



**lexicon**

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12T System User Guide

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## Unpacking and Inspection

After unpacking the 12T system modules, save all packing materials in case you ever need to ship the units. Thoroughly inspect the modules and packing materials for signs of damage. Report any damage to the carrier at once; report equipment malfunction to your dealer.

## Notice

This equipment generates and uses radio frequency energy and if not installed and used properly, that is, in strict accordance with the manufacturer's instructions, may cause interference to radio and television reception. It has been type tested and found to comply with the limits for a Class A computing device in accordance with the specifications in Subpart J of Part 15 of FCC Rules, which are designated to provide reasonable protection against such interference in a residential installation. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause interference to radio or television reception, which can be determined by turning the equipment OFF and ON, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient the receiving antenna
- Relocate the computer with respect to the receiver
- Move the computer away from the receiver
- Plug the computer into a different outlet so that the computer and receiver are on different branch circuits.

If necessary, the user should consult the dealer or an experienced radio/television technician for additional suggestions. The user may find the following booklet prepared by the Federal Communications Commission helpful:

"How to identify and Resolve Radio/TV Interference Problems."

This booklet is available from the U.S. Government Printing Office, Washington, DC 20402, Stock No. 004-000-00345-4.

Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques de la class A prescrites dans le Règlement sur le brouillage radioélectrique édicté par le ministère des Communications du Canada.

This triangle, which appears on your component, alerts you to the presence of uninsulated, dangerous voltage inside the enclosure... voltage that may be sufficient to constitute a risk of shock.



This triangle, which appears on your component, alerts you to important operating and maintenance instructions in this accompanying literature.

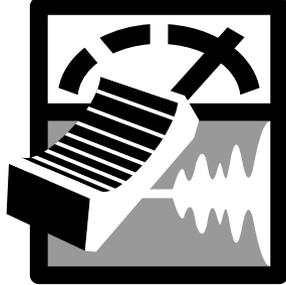
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Lexicon Part #070-12131

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**STUDIO**

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12T System User Guide

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# Table of Contents

## 1 Getting Started

Introduction .....	1
System Requirements .....	1
Versions of Windows .....	2
High Performance Systems .....	2
SCSI vs. IDE .....	3

## 2 Installation

Precautions .....	5
Hardware Installation .....	6
Setup .....	6
Mount the PC-90 Card onto the Core-32 Card .....	6
Attach the PCI Extender .....	5
Install the Core-32/PC-90 Assembly .....	7
Connect the LDI-12T Interface to Audio Sources .....	9
Software Installation .....	10
Install Supported Third Party Audio Software .....	10
Install the ASIO Lexicon Studio Driver for Windows 95 .....	10
Install the PC-90 Software for Windows 95 .	10
System Connections .....	11
Front Panel Connections .....	11
Rear Panel Connections .....	12
Typical Configurations .....	12

## 3 Core-32 Basics

Signal Routing .....	15
Core-32 Control Panel .....	16
Ctrl I/O .....	17
Reverb .....	19
Punch Record .....	22
Configuring the Punch Feature .....	22
The Mix Level control .....	23
Timecode and Synchronization .....	24
Setting Up Cubase to Sync to Timecode .....	25
Setting the Audio Sync Clock Reference .....	25
Sample Rates .....	26
Word Clock .....	26
Word Clock Sources .....	27
Recording .....	27

## 4 Using the PC-90

Basic Operation .....	29
The Interface .....	29
Inserting a Plug-In .....	30
Routing an Audio Channel through the PC-90 .....	30
Selecting PC-90 and Making Settings .....	31
Naming Effects .....	33
Saving Effects .....	33
Loading Effects .....	34
Editing Effects .....	34
Beyond the Basics .....	35
Routing a Plug-In .....	35
Automating the PC-90 .....	39

## 5 Reference

PC-90 Algorithms .....	41	Reverb Level .....	48
Ambience .....	41	Reflect Delays .....	48
Chamber .....	42	Reflect Levels .....	48
Concert Hall .....	43	Shape, Spread .....	48
Inverse .....	43	Size .....	48
Room .....	44	Slope .....	49
The Parameters .....	45	Spin .....	49
Chorus .....	45	Treble Dcy .....	49
Crossover .....	45	Preset Descriptions .....	50
Decay .....	45	Notes on Preset Design .....	50
Decay Lvl .....	45	<i>Appendix A</i>	
Definition .....	45	Third Party Support .....	A1
Diffusion .....	46	Steinberg Cubase VST .....	A1
Duration .....	46	<i>Appendix B</i>	
Echo Delays .....	46	Troubleshooting .....	B1
Echo Feedback .....	46	Installation Troubleshooting .....	B1
Effects Mix .....	46	PC Troubleshooting .....	B2
HF Rolloff .....	46	Optimizing Windows 95 for	
Infinite .....	47	Audio Applications .....	B4
Input Lvl .....	47	Common Problems and Solutions .....	B7
Link .....	47	Poor System Performance .....	B8
Mid RT and Bass Mult .....	47	Pops and Clicks .....	B8
Pre Delay .....	47	<i>Appendix C</i>	
Range and Rate .....	47	Specifications .....	C1



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Lexicon Part # 070-12131

# 1

## Getting Started

### *Introduction*

Thank you for your purchase of a Lexicon Studio system. Lexicon Studio provides a line of professional hardware components for computers that provides uncompromised audio quality and extensive processing capability. This family of products is designed to allow you to tailor your working environment to your needs. Lexicon Studio provides I/O options, DSP, signal routing, and synchronization for industry leading software.

Although written to accommodate both novice and expert users, this manual assumes some familiarity with the software application you have selected to interface with Lexicon Studio. If you have questions concerning the use of your software application, refer to the manufacturer's user manual before using your Lexicon Studio system.

A "Read Me" file containing late-breaking information is provided on diskette with your Lexicon Studio system, along with On-Line documentation for the PC-90.

### *System Requirements*

The following system is recommended as the minimum for working with Lexicon Studio and Steinberg Cubase Audio VST:

- 166 MHz Pentium Processor or compatible equivalent (Please visit our Web site for the latest information on compatibility.)
- 64 MB of RAM
- Audio-capable disk drive
- Microsoft Windows — 95™ or later

## ***Versions of Windows***

Lexicon Studio requires Windows 95 A or B. Studio is not compatible with Windows 95 V0. To determine the version of Windows you are running, follow these steps:

1. In the Control Panel, double-click **System**.
2. Click the General tab.
3. Locate the version number under the System heading, then refer to the following table:

<b>Version</b>	<b>Version of Windows</b>
4.00.950	Windows 95 V. 0
4.00.950A	Windows 95 plus the Service Pack 1 Update, or OEM Service Release 1.
4.00.950B	OEM Service Release 2 (OSR2)
4.00.950C	OEM Service Release 2.5 (OSR2.5)
4.10.1650 Beta 3	Current Beta version of Windows 98

As of the release of this manual, Windows 98 was not released. Lexicon has, however, tested with this version of Windows, and plans to support it on release. Upgrade to Windows 95 A, B or C, as necessary.

## ***High Performance Systems***

Popular software packages like Cubase VST, Logic Audio and Digital Performer all show performance improvements with high CPU speeds and additional RAM. A high performance system benefits from the following:

- Fast CPU — Intel is the best PC choice. CPUs from other manufacturers often don't perform as well at the same rated clock speeds and may have variant implementations of important sections like the FPU (Floating Point Unit).
- A system "boot" drive and a separate drive for audio
- A reliable video card with current drivers. Some cards offer acceleration done in such a way as to "hog" the PCI bus and create data logjams. A computer offering AGP (Advanced Graphic Port) capability for the graphics card is preferable as it gets the video off of the PCI bus.

### ***SCSI Vs. IDE***

Both SCSI and IDE offer distinct advantages:

	<b>Benefits</b>	<b>Liabilities</b>
SCSI	fast bus performance, removable media lots of choices of high performance drives	Use of a PCI card to interface to the computer increases bus traffic
IDE:	drives approach SCSI performance, inexpensive, using the IDE bus decreases activity on the SCSI bus	the IDE bus is slower (in most cases this is NOT the limiting factor) fewer model choices, no removable media

A system that uses SCSI (PCI) for audio I/O transfers, and IDE for data transfers to and from the hard drive, and the AGP port for graphics traffic wins the performance test. Adding a SCSI card and removable media to transport jobs (copying them to and from the internal IDE drive) wins on performance with the currently available technology. Stay tuned on our WEB site for the latest recommendations as technology changes.



# 2

## Installation

Depending on your computer, installation of your Lexicon Studio system should take about 20 minutes. Please read through the entire procedure before performing the installation.

### ***Precautions***

- Turn off and unplug your computer.
- Use the anti-static wrist strap provided with your system and make sure that you are grounded during the entire installation process to prevent static charges that can damage components.
- Put the anti-static bags containing your cards near your computer to avoid walking around with a card after it is removed from its bag (and generating static electricity).
- Before handling any Lexicon Studio cards, discharge any personal static electricity that may be on your clothes or body by touching a grounded metal surface, such as the power supply case inside your computer.
- Handle the cards only by their edges, as you would a CD. Avoid touching the pins on the bottom edge of the cards.
- Save the anti-static bags containing your Core-32 System card and PC-90 module. These prevent static electricity from damaging sensitive electronic components on the cards. Whenever cards are removed from your computer, they should be stored in these bags.

## *Hardware Installation*

### **Setup**

1. Put on the anti-static wrist strap.
2. Place the foam from your Lexicon Studio package on top of a work surface next to your computer. (Any flat piece of static-safe foam with similar dimensions to the Core-32 card can be used).
3. Make sure that your computer is properly shut down and unplugged.
4. Remove the computer top cover and PCI slot cover(s) for the slots you've chosen for your Lexicon Studio cards. If you do not know how to open your computer, consult your computer owner's manual.
5. Touch the computer power supply case to discharge any personal static electricity.
6. Remove the Core-32 from its anti-static bag, holding the card only by its edges. Be careful to avoid touching the pins on the bottom edge of the card. Place the Core-32 bag on top of the foam and place the card on the foam with the white PC-90 connector facing up as shown below.

### ***Mount the PC-90 card onto the Core-32 card***

The PC-90 module can only be mounted onto a Lexicon Studio Core-32 system card. Attempting to attach your PC-90 to any other type of PCI card may damage your PC-90 module.

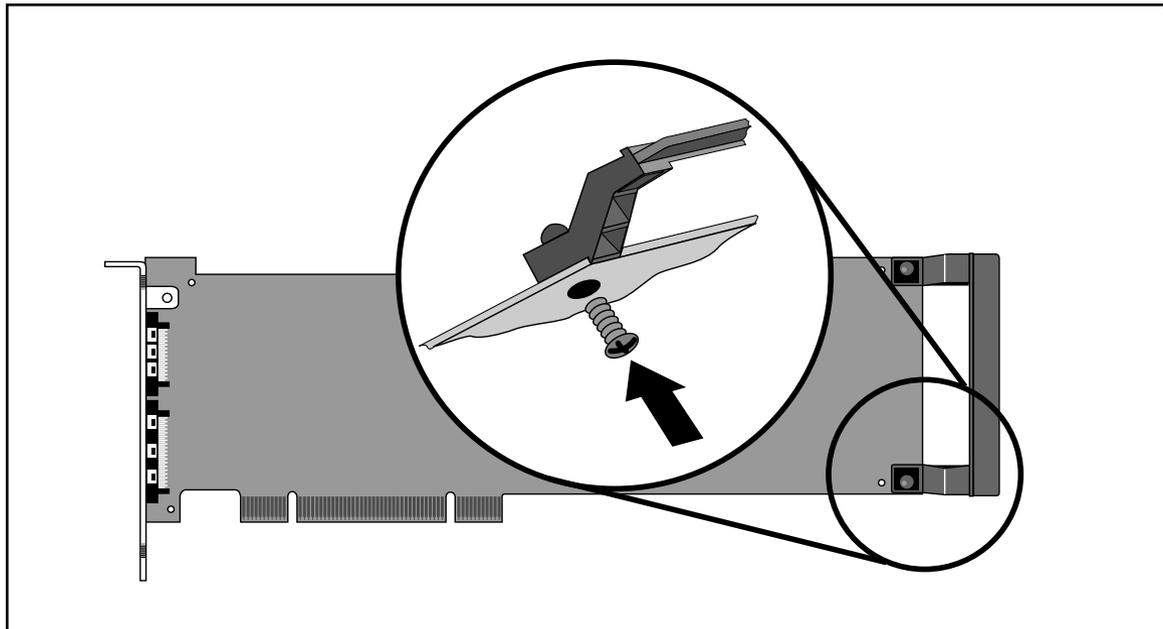
1. Touch the computer power supply case to discharge any personal static.
2. Remove the PC-90 module from its anti-static bag, holding the card only by its edges. Place the bag on your work surface next to your computer and place the PC-90 on it with its white connector facing up.
3. Pick up the PC-90 card and orient it so that the connector side faces away from the palm of your hand.
4. Pick up the Core-32 card in your other hand. Orient the cards with their connector sides facing each other and align the connectors at the top of each card. Firmly press the cards together until they connect fully. The holes on the edges of the Core-32 card should be aligned with the standoffs on the PC-90 card.

5. Once the cards are mated, place the assembly, with the Core-32 card on top, onto the Core-32 anti-static bag. Use the four screws provided to attach the PC-90 to the Core-32 card. Do not overtighten the screws as you may damage the PC board.

