



**lexicon**

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*V4.20*  
*User Guide*

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## **Unpacking and Inspection**

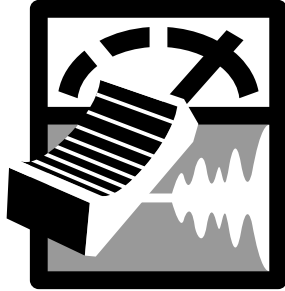
After unpacking the system modules, save all packing materials in case you ever need to ship the units. Thoroughly inspect the modules and packing materials for signs of damage. Report any damage to the carrier at once; report equipment malfunction to your dealer.

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**LEXICON**



**STUDIO**

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*V4.20*  
*User Guide*

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# 1

## ***Getting Started***

### *Introduction*

Thank you for your purchase of a Lexicon Studio system. Lexicon Studio is a line of professional hardware components designed for your computer 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, please refer to the user manual provided by the manufacturer of the software before using your Lexicon Studio system.

This manual is designed for both the Macintosh and PC versions of Lexicon Studio, which differ only in installation procedures. As pictures from both platforms are used throughout this manual, your screen graphics will vary depending on the system you are using.

Electronic PDF documentation containing late-breaking information is provided on CD-ROM with your Lexicon Studio system, along with electronic documentation for the PC-90. These PDFs have information that may be crucial to operating the latest version of the software you are using. The electronic documentation always supersedes the information in your printed manual. Please read it carefully, and always look to our website at [www.lexicon.com](http://www.lexicon.com) for the latest updates in both software and documentation.

## System Requirements

The following systems are recommended as the minimum for working with Lexicon Studio:

### Macintosh

- 200MHz PowerPC processor (PowerPC 750 G3 processors are highly recommended)
- 64MB of RAM
- Audio-capable disk drive
- MacOS 7.6.1 or later

### PC

- 200MHz Pentium Processor or compatible equivalent (Intel processors are also highly recommended.)
- 64MB of RAM
- Audio-capable disk drive
- Microsoft Windows — 95™, 98™ or later

### High Performance Systems

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

- Fast CPU — For the best Macintosh performance, use a G3 processor. 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 that "hog" the PCI bus and creates data logjams. For Macintosh, use on-board video or an Apple-approved video card.
- AGP (Advanced Graphic Port) capability for the graphics card is preferable as it gets the video off of the PC's PCI bus.



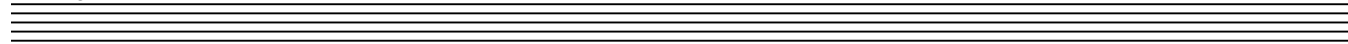
### **SCSI Vs. IDE**

Both SCSI and IDE offer distinct advantages and disadvantages:

	<b>Benefits</b>	<b>Liabilities</b>
SCSI	<ul style="list-style-type: none"><li>• Fast bus performance</li><li>• Removable media</li><li>• Choice of many high performance drives</li></ul>	<ul style="list-style-type: none"><li>• Use of a PCI card to interface to the computer increases bus traffic, especially if multiple PCI cards are in the system</li></ul>
IDE:	<ul style="list-style-type: none"><li>• Drives approach SCSI performance</li><li>• Inexpensive</li><li>• Using the IDE bus decreases activity on the SCSI bus</li></ul>	<ul style="list-style-type: none"><li>• The IDE bus is slower (in most cases this is NOT the limiting factor)</li><li>• Fewer model choices</li><li>• Limited removable media</li></ul>

For both Macintosh and PC systems it is always best to dedicate one disk drive for the operating system, and a second drive specifically for audio.

Stay tuned on our website for the latest recommendations as technology changes.



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# 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. Lexicon Studio works with third party software (such as Steinberg Cubase VST, Opcode VisionDSP, Emagic Logic Audio) which should be installed prior to installing the hardware and software for Lexicon Studio. Refer to your audio software user manual on how to accomplish this.

### ***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 and generating static electricity after it is removed from its bag.
- 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 Lexicon audio 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***

As some Lexicon Studio software is installed into directories of your audio control software (such as Cubase Audio VST), you should install and verify that your audio software launches properly before installing Lexicon Studio (even though it will not be audio-capable until installation of the Lexicon Studio system). Please refer to your audio software manual for details on installation, and then proceed with Lexicon Studio hardware and software installation.

Your Lexicon Studio system may contain either of two versions of the Core-32 audio card. The installation procedure is identical for both cards.

### ***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 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 the manufacturer's owner's manual.
5. Touch the computer power supply case to discharge any personal static electricity.
6. Remove the card 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 anti-static bag on top of the foam and place the card on the foam with the white PC-90 connector facing up.

### *Mount the PC-90 card onto the Audio card*

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

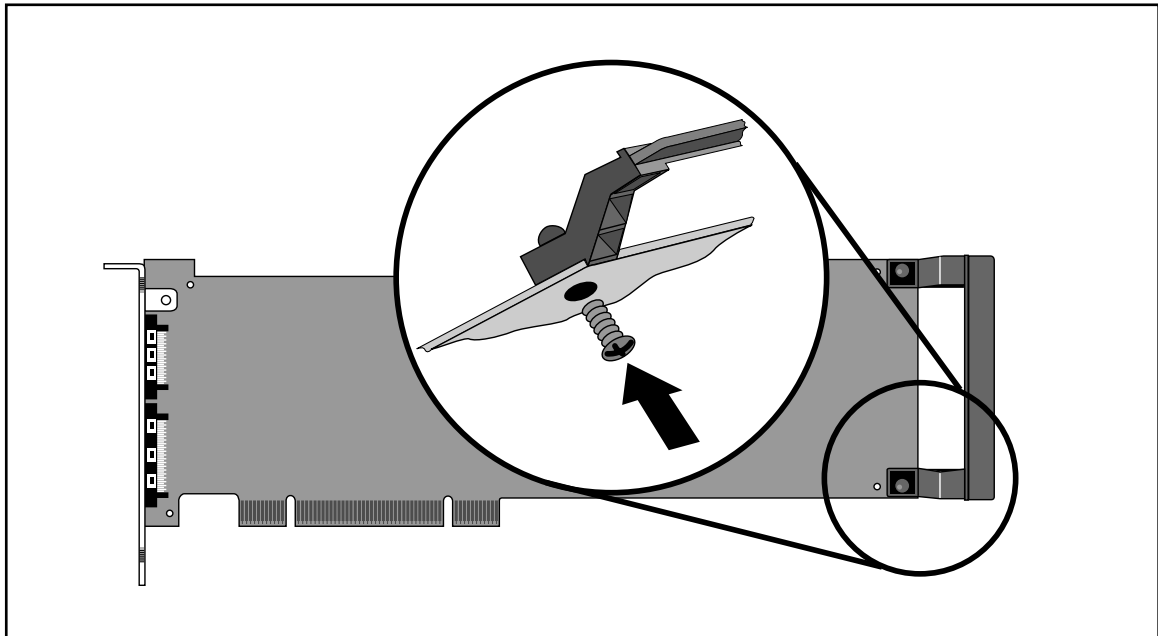
1. Touch the computer power supply case to discharge any personal static.
2. Remove the PC-90 module from its anti-static bag, holding the card only by its edges. Place the bag on your work surface next to your computer and place the PC-90 on it with its white connector facing up.
3. Pick up the PC-90 card and orient it so that the connector side faces away from the palm of your hand.
4. Pick up the audio 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 audio card should be aligned with the standoffs on the PC-90 card.
5. Once the cards are mated, place the assembly, with the audio card on top, onto the anti-static bag. Use the four screws provided to attach the PC-90 to the audio 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 (which can damage the cards). To attach the extender:

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

NOTE: On most new Macintosh models, the PCI bus is sized to fit full-length PCI cards solidly. On these models (9600, G3 Mini-tower) you will not need to install the PCI Extender.



### *Install the Card assembly*

The audio card must be installed in a full-size PCI slot which is a bus master slot to take advantage of its bus master capabilities. 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).

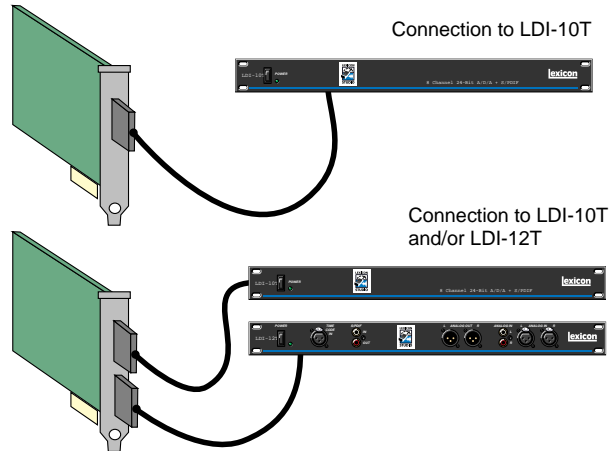
1. Verify that third party audio software is installed and launches correctly.
2. Touch the computer power supply case to discharge personal static.
3. Pick up the card assembly by the top of the metal bracket and the top of the other end of the card.
4. Align the card assembly over the slot you've chosen and insert it.
5. 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.
6. Once the card is fully seated, attach the bracket to the computer chassis with the screws provided with your computer. If the computer is a tower or mini-tower design, check to see that boards are not bending under their own weight and shorting against each other.
7. Replace the cover on the computer. **Do not operate the system without the cover.**



*One of the Core-32 Audio Card models*

## Connecting Audio Interfaces

Your Lexicon Studio system will contain a Core-32 audio card with one of the two connection configurations shown below.



### Cables/Connections

1. Use only the proprietary Lexicon cable (provided) to connect the Core-32 card to the Lexicon audio interface.
2. With both the computer and the audio interface powered down, attach the cable between the connector on the back of the Core-32 card and the Computer port on the audio interface rear panel (Refer to the illustration to locate the proper slot on your card).
3. Connect the power cable (either 9V connector port or standard IEC power depending upon the interface) between the rear panel of the audio interface and a wall socket.

The LDI-10T supports 24-bit analog I/O as well as 24-bit S/PDIF digital I/O. The 1/4" phone jacks accept TRS (Tip Ring Sleeve) cables for balanced +4dBu level, or TS (Tip Sleeve) cables for unbalanced -10dBV level, switchable within the Lexicon Studio Control Panel.

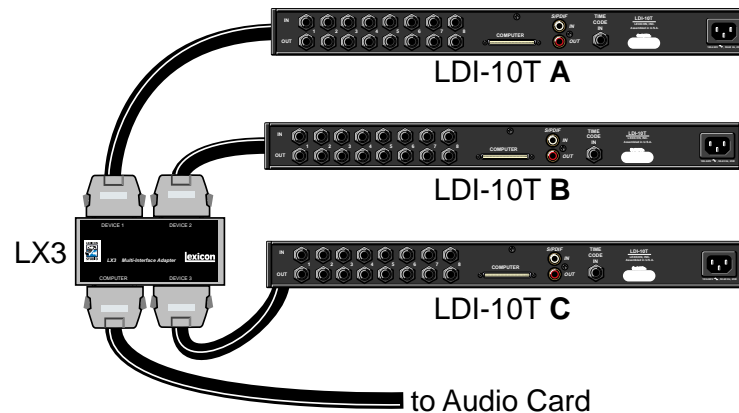


The LDI-12T supports various formats of analog and digital I/O. Make certain your cables are the correct types prior to connection. (Analog XLR connectors are +4dBu balanced (Pin 2 hot) , analog RCA connectors are –10dBV unbalanced). To select a source for input or output within the application software,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

When powering up your system, always make sure to turn the audio interface power on prior to turning on your computer. This ensures that diagnostics and calibrations necessary for optimum performance are performed on computer startup.

### **Connecting Multiple Interfaces**

The optional LX3 Multi-Interface Adapter allows you to connect as many as three LDI-10T interfaces to one single Lexicon audio card, creating a total of 24 channels of analog I/O.



1. Use only the proprietary Lexicon cable (provided) to connect the Core-32 card to the LX3 Multi-Interface Adapter.
2. With both the computer and the audio interfaces powered down, attach the cable between the connector on the back of the Core-32 card and the LX3 Multi-Interface Adapter.

## PC 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).

If you are installing a later version of Lexicon Studio software over your current Studio software, please skip to the next section entitled *Updating Driver and PC-90 Software*.

