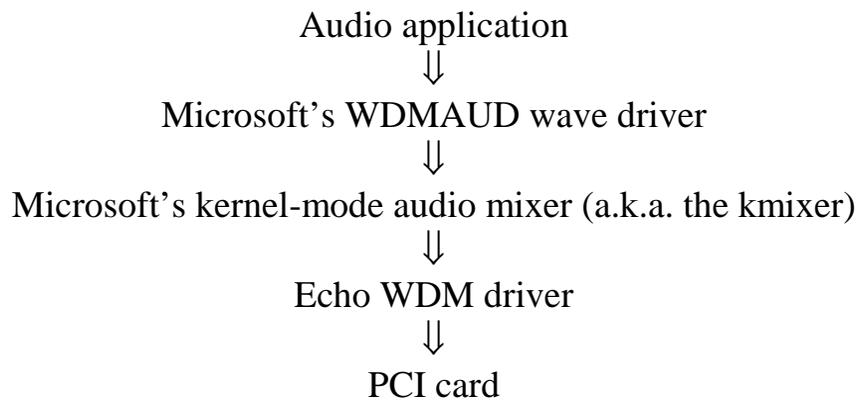
The previously released Windows 2000 beta drivers were actually Windows NT 4 drivers that could also be run under Windows 2000. They were not plug-n-play (since NT 4 does not support plug-n-play), and they worked the “old way.”

What’s the old way? Here’s a simplified diagram:



So, you can see the audio application would communicate more or less directly with our wave driver (it doesn’t really, but that’s close enough for this discussion). If an application wanted to use a specialized 24-bit format, we could see what it was asking for and add support for that. This also let us control exactly which wave devices were listed and how their names were listed.

The new driver is a WDM driver. This means it was natively developed for Windows 2000 (that is, it's not a legacy driver running on a newer OS). However, WDM drivers work differently (again, this is simplified):



So you can see that there are now several layers of software interposed between our driver and the application. This means we have had to give up some control over how this works.

Unfortunately, this causes some pain; the worst of it is the WDMAUD wave driver breaks most of the software out there that supports 24-bit audio. WDM audio, not our driver, is now in charge of which wave devices get listed and which formats we accept. It also means we have less control over the names of the wave devices, which is less painful, but still annoying.

As things stand, we are investigating the possibility of somehow combining the old and new approaches to get the best of both worlds.

### **Q: So if it causes all these problems, why bother with WDM?**

Because everyone asked us to. ☺

No, seriously, it has a number of benefits. The obvious ones are Plug-n-Play support (no more seventeen-step install process) and native support for Windows 2000 and XP, which we haven't had up 'til now.

It's also the wave (get it?) of the future; Cakewalk's SONAR gets excellent results by talking directly to the WDM driver and bypassing most of the layers in the above diagram. Other software vendors are pursuing the same approach.

## **Q: Why can't I record or play at 24-bit resolution?**

The short answer is that WDM audio drivers only support 24-bit audio if the application is using Microsoft's new "extensible" wave format. Many existing apps do not support this format. This only affects programs that use the wave/MME or DirectSound APIs.

The long answer to this question appears up above, in the general WDM discussion.

## **Q: Well, why can't you guys make 24-bit work? Your old drivers worked fine.**

Yep, they did. With the old drivers, we were a lot closer to the action- we got to see the audio format that the application was requesting. With WDM, we don't; there are now several layers of Windows code between us and the application. (Again, for more information about this, see the general WDM discussion above.)

Please be aware this would affect \*any\* WDM driver; it's not our bug, it's part and parcel of WDM audio. However, that being said, we are looking into a workaround. Stay tuned.

## **Q: Really, why don't you guys support WDM on Windows 98 or 98SE?**

One of the problems we have with running a WDM driver is we are largely at the mercy of the WDM audio subsystem in Windows. WDM audio in Windows 98 is more or less broken. Windows 98SE is better, but still has a significant number of problems. It's not worthwhile for us to try and support something like that.