### ***Attach the PCI Extender***

The extender provides support for cards installed horizontally and prevents boards from touching each other and shorting. To attach the extender:

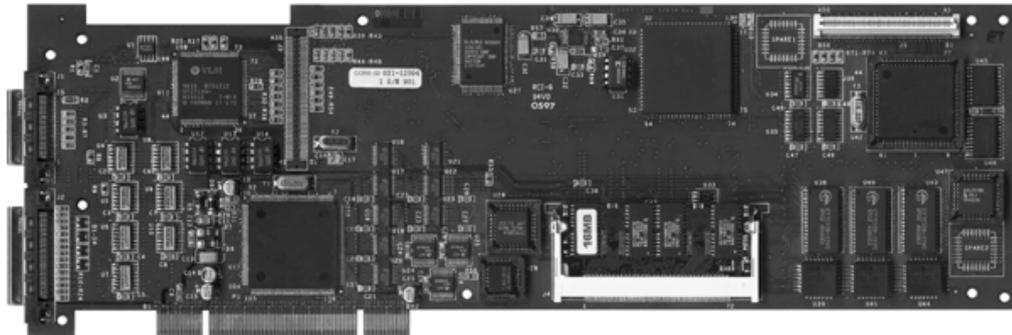
1. Holding the Core-32 assembly by its edges, align the holes of the board and the extender.
2. Insert two screws (provided) from the back of the Core-32 card as shown. Do not overtighten the screws, as you may damage the PC board.



### ***Install the Core-32/PC-90 assembly***

The Core-32 System card must be installed in a full-size PCI slot which is a bus master. Refer to your computer owner's manual to determine which slots are available as bus masters. (On some older motherboards, the PCI slot next to the ISA bus connector [the PCI/ISA shared slot] is not a PCI bus master slot.)

1. Touch the computer power supply case to discharge personal static.
2. Pick up the Core-32/PC-90 assembly by the top of the metal bracket and the top of the other end of the card.
3. Align the card assembly over the slot you've chosen and insert it.
4. Place one hand along the top edge of the card and push down firmly until the card is fully seated in the slot. Do not use excessive force.
5. Once the card is fully seated, attach the Core-32 bracket to the computer chassis with the screws provided with your computer. If you are standing the computer up, check to see that boards are not bending and shorting against each other.
6. Replace the cover on the computer. **Do not operate the system without the cover.**



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## ***Connect the LDI-12T Interface to Audio Sources***

### Cables/Connections

1. Use only the proprietary Lexicon cable (provided) to connect the Core-32 card to the LDI-12T Interface.
2. With both the computer and the LDI-12T powered down, attach the cable between the lower connector on the back of the Core-32 card and the **Computer** port on the LDI-12T rear panel.
3. Connect the 9V connector to the **AC Power** port on the rear panel of the LDI-12T, and plug the other end into a wall socket.

The LDI-12T supports various formats of analog and digital I/O. Make certain your cables are the correct types prior to connection. (Analog XLR connectors are +4dBu balanced (Pin 2 hot) , analog RCA connectors are –10dBV unbalanced). To select a source for input or output within the application software, refer to the documentation provided with your particular software package

## Software Installation

### ***Install Supported Third Party Audio Software***

Lexicon Studio requires compatible front-end software (such as Cubase Audio VST) to operate. Please refer to your particular software installation guide for assistance. As Lexicon Studio installs software within directories of your front-end software, you should install your audio software first and verify that it launches properly before installing Lexicon Studio (even though it will not be audio-capable until installation of the Lexicon Studio system).

### ***Install the ASIO Lexicon Studio Driver for Windows 95***

1. With the Core-32 assembly installed, power up your computer and your LDI-12T interface. Once your computer is fully booted, Windows 95 will identify a new PCI multimedia device (In the event of booting problems, refer to the troubleshooting section of this manual).
2. Insert Disk 1: Lexicon Studio Drivers. Windows 95 will search the floppy drive for the .inf file. Once this is found, click **Finish**.
3. Restart your computer.

In future versions of the Lexicon Studio drivers, the Installer will remove any prior versions of this software found in your system and place all components as required. Following is a list of files installed and their locations:

<b>File</b>	<b>Location</b>
asiolex.dll	C:\LxStudio
Core32.dll	C:\LxStudio
Coredsp.pci	C:\LxStudio
Corefpga.bin	C:\LxStudio
Corev40.bin	C:\LxStudio
Diagv40.bin	C:\LxStudio
lxstudio.inf	Installs as LexiconStudio.inf within C:\Windows\Inf\Other
Core32.vxd	C:\Windows\System
lexmtc.drv	C:\Windows\System

### ***Install the PC-90 Software for Windows 95***

1. Insert Disk 2: PC-90 Software Disk
2. Open the disk and double-click on **Setup.exe** to launch the Installer.
3. Make certain that the Installer is directed toward the desired plug-in folders (example: c:\.\Cubase Audio VST\vstplugins)
4. Click **Next** to install PC-90 software
5. Click **Finish** to complete the PC-90 software installation.
6. Restart your computer

Installing the PC-90 software will file it within the registry. To remove or update this software, remove the PC-90 install by using the **Add/Remove Programs** control panel, highlighting **PC-90 Install**, and selecting **Remove**. This control panel is located at Start/Settings/Control Panel/Add/Remove Programs.

## ***System Connections***

The Lexicon Studio 12T system has the following signal connections.

### ***Front Panel Connections***



- Two (2) analog +4dBu balanced XLR inputs (The stereo input is switchable between the +4dBu output and the -10dBV input pair.)
- Two (2) analog -10 unbalanced RCA inputs
- Two (2) analog +4dBu balanced XLR outputs (pin 2 hot)
- One (1) S/PDIF pair RCA (coaxial) digital input/output
- One (1) balanced XLR Time Code input

## Rear Panel Connections



Two (2) optical TOSLINK™ connectors for input/output connection to 8-channel ADAT-compatible format OR 2-channel Optical S/PDIF format

Two (2) ADAT sync D9 connectors

One (1) Word Clock input BNC connector (75 ohm termination)

One (1) Serial D9 connector

One (1) Power connection

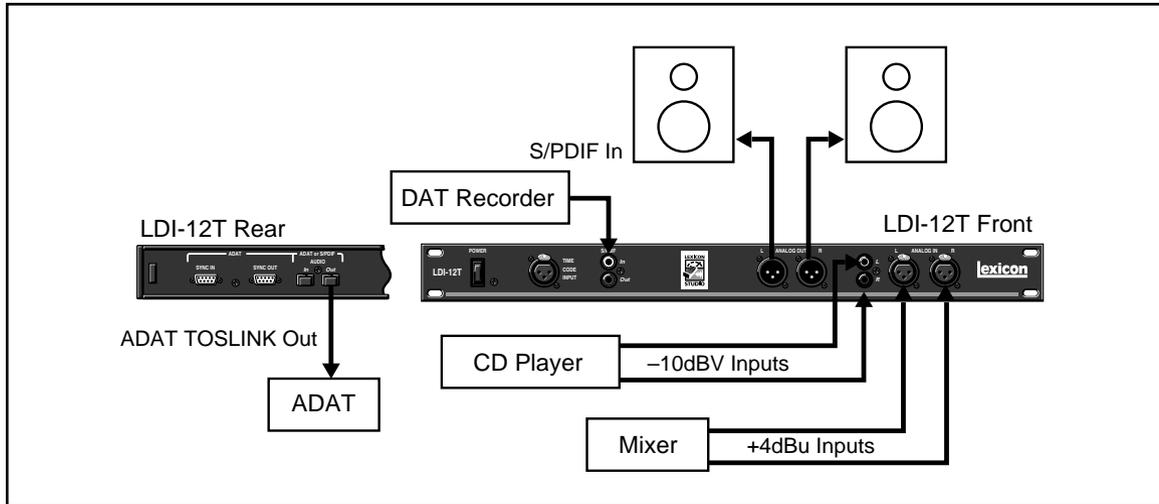
One (1) Multi-pin computer connector

NOTE: The BNC Word Clock Input has a 75 ohm termination. Therefore, if you are distributing Word Clock to a chain of devices, the LDI-12T must be last in the chain.

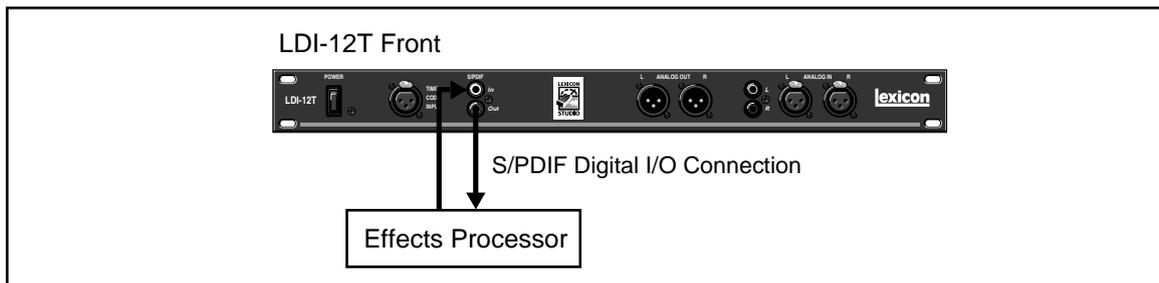
## Typical Configurations

Following are some typical LDI-12T configurations. These are provided as a starting point, and to familiarize you with the possibilities your system provides. Once your system is physically configured, signal routing is controlled by the Lexicon Studio Driver and the routing provided by your host application program (Cubase VST, for example). Use of the Lexicon Studio Driver and examples of typical host application routing is covered in the next section, *Signal Routing*.

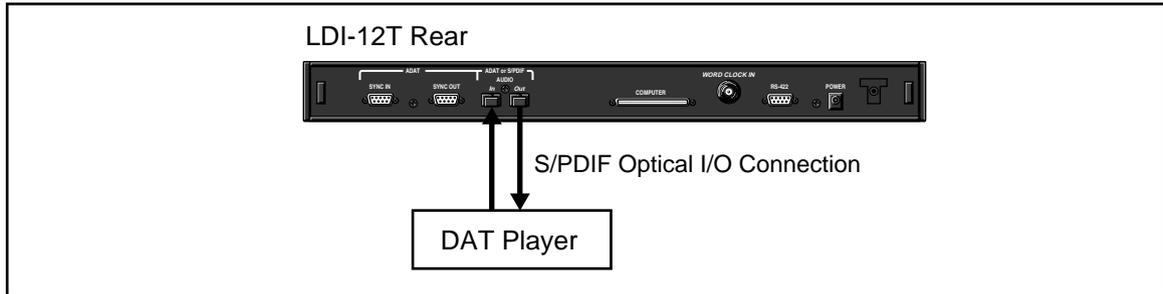
The configuration shown below has the analog outputs connected to speakers for monitoring, S/PDIF to a DAT for mixing, and ADAT connections for multi-channel recording. The XLR analog inputs could be from a mixer or microphone/instrument pre-amp, or the RCA inputs could be used for low-level devices such as a CD player or cassette deck.



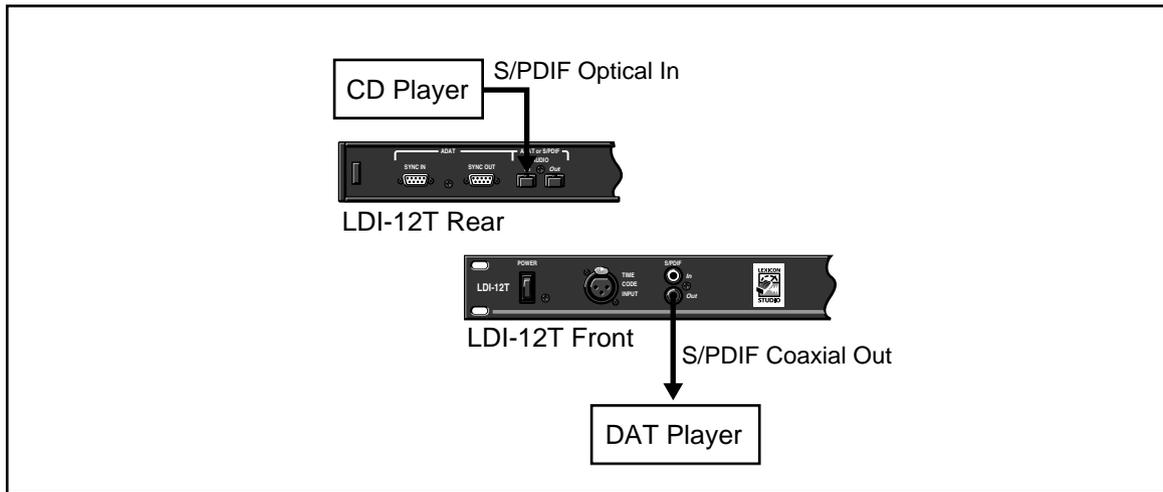
S/PDIF connections can be made to any digital device such as the send/returns of a digital signal processor.



Optical S/PDIF connections are provided as an alternative to the normal 8-channel ADAT format.



The LDI-12T can accept different input and output formats as shown below. This configuration can be used for format conversion from optical S/PDIF to coaxial (RCA) S/PDIF and to provide flexible routing.



# 3

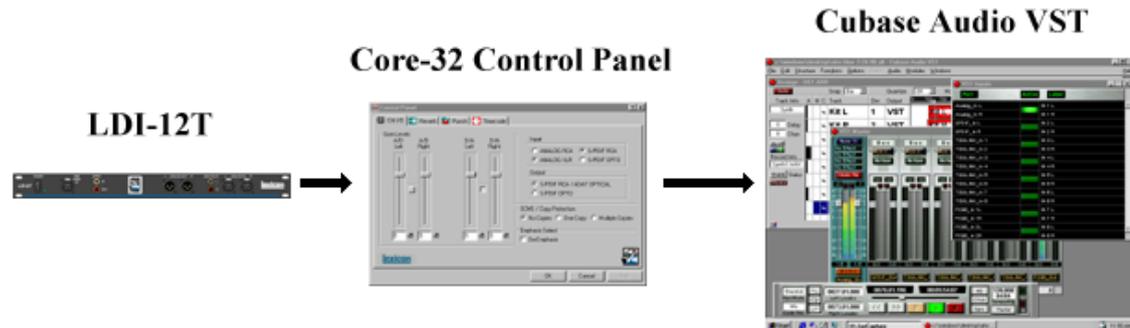
## Core-32 Basics

### Signal Routing

Using Cubase VST as an example of a host application, the following things determine signal routing.