### Install the Lexicon Studio Driver for Windows

1. With your Lexicon Studio hardware and third party audio software installed, power up LDI-12T interface and your computer. Once your computer is fully booted, Windows will identify your Core-32 as a new PCI multimedia device. In the event of booting problems, refer to the troubleshooting section of this user guide.
2. The Windows Update Device Driver Wizard will appear. Insert the “*Lexicon Studio: Drivers & Documentation*” CD-ROM , and click the **Next >** button, allowing Windows to perform a search for the Lexicon Studio driver.
3. Windows will find the driver information for the **Lexicon Studio Sound System** on the CD-ROM (if it does not, you may need to **Browse** to it. Additionally, you may also need to **Browse** to the CD-ROM for a file entitled ControlP.dll. Click the **Finish** button.)
4. When the drivers are loaded, you will be prompted to restart your computer. You should choose not to restart at this time and proceed to the instructions for installing PC-90 software within the next section. Windows will prompt you twice to restart. Select **No** at both prompts and move on to the next section, *Installing the PC-90 Software for Windows*.

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

1. Insert the “*Lexicon Studio: Drivers & Documentation*” CD-ROM if you haven’t already.
2. Open the contents of the CD-ROM and double-click on **Setup.exe** to launch the Installer.
3. Read the Directory Path Warning and then click **OK**.
4. Click **Next >** at the Installer welcome screen.
5. Make certain that the installer is directed toward the desired plug-in folder (example: C:\..\Cubase Audio VST\vstplugins) and click **Next >**.
6. At the summary screen, select the desired software and click **Next >** to choose it.
7. Click **Next >** to begin the installation.
8. Click **Finish** to complete the PC-90 software installation.
9. Restart your computer (even if you’ve already restarted after installing the Core-32 Drivers).

Installing the PC-90 software will file it within Windows program registry. To remove or update this software, remove the PC-90 install by using the **Add/Remove Programs** control panel, highlighting **Lexicon Studio PC90 PlugIn**, and selecting **Remove**. This control panel is located at Start/Settings/Control Panel/Add/Remove Programs.

## *Updating Driver and PC-90 Software*

If you are installing a newer version of Lexicon Studio software over your current software, follow the steps outlined below.

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

As PC-90 software is registered through the Windows Add/Remove Programs control panel, it must be completely removed to install a newer version. You should never have 2 different versions of PC-90 software installed simultaneously. To deinstall PC-90:

1. Go to Start/Settings/Control Panel /Add/Remove Programs.
2. Scroll down to **Lexicon Studio PC90 PlugIn**.
3. Click the **Add/Remove** button.

4. Click the **Yes** button when asked “Are you sure you want to completely remove **Lexicon Studio PC90 PlugIn** and all of its components?”.
5. Click **OK** to close the control panel



### **Update Studio Driver**

If you have a CD-ROM or have downloaded Lexicon Studio drivers with a higher revision number than those currently installed in your system, you should choose to update your driver. To do so, follow the directions below.

1. Go to Start / Settings / Control Panel / System.
2. Click on the **Device Manager** tab..
3. Open the directory for **Sound, Video and Game Controllers**.
4. Highlight **Lexicon IDS Sound System** by clicking on it.
5. Click the **Properties** button.



6. Select the **Driver** tab within the Properties page.
7. Click the **Update Driver...** button.
8. Windows will ask if you want to search for the driver. Select **Yes** (Recommended).
9. Windows will search the drive from which a driver was last loaded. If you installed drivers from floppy disk, it will check only the floppy drive. If you have loaded the updated drivers from different media, you will need to click the **Other Locations...** button and **Browse** to the CD-ROM for the new driver. If you encounter problems locating the first file, check to make sure the path is correct, then click **OK**.
10. Click the **Finish** button.
11. Windows will prompt you to restart. Select **No** when prompted and continue with these instructions.



## Reinstall PC-90 Software

1. Insert the “*Lexicon Studio: Drivers & Documentation*” CD-ROM if you haven’t already.
2. Open the contents of the CD-ROM and double-click on **Setup.exe** to launch the Installer.
3. Read the Directory Path Warning and then click **OK**.
4. Click **Next >** at the Installer welcome screen.
5. Make certain that the installer is directed toward the desired plug-in folder (example: C:\..\Cubase Audio VST\vstplugins) and click **Next >**.
6. At the summary screen, select the desired software and click **Next >** to choose it.
7. Click **Next >** to begin the installation.
8. Click **Finish** to complete the PC-90 software installation.
9. Restart your computer (even if you’ve already restarted after installing the Core-32 Drivers).

## Macintosh Software Installation

1. With your Lexicon Studio hardware and third party audio software installed, power up the LDI-12T interface and your computer. In the event of booting problems, refer to the troubleshooting section of this user guide.
2. Insert the “*Lexicon Studio: Drivers & Documentation*” CD-ROM, and double-click on the **Studio Installer** icon.
3. Click **Continue** at the splash screen. You will be given an opportunity to read the Lexicon software license agreement. Click **OK** again.
4. The Installer gives you the choice of **Easy Install**, **Custom Install**, and **Uninstall**. Choose **Easy Install** to load all the necessary components for Lexicon Studio. Choose **Custom Install** to load only some of the software components for Lexicon Studio. Click **Install**.
5. If you have installed multiple ASIO-capable audio programs, a screen will appear that allows you to choose which ASIO Drivers folder you want Lexicon Studio installed in. Click on a folder to highlight your choice, then click **OK**.
6. If you have multiple VST-capable audio programs installed, a screen will appear that allows you to choose which VstPlugins folder you want PC-90 installed in. Click on a folder to highlight your choice, then click **OK**.
7. Repeat this entire installation procedure for each application. When software loading is complete, click the **Restart** button.

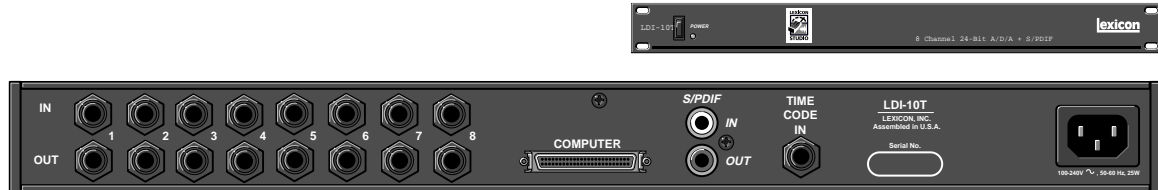
The Installer will remove any prior software versions found in your system and place all components as required. Following is a list of files installed and their locations:

<b>File</b>	<b>Location</b>
Core-32	Extensions Folder
ASIO Control Panel	Control Panel Folder
ASIO Core-32	ASIO Drivers Folder
Lexicon PC90 1	VstPlugins Folder
Lexicon PC90 2	VstPlugins Folder

NOTE: The Installer will not allow you to automatically install a previous software version over a later version. If you want to downgrade your software, you must Uninstall, then install the version you want.

## Interface Connections

### LDI-10T Rear Panel Connections



- Eight (8) analog balanced TRS 1/4" inputs (software switchable between +4dB and -10dB)
- Eight (8) analog +4dB balanced TRS 1/4" outputs
- One (1) S/PDIF pair RCA (coaxial) digital input/output
- One (1) balanced 1/4" timecode input
- One (1) IEC power connection
- One (1) multi-pin computer connection



### LDI-12T 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 dBV 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

### LDI-12T Rear Panel Connections

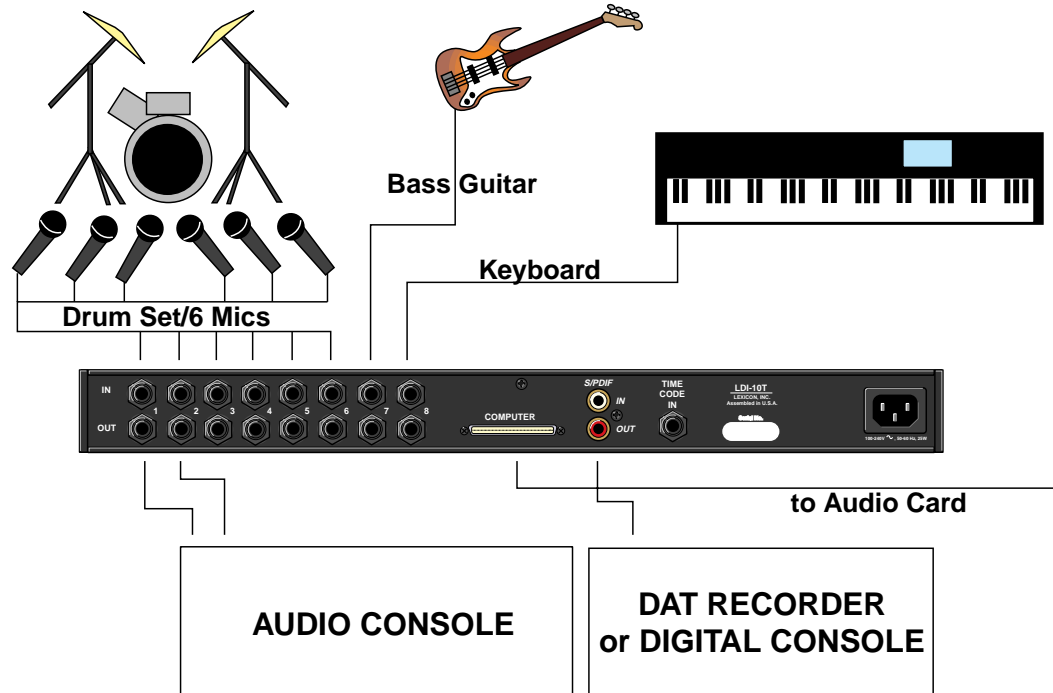


- Two (2) optical TOSLINK™ connectors for input/output connection to 8-channel ADAT®-compatible format OR 2-channel optical S/PDIF format
- Two (2) ADAT sync D9 connectors
- One (1) Word Clock input BNC connector (75 Ω termination)
- One (1) Serial D9 connector
- One (1) Power connection
- One (1) Multi-pin computer connector

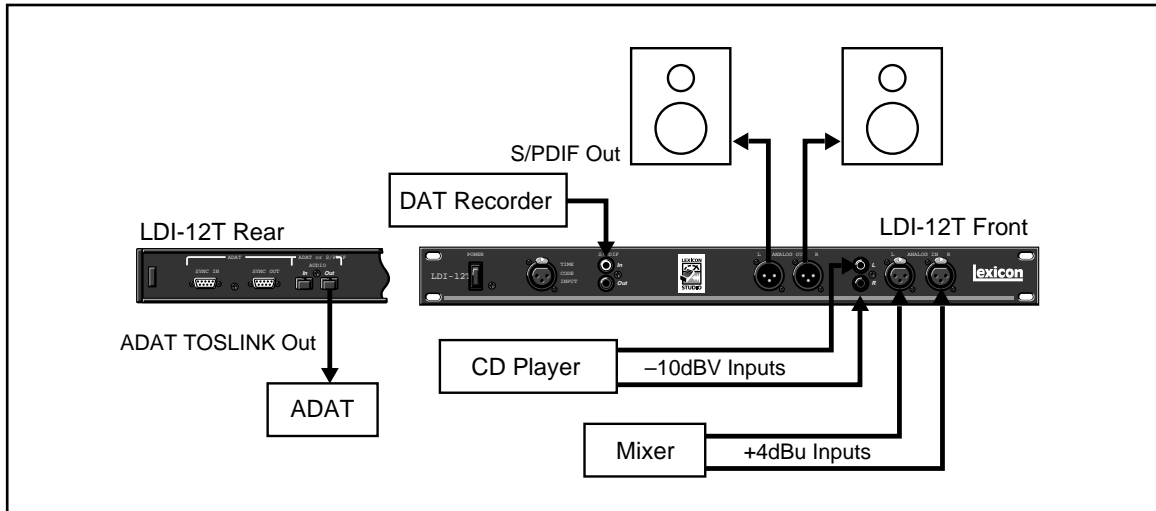
NOTE: The BNC Word Clock Input has a 75 Ω termination. Therefore, if you are distributing Word Clock to a chain of devices, the LDI-12T must be last in the chain.

