LAYLA® 3G

GINA® 3G

Owner's Manual Version 1.0 for Windows

FC Tested To Comply
With FCC Standards
FOR OFFICE USE

CE

N222
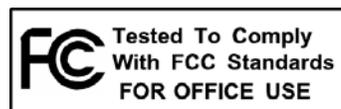
Important Safety Instructions

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. 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.

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

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

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

Limited Warranty

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

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

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

To obtain factory service:

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

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

DISCLAIMER AND LIMITATION OF WARRANTY

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

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

Introduction

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

What You Should Have Received in the Box

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

- A PCI card wrapped in an anti-static cover
- An audio interface box
- Four mounting screws with collars (Layla3G only)
- Two removable rack-mount brackets (Layla3G only)
- An interface cable

  Please note: *The cable included with your Echo3G system is a shielded data cable that has been custom manufactured to exacting standards. Use of any other cable will substantially reduce the system's overall audio quality and could damage the system.*

- A CD-ROM containing:

  - Windows Me/2000/XP Drivers
  - Mac OS X Drivers
  - PDF Owner's Manual
  - Tracktion multitrack recording, MIDI, editing, and sampling software for Windows ME/2000/XP & Mac OS X (10.2 & above)
  - HTML Tracktion User's Guide

- Quick Start Guide

System Requirements

In order to use Layla3G or Gina3G you'll need the following:

- A desktop computer with one of the following:

  - A genuine Intel Pentium III processor (or better) and a motherboard with a genuine Intel chipset **OR**
  - A genuine AMD Athlon processor and a motherboard with an AMD, Via, or Nvidia chipset

- An empty PCI slot

- A minimum 256Mb of RAM (512Mb highly recommended)

- A fast, high-capacity IDE or SCSI hard disk drive

- Windows Me/2000/XP

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

Installation

Complete installation consists of logging in as an administrator, disabling Windows system sounds, installing the Windows drivers into your system, installing the PCI card, connecting the audio interface to the PCI card and to your other audio hardware, and installing Tracktion multitrack audio recording software.

Log In as Administrator

If you are using Windows 2000 or XP, you must be logged into your computer as an Administrator. You can either log in with the account called "Administrator" or with any other account that has administrative privileges. Otherwise, you will not be able to install the drivers.

Disable Windows 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 from the "Start" button. On the control panel you will find an icon titled "Sounds" or "Sounds and Audio Devices". After double clicking on this icon, you will see a window labeled "Sounds and Audio Devices Properties." Click the "Sounds" tab and find the section labeled "Sound Scheme." Click on the drop down box, and select the "No Sounds" option. Then, click the Ok button.

Installing the PCI Card

Once you have logged in as an administrator, and disabled the Windows system sounds, it is time to install the Echo3G 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 Windows driver, shut down Windows and **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. You may use any available PCI slot 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. **Replace the computers cover**, and secure it. Reattach its power supply cord and reconnect any peripherals that you may have removed prior to beginning the installation.

9. Locate the audio interface box and the interface cable. **Securely mount the interface** into your equipment rack (Layla3G only). If you will not be mounting the box in a rack, then be sure to place the interface 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 connector on the rear of the interface labeled "computer" and fasten the cable securely with the screws.

Caution: Never connect the audio interface box to the computer while the computer is turned on. This could damage the interface box or PCI card.

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

11. **Turn on your computer.** Once Windows starts, it should find new hardware called "Echo3G" 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.

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Connecting to the Audio Interface

The Layla3G and Gina3G audio interfaces contain a wide variety of connections that allow great flexibility in operation. For optimal performance, it is critical to use the appropriate cabling and connectors.

Analog Inputs and Outputs – Layla3G

Layla3G's front panel has two universal inputs (#s 1 – 2) with meters and trim knobs, a phantom power switch and light (+48v), and an amber power indicator light. There is also a ¼" headphone output jack, with a corresponding volume knob, which monitors analog outputs 1 and 2.
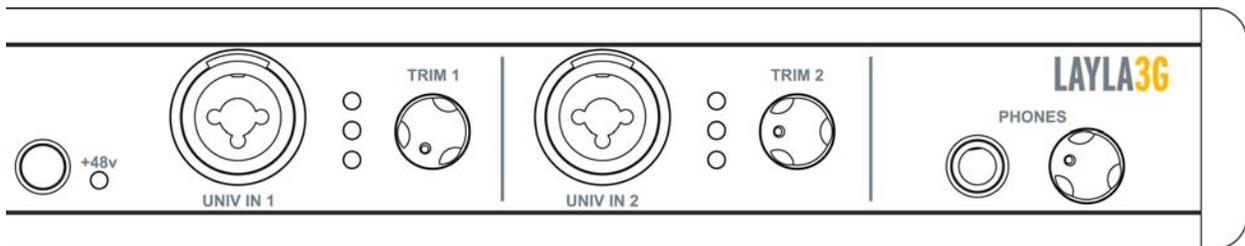
Layla3G has six analog line inputs (#s 3 – 8) and eight analog line outputs on the rear panel. These analog inputs (#s 3 – 8) operate with an input impedance of 10Kohms. The input and output connections can be used with balanced or unbalanced cables via the ¼" connectors. By default, the inputs and outputs are set to receive or send a +4dBu signal. You can switch between +4dBu and –10dBV for each individual input and output via the console software (see section below). There are also two channel inserts on the rear panel where you can connect a serial effects device (compressor, EQ, etc.) for the universal inputs (#s 1 – 2).

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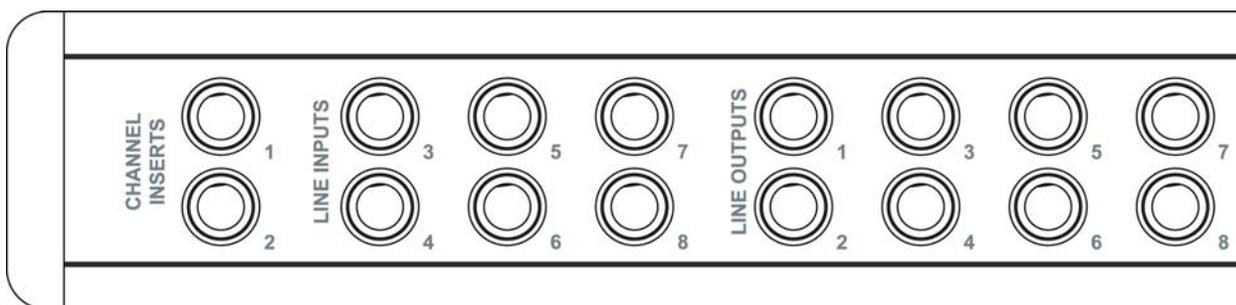
Analog Inputs and Outputs – Gina3G

Gina3G's front panel has two universal inputs (#s 1 – 2) with meters and trim knobs, a green phantom power light (+48v), and an amber power indicator light. There is also a ¼" headphone output jack, with a corresponding volume knob, which monitors analog outputs 1 and 2.
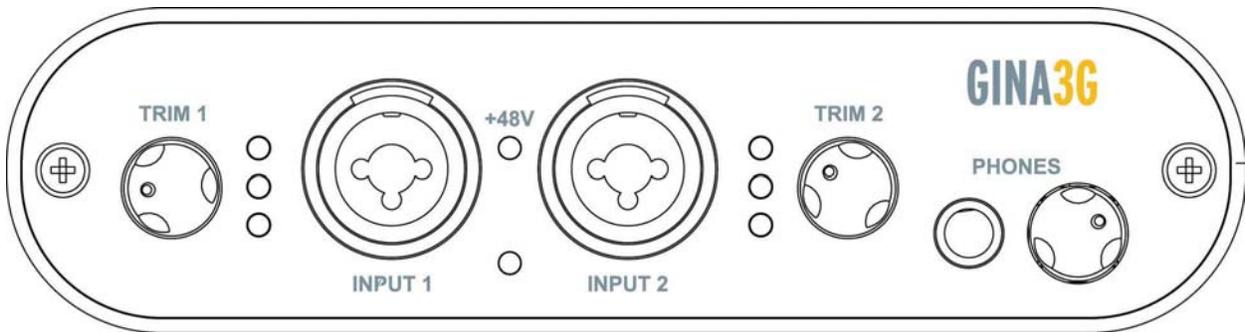

Gina3G has six analog line outputs on the rear panel. The input and output connections can be used with balanced or unbalanced cables via the ¼" connectors. By default, the outputs are set to send a +4dBu signal. You can switch between +4dBu and –10dBV for each individual output via the console software (see section below).
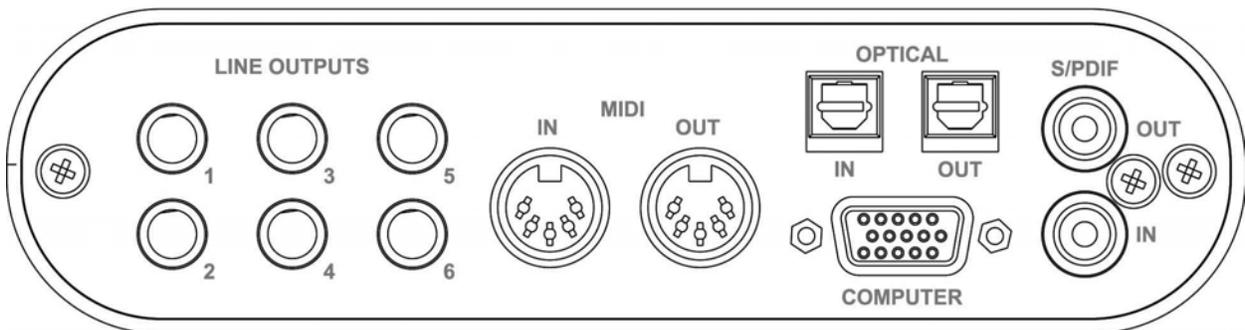
Universal Inputs

The universal input connections can accept balanced or unbalanced signals via ¼" or XLR connectors. Since Layla3G and Gina3G have built-in preamps, you can just plug your microphone or guitar cable right into the universal input. No external preamp, direct box, or mixer is needed!!