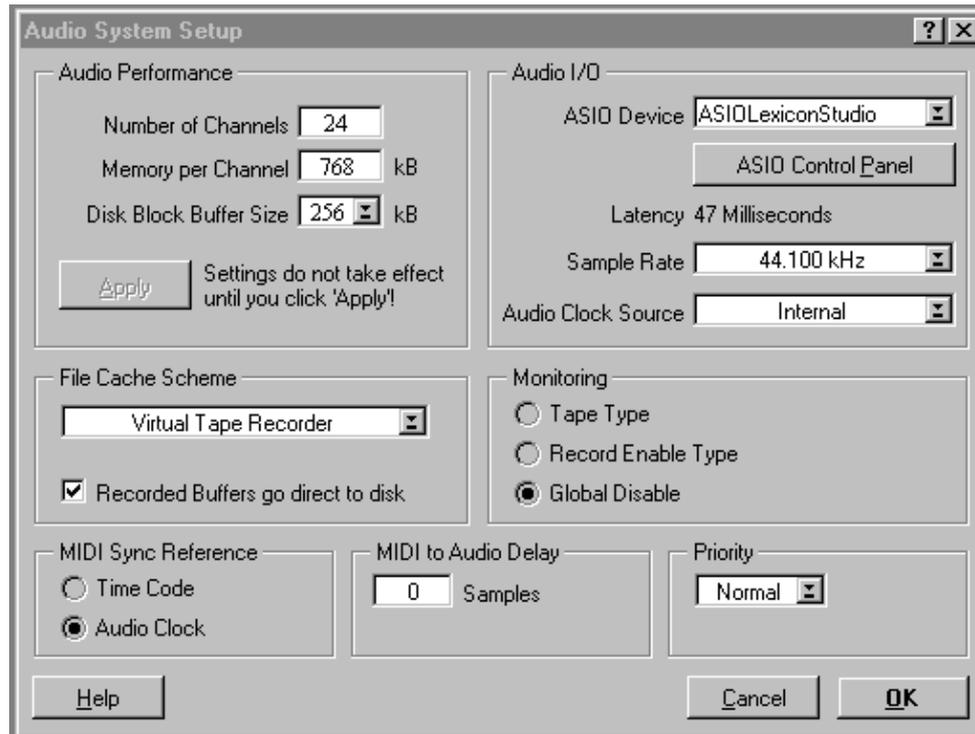
1. LDI-12T connections.
2. Configuration of the Core-32 Control Panel.
3. Inputs and outputs selected and enabled in Cubase Audio VST.

Typically, all connections will appear directly in the application as available inputs and outputs along with the Core-32 Control Panel routing for the PC-90 and Punch Record. Details on these are covered in the following section.



## Core-32 Control Panel

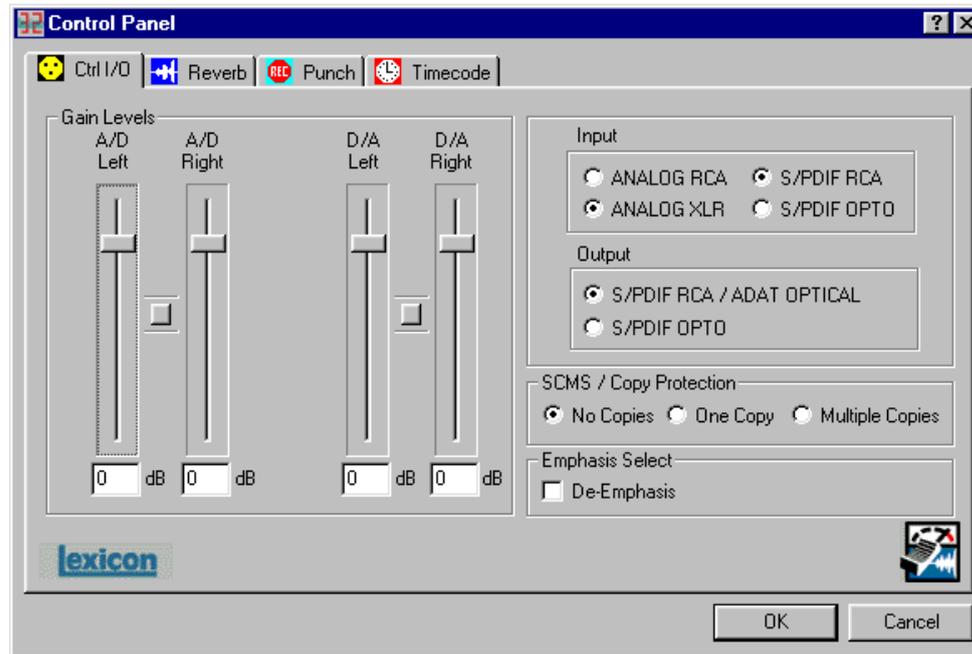
Several features of your Lexicon Studio System are set up for the specific program you are using in the Core-32 Control Panel. To access the Control Panel, pull down the **Audio** Menu, select **System** and click on the **ASIO Control Panel** button.



The method of selecting **Tab**s for different pages, will be familiar to users of Windows and other Microsoft products. The selections are: Ctrl I/O, Reverb, Punch and Timecode

## Ctrl I/O

Items on the **Ctrl I/O** tab control Input/output converter gain, Digital I/O format, Digital emphasis/de-emphasis and SCMS (Serial Copy Management System) copy protection code



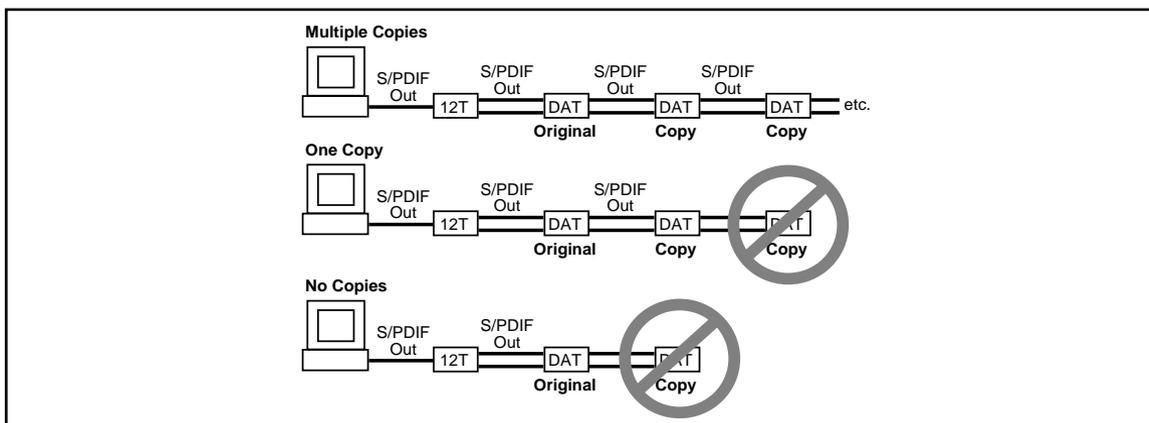
**Gain Levels** are controlled by the on screen faders with the gain displayed numerically at the bottom. A gain of 0dB sets unity gain from XLR input to XLR output, where +4dBu is 14dB below digital full-scale. (For the RCA inputs, 0dB corresponds to -10dBV, 14dB below digital full-scale.)

Clicking the button between fader pairs links them as a stereo pair for convenient identical gain setting. Control clicking on the **Link** button will link the fader heads to one another while maintaining their relative levels. Clicking with the mouse on the fader head while holding down the **ctrl** key on the computer keyboard will set the selected fader or stereo grouped fader to 0dB. Gain levels can also be entered numerically by double-clicking on a numeric field, then entering a value.

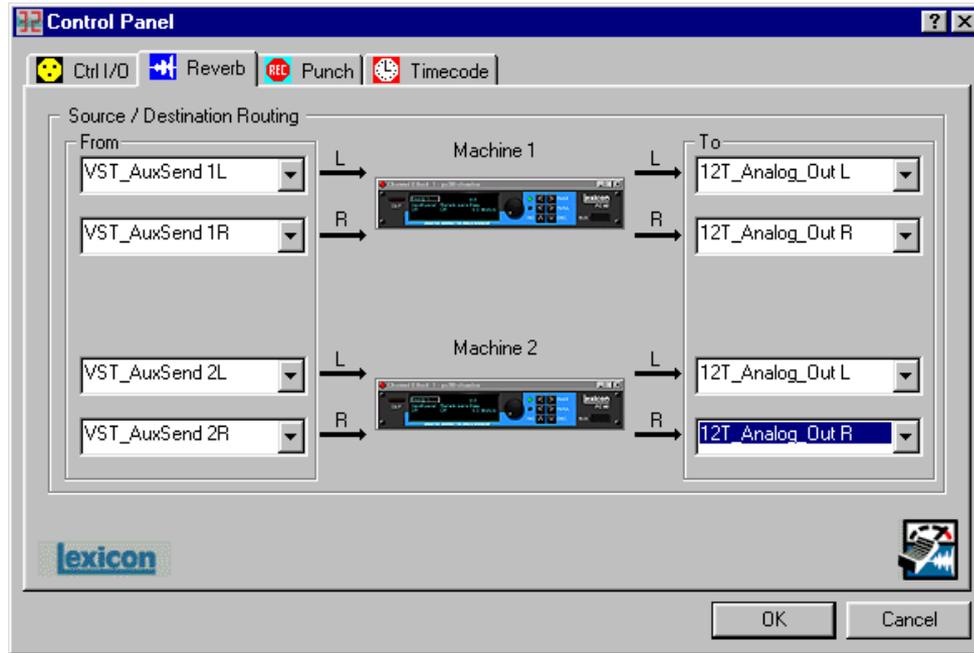
**Input/Output** controls allow you to select S/PDIF or ADAT as the digital format for the optical input and output connections, and to select between XLR or RCA connectors for the analog inputs. Conversion between Optical and Coaxial (RCA) S/PDIF is possible by selecting Optical input and Coaxial output, or the reverse. When Optical S/PDIF output is chosen, the RCA output remains active, creating a digital Y-cord or signal splitter. (NOTE: this is not true of the inputs —both S/PDIF inputs *cannot* be used at the same time.)

**Emphasis Select** enables de-emphasis to be applied to the analog output. This function applies only to the analog converter, and does not affect signals from the S/PDIF or ADAT interface.

**Security/Copy Protection** applies SCMS (Serial Copy Management System) coding to the S/PDIF output. Choose **Multiple Copies** to encode the output for reproduction, **One Copy** to allow a single digital 1-to-1 copy to be made from the S/PDIF output, or **No Copies** to prohibit any copies.



## Reverb



The reverb page allows you to route the PC-90's two DSP engines independently through any of the connectors of your audio interface.

Each PC-90 engine has two input sources (left and right), and two output destinations (left and right). To assign a source to any of the input channels in your PC-90:

1. Click the left mouse button on the down-arrow of an input selector. A menu will open, displaying the sources available to route into your PC-90.
2. Use the scrollbar to view all of the sources.
3. Click on a source to route it as an input source. The input selector will display the selected input source.

NOTE: Input sources must be routed one at a time.

To assign a destination to any of the output channels in your PC-90:

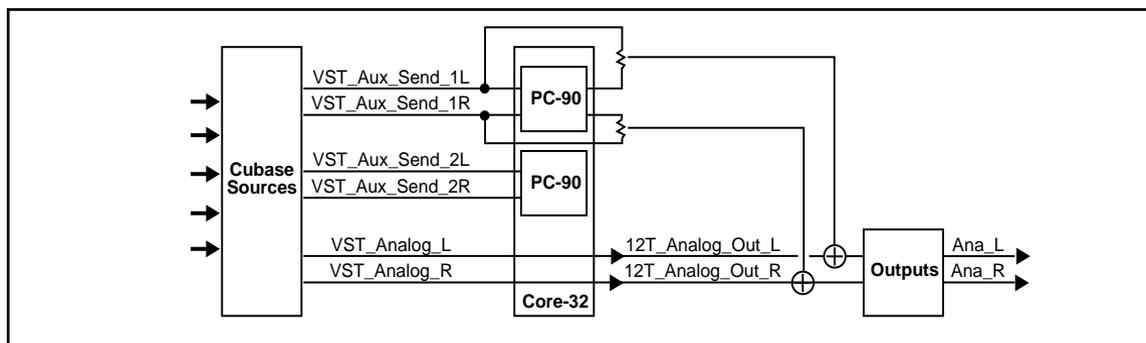
1. Click the left mouse button on the down-arrow of an output selector. A menu will open, displaying the destinations available to your PC-90.
2. Use the scrollbar to view all of the destinations.
3. Click on a source to select it as an output destination. The output selector will display the selected output destination.

Following are lists of each source and destination in the PC-90 Reverb Control Panel:

<b>Source</b>	<b>Description</b>
<Clear All>	Clears all source assignments
12T_Analog_In L & R	Patches the LDI-12T analog inputs into the PC-90
12T_SPDIF_In L & R	Patches the LDI-12T S/PDIF inputs into the PC-90
12T_ADAT_In 1 - 8	Patches the LDI-12T ADAT inputs into the PC-90
AuxRet 1L & R	To cascade PC-90 engines
AuxRet 2L & R	To cascade PC-90 engines
VST_Analog_Out L & R	Patches the sum of the Cubase analog bus out into the PC-90
VST_SPDIF_Out L & R	Patches the sum of the Cubase S/PDIF bus out into the PC-90
VST_ADAT_Out 1 - 8	Patches the Cubase ADAT bus out into the PC-90
VST_AuxSend 1L & R	Patches Cubase Aux Send 1 into the PC-90
VST_AuxSend 2L & R	Patches Cubase Aux Send 2 into the PC-90

Destination	Description
<Clear All>	Clears all destination assignments
VST_Analog_In L & R	Sends the PC-90 out to the analog in on the VST Input page
VST_SPDIF_In L & R	Sends the PC-90 out to the S/PDIF in on the VST Input page
VST_ADAT_In 1 - 8	Sends the PC-90 out to the ADAT in on the VST Input page
VST_AuxRet_In 1L & R	Sends the PC-90 out to Aux 1 in on the VST Input page
VST_AuxRet_In 2L & R	Sends the PC-90 out to Aux 2 in on the VST Input page
12T_Analog_Out L & R	Patches the PC-90 output to the LDI-12T analog outputs
12T_SPDIF_Out L & R	Patches the PC-90 output to the LDI-12T S/PDIF outputs
12T_ADAT_Out 1 - 8	Patches the PC-90 output to the LDI-12T ADAT outputs

The following diagram shows a flow chart of the audio routing from the picture of the control panel above. Note that the PC-90 receives its input from Cubase VST via the **VST\_Aux\_Send\_1L** and **VST\_Aux\_Send\_1R** sources. These sources route any audio assigned to that bus into the PC-90.