### Typical Configurations

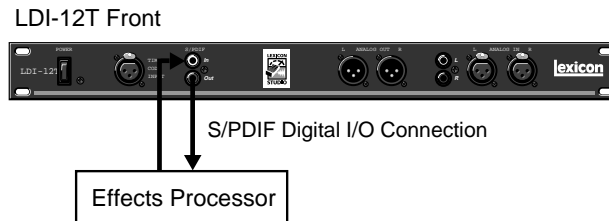
Following are some typical interface configurations. These are provided as a starting point, and to familiarize you with the system. Once your system is physically configured, signal routing is controlled by the Lexicon Studio Driver and the routing provided by your host application program (Cubase VST,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



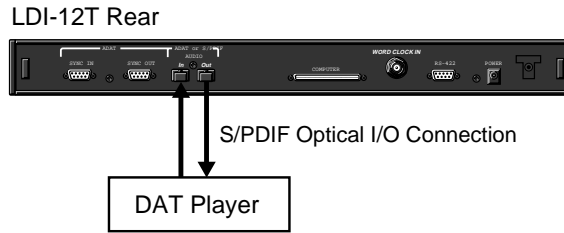
The configuration shown below has the analog outputs connected to powered speakers for monitoring, S/PDIF to a DAT for mixing, and ADAT connections for multi-channel recording. The XLR analog inputs could be from a mixer or microphone/instrument preamp, or the RCA inputs could be used for low-level devices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



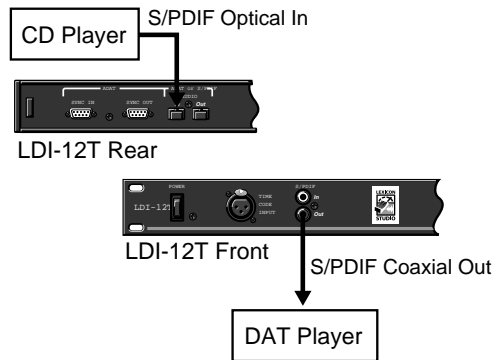
S/PDIF connections can be made to any digital device such as the send/returns of a digital signal processor.



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***

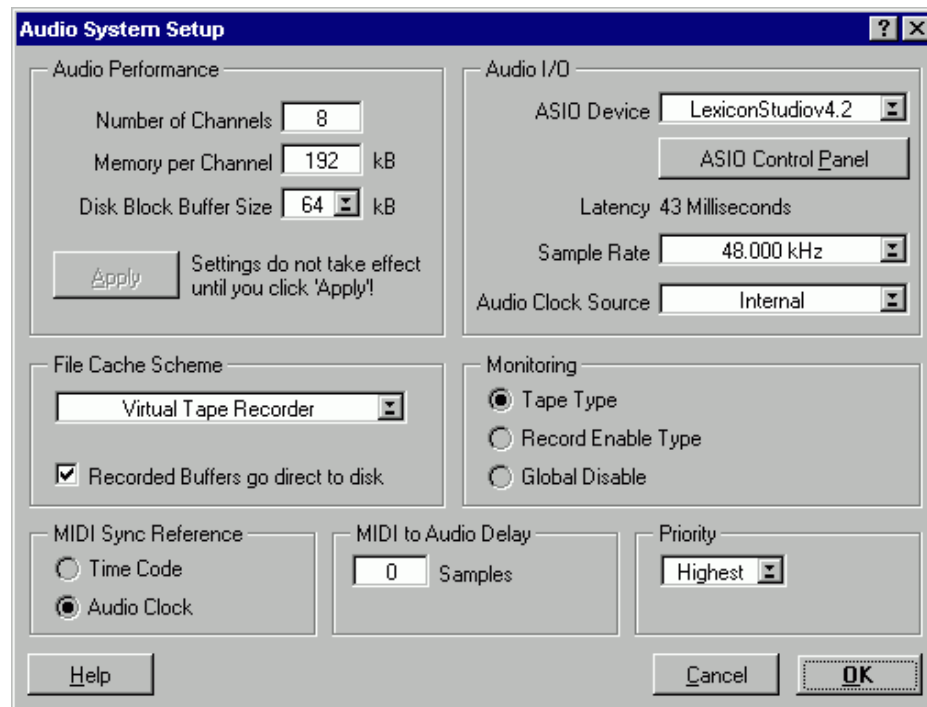
Using Cubase VST as an example of a host application, the following things determine signal routing.

1. Interface (LDI-12T/LDI-10T) connections.
2. Configuration of the Core-32 Control Panel 12T tab and Routing tab.
3. Inputs and outputs selected and enabled in Cubase Audio VST.

Typically, all connections enabled in the Routing tab 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

Certain portions of the Lexicon Studio System should be specifically set up for the program you are using. This setup is performed 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. Our example is from Cubase VST. The actual procedure will vary depending on your audio application.



The method of selecting **Tabs** for different pages, will be familiar to users of Windows and other Microsoft products. The selections are: Routing, Reverb, Punch, Timecode, LDI-12T and Clock Source.



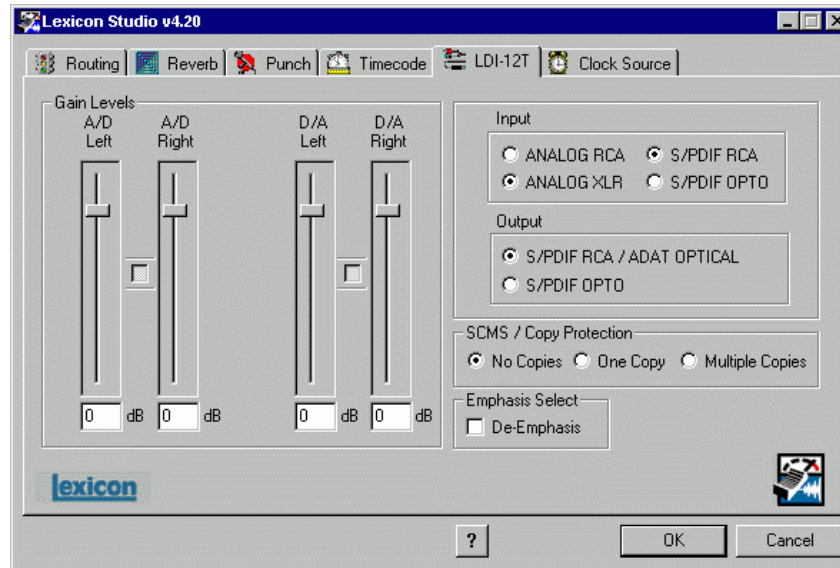
## ***Routing***

This tab lets you determine which of the connected devices' I/O streams are active. Disconnected devices will be unavailable for selection (greyed out). I/O streams which are activated here become available as a clock source, as a source/destination for reverb and punch, and for use in the audio application.

1. No more than 32 streams of I/O can be selected. When an attempt is made to select more than 32, a pop-up message box will report that this is not allowed.
2. The high gain buttons beside the 10T analog inputs will boost the gain from -10 to +4dB.
3. Press **Cancel** to preserve the current configuration (ignoring any routing changes made thus far).
4. Press **Apply** or **OK** to save routing changes. A dialog box will inform you that the audio application must be restarted for your changes to take effect.

## LDI-12T

Items on the **LDI-12T** tab control include Input/output converter gain, Digital I/O format, Digital de-emphasis and SCMS (Serial Copy Management System) copy protection code.



**Gain Levels** are controlled by on screen faders with the level 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  $-10\text{dBV}$ , 14dB below digital full-scale.)

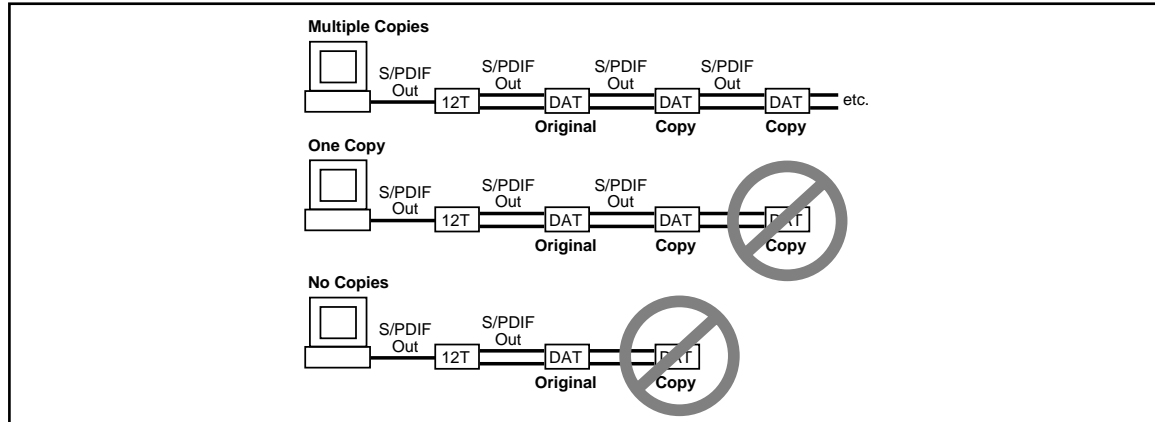
Clicking the **Link** button between fader pairs links them as a stereo pair for convenient identical gain setting. The left fader's value is always favored when Link is used to make the faders operate at the same level. Control clicking on the **Link** button will link the fader heads while maintaining 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.



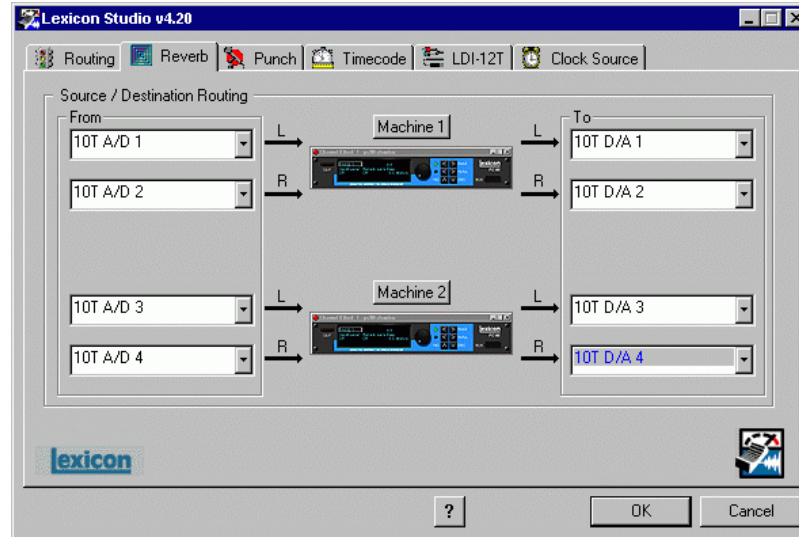
**Input/Output** controls allow you to select S/PDIF or ADAT as the digital format for the optical input and output connections, and to select between XLR or RCA connectors for the analog inputs. Conversion between optical and coaxial (RCA) S/PDIF is possible by selecting **S/PDIF OPTO** and **S/PDIF RCA** for input or output and vice versa. 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.) You can however use **ADAT OPTICAL** and **S/PDIF RCA** simultaneously.

**De-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.

**SCMS/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 tab allows you to route the PC-90's two DSP engines independently through any of the connectors of your audio interface. Note that only those I/O selections which are currently enabled in the Routing tab will appear as Source and Destination selections. (Sends and Returns from Cubase will also appear as selections if they have been enabled.)

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 pop-up menu 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.

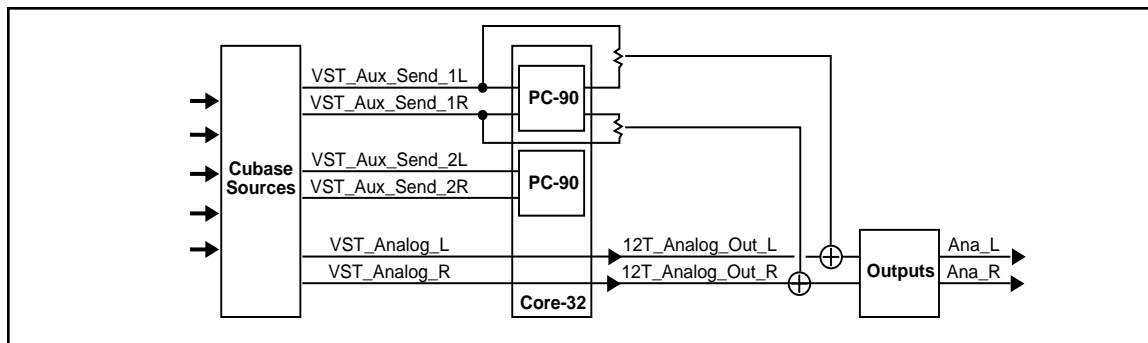
NOTE: Input sources must be routed one at a time.

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.

All I/O selections which have been enabled in the Routing tab are available as Sources/Destinations to each PC-90.