To use a microphone, plug the microphone's XLR cable into any of the universal inputs. Whenever an XLR cable is connected to a universal input, the microphone preamp is enabled and the gain range of the trim knob is set to range from +10 to +58dB. The input impedance of each microphone preamp is 1.5Kohms. Full-scale input is 9dBu with the trim set to minimum gain.

Layla3G and Gina3G provide 48 volts of phantom power for microphones that require it. To enable phantom power for Layla3G, press the button labeled +48v (the LED next to it will light up) and phantom power will be provided to all microphone inputs. Gina3G has a switch in the console software for this purpose.

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

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

If you want to adjust the input level of a universal input, just turn the trim knob and watch the corresponding level meter. The red light at the top will light anytime the input signal is greater than –6.5dBFS (or within 6.5dB of clipping). The yellow light represents a signal greater than –12.5dBFS, and the green light represents a signal greater than –24.5dBFS.

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

The Computer Connector

On the rear panel of the interface box is a connector labeled **COMPUTER**. It is the point at which the audio interface connects to the Echo3G PCI card inside your computer.

COMPUTER
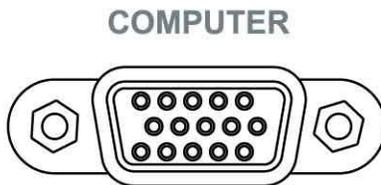
An Echo3G connector cable was supplied to you for this purpose. This custom made cable is manufactured to certain specifications; if you need to replace it, you should only use a replacement cable from Echo.

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 audio data via an optical signal using special digital optical cables with Toslink connectors.

OPTICAL

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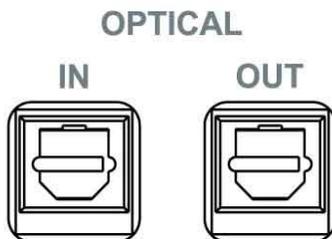
You can use this port for ADAT (8 channels) or for optical S/PDIF (stereo) I/O.

> **Note:** Your Echo3G product 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. This is set using the **Digital Mode Switch**, which can be found in the Echo Console. For more information, refer to the section of this manual that discusses the console software.

S/PDIF

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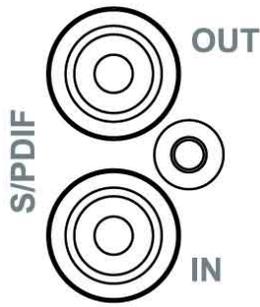
When connecting devices to the S/PDIF jacks, the use of standard analog RCA audio cables is not recommended. For reliable S/PDIF operation, 75-ohm coaxial (RG59) video cables are recommended.

MIDI

Both Layla3G and Gina3G are equipped with MIDI I/O connectors. The MIDI ports can be used for receiving MIDI time code (MTC), or sending and receiving MIDI signals between your digital audio/MIDI sequencing software and external sound modules, keyboards, MIDI controllers, etc.
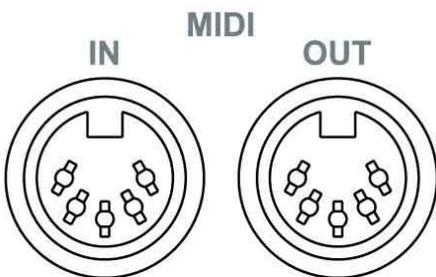
MIDI cabling is highly standardized and widely available.

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Word Clock (Layla3G only)

Next to the S/PDIF ports on Layla3G's back panel are the **Word Clock** connectors. These connectors allow you to synchronize one Layla3G to another Layla3G or to other digital audio devices equipped with a Word Clock connector.
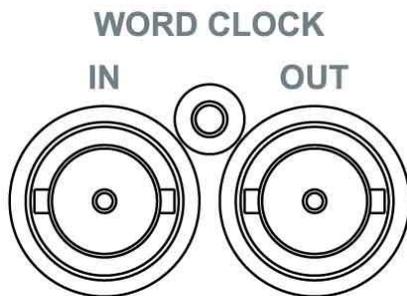
The Word Clock I/O uses a BNC connector. As with the S/PDIF I/O, a shielded 75-ohm (RG-59) coaxial video cable should be used. BNC connectors have been widely used in the electronics industry for both video and computer networking. However, computer networks use 50-ohm (RG-58) coaxial cables and not the 75-ohm (RG-59) cable used by video. For reliable word clock operation, use only video grade cables with Layla3G.

Installing Tracktion

Both Layla3G and Gina3G come bundled with the full version of Tracktion, audio multitrack recording, MIDI, editing, and sampling software.

Install Tracktion from the Echo3G CDROM

To install the software:

1. Insert the Echo3G 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 Tracktion." Then, simply follow the Install Wizard.

Tracktion Setup will copy all of the necessary files to your hard drive. It will also add a program group called "Tracktion" to the "Program" menu available from the "Start" button menu.

Tracktion comes with an HTML "User's Guide" that you can open from the "Help" button inside Tracktion. So, you'll be able to get up and running with a minimum of hassles (hopefully none at all!). You can also access the Tracktion "User's Guide" from the same program group in the "Start" menu.

Authorize your copy of Tracktion

Your copy of Tracktion will run in demo mode until you authorize it. Your Echo3G install CDROM came with a sticker that has an authorization code and a website address printed on it. Go to that website and follow the instructions to authorize your copy of Tracktion.

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

Launching Tracktion and Assigning Inputs & Outputs

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

1. Locate the Tracktion 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.

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

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

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

Audio Settings

1. Click the "Settings" tab at the top of the screen.
2. Select "Audio Devices" on the left.
3. Where it says "Wave device:" click on the drop down menu and select "ASIO Echo WDM".
4. Now you should be able to see a list of the available audio devices for your Echo product. Go ahead and enable the inputs and outputs you want to use with Tracktion by clicking on the red X and waiting for it turn into a green checkmark.

NOTE 1: If you can't find some of your Echo product's inputs or outputs, that means they are currently being used by another software program. You will have to shut down Tracktion, and disable them from the other software program. Most commonly, this is because you selected one of your Echo products' output devices as the default playback device for Windows. If this is the case, then you will need to change the default playback device from the "Sounds and Audio Devices" control panel in Windows.

NOTE 2: If you have several different Echo products and you want to use them simultaneously in Tracktion, you need to synchronize them first or else audio will not playback properly. Refer to "Synchronizing Multiple Devices" in the "Console3" section of this manual for more information.

5. Make sure one of your Echo product's outputs is the "default wave output", and that it is hooked up to some speakers or headphones. You can check this setting on the right side of the "Settings – Audio devices" screen of Tracktion.
6. Now you're ready to play some music. Click on the "Projects" tab at the top of the screen and select one of the sample projects that come with Tracktion.
7. You will see a listing of all the files for that project in the large area to the right. Double-click on the file with the same name as the project. This will open the project.
8. Now click on the play button in the bottom right section of the screen and you'll hear the sample project with your new Echo product.
9. Enjoy the music!!

There are more detailed instructions for Tracktion in the HTML "User's Guide" which is available by hitting the F12 key while in Tracktion. You can also find hints and tips in the Tracktion user's forums available through the Tracktion webpage here: http://www.mackie.com/products/tracktion/

This completes the installation of the hardware and software for Layla3G and Gina3G.

Using Audio Software

Understanding Wave Devices

To audio software the Echo hardware appears initially as a group of stereo wave devices. (Using the SuperChannel feature will change this – for more info see Appendix E: PureWave and SuperChannel.) 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 which product you installed.

For playback devices, you should see:

Layla3G	Gina3G
Layla3G 1/2 Analog Out	Gina3G 1/2 Analog Out
Layla3G 3/4 Analog Out	Gina3G 3/4 Analog Out
Layla3G 5/6 Analog Out	Gina3G 5/6 Analog Out
Layla3G 7/8 Analog Out	Gina3G 1/2 Digital Out
Layla3G 1/2 Digital Out	Gina3G 3/4 Digital Out
Layla3G 3/4 Digital Out	Gina3G 5/6 Digital Out
Layla3G 5/6 Digital Out	Gina3G 7/8 Digital Out
Layla3G 7/8 Digital Out	

And for recording:

Layla3G	Gina3G

Layla3G 1/2 Analog In   Gina3G 1/2 Analog In
Layla3G 3/4 Analog In   Gina3G 1/2 Digital In
Layla3G 5/6 Analog In   Gina3G 3/4 Digital In
Layla3G 7/8 Analog In   Gina3G 5/6 Digital In
Layla3G 1/2 Digital In   Gina3G 7/8 Digital In
Layla3G 3/4 Digital In
Layla3G 5/6 Digital In
Layla3G 7/8 Digital In

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

Digital Mode

Layla3G and Gina3G 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. We'll use Layla3G for our examples below, but Gina3G works exactly the same.

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

Device	S/PDIF RCA	S/PDIF optical	ADAT optical
Layla3G 1/2 Digital	S/PDIF	S/PDIF	ADAT 1/2
Layla3G 3/4 Digital	not used	not used	ADAT 3/4
Layla3G 5/6 Digital	not used	not used	ADAT 5/6
Layla3G 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 "Layla3G 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.

Console3

A "virtual control surface" application called Console3 is installed with the 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 Console windows" and "Start the tray icon."

If you select "Start the tray icon," a red "Echo" icon will be placed in 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. If you want the launcher icon to appear automatically in the icon tray every time you start Windows, then select "Auto-start the tray icon."