Note also that, as **12T\_Analog\_Out\_L** and **12T\_Analog\_Out\_R** are selected as PC-90s destinations, the audio is summed with the outputs of Cubase VST, which are also routed to the analog outputs. This allows monitoring of PC-90 reverb and all audio from VST through the same outputs of the interface.

For more information on routing and control of your PC-90, see *PC-90 Basics: Basic Operation* later in this manual.

## *Punch Record*

This page lets you select Input sources for Punch or overdub recording and Output monitor selections for the sources selected for Punch or overdub recording. The Punch feature lets you perform overdub or punch recording by directly connecting to a Lexicon Studio interface — No external mixer is required.

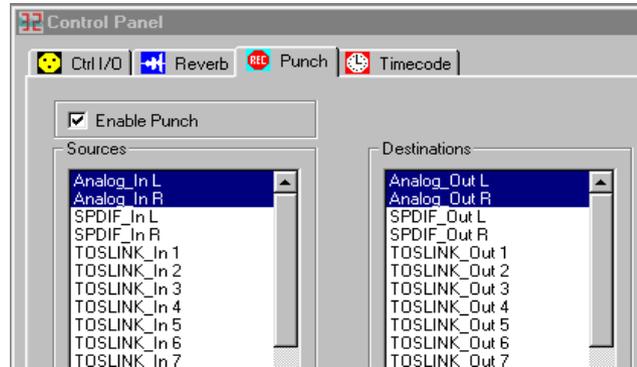
The Windows 95 operating system has a delay when applications use host processing. (You can observe the latency of your particular computer configuration on the System Setup page in Cubase.) This means that, when playing along with previously recorded tracks, the track you are playing will sound delayed while you are performing. This delay can be overcome by using an external mixer, or by using the Punch feature.

The punch feature mixes the input signals directly with the outputs selected for your monitor system (speakers or headphones) to avoid the inherent processing delay from the card to the computer application and back.

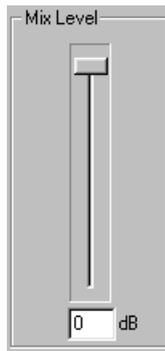
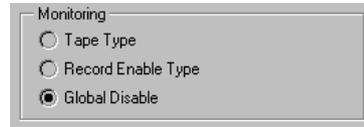
Unlike monitoring through a tape deck the input signal is monitored *all the time*. (With tape, the input is typically monitored when the transport is in Stop or Record and muted during Play.)

### ***Configuring the Punch feature***

1. Select **Enable Punch** to turn on monitoring.
2. Select as many input sources as you like, using the **shift** or **Ctrl** key and mouse for multiple selections.
3. Select the outputs you are using for monitoring. (Often the analog left and right outputs will be connected to an amp and speakers — digital connection to a mixer is, of course, also possible.) A mono mix of the selected inputs will be sent to the outputs.



4. Select **Global Disable** from the **System Setup** dialog to turn off the monitoring through the application so that a delay is not heard from the input signal.



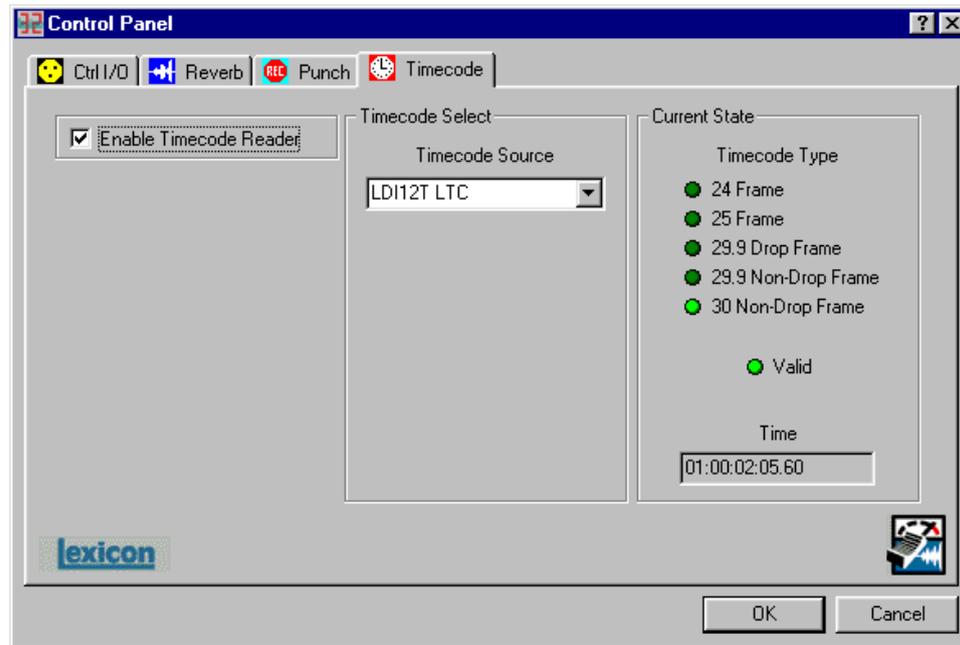
### ***The MIX Level Control***

**Mix Level** allows you to adjust the volume of the selected Punch sources that are mixed with the VST output assigned to the same outputs (Usually Master).

**NOTE:** If multiple sources are being summed, the **Mix Level** may need to be lowered to avoid distortion. In such cases, try an initial setting of 40.

## Timecode and synchronization

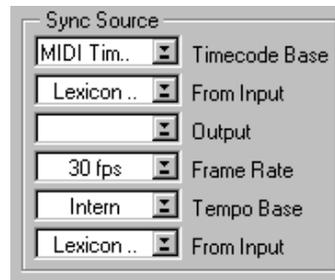
On this tab you can enable Timecode reading, select the Timecode source and display the Timecode type, validity and number. As these controls are independent of the host application, they can help you set up your software for proper frame rate and start time.



### ***Setting Up Cubase to Sync to Timecode***

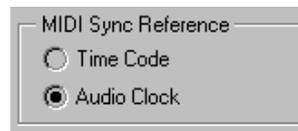
The LDI-12T uses a MIDI driver to convert the Longitudinal Timecode (LTC) supplied by the front panel XLR to MIDI Timecode (MTC).

1. In the **Options** menu, select **Synchronization** and set **Timecode Base** to **MIDI Timecode**.
2. From the menu just below Timecode base, select the input which is receiving MTC. Cubase VST will recognize time code synchronization signal from the selected port.
3. Set **Tempo base** to **Intern**. This tells Cubase VST to not expect MIDI Clock signals.
4. Use the pop-up in the dialog box to specify the Frame Rate of the incoming code.



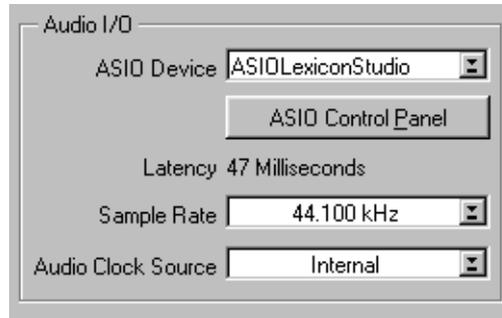
### ***Setting the Audio Sync Clock Reference***

In the Audio System dialog, set **MIDI Sync Reference** to **Audio Clock** so that, when Cubase VST plays back, the MIDI part of the program is internally synchronized to the digital audio.



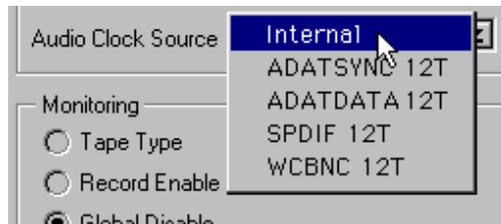
## Sample rates

Sample rates of 44.1 and 48 kHz are selected from the **Audio I/O** pull down menu.



## Word clock

The word clock sources available on your Lexicon Studio interface (**Audio Clock Source**) are also selected from the **Audio I/O** menu.



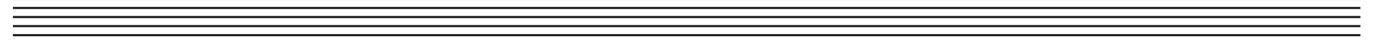
### ***Word Clock Sources***

INTERNAL	Selects the word clock generated by Cubase as the system master.
ADATSYNC 12T	Selects the word clock provided on the ADAT Sync connector as the system clock.
ADATDATA 12T	Selects the word clock provided on the ADAT optical data connector as the system clock.
SPDIF 12T	Selects the word clock provided by the S/PDIF signal (RCA or optical) as the system clock.
WCBNC 12T	Selects the word clock provided on the Word Clock BNC connector as the system clock. This connector provides 75 ohm termination to the Word Clock chain.

### ***Recording***

Generally, the system is locked to the incoming word clock source. For example, if you were transferring 8 tracks from an ADAT, you would select the ADAT data as the word clock source. The same would be true for an S/PDIF source. With analog input, the internal setting is used, as there is no incoming clock.

In larger systems with a digital mixer (such as Yamaha O2R or O3D), it is convenient to use the word clock output and connect it to the BNC word clock input on the LDI-12T, selecting **WCBNC 12T** as the word clock source. This provides centralized clocking of the connected units.



# 4

## Using the PC-90

### *Basic Operation*

The PC-90 combines the quality and power of Lexicon reverb on dedicated hardware with the ease and elegance of a software plug-in interface. Unlike software-based plug-ins, PC-90 provides outstanding sonic quality with no burden on the system performance of your computer's processor.

### *The Interface*



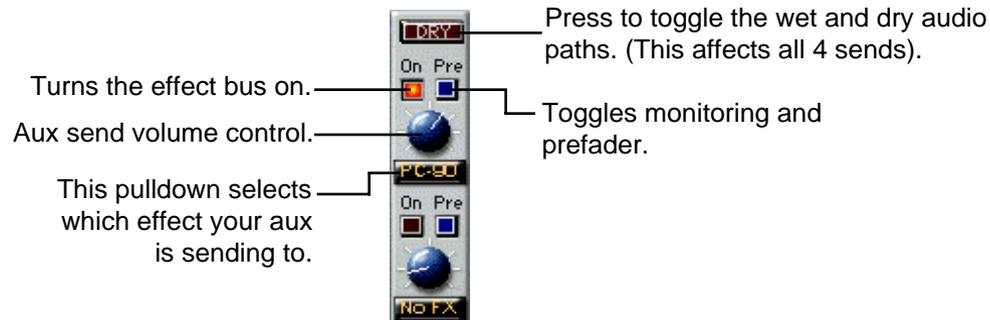
- **Clip** Indicates audio overload in the PC-90.
- **Rotary Encoder** The main control for adjusting a selected value.
- **Page Buttons** Scrolls three-at-a-time through the available parameters of the current algorithm.
- **Parameter Buttons** Scrolls through the available parameters of the current algorithm.
- **Increment/Decrement** Adjusts the value of a selected parameter by a single value. Very useful for fine manipulation of a parameter.

## Inserting a Plug-in

The interface for the PC-90 is available as a plug-in. Plug-ins are accessed differently in different software applications. To open a PC-90 plug-in in Cubase VST, you need to follow these steps:

## Routing an Audio Channel through the PC-90

1. Open the Monitor window in the **Audio** menu.
2. Click on the **EQ** button for the audio channel where you want effects. This will open the selected channel's EQ and effects bus.



3. Click on the pop-up menu below the Effect Send knob, to route the effect to an effects processor ( PC-90 Machine1 or PC-90 Machine2).
4. The available choices depend on selections made in the Effects window (described later in this chapter). However, the first four items correspond to the four internal effects while the following items route the effect sends to buses.
5. Make sure the **Dry** button (Click on/click off.) When this button is pressed, all four effect sends are deactivated for the channel.

6. Click **On** for each effect send you want to activate and turn the corresponding Send Level knob to a moderate value.
7. If you want the signal to be sent to the effects before the faders, click on the **Pre** button for the send.
8. With Pre-fader effect sends, the amount of effect for the channel is not affected by the volume fader. With Post-fader effect sends (**Pre** button not pressed), the amount of effect is proportional to the channel volume, and will change with the volume fader movements.

The next step is to select programs and set the parameters for the effect processors. Since you will probably need to adjust the send levels while doing this, leave the EQ window open.

### Selecting PC-90 and Making Settings

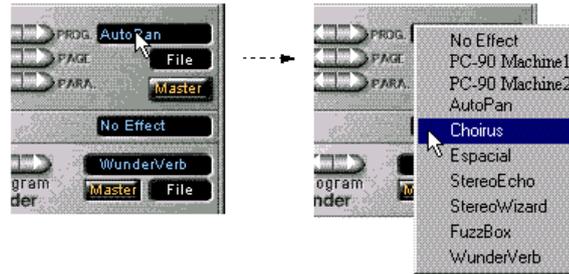
Effect types and programs are selected in the **Effects** window:

1. Pull down the **Audio** menu and select Effects. This window resembles an effects rack, with separate processors arranged on top of each other.



2. Activate an effect processor by clicking on its **Power** button.
3. Make sure the effect output is routed to the desired source, by checking the bus pop-up in the lower right corner. The default setting for this is **Master**.

Pull down the pop-up menu in the processor's upper right corner and select PC-90 Machine 1 or 2.



Selecting a PC-90 machine in this pulldown will activate a PC-90 Rack Xpander within VST.