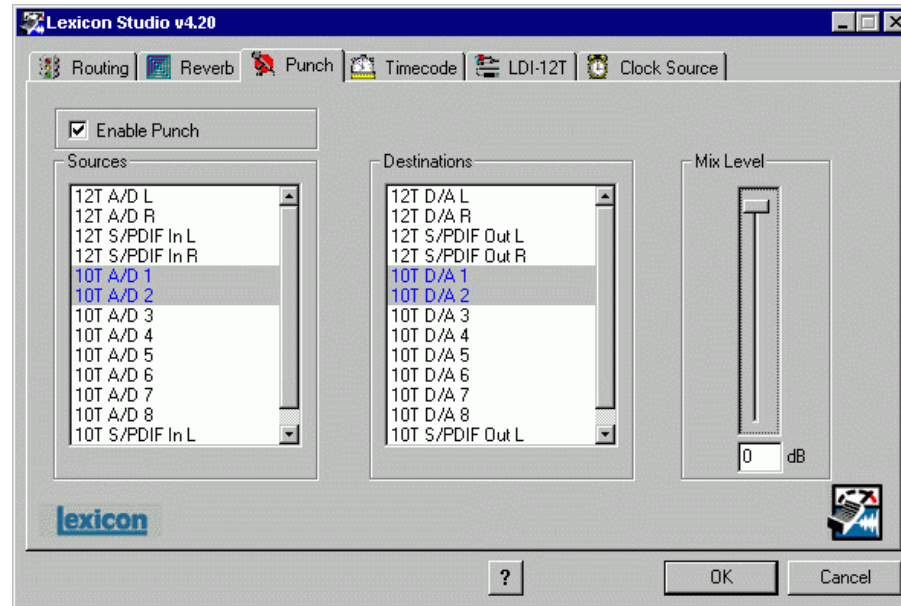
The following diagram shows a flow chart of a possible routing example. Note that the PC-90 receives its input from Cubase VST via the **VST\_Aux\_Send\_1L** and **VST\_Aux\_Send\_1R** sources. These sources route any audio assigned to that bus into the PC-90.



Note also that, as **12T\_Analog\_Out\_L** and **12T\_Analog\_Out\_R** are selected as PC-90 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 Destinations 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.

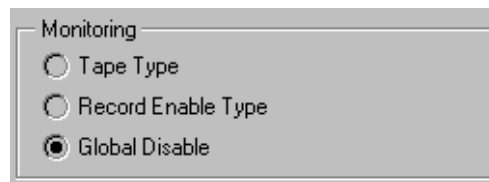
All computer operating systems have a delay when applications use host processing. The delay will vary depending on the buffer size of the audio hardware. (You can observe the latency of your particular computer configuration on the System Setup page in Cubase.) This means that, when playing along with 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 where the input is typically monitored when the transport is in Stop or Record and muted during Play, the input signal is monitored *all the time*.

### **Configuring the Punch feature**

1. Select **Enable Punch** to turn on monitoring.
2. Select as many input sources as you like, clicking multiple selections with the **shift** or **Ctrl** key. Note that only those I/O selections which are currently enabled in the Routing tab will appear as Source and Destination 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.



In order for audio to be heard through Punch, the corresponding I/O bus must be active in Cubase. For example, to hear audio from an LDI-10T in 1 & 2, that input bus must be activated in the Cubase Inputs window (in the Audio menu).

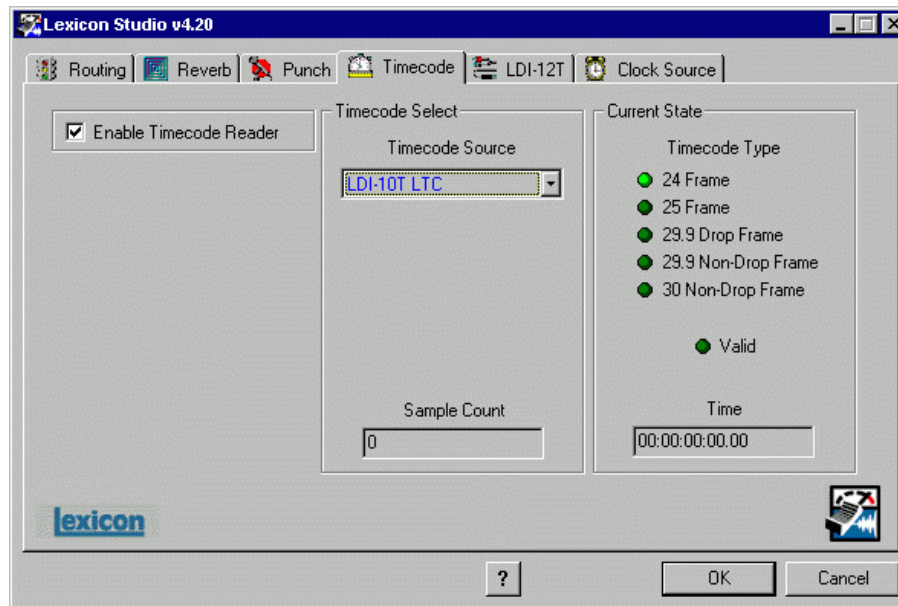
### The MIX Level Control

**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.

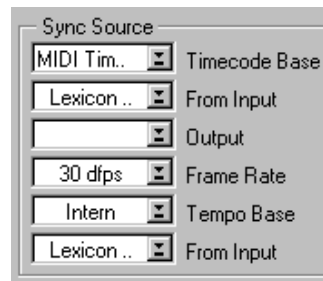


The Timecode Source list includes: LDI-12T LTC, LDI-12T ADAT and LDI-10T LTC. For systems running multiple 10T's, note that only the LDI-10T **A** can be designated as a Timecode Source.

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

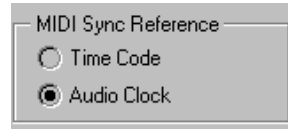
The LDI-12T and LDI-10T use a MIDI driver to convert the Longitudinal Timecode (LTC) supplied by the front panel Timecode connector to MIDI Timecode (MTC). Note that, as of this writing, the ASIO Driver specification is being updated to incorporate direct support of timecode. The following procedure describes the current procedure for setting up Cubase for use with timecode on the PC — as audio applications grow to support this new specification, this procedure may change slightly.

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.



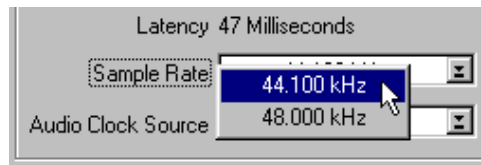
### ***Setting the Audio Sync Clock Reference***

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 48kHz are selected from the **Audio I/O** pull down menu.

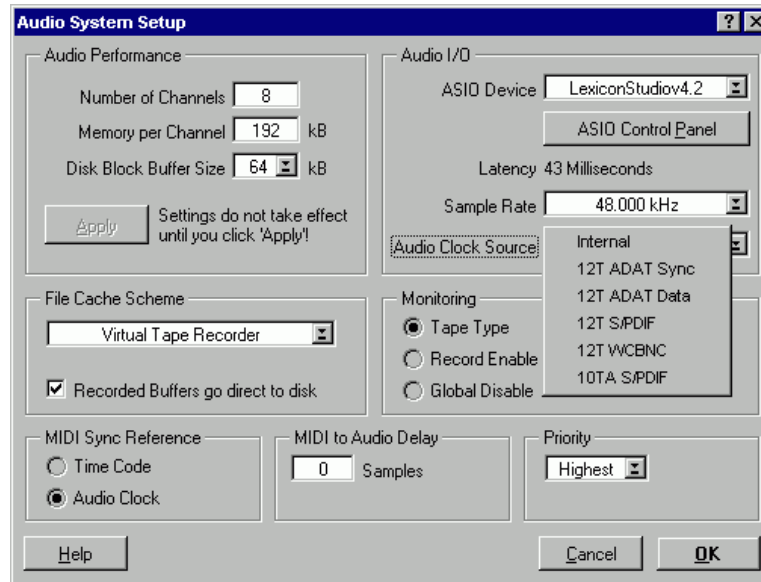


### ***Word clock***

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

Only connected devices will appear as available clock sources.





### Word Clock Sources

Internal	Selects the word clock generated by Cubase as the system master.
12T ADATSync	Selects the word clock provided on the ADAT Sync connector as the system clock.
12T ADATData	Selects the word clock provided on the ADAT optical data connector as the system clock.
12T S/PDIF	Selects the word clock provided by the S/PDIF signal (RCA or optical) as the system clock.
12T WCBNC	Selects the word clock provided on the Word Clock BNC connector as the system clock. This connector provides 75 $\Omega$ termination to the Word Clock chain.
10TA S/PDIF	Selects the word clock provided by the S/PDIF signal as the system clock.
10TB S/PDIF	Selects the word clock provided by the S/PDIF signal as the system clock.

10TC S/PDIFSelects the word clock provided by the S/PDIF signal as the system clock.

NOTE: If you are inputting audio data via ADAT or S/PDIF, but leave the Clock Source set to **Internal**, you will still hear audio but because the digital audio stream has no clock reference, you will get periodic audio ticks due to sample slip. Make sure your clock is set for your specific input source.

### ***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.

### ***MIDI, Automation, and Audio Editing***

Functions of Lexicon Studio such as panning and PC-90 parameter values are open to MIDI control and automation. The level of this functionality available depends upon the level of implementation provided by your audio software manufacturer.

Consult your software manufacturer's user guide for information regarding the functionality of these features.

A musical staff consisting of five horizontal lines. A large, bold, black number '4' is positioned on the right side of the staff, overlapping the top two lines.

# 4

## **MMIO**

### *Compatibility*

Lexicon Studio is compatible with two types of driver architecture, ASIO and MMIO.

ASIO (Audio Streaming Input Output) developed by Steinberg GmbH, allows compatibility between Lexicon Studio and ASIO-capable programs such as Steinberg Cubase VST, Opcode Vision DSP and Emagic Logic Audio.

MMIO (Microsoft Multimedia Input Output) is a standard developed by Microsoft on the PC that virtually every audio software developer has adopted for broad band compatibility between audio software and audio hardware.

These make Lexicon Studio compatible with a host of PC audio applications, such as:

- Steinberg Cubase VST
- Steinberg WaveLab
- Steinberg ReCycle
- Steinberg ReBirth
- Emagic Logic Audio
- Syntrillium Cool Edit Pro
- CakeWalk Pro Audio
- SEK'D Samplitude 24/96
- Sonic Foundry Sound Forge
- Sonic Foundry Acid
- JBL Smart

## LexPanel.exe

Configuration of your Lexicon Studio System for MMIO applications is done using the LexPanel.exe. Access to the Control Panel can be achieved in several ways.

1. A Lexicon Studio launcher icon is installed at the bottom left of your Windows taskbar. To launch LexPanel.exe, double-click on this icon. (NOTE: If you ever wish to remove this icon from your taskbar, simply remove it from C:\Windows \Start Menu\Programs\Startup).



2. LexPanel.exe is located in C:\Windows\System. You may launch the LexPanel.exe by double-clicking on the icon in the taskbar.

Once launched, the method of selecting Tabs for different pages, will be familiar to users of Windows and other Microsoft products. The selections are: Routing, Reverb, Punch, Timecode, LDI-12T and Clock Source.

NOTE: Some of the functions of Lexicon Studio (such as Timecode) do not get passed in MMIO until the LexPanel.exe has been booted once. Before each audio session, make sure that you boot the Lexicon Control Panel before using your audio software.



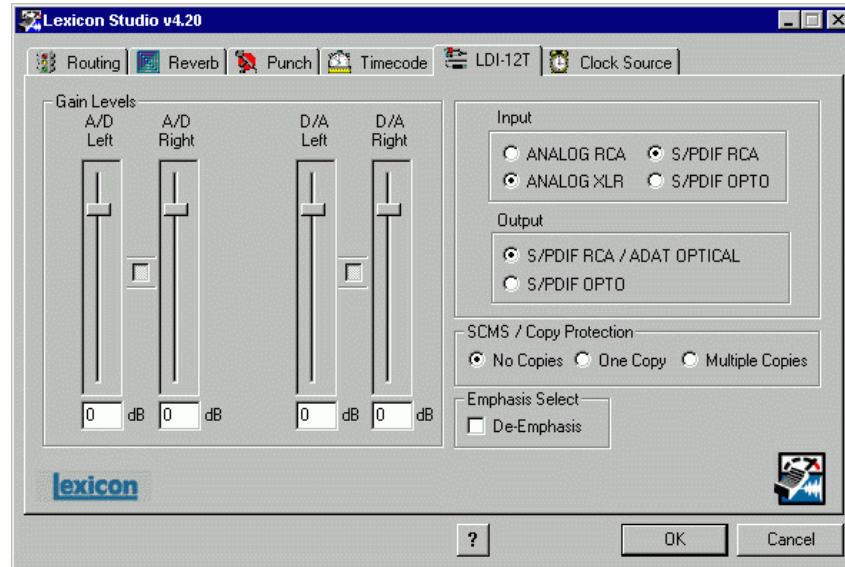
## Routing

This tab lets you determine which of the connected devices' I/O streams are active. Disconnected devices will be unavailable for selection (greyed out). I/O streams which are activated here become available as a clock source, as a source/destination for reverb and punch, and for use in the audio application.