There is a patch for Windows 98SE which corrects many of the problems with WDM audio; however, it is not directly available for download. If you want to pursue getting this patch, log on to support.microsoft.com and read article Q242937, or just follow this link:

[http://support.microsoft.com/support/kb/articles/Q242/9/37.ASP?LN=EN-US&SD=gn&FR=0&qry=WDM%20audio&rnk=7&src=DHCS\\_MSPSS\\_gn\\_SRC\\_H&SPR=W98SE](http://support.microsoft.com/support/kb/articles/Q242/9/37.ASP?LN=EN-US&SD=gn&FR=0&qry=WDM%20audio&rnk=7&src=DHCS_MSPSS_gn_SRC_H&SPR=W98SE)

# Appendix E: 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:

<u>BINARY</u>	<u>DECIMAL</u>
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).

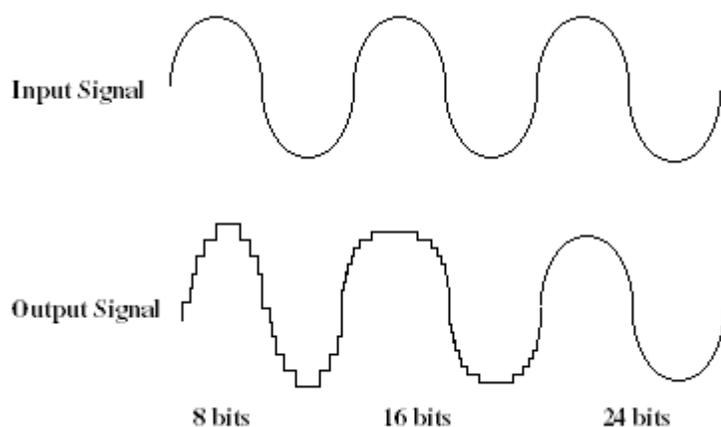


Figure B. The more bits there are available, the more accurate the representation of the signal and the greater the dynamic range.

Your Echo card'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 card 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 multiplies 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 96 dB.

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 Your Echo card allows you to sample sound at up to 96,000 times per second.

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 +4 dBu 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 +4 dBu signal has a voltage level of 1.23 volts and a –10 dBV 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 card provides approximately 14dB of headroom allowing an 18dBu signal to be recorded.

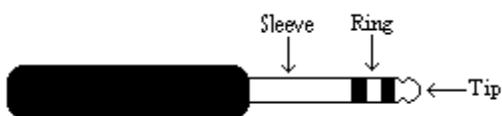
## Unbalanced and Balanced Inputs and Outputs

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 signals connect with either XLR connectors or TRS (tip, ring sleeve) connectors. Your Echo card uses TRS connectors for connecting balanced 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). Gina24 will also accommodate the two conductor unbalanced style connector.

## 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 card 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 card only have a dynamic range of 106 dB?
- 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 -137 dBu (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.

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 compressing 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 card has enough dynamic range to allow sufficient headroom and post-processing to be applied while still keeping the noise either off the CD completely or down as far as possible.

# Appendix F: Specifications

## **Gina24 - Audio Performance**

Analog Inputs (x2 balanced TRS):

- Frequency Response: 10Hz – 22kHz,  $\pm 0.25$ dB
- Dynamic Range: 108dB A-weighted
- THD+n: <0.001%, 20Hz–22kHz A-weighted
- Nominal Input Level: +4 dBu
- Maximum Input Level: +18dBu
- Input Impedance: 10K

Analog Outputs (x8 balanced TRS):

- Frequency Response: 10Hz – 22kHz,  $\pm 0.25$ dB
- Dynamic Range: 110dB A-weighted
- THD+n: <0.002%, 20Hz–22kHz A-weighted
- Nominal Output Level: +4 dBu
- Maximum Output Level: +18dBu

## **Gina24 - Hardware**

- Host Interface: PCI bus mastering card
- Two 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
- ADAT optical digital I/O
- Headphone output with volume control
- On-board 24-bit Motorola DSP
- 24-bit data resolution maintained throughout entire signal path
- Multiple sample rates from 8kHz to 96kHz
- Esync clock I/O

## **Layla24 - Audio Performance**

Analog Inputs (x8 balanced TRS):