1. Select other effect programs by clicking on the left or right **Program** buttons. These controls will allow you to cycle through all of the factory presets for the current algorithm. (You may want to set up a cycle and have the audio channel play back while you are making settings, to hear the effect of your selected effect programs).
2. Use the Send Level knob in the EQ window to control the amount of effect for the audio channel.
3. Click the **Edit** button to activate the PC-90 interface.
4. Use the Aux Send knob assigned to the PC-90 to set the input level to the PC-90. High values may cause clipping (distortion) in the PC-90. If the output of the PC-90 is clipped, the CLIP indicator on the Rack Xpander will light.

NOTE: As the Cubase VST Rack Xpander was designed specifically for software plug-ins, some of its controls are not designed to work with the PC-90. Most notably, the Input Level and Bus Select Button are not operational on the PC-90. PC-90 Input Level is controlled by the Aux Send knob in the channels monitor section. PC-90 bus selection is performed on the Reverb page of the Lexicon Studio control panel.

### ***Naming Effects***

1. Click on the Program name in the top center of the effect processor display to open a name dialog.
2. Type in a new name.
3. Press [Return].

### ***Saving Effects***

Save your edited effects in the File pop-up menu to the right, under the Effect Type pop-up.



1. To save the current Program, select **Save Effect**.
2. To save all Programs of the same Effect Type, select **Save Bank**.
3. In the file dialog that appears, select a name and location for the file. (You might want to create a special folder for your PC-90 effects.)
4. Click **Save**.

## Loading Effects

Load effects from disk into the PC-90 as follows:

1. Pull down the **File** menu.
2. To load a single Program, select **Load Effect**.
3. To load a complete Program Bank, select **Load Bank**.
4. In the file dialog that appears, find and click on the file you want to load.

The Effect settings are saved with your Song. If you want to use your edited effects in other Songs, you can save and load them separately.

## Editing Effects

You can create your own programs or edit any of the presets with the editor. To open the PC-90 interface, click the **Edit** button on the PC-90 Rack Xpander. The PC-90 Interface will open.

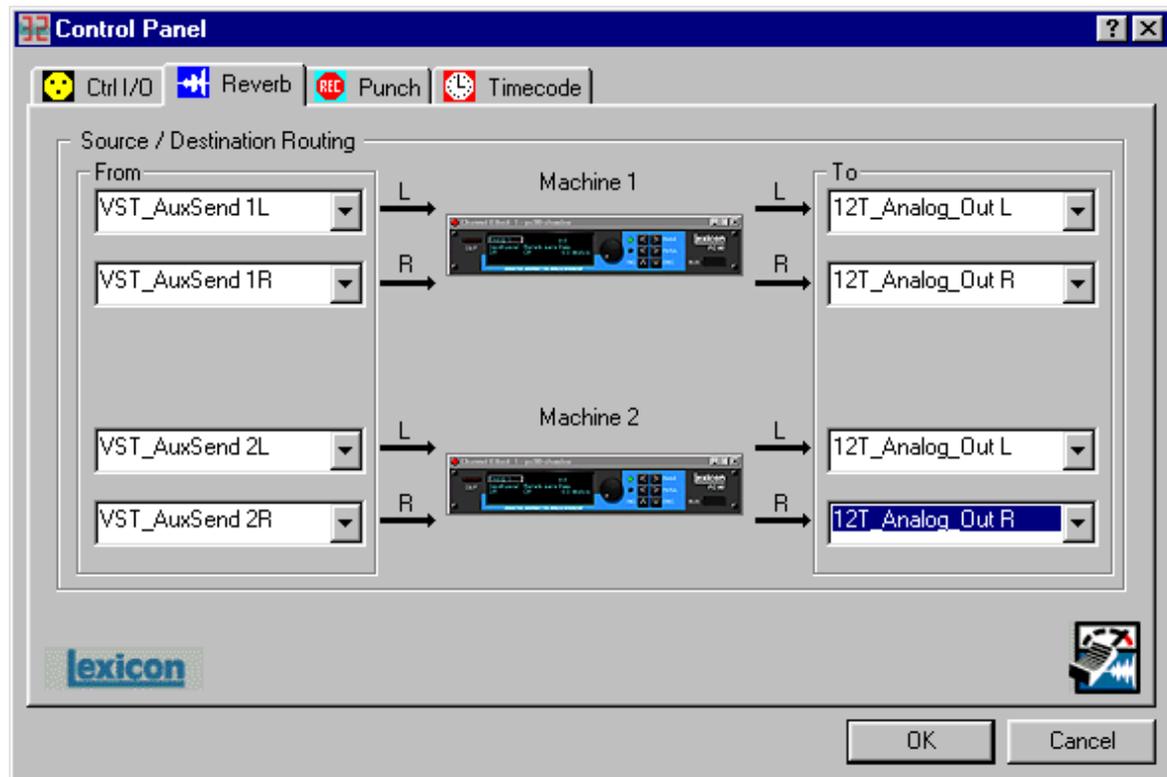


1. Click on the **PARA** button, to activate a display of three parameters. The parameter currently selected for editing is displayed with a border.
2. If the parameter you want to edit is not visible in the display, click on the left or right part of the Page button to go to another parameter page. To return to Parameter mode, click on the Para. button again.
3. Set a value for the selected parameter with the Value dial. To make small adjustments, click on the dial, hold down the mouse button and move the pointer away from the dial before turning it. The further away from the dial the pointer is, the finer the value adjustments.
4. Use the **INC** and **DEC** buttons for super fine manipulation of the active parameter.

## Beyond the Basics

### Routing a Plug-In

A Default routing is set in your Lexicon Studio system, to allow you to use your PC-90 as an auxiliary effect, monitoring the audio through the analog outputs on your audio interface as soon as it is Installed. This default routing configuration can be easily changed for your particular studio setup with the following procedure.



The **Reverb** page on the **Core-32 Control Panel** allows you to route the two PC-90 DSP engines independently through any of the connectors of your audio interface.

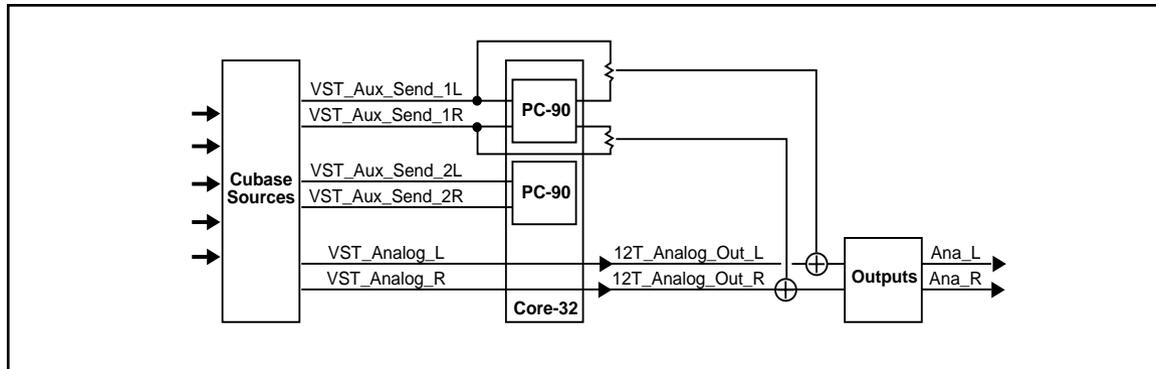
For each PC-90 engine, there are two input sources (left and right), and two output destinations (left and right). To assign a source to any of the input channels in your PC-90:

1. Clicking the left mouse button on the down-arrow of an input selector will open a menu displaying all sources available to route into your PC-90.
2. Use the scrollbar to view the available sources.
3. Click on a source to route it as an input source. The input selector will display your selection as the input source.  
NOTE: Input sources can only be be routed one at a time.

To assign a destination to any of the output channels in your PC-90:

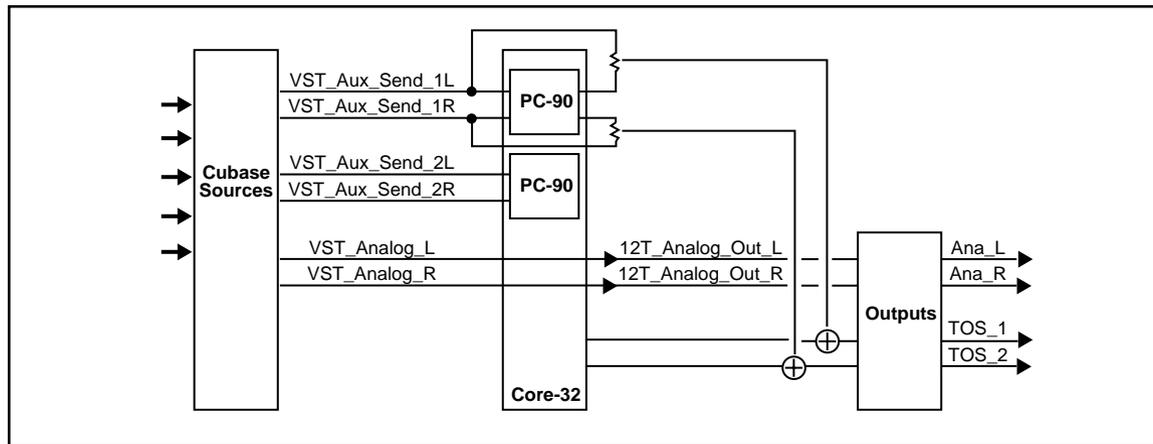
1. Click the left mouse button on the down-arrow of an output selector to open a menu of available destinations.
2. Use the scrollbar to view the available destinations.
3. Click on a source to route it as an output destination. The output selector will display your selection as the input source.

The diagram below is a flow chart of the audio routing from the previous picture of the control panel. Notice that PC-90 receives its input from Cubase VST via the **VST\_Aux\_Send\_1L** and **VST\_Aux\_Send\_1R** sources. These sources route any audio assigned to that buss into the PC-90.



As **12T\_Analog\_Out\_L** and **12T\_Analog\_Out\_R** are selected as destinations, PC-90 audio is summed with the outputs of Cubase VST, which are also routed to the analog outputs. This allows monitoring of PC-90 reverb and all audio from VST through the same outputs of the interface.

The reverb routing matrix is flexible, and can provide a wide array of configurations. The above configuration would not be ideal if, for example, you were mixing audio simultaneously from a digital tape deck and your computer to a digital mixer, and wanted to send your reverb into your digital mixer on independent channels. In this case, you can easily reroute your destinations to other channels, such as TOSLINKs 1 & 2 simply by changing the output destinations on the Control Panel.



### ***Automating the PC-90***

The PC-90 Digital Reverberator can be completely automated within any control software that supports automation. Refer to the user guide for your particular software package for support offered for this functionality. The following section describes PC-90 automation within Cubase VST.



To write enable the channel that PC-90 is assigned to, open the Cubase VST Mixer window and click the Write button.

Changes made to PC-90 parameters will be recorded to an Audio Mix Part. Any changes (even those made while the transport is in Stop) are recorded, as long as the Write button is enabled.

If you check the Arrange window after you stop playback, you will note that a special Mixer Track called Audiomix has been created. This Track contains one long Part named Audiomix, in which all your Monitor mixer actions are stored. Don't worry about the length of this Part — it will automatically be lengthened if you record past its end.

Please note that there is only one Audiomix Part/Track, created the first time you use the Write function in your Arrangement. No new Parts are created the next time you use the Write function — information is added to the existing Part.



To stop writing automation events, click the Write button a second time, or close the Mixer window. All changes to this track and its plug-ins will cease to record.

To playback automation in your mix:



1. Check that the Audiomix Track or Part are not muted.
2. Activate Automated playback by clicking on the Read button in the upper left corner of the Monitor window.

If you want to watch and listen to your recorded mixer actions while you're recording fader movements for another mixer channel, etc., you can activate Read and Write simultaneously.

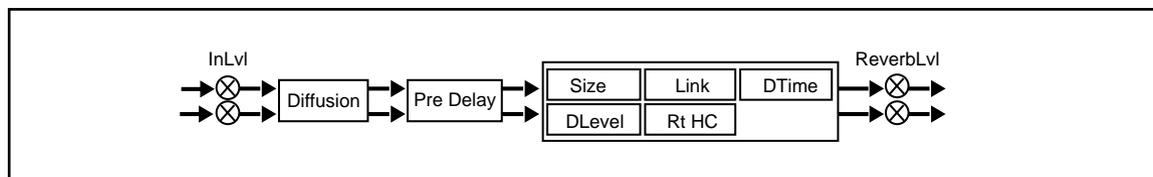
Begin playback as usual. The monitor faders and controls will move automatically, following your recorded actions. You will also hear your plug-in automation events playing back with your audio.

# 5

## Reference

### PC-90 Algorithms

#### Ambience



Unlike traditional reverb, Ambience is intended to become a part of the direct sound – to give it both better blend and a definite position in space. Ambience gives warmth, spaciousness and depth to a performance without coloring the direct sound at all.

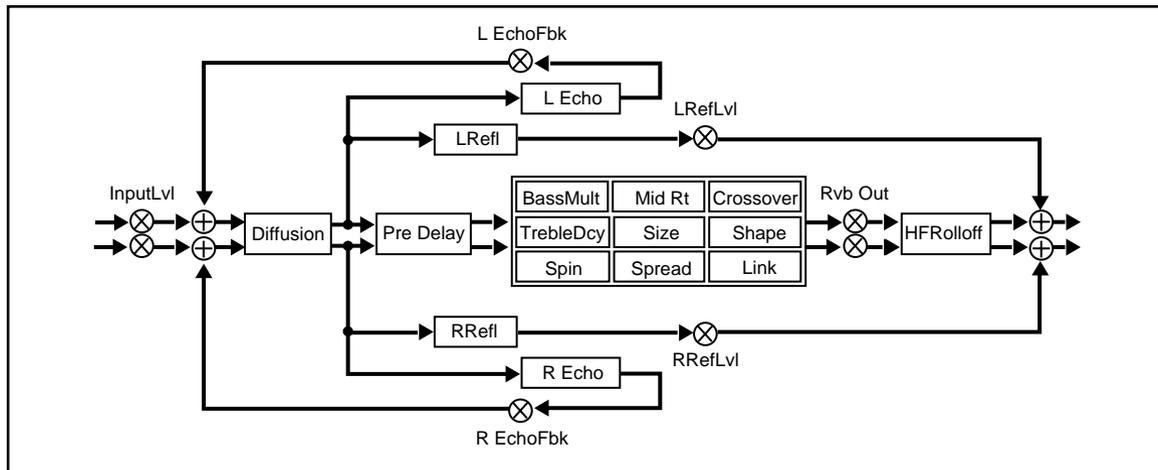
This algorithm generates primarily the strong reflections which appear in the first few hundred milliseconds of the reverberation process. These early reflections constitute the primary audible effect, giving you the impression of a hall surrounding you while the music is playing. To avoid any coloration from these strong reflections, the time delays and amplitudes are random functions.