1. No more than 32 streams of I/O can be selected. When an attempt is made to select more than 32, a pop-up message box will report that this is not allowed.
2. The high gain buttons beside the 10T analog inputs will boost the gain from -10 to +4dB.
3. Press **Cancel** to preserve the current configuration (ignoring any routing changes made thus far).
4. Press **Apply** or **OK** to save routing changes. A dialog box will inform you that the audio application must be restarted for your changes to take effect.

## LDI-12T

Items on the **LDI-12T** tab control include Input/output converter gain, Digital I/O format, Digital de-emphasis and SCMS (Serial Copy Management System) copy protection code.



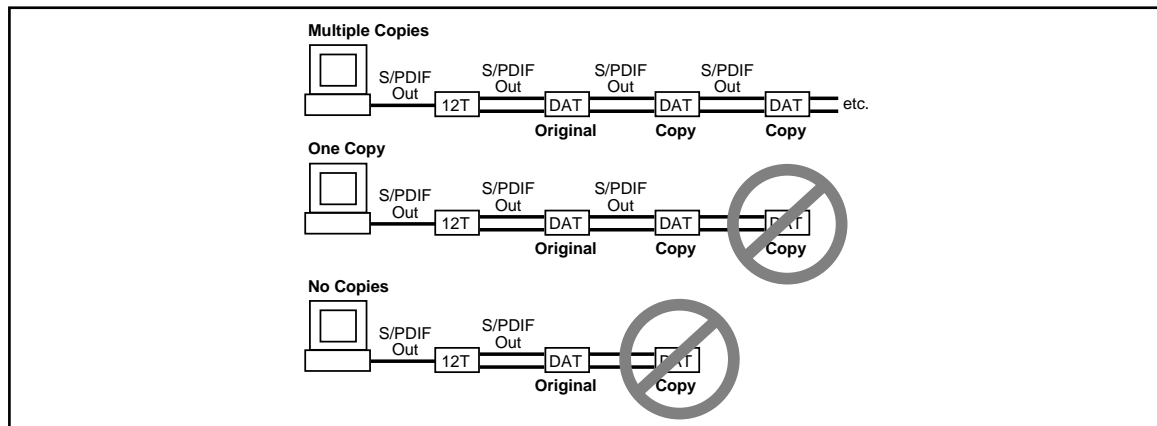
**Gain Levels** are controlled by on screen faders with the level displayed numerically at the bottom. A gain of 0dB sets unity gain from XLR input to XLR output, where +4dBu is 14dB below digital full-scale. (For the RCA inputs, 0dB corresponds to  $-10\text{dBV}$ , 14dB below digital full-scale.)

Clicking the **Link** button between fader pairs links them as a stereo pair for convenient identical gain setting. The left fader's value is always favored when Link is used to make the faders operate at the same level. Control clicking on the **Link** button will link the fader heads 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 **S/PDIF OPTO** and **S/PDIF RCA** for input or output and vice versa. 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.) You can however use **ADAT OPTICAL** and **S/PDIF RCA** simultaneously.

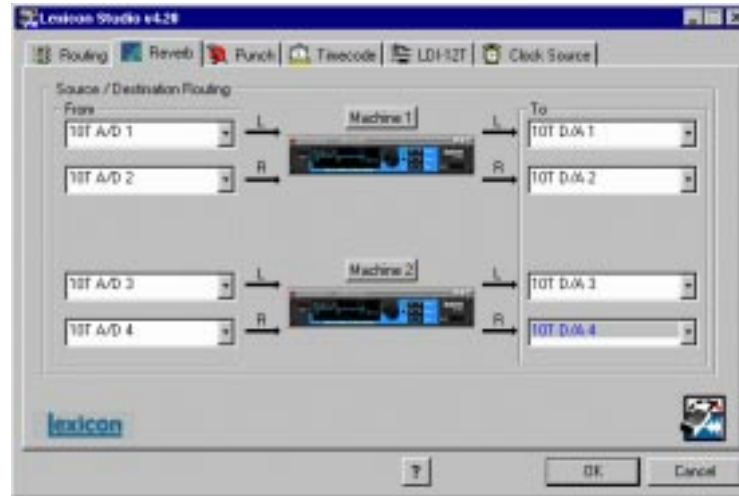
**De-Emphasis Select** enables de-emphasis to be applied to the analog output. This function applies only to the analog converter, and does not affect signals from the S/PDIF or ADAT interface.

**SCMS/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.



## PC-90 Routing

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. Note that only selections currently enabled in the Routing tab will be available as Sources and Destinations. (Cubase Sends and Returns will also be available if they have been enabled.)
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.

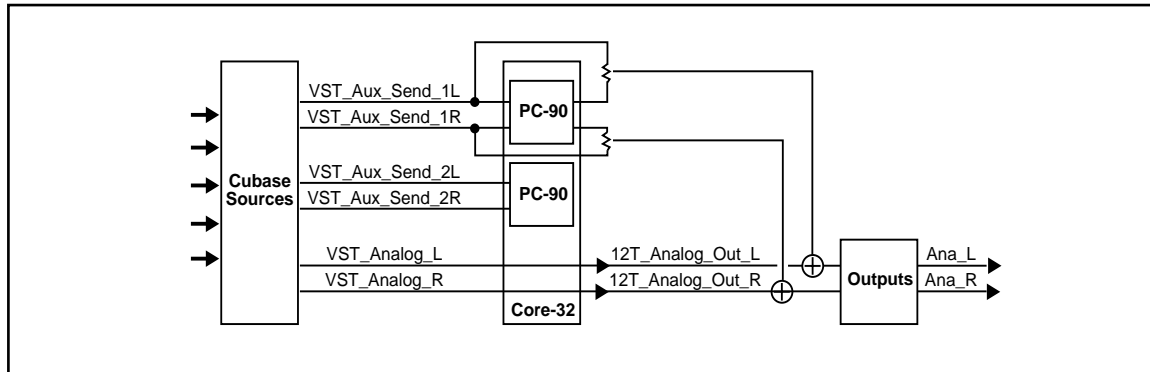


To assign a destination to any of the output channels in your PC-90:

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.

All resources that have been enabled in the Routing tab are available as Sources/Destinations to each PC-90.

The following diagram shows a flow chart of a routing example. Note that the PC-90 receives its input from the MMIO application via the **MMIO\_Analog\_Out\_L** and **MMIO\_Analog\_Out\_R** sources. These sources route any audio assigned to the analog outs 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 the audio application, which should also be routed to the analog outputs. This allows monitoring of PC-90 reverb and all audio from your audio software through the same outputs of the interface.

### ***PC-90 Interface***

The PC-90 can be fully integrated with any MMIO audio application. Unlike ASIO applications where the PC-90 is plugged in, much like a VST plug-in, the PC-90 only needs to be routed within MMIO. The reverb programs may be edited by clicking on the **Machine 1** or **Machine 2** buttons to engage the PC-90 user interface.

### ***PC-90 Presets***

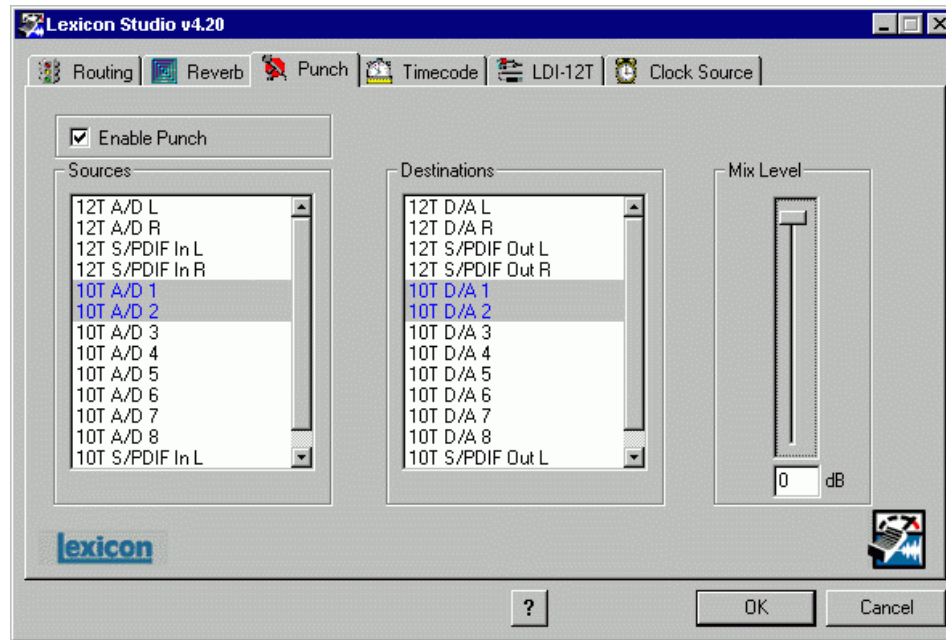
The PC-90 has 100 factory loaded programs, and 100 open slots for user preset storage. The factory presets are fully editable, however if you wish to save your edits, you must copy the preset into a user slot. In future versions of Lexicon Studio MMIO software, you will be able to create and save effects within banks that can be saved and reloaded.

The programs designed for MMIO have been saved with a Mix value of 50% wet. The programs have been saved this way because most MMIO applications do not support VST plug-ins, and therefore the PC-90 must be used as a master effect over the entire mix. You may need to tailor the Mix setting according to the needs of your particular recording. For full descriptions of the factory presets, see *Chapter 6: Reference*.

For more information on routing and control of your PC-90 and its parameters, 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 Destinations 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. This means that, when playing along with previously recorded tracks, the track you are playing live 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. Note that only selections currently enabled in the Routing tab will be available as Sources and Destinations.
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. Disable monitoring within the audio application that you are using so that a delay is not heard from the input signal.

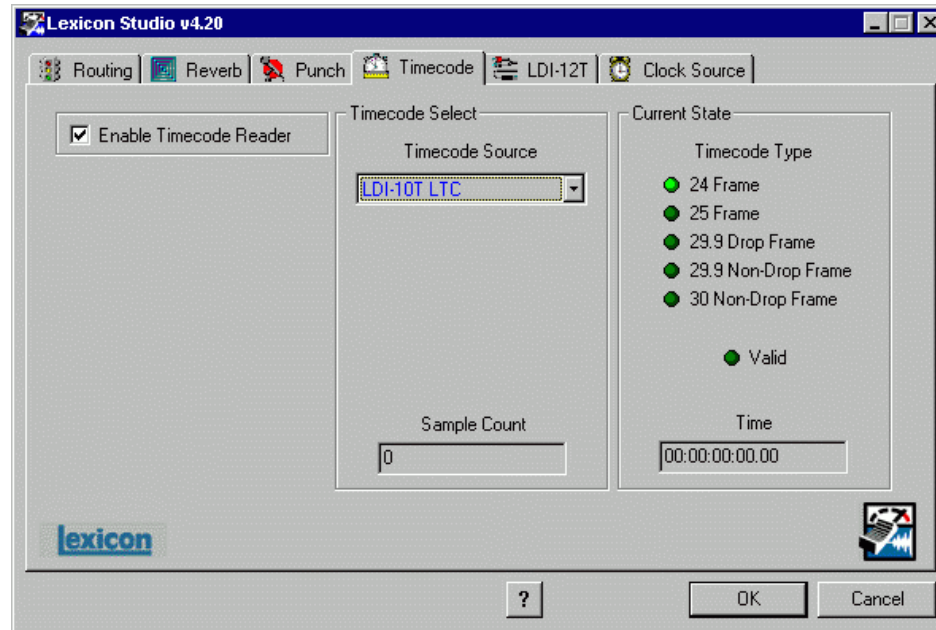
### **The MIX Level Control**

**Mix Level** allows you to adjust the volume of the selected Punch sources that are mixed with the MMIO output assigned to the same outputs.

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.



The Timecode Source list includes: LDI-12T LTC, LDI-12T ADAT and LDI-10T LTC. For systems running multiple 10T's, note that only the LDI-10T **A** can be designated as a Timecode Source.