- Frequency Response: 10Hz – 22kHz,  $\pm 0.25$ dB
- Dynamic Range: 110dB A-weighted
- THD+n: <0.001%, 20Hz–22kHz A-weighted
- Nominal Input Level: +4 dBu
- Maximum Input Level: +22dBu
- Input Impedance: 10K

Analog Outputs (x8 balanced TRS):

- Frequency Response: 10Hz – 22kHz,  $\pm 0.25$ dB
- Dynamic Range: 115dB A-weighted
- THD+n: <0.002%, 20Hz–22kHz A-weighted
- Nominal Output Level: +4 dBu
- Maximum Output Level: +22dBu

## **Layla24 - Hardware**

- Host Interface: PCI bus mastering card
- 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
- ADAT optical digital I/O
- Headphone output with volume control
- On-board 24-bit Motorola DSP
- 24-bit data resolution maintained throughout entire signal path
- Multiple sample rates from 8kHz to 96kHz
- MIDI In/Out/Thru
- Word clock I/O
- Esync clock Output

## **Mona -Audio Performance**

### Analog Inputs (x4):

- Frequency Response: 10Hz – 22kHz,  $\pm 0.25$ dB
- Dynamic Range: 110dB A-weighted
- THD+n:  $<0.001\%$ , 20Hz–22kHz
- Nominal Input Level: +4 dBu
- Maximum Input Level: +22dBu
- Microphone Input Level (balanced XLR):
  - EIN: -129dBV
  - Input Impedance: 1.5K
  - Gain Adjustment: 20 – 60dB
- Line Input Level (balanced TRS):
  - Input Impedance: 10K
  - Gain Adjustment: 0 – 40dB
- Guitar Input (unbalanced ¼”):
  - Input Impedance: 107K
  - Gain Adjustment: 10 – 50dB

### Analog Outputs (x6):

- Frequency Response: 10Hz – 22kHz,  $\pm 0.25$ dB
- Dynamic Range: 115dB A-weighted
- THD+n:  $<0.002\%$ , 20Hz–22kHz
  
- +4 dBu Outputs (balanced XLR):
  - Nominal Output Level: +4 dBu
  - Maximum Output Level: +22dBu
  
- -10 dBV Outputs (RCA):
  - Nominal Output Level: -10 dBV
  - Maximum Output Level: +8dBV

## **Mona - Hardware**

- Host Interface: PCI bus mastering card
- Four high quality mic preamps with phantom power (+48v)
- Four universal analog inputs with precision 24-bit 128x oversampling analog-to-digital converters
- Six analog outputs (both XLR & RCA) with high performance 24-bit 128x oversampling digital-to-analog converters
- S/PDIF digital I/O with up to 24-bit resolution
- ADAT optical digital I/O
- Trim control and level meter for each analog input
- Headphone output with volume control
- On-board 24-bit Motorola DSP
- 24-bit data resolution maintained throughout entire signal path
- Multiple sample rates from 8kHz to 96kHz
- Word clock I/O
- Esync clock Output

## **Mia - Audio Performance**

Analog Inputs (x2 balanced TRS):

- Frequency Response: 10Hz – 22kHz,  $\pm 0.5$ dB
- Dynamic Range: 106dB A-weighted
- THD+n: <0.001%, 20Hz–22kHz
- Nominal Input Level: +4 dBu
- Maximum Input Level: +18dBu
- Input Impedance: 10K

Analog Outputs (x2 balanced TRS):

- Frequency Response: 10Hz – 22kHz,  $\pm 0.5$ dB
- Dynamic Range: 106dB A-weighted
- THD+n: <0.002%, 20Hz–22kHz
- Nominal Output Level: +4 dBu
- Maximum Output Level: +18dBu

## **Mia - Hardware**

- Host Interface: PCI bus card
- PCI bus master interface
- Two balanced ¼” analog inputs with precision 24-bit 64x oversampling analog-to-digital converters
- Two 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
- On-board 24-bit Motorola DSP
- 24-bit data resolution maintained throughout entire signal path
- Multiple sample rates from 8kHz to 96kHz

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