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.
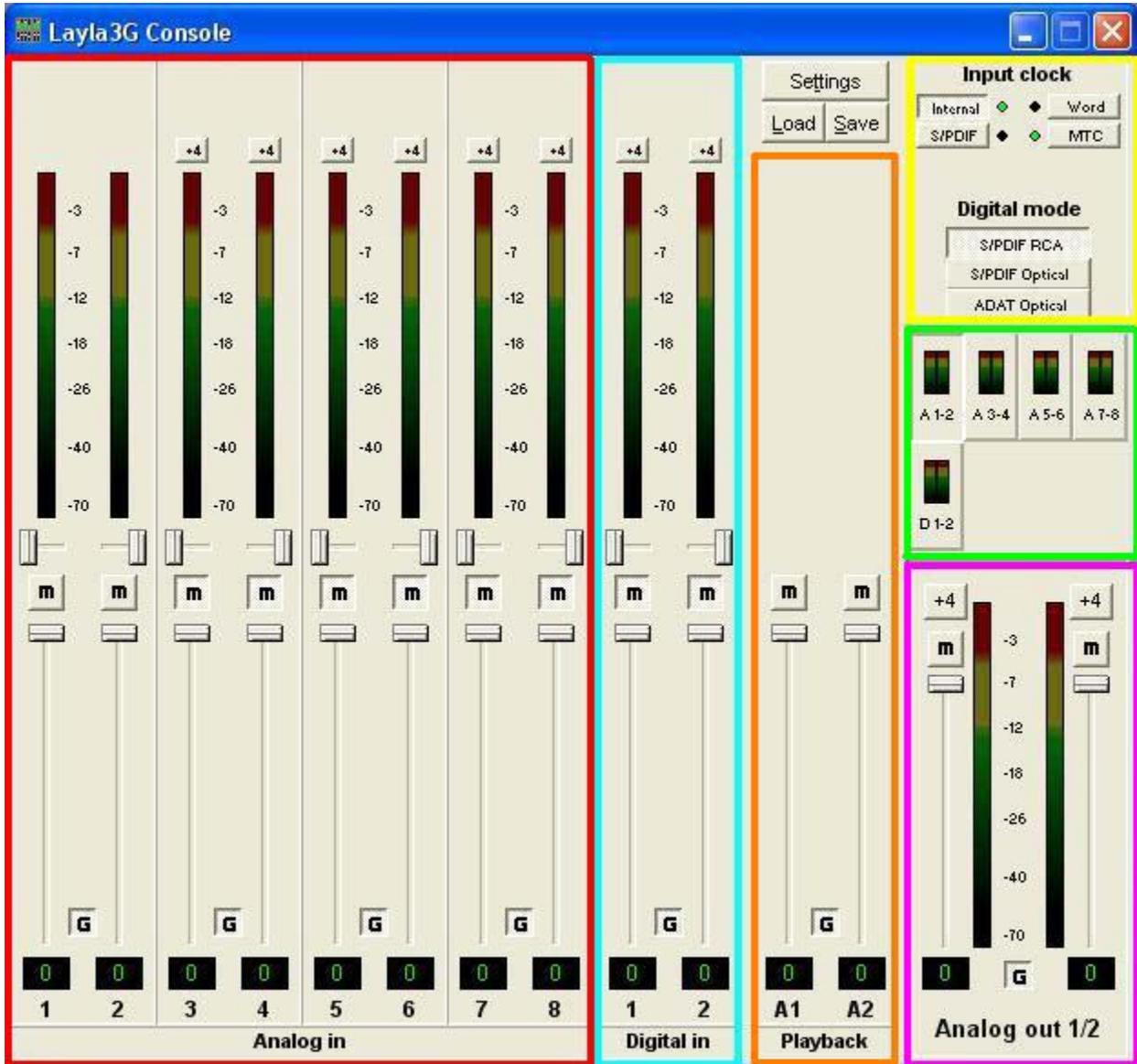
Screenshot of Layla3G Console3 main window – Gina3G will work similarly.

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Bus Select

The output bus select panel is the area outlined in green. Five different output busses are shown here: four analog busses in the top row and one digital bus 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. In this case Analog 1-2 (A 1-2) is the active bus. Clicking on an output bus button simply selects the **settings that are displayed**. This means all input monitor parameters such as volume, pan, and mute **only apply** to the audio coming out **Analog outputs 1 and 2** of Layla3G. To select, view and change the input monitoring and playback settings for another set of outputs you must select a different output bus by clicking on one of the output bus buttons. Alternatively, you can use the function keys F1 through F5. In the example above there is only one digital output bus because the console is in S/PDIF mode. When the console is in ADAT mode, four digital busses will be displayed representing all 8 ADAT channels.

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 +4dBu (professional level) signal. To send a –10dBV (consumer level) signal, click the button. This feature allows you to connect either professional or consumer gear to each output.

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 the left and right channels; 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 temporarily reverses the state of the gang button; that is, if the gang button is down, holding down shift puts you in non-ganged mode and vice versa.

Analog Inputs

The controls for the analog inputs are outlined in red and have the label "Analog in" at the bottom. There is one strip, or *input channel*, corresponding to each of Layla3G's eight physical analog inputs. The input level meters and nominal level buttons remain viewable at all times, whereas the input monitor controls (pan, mute, faders and gang) that are viewable will change depending on which output bus is currently selected.

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 dBu signal. Click the button to change this input to –10 dBV mode. This button will affect the level of what you record as well as the monitor level. Below the "nominal level" button is the peak meter for this channel, labeled in decibels. Once again, these are visible at all times. Please note that there are no nominal level buttons for inputs 1 and 2 since they have trim knobs on the front of the audio interface box.

For Gina3G, you will find a phantom power button labeled "+48v" at the top of the input channel for analog inputs 1 and 2. This button toggles the phantom power on and off for the universal inputs (1 and 2).

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

Below the peak meters are the input monitor controls starting with the monitor pan slider. This control changes how the input channel is monitored between the two channels of the currently selected 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, which controls the input monitor level. Like the pan and mute controls, this does not affect your recording level. Also, clicking on the readout below the fader will allow you to numerically enter a fader setting. The faders also have gang buttons, and their operation is the same as the gang button for the master output bus controls.

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 inputs.

Whether you are monitoring the S/PDIF or the ADAT inputs depends upon the digital mode you have selected. If you are in one of the S/PDIF modes (RCA or optical), the S/PDIF inputs will be digital inputs 1 and 2. If you are in ADAT mode, the console expands to display the eight ADAT inputs as digital inputs 1 through 8.

Playback Strips

The controls for the level of audio being played by an application out the currently selected output bus are outlined in orange and have the label "Playback" at the bottom. These controls operate the same as the input monitor controls, except there are no pan sliders.

Adjusting Playback Volume

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

Input Clock and Digital Mode Settings

The area within the yellow outline on the upper right lets you select your input clock source and digital mode. If a given synchronization source (input clock) is detected, the indicator next to the input clock button will turn green. You will only be allowed to select input clocks that have been detected. In order for a digital input clock (S/PDIF or ADAT) to be detected the console must be set to the appropriate digital mode. Also, you may only select the digital mode if there aren't any applications actively playing or recording. The console defaults to S/PDIF RCA digital mode. To use the S/PDIF Optical I/O or ADAT optical I/O on your audio interface you will need to select the corresponding digital mode. Your Echo product is only capable of using one digital mode at a time.

> **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

Layla3G and Gina3G are designed to work alongside other audio equipment. If you are planning on using Layla3G or Gina3G with other audio equipment, please note the following:

The Windows drivers included in this package support multiple Echo products 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.

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

Word Clock – (Layla3G only) The **Word Clock** connectors can be found on Layla3G's back panel. When Layla3G is set to "Internal" this synchronization clock runs at Layla3G's selected sample rate. Think of it as a kind of electronic metronome, which clicks back and forth at the digital sample rate. It is one of the most widely used forms of synchronization in digital audio. Layla3G 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, Layla3G 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.

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

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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, Layla3G & Gina3G will utilize the synchronization clock that is embedded in the S/PDIF while they decode the bit-stream. Layla3G & Gina3G can only sync to S/PDIF clock if the master device is set to a sample rate between 30kHz and 100kHz. Otherwise you will get noise and/or loss of sync.

> **<u>Note:</u>** 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 – 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. Layla3G & Gina3G can send out (or "master") ADAT clock as well as slave to it. If you want Layla3G or Gina3G to slave to an ADAT clock, just set the input clock to the **ADAT** setting. Layla3G/Gina3G 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.

> **<u>Note:</u>** To preserve the high audio quality of Layla3G/Gina3G'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.

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

Let's start with a simple example. Suppose that Layla3G is the only audio device used in your system. Since you have no other devices to synchronize with, simply select **Internal** for Layla3G's input clock. Layla3G will then use its own clock to control its operation.

Now a little more complicated set-up: You have two Layla3Gs connected. Simply set Layla3G #1 to **Internal** for its input clock. Now connect the first Layla3G to

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the second one via a BNC cable running from Word Clock Out on Layla3G #1 to Word Clock In on Layla3G #2. Now select **Word** for Layla3G #2's input sync. The second Layla3G will slave to the first one and your Layla3Gs 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 six different tabs to choose from: "Driver," "Console," "Digital," "GSIF," "Surround," and "About."

Driver

Here you can do two things: lock the sample rate and tell the driver whether or not to synchronize wave devices. For information regarding the **Advanced…** button, please refer to "Appendix E: PureWave and SuperChannel" towards the end of this document.
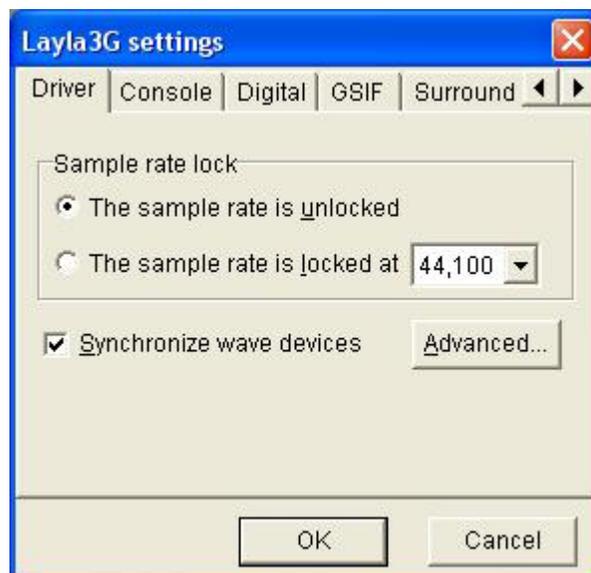
Locking the Sample Rate

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 option is here to support software that wants 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 audio interface and you can't get it to work, try un-checking this box.