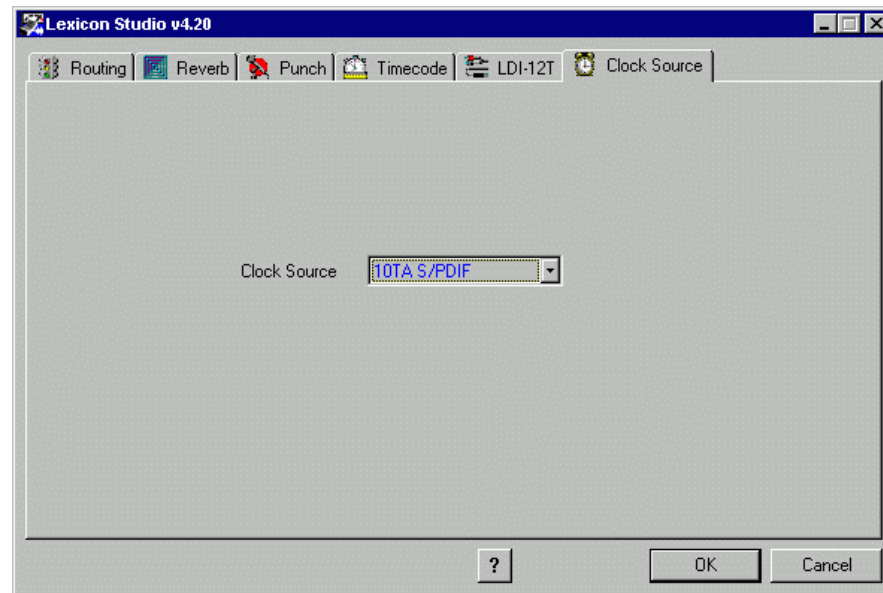
### ***Setting Up the Audio Application to Sync to Timecode***

The LDI-12T and LDI-10T use a MIDI driver to convert the Longitudinal Timecode (LTC) supplied by the front panel XLR to MIDI Timecode (MTC).

Each audio application deals with setup for Timecode differently. Please refer to your software applications user manual for instruction on how to configure your software to work with Timecode.

### ***Clock Source***

This control panel (active only for MMIO) enables the configuration of where Lexicon Studio accepts its clock source from.



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### Word Clock Sources

There are five different clock sources that Studio can lock to. Remember that Lexicon Studio can only lock to one digital source at a time, so if you plan to send in streams of different types of digital audio, make sure they are all utilizing the same clock (such as word clock). The available clock sources are:

Internal	Selects the word clock generated by Cubase as the system master.
12TADATSync	Selects the word clock provided on the ADAT Sync connector as the system clock.
12TADATData	Selects the word clock provided on the ADAT optical data connector as the system clock.
12TS/PDIF	Selects the word clock provided by the S/PDIF signal (RCA or optical) as the system clock.
12TWCBNC	Selects the word clock provided on the Word Clock BNC connector as the system clock. This connector provides 75 $\Omega$ termination to the Word Clock chain.
10TA S/PDIF	Selects the word clock provided by the S/PDIF signal (RCA or optical) as the system clock.
10TB S/PDIF	Selects the word clock provided by the S/PDIF signal (RCA or optical) as the system clock.
10TC S/PDIF	Selects the word clock provided by the S/PDIF signal (RCA or optical) as the system clock.

### 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.

NOTE: Some MMIO applications close their channels when the transport is in Stop mode. This will result in loss of clock or timecode when syncing to digital S/PDIF or word clock. Most applications will allow you to leave the channels open in its Preferences. You should do so, if this functionality is provided.



# 5

## 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, the PC-90 provides outstanding sonic quality with no burden on your computer processor's performance.

### The Interface



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

### ***Inserting PC-90 within a VST Application***

The interface for the PC-90 is available as a plug-in. Plug-ins are accessed differently in different software applications. The following procedure details the steps to open a PC-90 plug-in in a VST-capable program such as Cubase VST or Opcode Vision DSP. This procedure will vary depending on your application.

### ***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 is off (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. As you will probably need to adjust the send levels while doing this, leave the EQ window open.

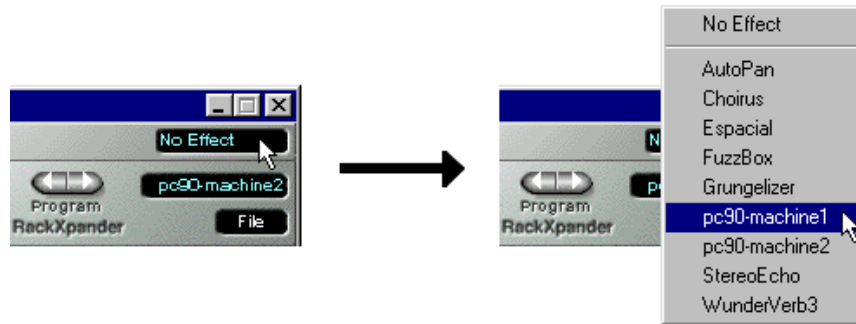
### Selecting PC-90 and Making Settings

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 stacked processors.



Pull down the pop-up menu in the processor's upper right corner and select PC-90 Machine 1 or PC-90 Machine 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 in order 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 Buttons. 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. On the keyboard, 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.

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** knob. To make small adjustments, click on the knob, hold down the mouse button and move the pointer away from the knob before turning it. The further away from the knob the pointer is, the finer the value adjustments.
4. Use the **INC** and **DEC** buttons for super-fine manipulation of the active parameter.

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## Using PC-90 within an MMIO Application

### *Routing an Audio Channel through the PC-90*

Like traditional effects devices, the PC-90 can be used in two ways: as auxiliary effects or as line effects.

Most effects processors are used as *auxilliary effects*. On a typical console, this is achieved by using the aux send and returns to send specific channels to the effects devices, then back to the board where the end result is mixed with the dry outputs of the board. Typically, the Mix ratio is kept at 100% wet, since the signal is mixed with the board outputs. Most of the multi-channel applications available today (such as Cubase VST, Logic Audio, Cakewalk Pro Audio) simulate this functionality with virtual aux sends and returns. For an example on how to set up PC-90 within an application that supports VST plug-in architecture, see the previous section: *Using PC-90 within an ASIO Application*.

Effects devices can also be used as line effects by placing them in series with the signal chain. Steinberg WaveLab is capable of doing this through its VST plug-in architecture, which does not require additional routing. MMIO programs that do not support VST plug-ins must route audio to the PC-90, where it is mixed into the entire signal. Typically, the Mix ratio for reverb in this situation should be set at 50%, to have a mixture of direct and reverberated signal.

Routing MMIO audio to and from the PC-90 is done in the Reverb tab of LexPanel.exe. To route audio:

1. Boot LexPanel.exe by double-clicking on the Lexicon Studio icon at the bottom right corner of your task bar.
2. Click the Reverb tab to view all MMIO routing and control options for the PC-90.



- Using the **To** and **From** pulldown menus for each PC-90 machine, route your audio according to which MMIO channels and interface outputs you would like to use. The picture below shows a typical setup for Machine 1, mixing the reverb and MMIO dry audio all through the analog outputs.



For more information on how to route MMIO audio to and from your PC-90, refer to *Chapter 4: MMIO*.

### Making Settings for the PC-90

The MMIO presets for the PC-90 are virtually identical to those available when using PC-90 with ASIO applications such as Cubase VST. The only differences are in the Mix values. As most MMIO applications will use the PC-90 as a line effect, the Mix values have been saved at 50% wet. If you are using an MMIO program that supports VST plug-ins (such as Steinberg WaveLab or Emagic Logic Audio), remember to change the Mix values in the presets you are using to 100% wet.

### Loading Effects

The MMIO version of PC-90 comes with 100 factory presets and 100 user slots where you can store your own presets or save edited versions of the factory presets.





A preset is loaded from the PC-90 interface by highlighting the preset name area, and using the rotary knob to scroll through the 100 factory and 100 user slots available. A preset is automatically loaded once the rotary knob is used to move to the new preset.

### Editing Effects

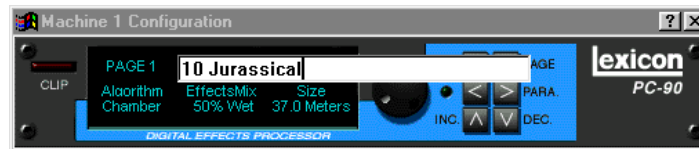
You can create your own programs or edit any of the presets with the editor. To open the PC-90 interface, click the **machine 1** or **machine 2** buttons in the Reverb tab on LexPanel.exe. The PC-90 Interface will open.



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** knob. To make small adjustments, click on the knob, hold down the mouse button and move the pointer away from the knob before turning it. The further away from the knob the pointer is, the finer the value adjustments.
4. Use the **INC** and **DEC** buttons for super-fine manipulation of the active parameter.

## Naming Effects

To name a preset, simply double-click on the area where preset names are displayed. An editable text field will appear that allows you to select any preset name you desire. Note that the editor will allow you to change the names of any factory preset, however you must save that edited preset into a user slot if you wish to permanently change the name of the preset. For descriptions of all presets, refer to *Chapter 6: Reference*.



## Saving Effects

The MMIO version of the PC-90 provides 100 user slots for your edited presets. Once you have tweaked an effect to the point you are happy with it, and have named it, you can store it as follows:

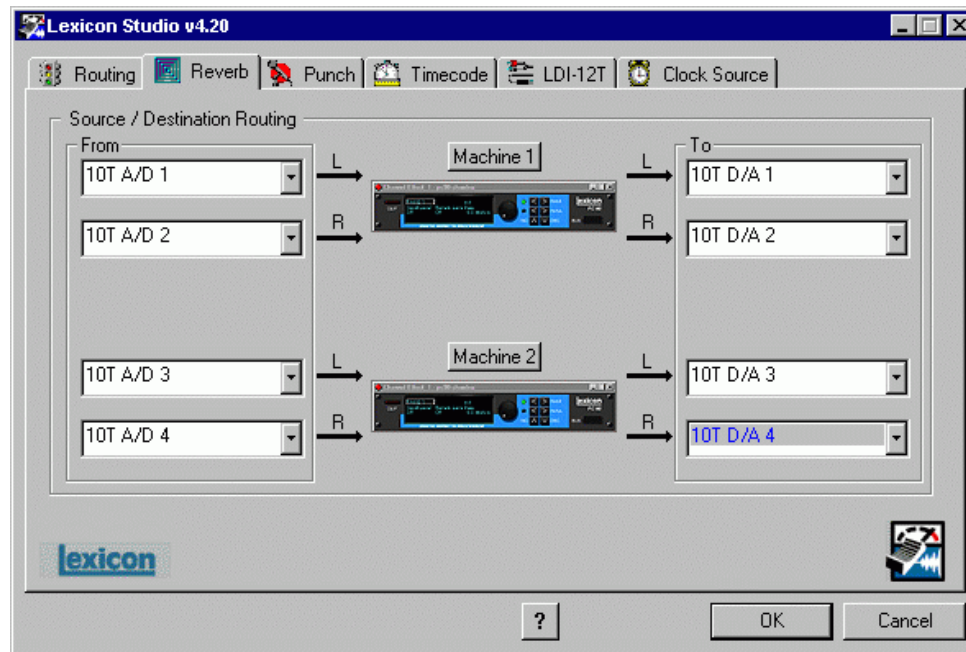
1. Click on a preset to highlight it.
2. Move the rotary knob to enter Save mode. A window will appear displaying a complete list of the user slots, including any programs currently residing in the slots. To save a preset, highlight a slot by clicking on it, and clicking the **Save** button.
3. To load a saved preset, use the rotary knob to scroll through all 200 presets. The 100 user slots appear after the factory presets, so user slot 1 is at roughly the 12 o'clock position.



## 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. Click the left mouse button on the down-arrow of an input selector to open a menu displaying all Sources available for routing to the 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.

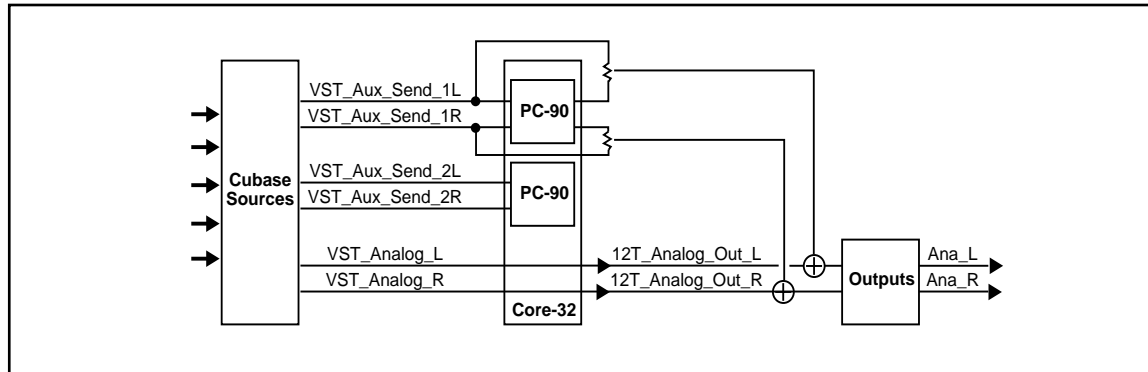
NOTE: Input sources can only be be routed one at a time.

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 an output to designate it as an output destination. The output selector will display your selection.