Ambience is very useful for adding a room sound to recorded music or speech. It is particularly easy to match a studio recording of dialog to a typical room environment. In music recording, it allows you to realistically add distance to a close-miked signal. If an ensemble has been recorded with close-mikes and pan pots, Ambience can provide the missing blend and depth. The apparent position of the instruments is preserved in the reverb while the apparent distance is increased. This algorithm is also useful in matching a closely miked accent microphone to the overall ambience of a recording.

This allows a soloist to be increased in level without changing the apparent distance. Ambience can be used in a recording situation any time a close-miked sound is undesirable.

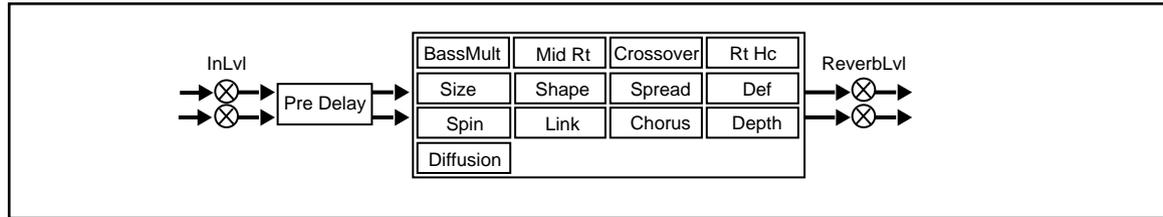
When using Ambience in a mix, it is best to use a stereo send to the PC-90, carefully matching the panning of the various close-miked sources to their positions in the mix. Leave the Mix control at 100% wet. the apparent distance of each source can be controlled by the level of its feed.

## Chamber



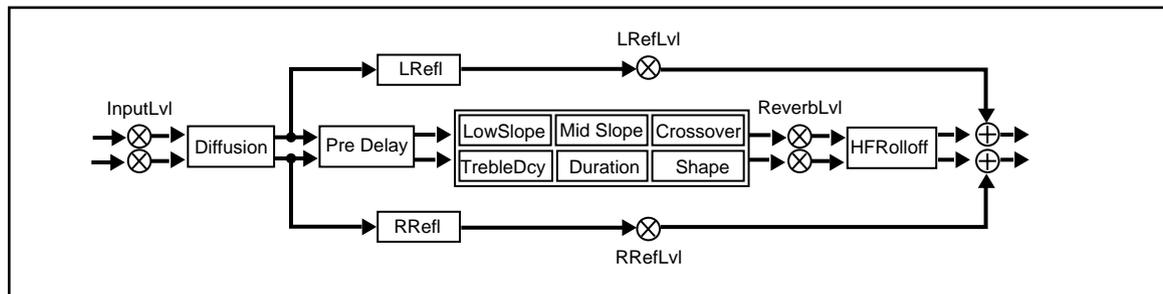
Chamber provides an even, relatively dimension-less reverberation, with little change in color as the sound decays. The initial diffusion is similar to the Concert Hall algorithm, but the sense of size and space is much less obvious. This characteristic, along with low color in the decay tail, makes Chamber useful on a wide range of program material. It is especially useful on spoken voice, giving a noticeable increase in loudness with very low color.

## Concert Hall



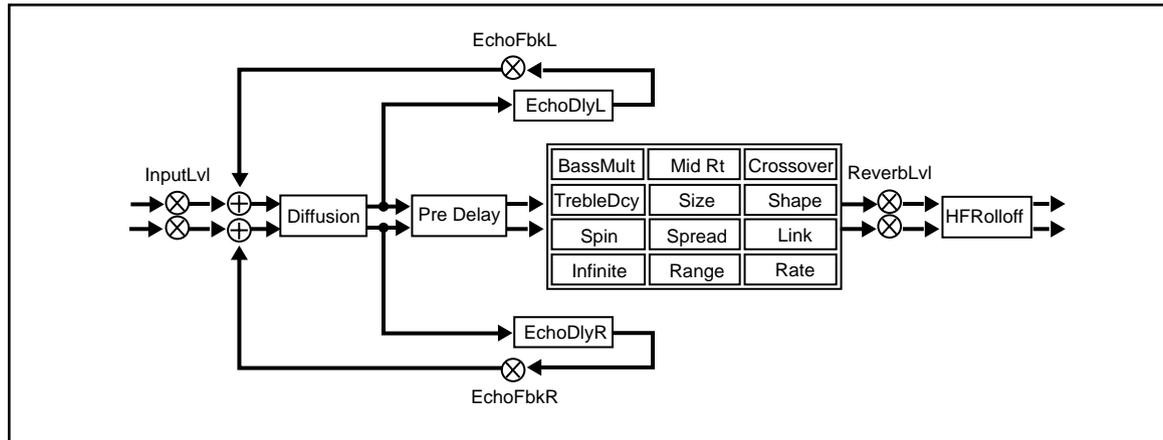
This algorithm emulates a real concert hall. The reverberation is very clean, and designed to remain behind the direct sound – adding ambience, but leaving the source unchanged. This effect has a relatively low initial echo density which builds up gradually over time.

## Inverse



This algorithm allows you to vary the slope of the initial portion of the reverb envelope. The slope can decay, remain level, or rise over a variable time interval. When the time interval is up, the reverberation cuts off abruptly. The resulting effect is similar to a gate, but is not at all dependent on the level or complexity of the input signal. Slopes are adjustable over a negative, even, or positive slope. Positive slopes create inverse effects, while more even slopes create gated effects. Negative slope values have rather natural reverb tails.

## Room



Room produces an excellent simulation of a very small room which is useful for dialog and voice-over applications, but which may be too colored for some sustained musical tones. Increasing the Size/Shape/Spread parameters produces an excellent reverb.

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## *The Parameters*

The PC-90 has a wide range of parameters to control the sound of your algorithms. Not all parameters are found in all algorithms.

### ***Chorus***

In the Concert Hall algorithm, Chorus randomizes delay times and introduces modulation to make reverberation sound less metallic. Increasing Chorus increases the rate of modulation. Because Chorusing can cause pitch variation, this parameter should be set with care when using sources with very little pitch wobble (such as guitar or piano). A good practice is to increase the setting until the pitch wobble becomes noticeable, then lower it slightly.

### ***Crossover***

Crossover sets the frequency at which the transition from Mid Rt to Low Rt takes place. This control should be set at least two octaves higher than the low frequency you want to boost. For example, to boost a signal at 100Hz, set Crossover to 400Hz. (This setting works well for classical music). Crossover works best around 400Hz for boosting low frequencies, and around 1.5 kHz for cutting low frequencies.

### ***Decay***

In Ambience, Decay controls the length of the ambience “tail”.

### ***Decay Level***

In Ambience, Decay Level controls the level of the ambience “tail”. When Decay Level is off, ambience consists entirely of the early reflection signal.

### ***Definition***

In the Concert Hall algorithm, Definition affects the echo density buildup rate during the latter part of the decay period. When set to Off, the rate is determined by the program material. Raising Definition through its range (1-99%) causes the sound to become choppy - the decrease in echo density creates increasingly distinct, repetitive echo trails.

### ***Depth***

In the Concert Hall algorithm, Depth sets the output amplitude envelope, changing the listener's perspective from the front to the rear of the hall.

### ***Diffusion***

A Diffusion control is provided in all algorithms. It controls the degree to which initial echo density increases over time. High settings of Diffusion result in initial build-up of echo density, and low settings cause low initial build-up. Echo density is also affected by Size; smaller spaces will sound denser. To enhance percussion, use high settings of Diffusion. For clearer, more natural vocals, mixes, and piano music, use low or moderate settings of Diffusion. Note that, at some extreme input levels, high settings of Diffusion may trigger the overload indicators on the Clip display.

### ***Duration***

In the Inverse algorithm, Duration determines the length of time, in milliseconds, which passes before the cutoff in Inverse effects.

### ***Echo Delays***

Echo Delay L and Echo Delay R provide echoes to the left and right channels. Unlike Delay and Reflect, which are isolated right and left delays, left and right echoes are blended in the diffusor. The echoes are routed both to the outputs and through a feedback path.

### ***Echo Feedback***

Fbk L and Fbk R modify the levels of the echo feedback path. The range is from -100% to 0 to +100%.

### ***Effects Mix***

Mix controls the ratio of dry and wet signal present at the PC-90 outputs. When the PC-90 is used as an effects loop, this control should always be set for 100% wet.

### ***HF Rolloff***

HF Rolloff sets the high frequency cutoff of a low-pass filter. This parameter affects both channels.

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### ***Infinite***

In the Room algorithm, this control is provided to turn the Infinite effect On or Off.

### ***Input Lvl***

InLvl controls the level of the unprocessed (dry) signal into the effect.

### ***Link***

When Link is set to On, the reverb time (Mid RT) and Spread scale linearly as the Size control is varies. For some special effects, Mid RT, Spread and Size can be unlinked.

### ***Mid Rt and Bass Mult***

Mid Rt sets the reverb time for mid-frequency signals. Because low frequency reverb time (Low Rt) is a multiplier of Mid Rt, Mid Rt acts as a master control for the reverb time. Bass Mult sets the reverb time for low-frequency signals, as a multiplier of the Mid Rt parameter. For example, if Bass Mult is set to 2X, and Mid Rt is set to two seconds, the low frequency reverb time will be four seconds. For a naturally sounding hall ambience, we recommend values of 1.5X or less.

### ***Pre Delay***

Pre Delay adjusts an additional time delay between the input of signal and the onset of reverberation. This control is not intended to mimic the time delays in natural spaces. In real rooms, the build-up of reverberation is gradual, and the initial time gap is usually relatively short. Natural spaces are best emulated by adjusting Spread for the desired effective predelay.

### ***Range and Rate***

In the Room algorithm, these controls are used to reduce coloration for small room sizes or to reduce the sense of periodicity when the Infinite control is on. These controls allow you to set the range of a moving delay and the speed at which it moves. High settings of either control may be unsuitable for sustained tones, like piano.

### ***Reverb Level***

Reverb Level sets the amount of reverberation in the processed signal. It is normally FULL, but may be reduced for effects where the pre-echoes should dominate.

### ***Reflect Delays***

In the Chamber and Inverse algorithms, L Refl and R Refl provide pre-echoes to the left and right channels. In Chamber, the maximum delay value is 1.2 seconds. In Inverse, the maximum delay value is 800 milliseconds.

### ***Reflect Levels***

L RefLvl and R RefLvl control the level of the reflections (L Refl and R Refl). The range of each level is from Full (0dB) to -85dB, and to Off.

### ***Shape, Spread***

Shape and Spread work together to control the overall ambience of the reverberation. Shape determines the contour of the reverberation envelope. With Shape all the way down, reverberation builds explosively, and decays quickly. As Shape is advanced, reverberation builds up more slowly and sustains for the time set by the Spread. With Shape in the middle, the build-up and sustain of the reverberation envelope emulates a large concert hall (assuming that Spread is at least halfway up, and that Size is 30 meters or larger). Low Spread settings result in a rapid onset of reverberation at the beginning of the envelope, with little or no sustain. Higher settings spread out both the buildup and sustain.

### ***Size***

Size sets the rate of build-up of diffusion after the initial period (which is controlled by Diffusion). The Size control changes a reverb sound from very large to very small. Generally, you should set this control to approximate the size of the acoustic space you are trying to create, before adjusting anything else. The size in meters is roughly equal to the longest dimension of the space. Audio is temporarily muted when Size is changed.

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### ***Slope***

In the Inverse algorithm, Slope determines the shape of the reverb envelope. When set to 0, the level of reverb remains unchanged over its duration, then cuts off abruptly (depending upon the amount of Diffusion in use). Setting Slope above 0 causes the level of reverb to rise smoothly from soft to loud until the sound is cut off. The greater the slope, the softer the initial reverberation and the more pronounced its rise. With negative values, the reverb drops from its initial level to a quieter one before cutoff. The lower the slope, the more pronounced the drop-off.

### ***Spin***

Spin affects the movement of the reverberation tail. The object of Spin is to continuously alter the timbre of the reverberant sound. This makes the result more natural, without making the position of the instruments unstable. Spin should typically be set to values between 10% and 50%. Higher values may make the timbre of piano, guitar and other precisely pitched instruments unstable.

### ***Treble Dcy***

Treble Dcy sets the frequency above which a 6dB/octave low-pass filter attenuates the reverberated signal. It does not attenuate Reflection Delays. High frequencies are often rolled off with this parameter, resulting in more natural-sounding reverberation. Setting a low frequency for this parameter can actually shorten the reverb time, as it damps the audio as it recirculates.

## *Preset Descriptions*

### **Notes on Preset Design**

PC-90 incorporates the results of a great deal of research into acoustics and reverberation. Reverberation, or reflected sound energy, gives recorded music a sense of being performed in a real acoustic location.

### **Using the Size and Spread Parameters**

In the PC-90, the Size and Spread controls allow adjustment of the buildup and decay of the initial part of the reverberation envelope. In the Chamber algorithm, Size acts as a master control for the apparent size of the space being created by PC-90. Both Spread and Mid Rt vary linearly with the setting of Size. Thus, maximum reverb time and spread may require high settings of Size. To find an appropriate reverb sound, start with a preset with a similar sound to what you want to end up with. Simply varying Size is often sufficient to arrive at the exact sound you are seeking.