Console Sessions

Console settings can be saved as a console session to be reloaded later. To save a console session, click the "Save" button in the console. To load a previously saved session, press the Load button in the console. The current session can be "fast saved" by pressing the F key; this will overwrite the most recently opened session file with the current settings.

By default, a directory with your product name is created, and all sessions for that product are saved to it. Sessions are not cross compatible between different types of hardware; that is, a Gina3G session cannot be used with a Layla3G session. All console settings are saved, including:

- Volume and pan sliders
- Mute, gang and nominal level settings
- Digital mode
- Input clock
- Wave Device Configuration

The various options available in the "Settings" dialog are not saved as part of the session.

Digital I/O

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

Layla3G and Gina3G can transmit digital information in either of two formats: "professional" or "consumer." The primary difference between the two is in the implementation of the SCMS copy protection bit, which, in the Consumer format, prevents the user from making digital copies of a digital copy.

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

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

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

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. The driver that comes with Layla3G/Gina3G fully supports GigaStudio 3, including support for GSIF 2 audio input and low-latency kernel-level MIDI input.
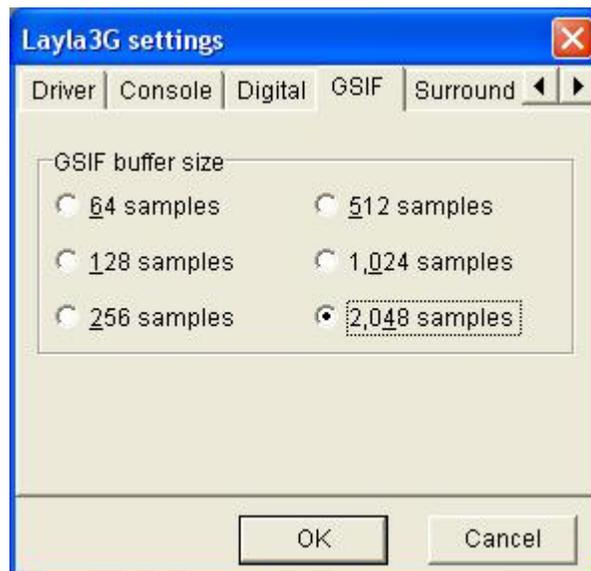
Surround

Here you can find the controls for surround sound delay management. Please refer to "Appendix F: Surround Sound" under the heading "Delay Management" for more information. Windows XP Service Pack 2 may alter some of these settings.
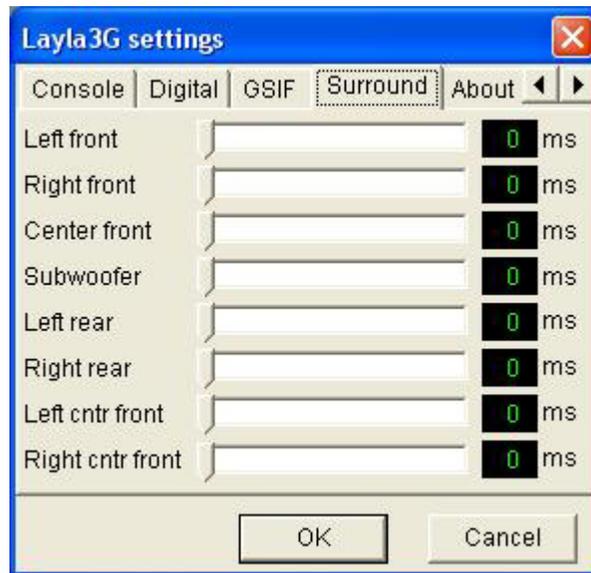
About

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

Console Hot Keys

The console supports several hot key functions:

F1 - F12	Selects the output bus
F	Fast-saves the current session
L	Loads a session
S	Saves a session
T	Opens the settings dialog

Ctrl-click a fader to set to 0 dB
Ctrl-click a pan slider to set to center
Shift-click a fader to negate the gang button
Page Up/Down, Home/End, & up/down arrow keys control the selected fader(s)

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, and check out the "QuickTips" & "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 hardware, and the problem you are having.

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

Thank you for buying an Echo product!

Contacting Mackie Customer Service

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

You can find helpful information about Tracktion and links to user and support forums on the Internet at: http://www.mackie.com/products/tracktion/

If you are still having problems with Tracktion you can contact Mackie Technical Support here: http://www.mackie.com/support/index.html

Appendix A: General Troubleshooting Guide

Problem: When you try to install the drivers from the CDROM, Windows gives you an error message.

Solution: There are several things that could be causing the driver installation to fail. First, if you are trying to install the drivers on Windows 2000 or XP you *must* be logged in as an Administrator. If you are, and you're still getting an error message, then it's probably an issue with InstallShield. What you should do to fix it depends on the error code:

Error code 1607 or 1608

For this error, you should try to set up the Windows Installer separately and see if that works. First, install the base Windows Installer. To do that, download and run one of the following:

Windows Me:
> http://www.echoaudio.com/downloadfiles/InstallShield/instmsiA.exe

Windows 2000:
> http://www.echoaudio.com/downloadfiles/InstallShield/instmsiW.exe

Now, you need the InstallScript engine. Download and run the following:

All versions of Windows:
> http://www.echoaudio.com/downloadfiles/InstallShield/ISScript.msi

If all this is successful, you should then be able to run our installer from the CDROM. If you are still stuck on error code 1608, you may want to try fixing the users' DCOM permissions. You can find directions on this website:

> http://consumer.installshield.com/kb.asp?id=Q108324

Error code 1628

Follow the instructions at the website below:

http://consumer.installshield.com/kb.asp?id=Q108464

Other error codes

The ones listed above are the most common, but if you get a different error code try looking at this list for help:

http://consumer.installshield.com/common.asp?source=all

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 OR you find that one of your other hardware devices is simply not working. 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 PCI 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 PCI 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 are using more than one Echo product at the same time with Tracktion, but the audio playback doesn't sound right.

Solution: This will happen if you have not properly synchronized all of the Echo products together. For more information on how to synchronize your Echo products with each other or with other audio hardware, please refer to "Synchronizing Multiple Devices" in the "Console3" section of this manual.

Problem: You are running Windows 2000 or XP and your computer spontaneously reboots on you.

Solution: You have probably experienced what Microsoft calls a "bug check", but what everyone else calls the Blue Screen of Death (BSOD). The default setting for the BSOD is not to show the BSOD, but to reboot the computer. This isn't very helpful for tracking down problems.   If you are experiencing blue screens, here's how you can help us track it down (these are for Windows XP, but Windows 2000 is similar):

- Select Start/Control Panel/System
- Go to the Advanced tab and click on the "settings" button in the "Startup and Recovery" section
- Uncheck "Automatically restart"
- In the "Write debugging information" section select "Small memory dump" from the dropdown box
- Click OK

Now, next time you get a blue screen, look at it. See if the crash occurred in echo3g.sys; if it did, then it's probably something we need to fix. Restart your computer and find the most recent .dmp file – this is the memory dump. It's probably in C:\WINDOWS\Minidump. Zip up this .dmp file and send it to techsupport@echoaudio.com along with a description of how it happened. This will really help us track down problems. Unfortunately, Windows Me does not save crash dumps.

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 product, 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**. Click on the **Settings** button and select the **Digital** tab. In the window that appears you'll see a pair of check boxes in an area labeled **S/PDIF Format**; one check 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 product 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 product 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 product for playback of any of Windows' system sounds.

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

Solution: Although it may seem obvious, the first thing to check is that there is a physical connection between the device generating the clock and your 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 for your Echo product.

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

Solution: When recording with the S/PDIF input, you must manually select the S/PDIF clock as the input clock. This can be done from 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. Older versions of Windows always set 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, 48 or 96kHZ). You can do this by playing a wave file of the appropriate sample rate through your Echo product's outputs. Then you will be able to sync properly.

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

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

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

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

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

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

Appendix B: 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 product with different software applications at the same time. This way, you can use several different audio applications with just one Echo product. Otherwise, you would have to use a different audio interface for each audio application.

For example, suppose you own both SONAR (multitrack audio) and GigaStudio (software sampler). You can set SONAR to play out through outputs 1 through 6, and set GigaStudio to play out through outputs 7 and 8. With some other hardware products, you have 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.

Q: So what's the catch?

The catch is this: all the software applications using your Echo product 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 SONAR and GigaStudio set up for 44.1 kHz - so far, so good. Then you load a song into SONAR that is set up for 48 kHz. SONAR will change the sample rate of the hardware. Unfortunately, this means that GigaStudio is now also coming out at 48 kHz, so all the sounds played out of GigaStudio 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 your Echo product and write down what audio application is going to use what input or output. Go through all the applications you plan to use this way, 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. Open the "Sounds and Audio Devices" control panel and click the "Audio" tab. Set the default playback device to the cheapo sound card and check "Use only default devices."

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 GigaStudio, Reason, or Absynth. What's the best way to use this with an audio and MIDI sequencer like Cubase or SONAR?

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.

Q: Can I run a DirectSound 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 C: Windows Driver Model (WDM) FAQ

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

WDM stands for "Windows Driver Model." A few years ago 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.

Q: How does the WDM driver talk to the Echo hardware?