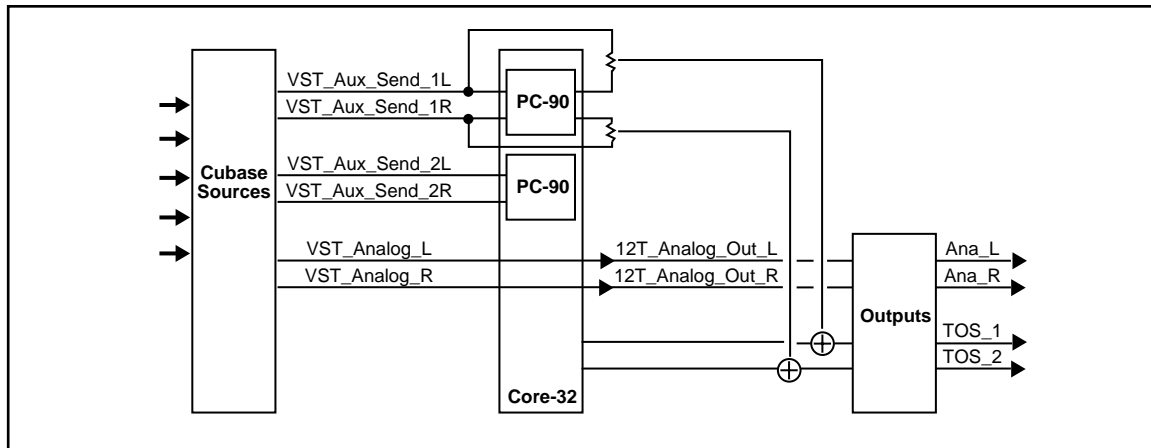
NOTE: Only resources that have been enabled in the Routing tab are available as sources/destinations to each PC-90.

The diagram below is a flow chart of a routing example. 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 bus 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 previous 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 the 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 when playback is stopped, 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. 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 play back automation in your mix:



1. Check that the Audiomix Track or Part is 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, for example, while you're recording fader movements for another mixer channel, 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.

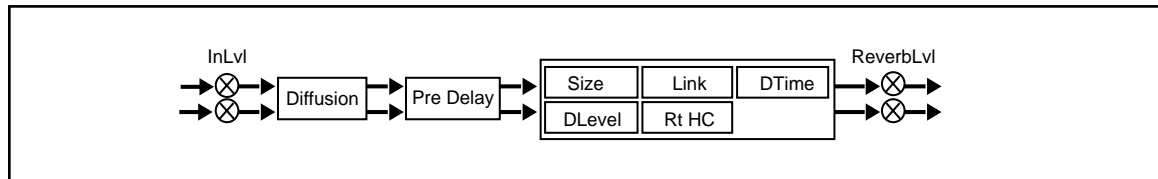


# 6

## 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.

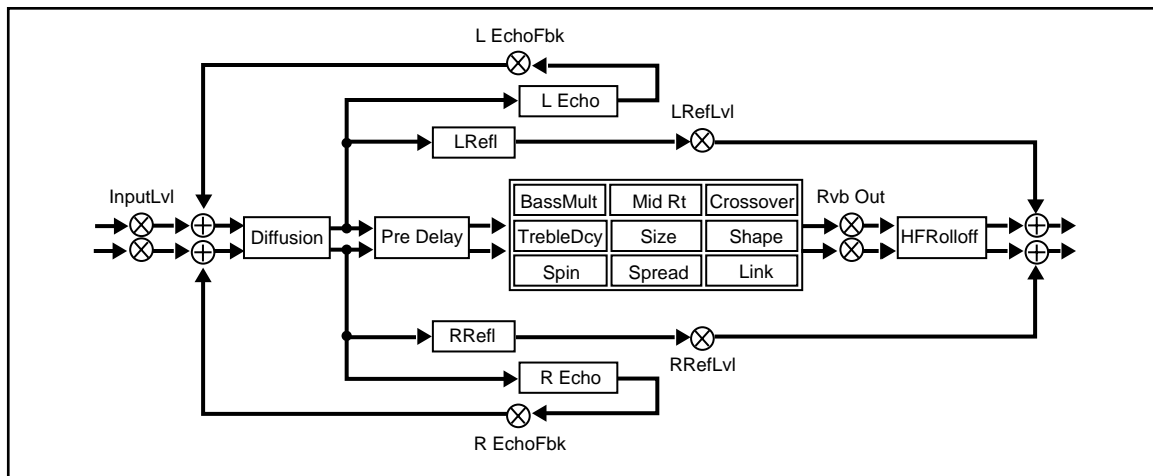
This algorithm generates primarily the strong reflections which appear in the first few hundred milliseconds of the reverberation process. These early reflections constitute the primary audible effect, giving you the impression of a hall surrounding you while the music is playing. To avoid any coloration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

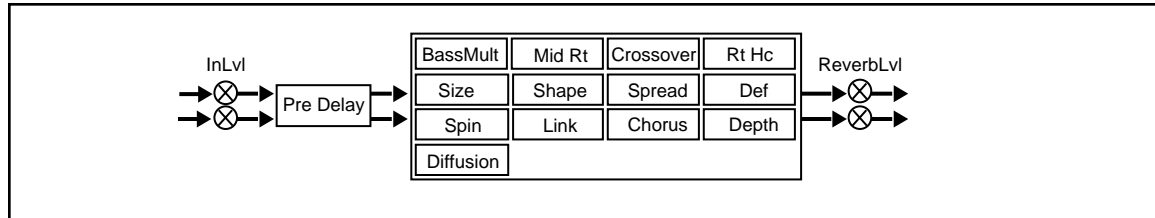
When using Ambience in a mix, it is best to use a stereo send to the PC-90, carefully matching the panning of the various close-miked sources to their positions in the mix. Leave the Mix control at 100% wet. the apparent distance of each source can be controlled by the level of its feed.

## Chamber



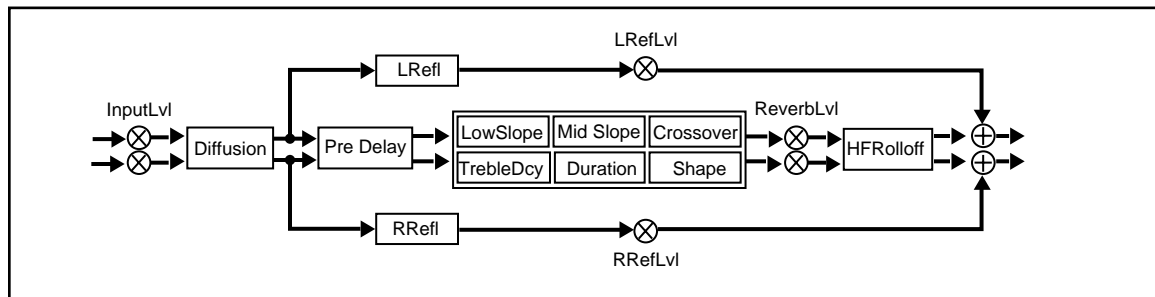
Chamber provides an even, relatively dimension-less reverberation, with little change in color as the sound decays. The initial diffusion is similar to the Concert Hall algorithm, but the sense of size and space is much less obvious. This characteristic, along with low color in the decay tail, makes 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



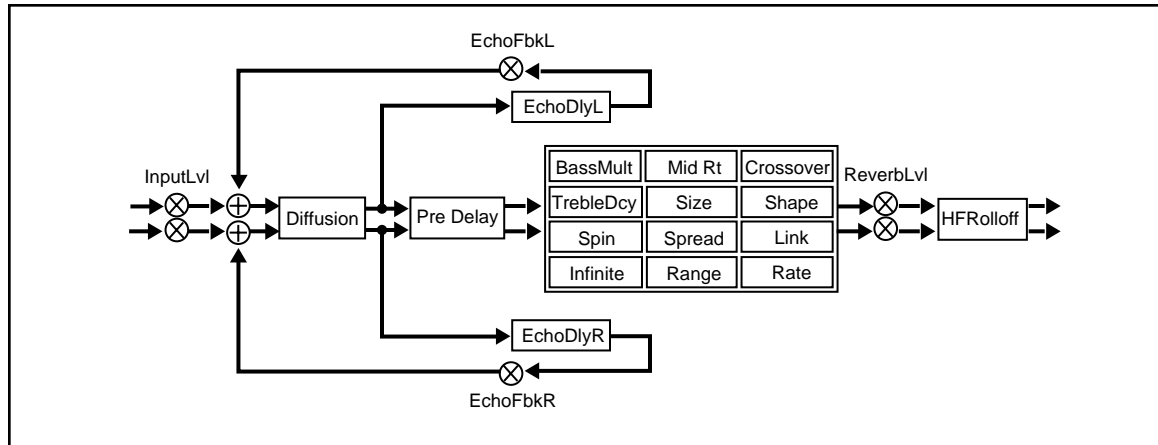
This algorithm emulates a real concert hall. The reverberation is very clean, and designed to remain behind the direct sound – adding ambience, but leaving the source unchanged. This effect has a relatively low initial echo density which builds up gradually over time.

## 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.

## *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.

### ***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.



### ***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.

## *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).

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**

## **PC-90 Application Notes**

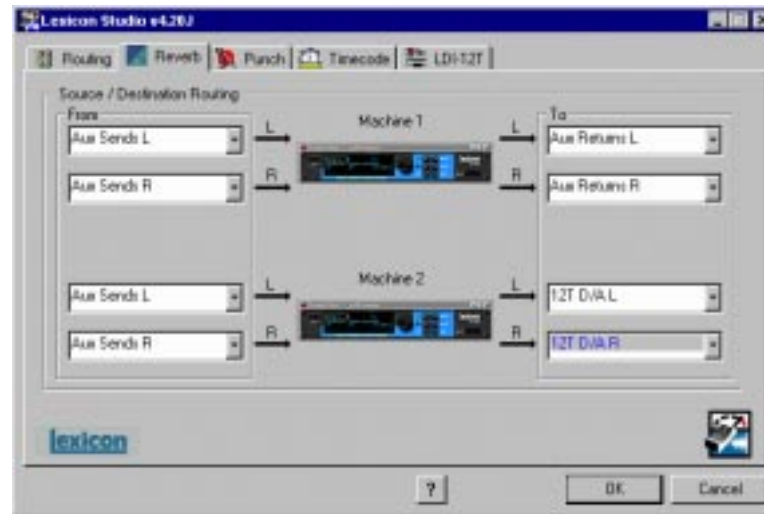
### *Routing PC-90 Reverb via the Aux sends in VST*

1. From the ASIO control panel select the Reverb page.
2. In the “From” section for the left input to Machine 1 assign **Aux Send 1L**
3. In the “From” section for the right input to Machine 2 assign **Aux Send 2R**
4. In the “To” section for the left output of Machine 1 assign **D/A L**
5. In the “To”section for the right output of Machine 2 assign **D/A R**
6. From Audio menu select **Effects** and assign PC90-machine 1 to **effect# 1**
7. Turn the power on and the effect master knob to full CW position
8. From the master window, verify that the PC90-1 and PC90-2, bus 7 and 8, are active. Verify that the output of those buses are set to AuxSend 1 and AuxSend 2.
9. From the VST mixer window select **FX**.
10. Verify that send 1 is assigned to **PC-90 machine 1**
11. Turn on Send 1 and turn the volume knob to approximately 3:00 o'clock. Adjust according to recording needs.

### ***PC-90 Cascade Routing***

PC-90 routing allows you to daisy-chain the PC-90 reverb engines to make a dual-engine reverb. This is useful for creating your own “super-reverbs”, for example, using Machine 1 for early reflections and Machine 2 for the main reverb. In the early days of digital reverberation, creating your own reverbs was essential to get great-sounding reverb. Today, this is still common practice for many people, so we have enabled the PC-90 to be a great reverb construction kit.