Once a size has been selected, Spread and Shape are used to adjust the shape and duration of the initial reverb envelope, which together provide the major sonic impression of room size. The density is set by the size control, and the rate of decay is set by Mid Rt.

As Shape is raised to about 1/8 of its range, the initial sharp attack of the reverberation is reduced, and reverberation builds more slowly. The envelope then sustains briefly before it begins to die away at the rate set by Mid Rt. Spread has little or no effect on this shape.

When Shape is at 1/4 of its range, buildup is even slower and the sustain is longer. Now Spread affects the length of both the buildup and sustain. As a rough estimate, the sustain will be approximately the time value indicated by the Spread display (in milliseconds).

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As Shape is raised further, the buildup and sustain remain similar, but now a secondary sustain appears in the envelope, at a lower level than the first. This secondary plateau simulates a very diffused reflection off the back wall of a hall, and is effective in creating a sense of size and space. This reflection becomes stronger and stronger, reaching an optimal loudness when Shape is at about 1/2 of its range.

The highest Shape settings are typically used for effects. Near the top of the scale the back wall reflection becomes stronger than the earlier part of the envelope, resulting in an inverse sound.

NOTE: None of these effects are audible unless Mid Rt is set short enough. Generally, Mid Rt should be set to a value of about 1.2 seconds for small rooms, and up to 2.4 seconds or so for “halls”. Size should also be set to a value appropriate to the desired hall size (note, however, that small sizes color the reverberation). 15 meters makes a very small room, and 38 meters is useful for a large hall.

Used with care, Shape and Spread allow PC-90 to produce superior ambience – a sound which is spacious and has great depth – without the long reverberation of a church.

### ***Random Delay Elements***

PC-90 incorporates random delay elements in its reverb. These elements have several effects. First, there is a reduction of long-lived modes in the reverberant decay, which makes the decay less metallic and reduces the apparent reverb time. The random elements also improve the steady-state timbre of the effect.

The speed at which the delay elements move is controlled by Spin. Settings higher than about 40-40% can cause audible pitch wobble in very critical material such as classical guitar or piano and can also cause noise on pure tones. This noise is not audible in speech, however, and, for mixed music or speech, values up to 48% can give an improved sound.

### ***Creating a Realistic Sound***

When you set out to create a sound, the first and most important decision is how big a space you want. The best way to start is to listen to several presets and choose the one which sounds closest to what you have in mind. If necessary, use Size to make a slightly larger or smaller sound, as needed.

Next use Mid Rt to fine-tune the amount of time the reverberation takes to die away at the end of musical phrases. Actual halls vary a great deal in their Mid Rt values. The setting of Bass Mult is also critical in matching the sound of an existing hall. An ideal concert hall would have a Bass Mult setting of 1.2. It is rare when actual physical spaces exceed 1.5. many (if not most) good recording environments have values of 1.0 or less, so a value of 0.8 could be tried when attempting to match an existing hall.

Spread also adjusts the effective reverb time when the music is running. Higher values of Spread produce a longer effective reverb time, which, in turn, gives greater spaciousness to the sound.

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# Appendix A

## Third Party Support

This section provides user notes and excerpts from user guides from supported third party applications. Documented below are several important features for Steinberg Cubase VST, such as automation.

### *Steinberg Cubase VST*

#### **Automation**

##### **Volume and Pan Automation – Dynamic Events or Monitor Mixer**

Cubase VST offers two ways of automating volume and pan: the Dynamic Events set in the Audio Editor and the Write function in the Monitor Mixer. The two methods have their specific properties and uses, as described below:

##### **Dynamic Events**

The Dynamic events allow you to control the volume and panning individually for *each segment*. This allows you to create fade-ins, fade-outs and auto-pan effects, which are then integral parts of each segment. This is all done in the Audio Editor.

What this means is that if you for example create a volume curve for one segment, and later copy this segment for use in other places in the song, all copies will “inherit” the volume curve of the original segment. You can later adjust the curve for each individual segment, if needed.

A more advanced possibility is to use Ghost events. These are audio events which all *share* a segment, and thereby share a volume or pan curve. This means that if one is adjusted, they are all affected.

If you use the Dynamic Events to control volume, you cannot go above unity gain (0.0 dB).

### ***Monitor Mixer Automation***

Using the Write function in the Monitor mixer you can automate Volume and Panning (actually all mixer settings and changes) for each *audio channel*. All Tracks in the Arrange window that are set to play back via this audio channel will be affected likewise by these controls. This method is described on the following pages.

Let's say, as an example, that one channel is first used for a guitar and later in the song for vocals. If you then use the Monitor window to lower the volume in the guitar intro, you will get the same lower volume for the vocal, since they are on the same channel.

When you are using the Monitor mixer to control volume, you can go above unity gain (actually amplify the sound). Furthermore, you can automate Mutes and Solos.

### **Conclusion**

So, when to use Dynamic Events and when to use the Monitor Mixer depends on what is needed at the moment; if you need to make minute adjustment of shorter segments and repeat those in the Song, you should choose Dynamic Events. If you make longer non-repeating recordings and prefer to adjust volumes via (on-screen) faders, you use the Monitor window.

Do not hesitate to combine the two methods of automation, simultaneously or during different parts of the Song. The settings you make in the Monitor window are relative to the volume value delivered by the Dynamic Events automation. For example, if the Dynamic Event volume has a value of -2.0 dB, and you set the Monitor window fader to -0.4 dB, the resulting volume will be -2.4 dB. |

Please note that the Monitor Mixer automation is not restricted to Volume and Pan - Mutes, Solo, EQ and Effect settings can also be automated, as with the Lexicon PC-90.

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### **Automating the Monitor mixer**

Your actions in the Monitor mixer window can be automated in a very straightforward way: by “writing” them into a special Audiomix Part. When played back, this Part will repeat your fader movements and button presses just like you performed them. You will even see the faders and buttons move on the screen, like on a physical mixer with motorized controls.

What can be automated?

Mixdown in Cubase VST can be completely automated! The following parameter settings are recorded with the Write function:

For each channel:

- Volume
- Pan
- Mute
- Solo
- EQ On switch
- 4 x Effect Send Active switches
- 4 x Effect Send levels
- 4 x Effect Send PRE switches
- Effect DRY switch
- Settings for 4 EQ modules

Global for all channels

- Master volume Left and Right
- 4 x Master Effect Send levels
- 4 x Effect On/Off
- 4 x Effect Program selection
- 4 x Effect parameters (the 16 first parameters for each Effect “processor”)

- 4 x Master Effect On/Off
- 4 x Master Effect Program selection
- 4 x Master Effect parameters (the 8 first parameters for each Master Effect)

The PC-90 has been enabled for complete automation. As Steinberg Cubase VST provides automation for only the first 16 parameters of a channel effect, and the first 8 parameters for a master effect. Lexicon has prioritized the PC-90 parameters to take the fullest advantage of this feature.

### Recording your actions

1. Open the Monitor window.



2. Click on the Write button in the upper left corner to activate recording.

While the Write button is “lit” (activated), every volume, pan, mute or solo movement you make will be recorded.

3. Start playback.

4. Move the faders and pan controls, mute and solo as you would during a manual mixdown. As you can repeat this recording several times, it is probably easiest to mix one or a couple of channels at a time, and stop and deactivate the Write function in between. That way, you can also Undo your last run if you’re not satisfied, using the Undo command on the Edit menu.

5. Stop playback.

6. If you check the Arrange window, you will note that a special Mixer Track called Audiomix has been created. This Track contains one long Part named Audiomix, in which all your Monitor mixer actions are stored. Don’t worry about the length of this Part; it will automatically be lengthened if you record past its end. Please note that there is only one Audiomix Part/Track, created the first time you use the Write function in your Arrangement. No new Parts are created the next time you use the Write function; information is added to the existing Part instead.

Drums	10	Drums	Drums
Track 11	11		
Track 12	12		
Audio Mix		Audio Mix	

7. Deactivate the Write function by clicking on the button.
8. Exiting the Monitor window will automatically deactivate the Write function.

### Recording Mixer settings in Stop mode

The Write function works in Stop mode as well as during playback. If you activate Write when Cubase VST is stopped, all changes you make to your mixer parameters are recorded at the current Song Position. This feature can be used creatively if you need initial mixer settings, abrupt changes, etc.

### Undoing your recording

If you select Undo from the Edit menu, all actions recorded *since you last activated Write* will be undone. Therefore, make it a habit to always deactivate Write after each recording “pass” and listen to what you recorded.

- If you find this too cumbersome, you can make separate Audiomix Parts for the different audio channels or recording “passes”, and edit/delete these afterwards.

### Playing back your recorded Mixer actions

1. Check that the Audiomix Track or Part are not muted.
2. Activate the Automated playback by clicking on the Read button in the upper left corner of the Monitor window.

You may have Write and Read activated simultaneously, if you want to watch and listen to your recorded mixer actions while you’re recording fader movements for another mixer channel, etc.

3. Start playback as usual. The monitor faders and controls will move automatically, following your recorded actions.



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### **Moving and Muting the Audiomix Part**

The Audiomix Part can be handled as any Mixer Part:

- If you want to turn off Monitor mixer automation from the Arrange window, you can mute the Audiomix Part or Track.
- You can copy recorded mixer actions to several places in your Arrangement by cutting out the relevant section of the Audiomix Part with the Scissors tool and duplicating it, placing the copies wherever you want them.

### **Working with one Audiomix Part for each Audio Channel**

Although Cubase VST only creates one dedicated Audiomix Track, there is a way to structure your Mixer Automation into different Parts for different channels, which makes it easier to edit and redo mixer recordings you're not satisfied with:

1. Record your fader movements and other mixer actions for the first audio channel, as described on the previous page.
2. Deactivate the Write function and go to the Arrange window.
3. Make a new Mixer Track. It might be a good idea to name it after the audio channel you just mixed, to avoid confusion later on.
4. Move the entire Audiomix Part to the new Track.
5. Open the Monitor again, activate the Write function and record your mixer actions for the next audio channel. Now Cubase VST creates a new Audiomix Part on the original, empty Audiomix Track. If you activate the Read function, your previously recorded Part will be played back from its new Track, so you can watch your recorded fader movements while you continue with new audio channels.
6. Repeat steps 2 to 5 for as many channels you like.

You will end up with a number of Mixer Tracks, all playing back simultaneously, affecting different audio channels in the Monitor mixer. If you want to, you can keep it that way. If you like to clean up your Arrange window, you can Merge the Parts into one.

### Editing the Audiomix Part

The Audiomix Part contains Mixer Events, which can be edited in List Edit:

1. Select the Audiomix Part in the Arrange Window.
2. Open List Edit.
3. Pull the divider to the right so that you can see the Comment column.

This column shows which Mixer parameter (volume, pan etc.) and audio channel each Event controls.

Switch parameters (parameters with on/off values) are handled in a special way. All information about a switch Event (switch name, on/off) will be displayed in the Comments column, but do not try to edit a Switch value if you are not sure about how the Value 2 encoding works.

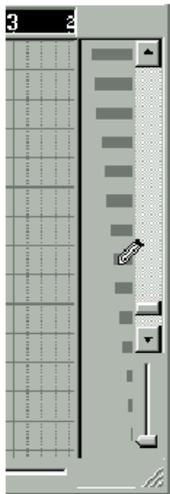
4. Select an Event that corresponds to the fader (or other Mixer control) you want to edit.
5. Pull down the Mask pop-up menu and select "Mask It!". This will hide all Events except those of the same type as the selected (that is, only Events of the same type and the same audio channel as the selected one will be shown).
6. Look at the "Value 2 display" to the right in the editor.

As described in the List Edit chapter, this display shows Value 2 for each Event as a horizontal bargraph. If you have recorded a fader movement, this will be displayed as a ramp, as shown to the side.

7. Use the Pencil tool to redraw the curve, or the Eraser tool to delete Events. You can of course edit the values in the Value 2 column in the actual list. Use this feature if for example you want to set a fader to a specific value at a certain position.

The value 101 for a fader corresponds to 0 dB (unity gain).

8. When you're done, pull down the Mask pop-up menu, select "No Mask" and press [Return] to leave List Edit.



### How List values work for Switches

The text in the Comments column corresponds to the numerical value in the Value 1 column. Each *continuous parameter* (such as volume, pan, etc.) on *each audio channel*, has a unique Event number (value 1), while the Value 2 column shows the parameter values. However, switches (on/off-parameters) are handled in a special way:

All switches on the same audio channel share the same Event number!

All other information (which switch is it and what is its state) is encoded into Value 2. If you want to edit the values of switch Events, you have to consult the tables below to find the appropriate values.

This is the Value 2 functionality for switches on each audio channel:

Switch Name	“On” value	“Off” value
Mute	64	0
Solo	65	1
EQ On/Off	66	2
Effect send DRY	67	3
Effect send 1 On/Off	68	4
Effect send 2 On/Off	69	5
Effect send 3 On/Off	70	6
Effect send 4 On/Off	71	7
Effect send 1 PRE switch	72	8
Effect send 2 PRE switch	73	9
Effect send 3 PRE switch	74	10
Effect send 4 PRE switch	75	11
EQ module 1 Enable switch	76	12
EQ module 2 Enable switch	77	13
EQ module 3 Enable switch	78	14
EQ module 4 Enable switch	79	15

### Combining manual Mixing and/or two kinds of Automation

Even if you automate the volume or pan for an audio channel, you can still make a manual adjustment by dragging the fader/pan control in the Monitor window. Just remember to leave the Write function off, or your movements will be recorded “on top of” the already recorded automation. Likewise, if you have automated volume or pan using Dynamic Events, you can still manually or automatically adjust the volume/pan in the Monitor mixer.

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# Appendix B

## Troubleshooting

### *Installation troubleshooting*

#### *Problems Starting Autodetection*

If there was a card installed in your system previous to this installation you should run uninstall to remove all the components that may be in your system. Uninstall will reset your system to the state that it was in before the Core-32 card was installed. Make sure that all other devices are installed and working properly. An existing conflict or device that is in a problem state (!) can cause problems when installing additional cards.