The WDM driver was natively developed for Windows 2000 and XP (that is, it's not a legacy driver running on a newer OS). Here's how it works:

<div align="center">

Audio application

⇓

Microsoft's WDMAUD wave driver

⇓

Microsoft's kernel-mode audio mixer (a.k.a. the kmixer)

⇓

Echo WDM driver

⇓

PCI card

</div>
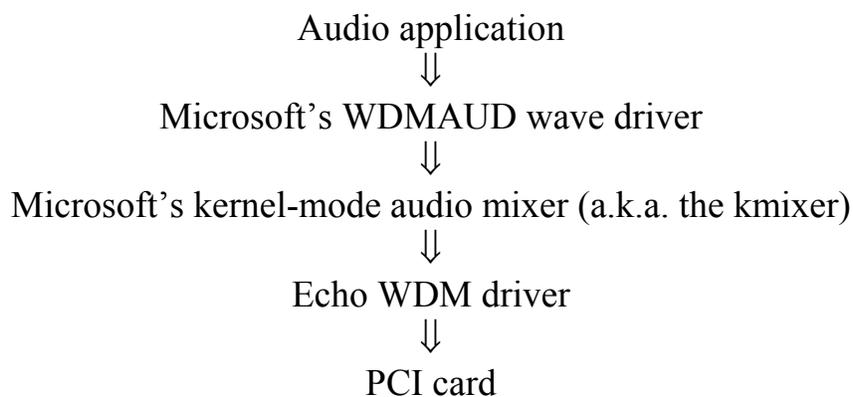
So you can see that there are several layers of software interposed between our driver and the application. This means we don't have as much control over the interaction between our hardware and the audio software as we'd like.

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 in charge of which wave devices get listed and which formats we accept.

That is why we created PureWave mode, which provides 24-bit support for applications that don't work with WDM drivers. For more information on PureWave, please refer to Appendix E: PureWave and SuperChannel.

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.

If your audio application doesn't work in 24-bit mode, you can try PureWave mode, which is documented in Appendix E: PureWave and SuperChannel.

Q: 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_SRCH&SPR=W98SE

Q: Does the WDM driver support the Windows volume control program?

Yes. When you run it (Start/All Programs/Accessories/Entertainment/Volume control), you should see a volume slider in the "Line out" section. This is the same as the "Playback" slider in the Echo console.

Adjusting the "Wave" and "Synth" sliders will set the volume levels for the Windows kernel mixer, not the Echo hardware. These settings are *not* the same as the volume sliders in the console. The console sliders adjust the levels in the hardware; these Windows volume controls are for adjusting the levels in the Windows mixer.

Appendix D: Audio Software Guide

You can use your Echo product with any audio software that supports the standard Windows audio APIs, WDM Direct KS, ASIO, and GSIF. That's a lot of audio software! For most of them, all you need to do is read the software manual to learn how to use them with your Echo product. We'll just touch on a few special subjects here.

ASIO Software

Your Echo product comes with a driver that includes full ASIO 2.0 support.

Help! I don't see all of my outputs or inputs. When you run Cubase SX 2, you may not see some of your outputs. This is probably due to the fact that another program is using them. The most likely culprit is the Microsoft wavetable synthesizer, which is being opened by Cubase as a MIDI output device. Exit Cubase and run the "ASIO Multimedia Setup" program. Click the "Advanced" button, select "Microsoft GS Wavetable SW Synth" and unclick the checkbox to set it as "Inactive". Run Cubase again.

If your outputs still don't show up, you may have some other software that's doing the same thing. Look in the "Setup MME" program for other programs that may be grabbing audio outputs or inputs. In addition, see what other audio software you have running.

Why is my first input monitor always muted when I run Cubase SX 2? This is because we support ASIO 2.0 Direct Monitoring.   Direct Monitoring gives Cubase control over the hardware monitors. When Cubase starts up, it uses Direct Monitoring to mute that monitor.

We realize that this is annoying.   As far as we can tell the concept behind Cubase is that you should have all your monitors muted at first, and then enable them within Cubase in the Channel Mixer window as you prepare to record tracks, etc.

If you don't like this behavior, you can always go to "Devices/Device Setup/VST Multitrack", click on "Control Panel", and uncheck the box labeled "Enable ASIO

2.0 Direct Monitoring". This will cause our ASIO driver to reject any Direct Monitoring commands and prevent the mute on startup. Of course, it also means that you can't use Direct Monitoring.

Using SONAR

Here are the required settings for using SONAR with our driver:

Note: The settings below will need to be reset every time you reconfigure your wave devices.

Go to Options/Audio.

- On the "General" tab, set "Audio Driver Bit Depth" to 24.

- On the "Advanced" tab, we suggest selecting "Trigger & Freewheel"

- On the "Driver Profiles" tab, uncheck "Access Driver In Mono". Be sure to set "Stream > 16 bit data as" to "32 bit PCM, left justified".

You may get a message that your audio devices are not compatible with the specified format; you may need to restart SONAR several times. Make sure each time that the settings are correct; once SONAR starts successfully without the "not compatible" message, be sure to run the Wave Profiler under Options/Audio/General.

To get the smallest possible latency, you should only enable input and output drivers that you actually intend to use.

We recommend that you try using SuperChannel mode in combination with SONAR; using SuperChannel mode can make SONAR more efficient.
You can find more info in Appendix E: PureWave and SuperChannel.

Appendix E: PureWave and SuperChannel

What is PureWave?

While the Windows WDM audio architecture is fine for consumer applications such as games and watching DVDs, it is not entirely suitable for professional audio applications. PureWave is an Echo-exclusive feature that gives your audio applications a more direct connection to your Echo hardware, allowing your audio data to pass through unmodified.

Here's a diagram to illustrate the difference:

WDM   **PureWave**

| Audio Application | → | Microsoft WDM wave driver |
| Audio Application | → | Echo PureWave driver |

| Microsoft Windows kernel mixer |

| Echo24 WDM driver |

| Echo hardware |

| Microsoft Windows kernel mixer |

| Echo24 WDM driver |

| Echo hardware |
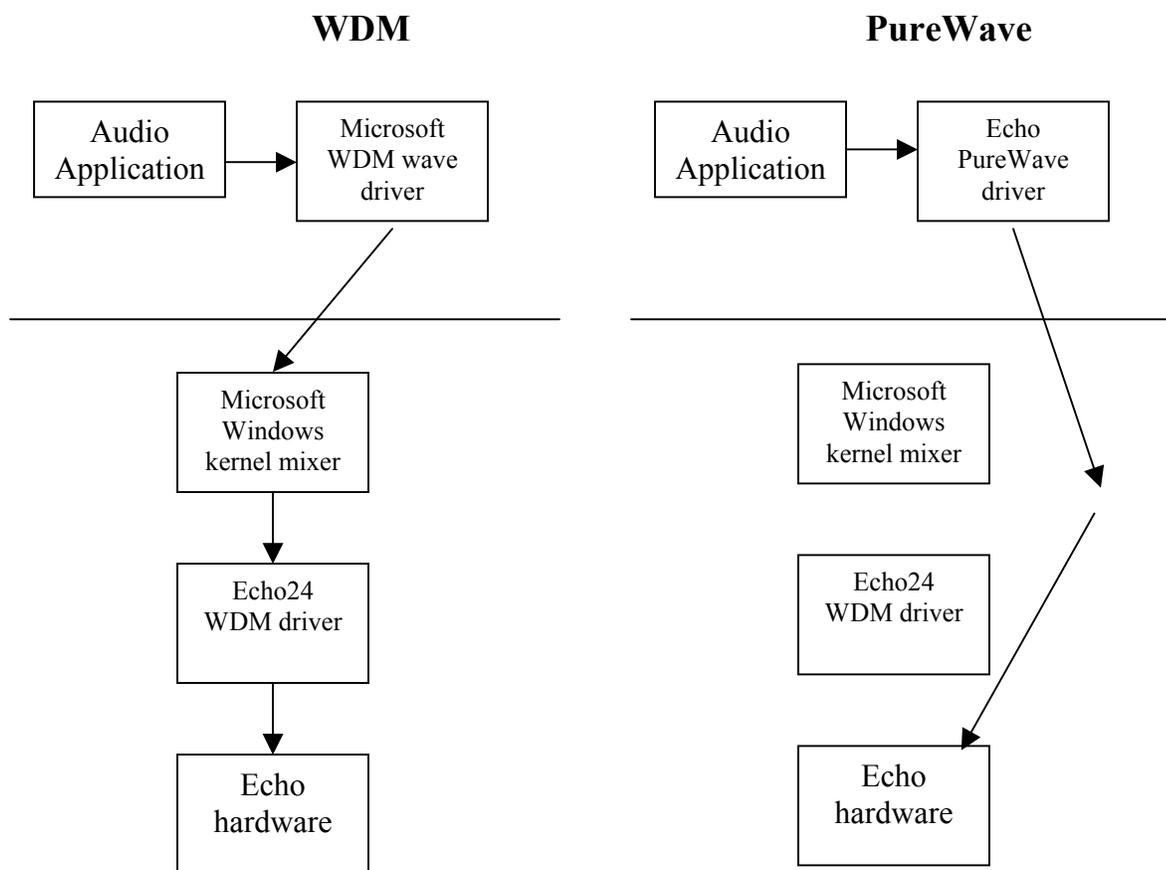
As you can see from the diagram, PureWave mode completely bypasses the Windows kernel mixer to deliver your audio data directly to the hardware.

PureWave versus WDM

With the WDM driver, you can run each wave device in either WDM mode or PureWave mode. Here's a chart explaining the differences between the two.

WDM wave	PureWave
Audio data may be resampled, dithered, or otherwise modified	Audio data is sent to the hardware unchanged
Not all 24 bit applications are supported	All 24 bit applications are supported
30 milliseconds of latency added for both playback and record	Latency may be adjusted to best suit your machine.

Note that this only applies to software that uses the wave/MME API to communicate with audio hardware. Applications that use other APIs, such as ASIO or GSIF, do not benefit from PureWave as they already enjoy a direct connection to the hardware.

Despite all this, you may still wish to use WDM mode. There are applications, most notably Cakewalk's SONAR, that take advantage of WDM Kernel Streaming. WDM Kernel Streaming allows the application to bypass the kernel mixer; in that case, you should definitely use WDM mode.

Surround sound playback applications such as WinDVD or PowerDVD should use the WDM mode; see below for details.

Please note that under PureWave mode, DirectSound is not supported.

PureWave only works in Windows 2000 and Windows XP.

PureWave application guide

The following table lists various professional audio software applications and how best to configure your Echo hardware for these various applications.

Software	Best mode
Acid 3	PureWave
Acid 4 or higher	PureWave or ASIO
Cakewalk Pro Audio	PureWave
Cool Edit	PureWave
Cubase	ASIO
eSkimmer	PureWave
GigaStudio	GSIF
Kontakt	ASIO or PureWave
Logic Audio	PureWave or ASIO
Nuendo	ASIO
PowerDVD	WDM
Reaktor	ASIO or PureWave
Reason	ASIO or PureWave
Rebirth	ASIO or PureWave
Samplitude	PureWave
SAW	PureWave
SONAR 2.1	WDM Direct KS with SuperChannel
SONAR 2.2 or higher	ASIO
Sound Forge	PureWave
Tracktion	ASIO

Vegas 3	PureWave
Wavelab	PureWave or ASIO
WinAmp	PureWave
WinDVD	WDM

This list is far from comprehensive. As a general rule, if your professional audio software offers ASIO or WDM Direct KS (kernel-streaming) modes then you should use those. If it only offers basic WDM mode, then try PureWave. If you are using an application that is not on this list, please contact Echo customer service, and we'll add it to the list.

What is SuperChannel?

SuperChannel is an Echo-exclusive feature that allows you to customize how the various inputs and outputs on your Echo hardware are presented to Windows.

When the WDM driver is first installed, the hardware is presented as a list of stereo inputs and outputs. Take Layla3G, for example: here's the list of wave devices that you'd see in the "Sounds and Audio Devices" control panel:

- Layla3G 1-2 Analog Out
- Layla3G 3-4 Analog Out
- Layla3G 5-6 Analog Out
- Layla3G 7-8 Analog Out
- Layla3G 1-2 Digital Out
- Layla3G 3-4 Digital Out
- Layla3G 5-6 Digital Out
- Layla3G 7-8 Digital Out

As you can see, there is one wave device for each stereo pair of outputs. Inputs are presented similarly.

This works well for many audio applications. However, there are some cases where you don't want a bunch of discrete stereo pairs; you want to be able to group multiple channels together into a single wave device.

Say you want to do 5.1 surround sound playback from a DVD; to do this, you need a single six-channel wave device. For instance, you could set up your wave-out devices like this:

- Layla3G 1-6 Analog Out
- Layla3G 7-8 Analog Out
- Layla3G 1-8 Digital Out

Now you have only three wave devices – a six channel device addressing the first six analog outputs, a stereo device addressing the next two analog outs, and finally an eight channel device for the eight digital outputs.

This is the basic idea of SuperChannel – it allows you to optimally configure your wave devices for your specific application.

When would I want to use SuperChannel?

SuperChannel is useful in the following situations:

- You are using Cakewalk's SONAR or some other application that supports multichannel wave devices. SONAR works more efficiently with a multichannel wave device than with multiple stereo wave devices. We recommend that you decide how many output and input channels you want to use from SONAR and then configure the wave devices accordingly.

- You want to do surround sound playback from an application such as Windows Media Player, WinDVD, or PowerDVD. Please refer to the surround sound setup example, below, for more information.

- You want to hide channels that you aren't using. This is especially useful on Windows 2000, where you can only have ten wave devices. SuperChannel lets you collapse unused channels into a single wave device.

SuperChannel has no effect on software that uses the ASIO or GSIF interfaces. In addition, SuperChannel is not supported for PureWave mode; there's no real need for it.

Wave device configuration

You can use the "Wave device configuration" window to set up both PureWave and SuperChannel mode. This allows you to customize how your audio hardware is presented to Windows applications.

To access the "Wave device configuration" window, open the console, click on the "Settings" button, select the "Driver" tab, and then click on the "Advanced…" button. Here's what the window looks like for Layla3G:
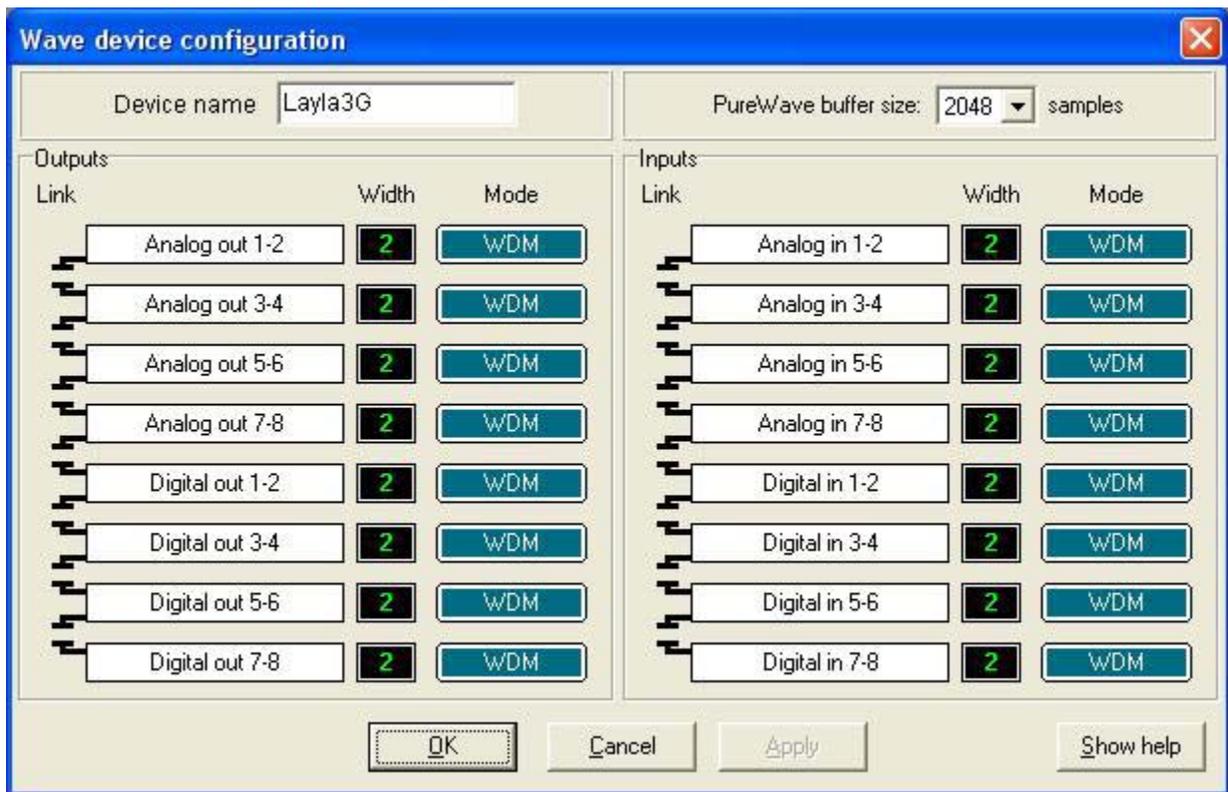
Device Name - In the box labeled "Device name", you can enter a name for your Echo product. This name will be used in applications such as the Windows "Sounds and Audio Devices" control panel.

PureWave buffer size – This box controls the overall latency of the PureWave driver. The lower the value, the lower the latency. You can try different values to find the best possible latency for your system.

The controls found in the **Outputs** and **Inputs** sections of the Wave device configuration window work identically.

Link buttons – Along the left hand side you'll see a group of funny-looking black connectors. Notice that this column is labeled "Link." These are the link buttons; they let you link channels together to form SuperChannel wave devices. Try clicking one of the link buttons; notice that it has changed shape and that two of the rows are now joined together. Click on it again; the two rows are now disconnected.

You can use the link buttons to build your desired set of Super Channel wave devices.

Width indicator – This just shows you how many channels there are in each wave device; these values are updated as you click on the link buttons.

Mode button – The blue mode buttons let you set either WDM or PureWave mode for each individual wave device. Just click the button to change the mode. Note that the interface won't allow you to use SuperChannel and PureWave simultaneously. Selecting PureWave will take the top stereo pair out of the SuperChannel. PureWave is not available on Windows Me.

Below you can see what the Wave device configuration window would look like for the surround sound playback example previously discussed in the SuperChannel section. The wave-out devices have been linked together like this:

- Layla3G 1-6 Analog Out
- Layla3G 7-8 Analog Out
- Layla3G 1-8 Digital Out

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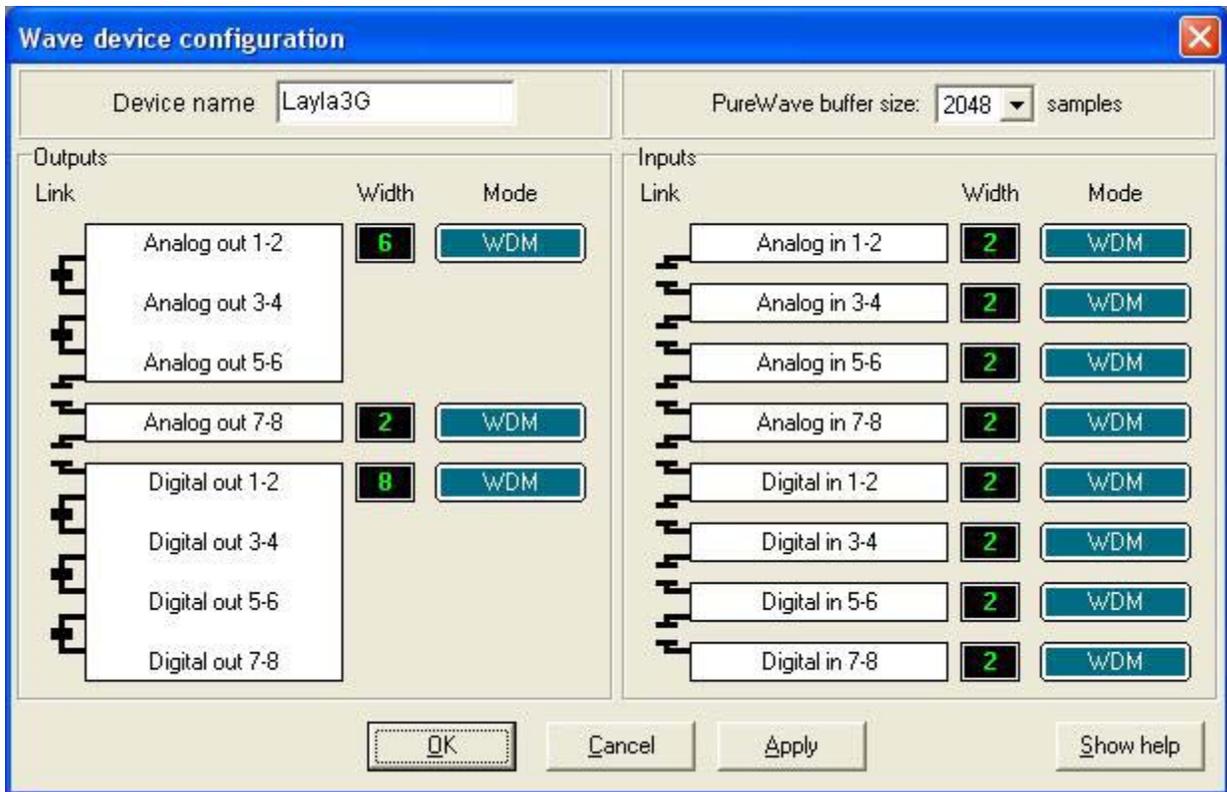
Once you have completed your changes, hit OK or Apply. On Windows 2000 and XP, you will be asked if you want to "Update now", or "Wait until the computer restarts". If you choose to "Update now", your computer will be unresponsive for a period of time after clicking OK while the driver restarts. The delay time will depend on the number of channels you are grouping together. More devices will mean a longer delay.

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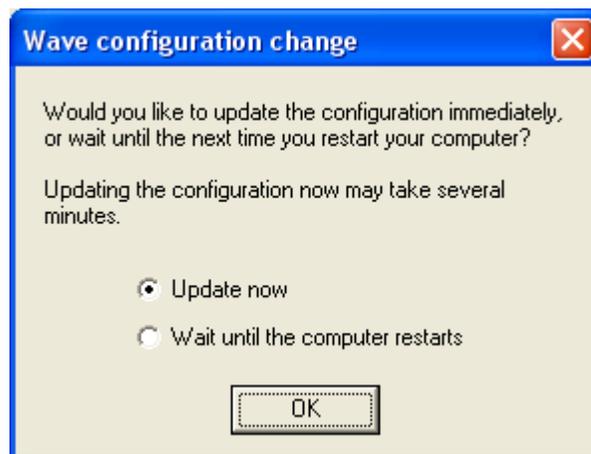
It is not always possible to Update on the fly. If you attempt to update on the fly and it fails you will see this message:
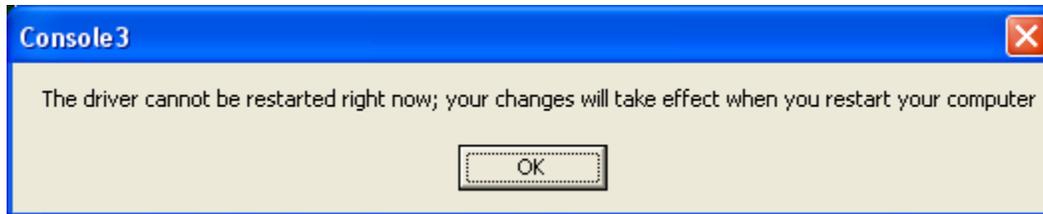
In either case, just close down any open applications and restart your computer and everything will work properly.

For Windows Me, you will need to reboot for your changes to take effect.

Known issues

In some cases only the first six channels of a device set to Super Channel greater than six will play back.

The Windows volume control application does not work when devices are using Super Channel.

For more information on how to use a SuperChannel for surround sound playback please refer to "Appendix F: Surround Sound".

Appendix F: Surround Sound

How to Playback a Surround Sound Mix

You can listen to surround sound mixes without special pro audio software with both Layla3G and Gina3G. You can play surround sound Dolby Digital, Windows Media Audio, and DTS encoded data, among others. The following section will explain how to do this with Windows Media Player 9 on Windows XP. Windows XP Service Pack 2 may change some of these settings.

First you need to be aware of how Microsoft has decided to decode surround sound mixes. While Gina3G can playback up to 6 channels through the analog outputs, and 8 channels via ADAT, Layla3G can playback up to 8 channels via analog or ADAT. This is how Microsoft decodes those 8 channels:

1 – front left
2 – front right
3 – center (front)
4 – sub
5 – rear left
6 – rear right
7 – center left
8 – center right

Both Layla3G and Gina3G allow you to decide how many channels you are going to use for the surround sound mix. Most multi-channel audio interfaces appear to Windows as a series of stereo device pairs. For surround sound playback, Windows wants to see one big multi-channel audio device instead of a bunch of stereo audio devices. This is where our exclusive SuperChannel feature comes in. SuperChannel allows you to change how Windows sees the inputs and allows you to play back surround sound mixes without pro audio software. All you need is Windows Media Player and a surround encoded WMA file.

> **NOTE:** The information and screen shots here pertain to Windows 2000 and Windows XP *without SP2*. Windows XP Service Pack 2 may make some minor changes to these screens and to the options that are available.

For instructions on using the console please refer to the "Console3" section of this manual. For instructions on how to use the SuperChannel feature, please refer to "Appendix E: PureWave and SuperChannel".
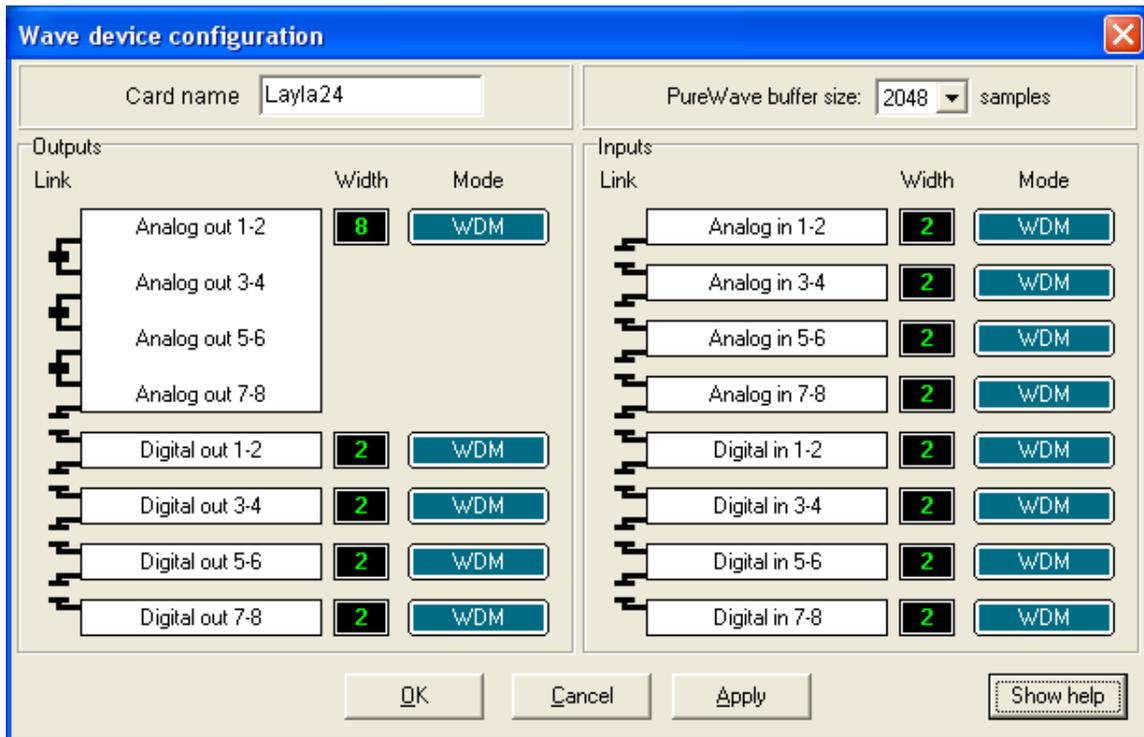

Here we've used SuperChannel to create an 8-channel analog output audio device for playback of a 7.1 surround mix.

You can select the new SuperChannel device from the Windows "Sound and Audio Devices" Control Panel by going to Start – Control Panel and selecting Sound and Audio Devices.

Select the Audio tab at the top of the window and select your newly created SuperChannel audio device from the Sound playback Default device dropdown box. In this example our SuperChannel device is called Layla3G 1-8 Analog Out. Click the Apply button.