#### *Problems During Autodetection*

If the computer freezes when detecting new hardware it may be necessary to reset the computer hardware. To accomplish this, turn the computer off, wait three seconds, then turn the computer on. Pressing Ctrl+Alt+Delete or pressing the reset button will **not** reinitialize your computer hardware.

#### *Manual Detection*

If auto detection does not work, you may have to use Manual detection by running Add New Hardware from the Control Panel.

#### *Resolving Hardware Conflicts*

Windows 95 may not be able to configure the Core-32 without creating conflicts with another device. This can happen as more devices are installed and system resources are depleted. Related error messages that may appear are:

- Turn off a device to make room for the new device
- Disable the new device

- Reconfigure a legacy device to make room for the new device

Use the Hardware Conflict Troubleshooter in Windows Help. The Troubleshooting Wizard walks you through step-by-step troubleshooting or will suggest solutions to fix the problem.

Tip: If you have a motherboard with five (5) PCI slots you may find that 2 of the slots share one IRQ. As sharing IRQs can be problematic on some devices, move the Core-32 to another slot. See your manufacturer's manual for more information.

### *Problematic Devices*

If the Core-32 seems to configure properly but is in a *problem state* it will be marked with an exclamation point (!) in the Device manager. This indicates that Windows 95 tried to load a driver but failed. Reinstall the Core-32 driver per the installation instructions.

### *Software*

LED Status on Core 32

There are 2 status LEDs mounted on the card. When the audio application is booted and the drivers are loaded, one LED will light and the other will blink repeatedly.

## *PC Troubleshooting*

### *Power Up Problems*

If the computer will not power up after initially installing a Core-32, the Core-32 or another expansion card may be partially dislodged from the expansion slot on the motherboard. Symptoms may include keyboard lights on, power indicator lit, or the hard drive spinning. Remove the Core-32 and verify that the system boots without it installed. Verify that the expansion cards are seated securely in the slots. Remove all peripheral devices from the Core-32 including PC-90 and cables, checking for re-booting each time.

### *VGA Cards*

Screen redraws may create pops, clicks or crashes. Many times, this can be either the bus being locked, or the CPU being interrupted too often to maintain a reliable stream of audio. Certain VGA cards appear to be the cause of this problem. To test for this, drag and drop one of your open windows while playing a multitrack session. Turn the VGA acceleration off and perform the same test. This may solve the problem. Contact the VGA card manufacturer and obtain the most recent drivers for that board and report the problem.

Suggested Mother board BIOS setting suggestions.

- Disable Video BIOS/ROM shadow
- Disable Video BIOS/ROM cacheable
- Disable PCI/VGA palette snoop
- Enable PCI 2.1 Compliancy

#### **TIP: Reporting problems to hardware manufacturers**

When reporting a problem to your hardware manufacturer, they will ask you if your motherboard bios, VGA card and other peripheral card drivers are up to date. It is a good idea to have this information ready when you first contact them.

### *AGP (Accelerated Graphics Port)*

Some PC motherboards can use an AGP graphics adapter. This standard was designed to take the graphics operations off of the PCI bus and onto a dedicated bus — the AGP port. The AGP port is physically and logically separate from the PCI bus. Using this port to handle graphics operations frees the PCI bus for audio and SCSI data activity. Contact your computer manufacturer for more information on the benefits of using an AGP port.

## *Optimizing Windows 95 for Audio Applications*

Following are some suggestions that may improve your computers performance as a dedicated Digital Audio Workstation. Some of these suggestions disable some Windows 95 optimization features and may slow the overall performance of your system when using non-audio applications.

### ***Add Memory***

Adding RAM to your computer will reduce the chance that Windows needs to use Virtual Memory.

### ***Defragment Your Hard Drives***

Defragment your drives on a regular basis.

### ***Run Scan Disk Regularly***

Scan disk can detect and fix problems before becoming fatal.

### ***Disconnect Any Mapped Drives That Are No Longer Connected To Your Computer***

Drives will appear in Explorer with a red "x" under them if the drive is no longer connected to your system.

### ***Verify no Drives are Compressed***

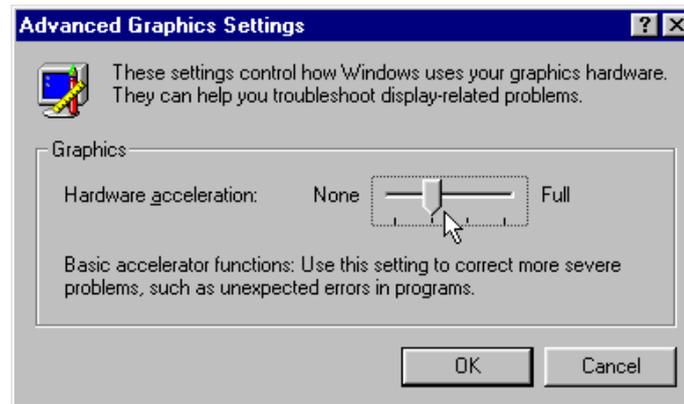
ControlPanel/System/Performance: Make sure it says "Not Installed" after Disk Compression

### ***Disable Write-Behind Disk Caching***

Sets the number of kilobytes that Win95 reads ahead. By reading ahead the audio drive is working harder than necessary.

## Graphics Hardware Acceleration

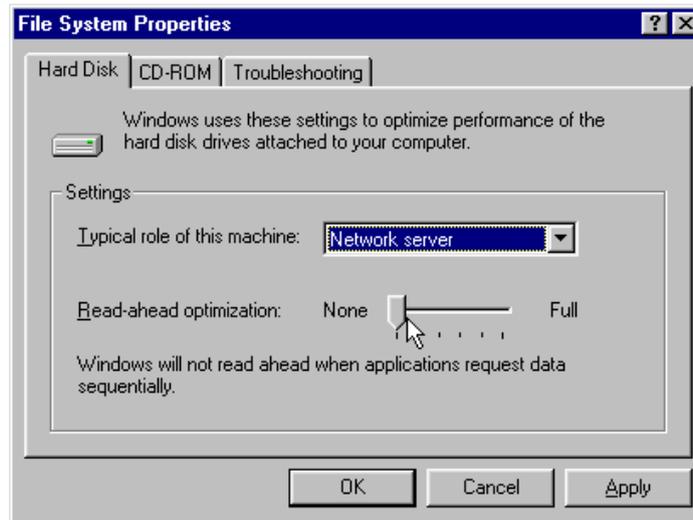
In non-AGP computer systems (computers using VGA cards), graphics acceleration settings can force your video card to hog the PCI bus. Crackling and zippering in the audio while opening and moving windows is a warning sign of this.



Try lowering the acceleration settings of your graphics hardware. This is done by going to **Start/Settings/Control Pane/System**, then selecting the **Performance** tab and clicking the **Graphics** button. A slider in this control panel allows you to select the amount of hardware acceleration. Try using **Basic Acceleration Functions** or **No Accelerator Functions** to reduce noise from PCI burden within your system.

### ***Read-Ahead Optimization***

If you find that you are getting noise when adding more tracks of hard disk audio, try this to optimize your system and increase the number of tracks within your system.



This control panel can be found by opening **Start/Settings/ Control Panel/System**, then selecting the **Performance** tab and clicking the **File System** button. This control panel optimizes the performance of your disk drives for particular applications. For audio systems, it is best to set the typical role for the machine to be a **Desktop System** or **File Server**. In either case, make sure your Read-Ahead Optimization is set for **None**.

### ***Set Virtual Memory To A Fixed Amount***

If you have sufficient memory it is possible to completely turn off Virtual Memory. If not set this to 50MB to 100MB.

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## *Common Problems and Solutions*

**The Lexicon Studio Sound System is not recognized by Windows 95 when first installed.**

Make sure you have a Plug and Play ready BIOS that complies with V2.1 of the PCI specification. Load the default values of the BIOS and try installing the card again.

**The Lexicon Studio Sound System appears in the Device Manager tagged with a yellow exclamation point.**

Windows tried to load a driver but failed. Reload the driver.

Check the resources page for a resource conflict. Resources in your computer may have been depleted before the Core-32 card was installed. Try moving the board to another PCI slot or remove a card to make more resources available.

A legacy device (a card with manually-set IRQs) may need to be disabled by removing the card.

**The Lexicon Studio Sound System appears in the Device Manager with no resources assigned.**

Make sure you have a Plug and Play ready BIOS that complies with V2.1 of the PCI specification. Load the default values of the BIOS and try installing the card again.

**Opening the Lexicon Studio ASIO Control Panel crashes the computer.**

This may indicate that you are running the first release of Windows 95. To determine the version of Windows 95 you are running:

1. In the Control Panel, double-click on **System**.
2. Click the **General** tab.
3. Locate the version number under the System heading and refer to the following table:

<b>Version number</b>	<b>Version of Windows 95</b>
4.00.950	Windows 95
4.00.950A	Windows 95 plus the Service Pack 1 Update, or OEM Service Release 1.
4.00.950B	OEM Service Release 2 (OSR2)

Upgrade to Windows 95 A or B if necessary.

## *Poor System Performance*

A computer with a fast CPU does not guarantee high performance. If your system is not performing as well as it should be, try some of these suggestions:

- Quit any open applications other than your audio software.
- Use a fast IDE drive for your audio. Getting audio off the PCI bus creates more bandwidth for the Lexicon Studio System. You might also try a fast SCSI drive with a PCI SCSI controller, which will yield performance almost as high as the IDE due to the faster speed of SCSI.
- Add memory. (Cubase VST recommends at least 64MB.)

## *Pops and clicks*

If you have a PCI-based VGA board, screen redraws may create pops, clicks or crashes. Try turning VGA acceleration off. The Graphics accelerator slider is accessed by pressing the Graphics button in the Performance page of the System properties box. This may solve the problem.

Contact the VGA card manufacturer and obtain the most recent drivers for that board and to report the problem.

Minimize any unnecessary windows that are constantly redrawing

Use an AGP (Accelerated Graphics Port) video card if your motherboard supports this.

Verify that the SCSI card drivers are up to date.

Verify all audio and data cables are securely fastened to their respective connectors.

Try another Word clock source. Switch the word clock from internal to external or vice versa.

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# Appendix C

## Specifications

### ***LDI-12T Interface***

#### **Audio Inputs (2)**

**Connectors:** Balanced: female XLR-3 type  
Unbalanced: RCA type (software selectable)  
**Impedance:** 100k $\Omega$ , balanced; 50k $\Omega$ , unbalanced  
**Levels:** -14 to +18dBu full-scale, balanced  
-20 to +12dBu full-scale, unbalanced

#### **A/D Performance**

**Frequency Response:** 20Hz to 20kHz,  $\pm 0.5$ dB ref 1kHz  
**Crosstalk:** <-76dB, 20Hz to 20kHz  
**THD:** <0.005%, 20Hz to 20kHz  
**Dynamic Range:** 104dB typical, 20kHz bandwidth;  
106dB typical, A-weighted  
**Resolution:** 24 bits

#### **Audio Outputs (2)**

**Connectors:** Balanced, male XLR-3 type  
**Impedance:** 600 $\Omega$  nominal, each side  
**Levels:** +22dBu full-scale, balanced  
+16dBu full-scale, unbalanced

#### **D/A Performance**

**Frequency Response:** 20Hz to 20kHz,  $\pm 0.5$ dB ref 1kHz  
**Crosstalk:** <-86dB, 20Hz to 20kHz  
**THD:** <0.01%, 20Hz to 20kHz  
**Dynamic Range:** 94dB typical, 20 kHz bandwidth  
97dB typical, A-weighted  
**Resolution:** 20 bits

**Digital Audio Interface**

**Input Connectors:** Coaxial RCA type (1)  
Optical TOSLINK (1), assignable as S/PDIF or TOSLINK

**Output Connectors:** Coaxial RCA type (1)  
Optical TOSLINK (1), assignable as S/PDIF or TOSLINK

**Format:** S/PDIF (IEC-958, CP-340) consumer audio interface

**System Sample Rates**

44.1kHz, 48kHz

**Control Interface**

**Timecode Input:** Balanced, female XLR-3 type connector, EIA-422,  
Formats supported @±15%: SMPTE non-drop/drop frame

**External Word Clock Input:** BNC connector, TTL compatible, unterminated

**Alesis ADAT Sync:** 9-pin D-sub female connectors, input/output

**Machine Control:** 9-pin D-sub female connector

**Host Computer:** 68-pin high-density connector

**General Dimensions**19.0" x 1.75" x 4.0" (WHD)  
(438 x 45 x 102mm), 19" rack mount standard, 1U high**Power Requirements**

9 VAC, 1A wall transformer provided

**RFI/ESD**

Conforms to FCC Class B; EN55022 Class B (CE), IEC 801-2, IEC 801-3

**Environment**

**Operating Temperature:** 32° to 104°F (0° to 40°C)

**Storage Temperature:** -22° to 167°F (-30° to 70°C)

**Humidity:** Max 95% non-condensing

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## ***Core-32***

<b>External Interfaces:</b>	LDI-12T proprietary multi-pin connector LDI-16S proprietary multi-pin connector
<b>Internal Interfaces:</b>	PCI Interface PC-90 daughterboard interface TBUS daughterboard interface
<b>Data Resolution:</b>	24-bit architecture
<b>Power Requirements:</b>	+5v± 5%
<b>Clock Range:</b>	44.1 kHz -5% to 48 kHz +5%
<b>FCC Rating:</b>	Class A
<b>Dimensions:</b>	12.25"L x 3.875"H x .0625"W "full length" (PCI)
<b>Operating Temperature:</b>	32-104 degrees F (0-40 degrees C)

## ***PC-90***

<b>Connections:</b>	Multipin connection to Core-32 card
<b>Processing:</b>	20 bit
<b>Sample Rates:</b>	44.1kHz, 48kHz
<b>Inputs:</b>	2 Stereo pairs to the Core-32 routing matrix
<b>Outputs:</b>	2 Stereo pairs to the Core-32 routing matrix

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