Now click the "Advanced…" button in the Sound Playback area to bring up the Advanced Audio Properties window:

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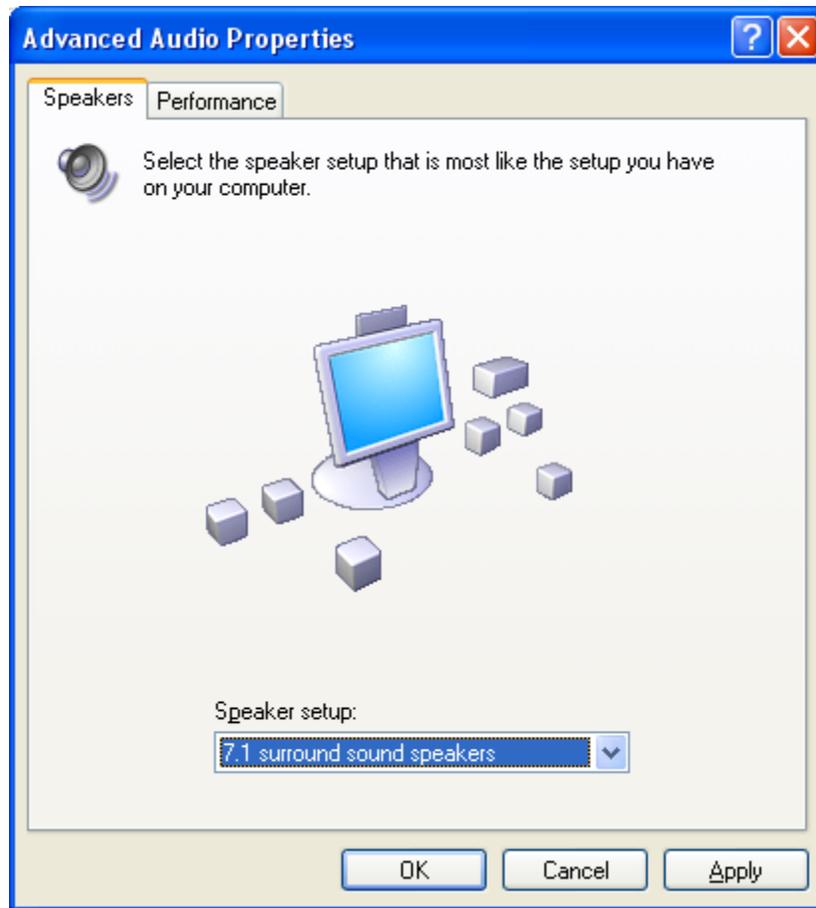
This window allows you to select the speaker configuration you'll be using. Here we've selected 7.1 surround sound speakers from the Speaker setup dropdown box. Click the Apply button and then click OK.

You're all done!! Now all you have to do is start up Windows Media Player 9, play a surround sound encoded WMA file, and enjoy!

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5.1 Surround Sound Example

SuperChannel can also be used to play surround sound Dolby Digital, Windows Media Audio, and DTS encoded data in 5.1 surround sound. Just follow the same steps as above with the following exception: you should create a 6 channel audio device instead of 8, and you should select "5.1 surround sound speakers" instead of 7.1. You can use the same directions above for playback with Windows Media player, but you can also use other DVD player software as well.

For this example we will use Intervideo's WinDVD. After you've followed the directions above:

- Open WinDVD.
- Select "Properties…" by right clicking in the WinDVD playback window or by clicking on the "wrench" button in the WinDVD player.
- Select the Audio tab.
- Select "6 speaker mode (5.1 channel)".
- Click OK.
- You may need to close WinDVD and open it again for the "6 speaker mode" option to be available.
- Verify playback by listening to the speakers or by looking at the output meters in the Echo console.

Other surround sound applications such as PowerDVD will work similarly.

If your speakers are connected correctly, you can now enjoy 5.1 surround sound playback with your Layla3G or Gina3G.

Delay Management

These controls can be found in the console by clicking the "Settings" button and selecting the "Surround" tab. They allow you to adjust the amount of delay added to the playback for each individual surround sound speaker channel. You can move the sliders anywhere from 0 to 250 ms; 0 ms means no delay. At sea level, sound travels at about 1 foot per millisecond (ms), so if your speakers are up to 250 feet away from each other then we've got you covered!

Keep in mind that these delay management controls don't apply to software applications that use WDM Direct kernel-streaming or ASIO, or if you are using PureWave mode. They only work with software that uses the standard Windows APIs (DirectSound, wave/mme, regular WDM, etc), and only with surround sound encoded audio.

NOTE: Windows XP Service Pack 2 may change some of these settings.

Surround Sound over S/PDIF

The WDM drivers support transmitting surround sound-encoded data in either Windows Media Audio or Dolby Digital formats via Layla3G and Gina3G's S/PDIF outputs. The stream will bypass the kernel mixer and will be transmitted with bit-for-bit accuracy. In order to use this feature, follow these steps:

- Select S/PDIF (RCA or Optical) digital mode in the Echo console.
- Select the digital output of your Echo product in the Windows "Sounds and Audio Devices" control panel, under the "Audio" tab.
- In your Dolby Digital playback software, enable digital streaming.

For example, in WinDVD select "Properties…" and under the audio tab select "Enable S/PDIF output". If it is grayed out, double-check the control panel settings, then quit and restart WinDVD.

Appendix G: An Introduction to Digital Recording

Converting Sound into Numbers

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

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

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

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

Figure A. Binary numbers and their decimal equivalents

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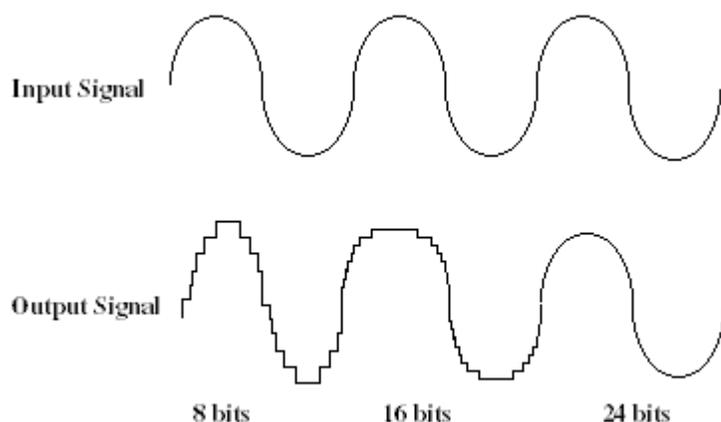

Figure B. The more bits there are available, the more accurate the representation of the signal and the greater the dynamic range.

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

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

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

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

Decibels

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

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

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

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

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

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

Nominal Signal Levels and Headroom

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

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

Unbalanced and Balanced Inputs and Outputs

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

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

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

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

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

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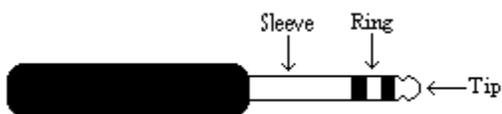
The three sections of a TRS connector are used to transmit the three components of a balanced signal (T = plus, R = minus, S = ground).

Dynamic Range

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

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

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

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

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 product 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 H: Specifications

<u>Layla3G - Audio Performance</u>

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

- Universal Inputs (x2 balanced XLR/TRS):
  - Frequency Response: 10Hz to 20kHz, ±0.1dB
  - THD+n: <0.002% A-weighted
  - Microphone Input Level (balanced XLR):
    - EIN: -130dBu
    - Dynamic Range: 112dB A-weighted
    - Input Impedance: 1.5Kohms
    - Gain Adjustment: +10 to +58dB
    - Maximum Input Level at Minimum Gain: +9dBu
  - Guitar/Line Input Level (balanced TRS):
    - Dynamic Range: 110dB A-weighted
    - Input Impedance: 102Kohms
    - Gain Adjustment: 0 to +45dB
    - Maximum Input Level at Minimum Gain: +18dBu
- Normal Line Inputs (x6 balanced TRS):
  - Frequency Response: 10Hz to 20kHz, ±0.1dB
  - Dynamic Range: 113dB A-weighted
  - THD+n: <0.0015% A-weighted
  - Linearity: ±0.01dB, 0dBFS to 100dBFS
  - Crosstalk: <0.0003%, 20Hz to 20kHz
  - Nominal Input Level: +4dBu or –10dBV
  - Input Impedance: 10Kohms
  - Maximum Input Level: +18dBu

Analog Outputs (x8 balanced TRS):

- Frequency Response: 10Hz to 20kHz, ±0.1dB
- Dynamic Range: 114dB A-weighted
- THD+n: <0.002% A-weighted
- Linearity: ±0.5dB, 0dBFS to 100dBFS
- Crosstalk: <0.0002% @ 1kHz
- Nominal Output Level: +4dBu or –10dBV
- Maximum Output Level: +17.5dBu

Layla3G - Hardware

- Host Interface: PCI bus mastering card
- Two high quality mic preamps with phantom power (+48v)
- Six 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
- Trim control, level meter, and channel inserts for analog inputs 1 and 2
- Headphone output with volume control
- On-board 56-bit / 100Mhz Motorola DSP
- 24-bit data resolution maintained throughout entire signal path
- Multiple sample rates from 8kHz to 96kHz
- Variable sample rate feature lets you select any rate between 25kHz – 100kHz
- MIDI In/Out
- Word clock I/O

Gina3G - Audio Performance

Analog Inputs (x2 balanced universal XLR/TRS):

- Frequency Response: 10Hz to 20kHz, ±0.1dB
- THD+n: <0.002% A-weighted
- Universal Inputs:
    - Microphone Input Level (balanced XLR):
        - EIN: -130dBu
        - Dynamic Range: 112dB A-weighted
        - Input Impedance: 1.5Kohms
        - Gain Adjustment: +10 to +58dB
        - Maximum Input Level at Minimum Gain: +9dBu
    - Guitar/Line Input Level (balanced TRS):
        - Dynamic Range: 110dB A-weighted
        - Input Impedance: 102Kohms
        - Gain Adjustment: 0 to +45dB
        - Maximum Input Level at Minimum Gain: +18dBu

Analog Outputs (x6 balanced TRS):

- Frequency Response: 10Hz to 20kHz, ±0.1dB
- Dynamic Range: 114dB A-weighted
- THD+n: <0.002% A-weighted
- Linearity: ±0.5dB, 0dBFS to 100dBFS
- Crosstalk: <0.0002% @ 1kHz
- Nominal Output Level: +4dBu or –10dBV

Maximum Output Level: +17.5dBu

Gina3G - Hardware

- Host Interface: PCI bus mastering card
- Two high quality mic preamps with phantom power (+48v)
- Six 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
- Trim control and level meter for analog inputs
- Headphone output with volume control
- On-board 56-bit / 100Mhz Motorola DSP
- 24-bit data resolution maintained throughout entire signal path
- Multiple sample rates from 8kHz to 96kHz
- Variable sample rate feature lets you select any rate between 25kHz – 100kHz
- MIDI In/Out

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