To do this, select the inputs and outputs in the Reverb Control Panel as indicated:



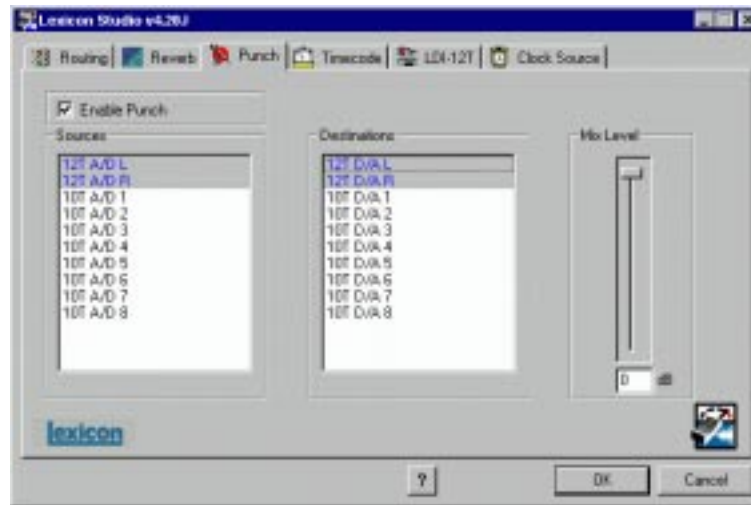
1. Set Machine 1 inputs to **Aux Send 1L** and **Aux Send 1R**.
2. Set Machine 1 outputs to **Aux Ret 1L** and **Aux Ret 1R**.
3. Set Machine 2 inputs to **Aux Ret 1L** and **Aux Ret 1R**.
4. Set Machine 2 outputs to **12T D/A L** and **12T D/A R**.
5. Make sure the channel of the audio you wish to process is sending from Aux Send 1 into the PC-90.

You now have created a cascade reverb within PC-90. This will give you immense flexibility when creating your own “super-reverbs”.

### ***Monitoring the PC90 with No Latency while Recording***

The following example shows how to add reverb to your monitor mix without adding the reverb to the input signal. The input signal and the reverb are monitored in real time with no latency. The input signal will record to disk dry. Reverb processing can be added to the dry signal during mix down.

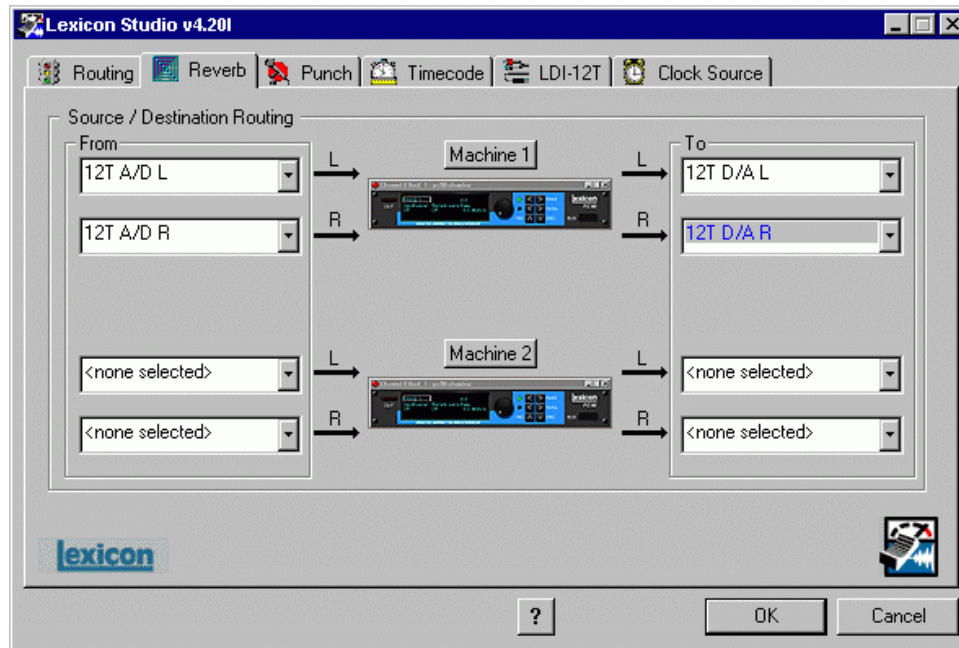
Set up the punch tab in the ASIO Control Panel:



1. From the Control Panel, select the Reverb Routing tab.
2. Set Reverb Routing as follows:
  - 12T A/D L -> PC90 machine 1 -> 12T D/A L
  - 12T A/D R -> PC90 machine 1 -> 12T D/A R

3. Set the channel you want to record to A/D 1 Left, Right or both on a stereo track.
4. Click on the **Edit** button to open the PC90 Control Panel and adjust **Reverb Mix** (wet to dry ratio) for monitoring.
5. Start your recording.

Set up the Reverb (for either Machine):

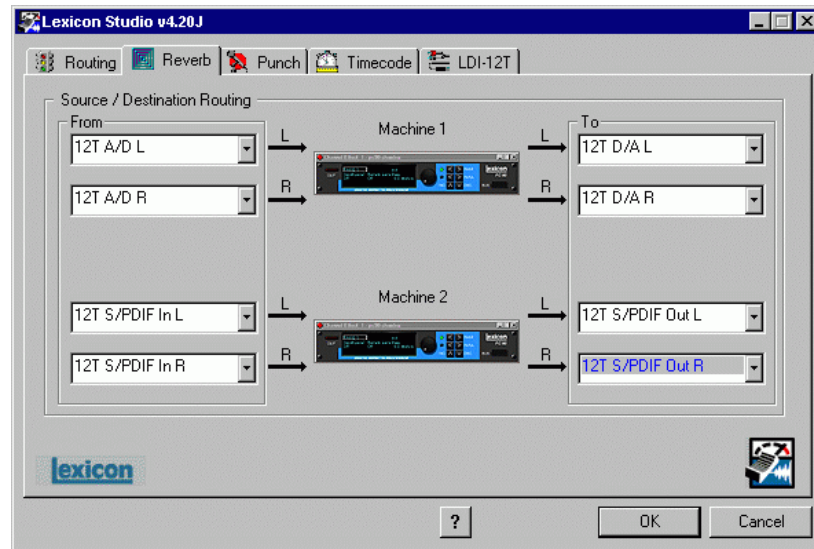


- Dry level can be adjusted by the **Mix Level** parameter in the ASIO Control panel Punch Tab.
- Reverb level is controlled by the **ReverbLvl** parameter in the PC90 interface.



## Using the PC-90 as Outboard Gear on the PC

Using MMIO, you can configure your PC-90 to be used as a stand alone reverb box.



The source (send from your console) is routed directly to the inputs of the PC90 engines and the outputs can be routed (analog or digitally) to the returns of the mixer. There is no latency.

You are not restricted to the routing in this example. It is possible to use any input as a Source and any output as a Destination.

### Notes:

This example requires the audio clock to be set to SPDIF12T. If only analog sources are being used this is not necessary.

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# **Appendix B**

## ***Troubleshooting for PC***

### *Installation troubleshooting*

#### ***Problems Starting Autodetection***

If there was an audio card from another manufacturer installed in your system previous to this installation but Lexicon Studio will now replace it, you should run **Uninstall** to remove all components and reset your system. 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 will instruct you to:

- 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 and suggest solutions.

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 the computer manufacturer's user guide for more information on its PCI architecture.

### ***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.

### ***Hardware Recognition***

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 contact them.

### ***AGP (Accelerated Graphics Port)***

Some PC motherboards can use an AGP graphics adapter. This standard was designed to move the graphics operations off of the PCI bus onto a dedicated bus — the AGP port which 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 for Audio Applications***

Following are some suggestions that may improve your computer's 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 they become 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***

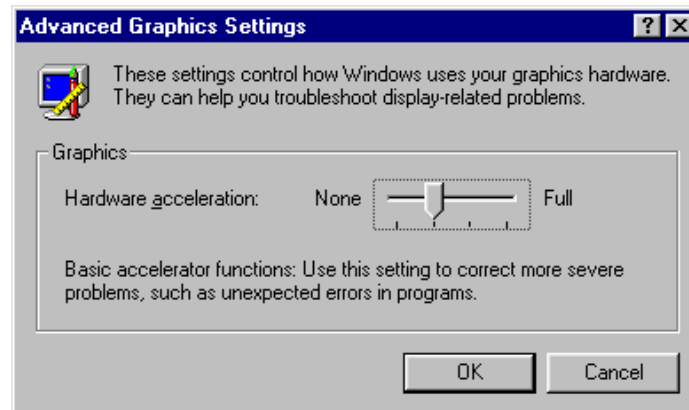
ControlPanel/System/Performance: Make sure it says "Not Installed" after Disk Compression

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

Sets the number of kilobytes that Win95 reads ahead. (Reading ahead makes the audio drive work harder than necessary.)

### ***Graphics Hardware Acceleration***

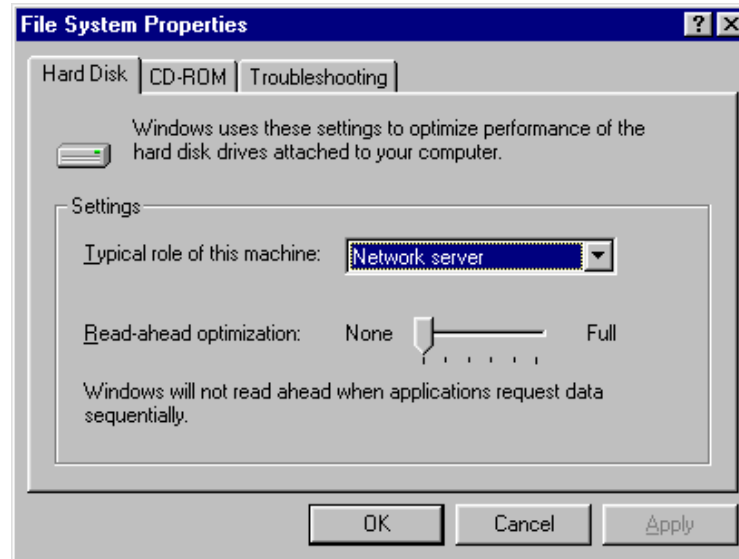
In non-AGP computer systems (computers using VGA cards), graphics acceleration settings can force your video card to hog the PCI bus. Crackling and zipping in the audio while opening and moving windows is a warning sign of this.



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 50-100MB.

### ***Use A Dedicated Drive For Audio***

Use the fastest drive available to you. Any drive sold as an AV drive will probably work. Try before you buy.

### ***Disable Screen Savers***

Screen savers, TSRs and Timeout features should be disabled. Disable any program or task that runs in the background.

### ***Disable Virus Protection Software***

Virus protection software typically runs in the background of the computer, taking up processing time from the CPU. Make sure your virus protection software is disabled while running your audio applications, but enable it periodically to check your system for viruses.

### ***Dedicate your Computer to Audio Work***

Computers are complex machines. The software that runs in them is equally complex. Dedicating a system to certain applications will minimize conflicts and other system problems. Try to dedicate your machine solely to audio purposes, and do your word processing, web browsing and financing on other machines.

### ***Keep Your Drivers Up To Date (vga, midi, etc.)***

Contact your card manufacturer for any updates that are available. Keep the old drivers on hand in the event that the new drivers cause more problems.

### ***Compatibility Mode***

In Control panel/System/Performance if Win95 tells you it is running in "Compatibility Mode" it means that Windows drivers are not being used for some device. Try to resolve the problem.

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## ***Edit System Files***

Editing of Windows and DOS system files should be attempted by experienced personnel only.

### ***Autoexec.bat & Config.sys -***

Settings for network drivers, CD-ROMS, video cards and the mouse may need to remain in these files but all other entries should be remarked out by placing REM in front of the line.

### ***win.ini -***

Load= or Run= should have no values after them.

### ***Set Disk Caching To A Fixed Amount***

Edit the system.ini [vcache] MinFileCache=x and MaxFileCache=x. Where x = 10-25% of the total system memory in Kilobytes.

### ***MS Plus***

- Disable “show window content while dragging” in Control Panel/Display/Plus.
- Disable low disk space notification in the System Agent and suspend the System Agent.
- Remove DriveSpace3 from Add and Remove programs

### ***Power Management Systems***

Some computers enable certain settings for power management of hard drives when they are not in use, dropping them into low power mode. Disable these settings for optimum performance. You will find these settings in Settings/Control Panel/Power.

Your particular BIOS may also have power management settings. These should also be disabled. Refer to your BIOS manual or contact its manufacturer for information on how to accomplish this.

**TIP: Back up your system on a regular basis**

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## Common Problems and Solutions

### **Installation & Hardware Recognition**

**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.

**Computer crash and reboot when Lexicon IDS is removed from the Device Manager.**

Go into the Device Manager, open the System devices category, highlight the PCI bus entry and click the **Properties** button. Under the IRQ steering tab make sure IRQ Steering is unchecked.

**Notes:**

Windows versions 4.00.950 and 4.00.950A will not exhibit this problem. These versions do not have an IRQ steering tab or the option to enable Windows to steer IRQs. IRQ steering is done by the BIOS.

Windows versions 4.00.950B and 4.00.950C will exhibit this problem only if the option to enable IRQ Steering is enabled. Windows, by default, does not have this option enabled.

(Refer to Article number Q182604 in the Microsoft knowledgebase for a complete description of PCI bus IRQ steering.)

**Installer can not find the first file during Lexicon Studio installation**

At the start of the installation process, after Windows has identified the new hardware, it searches for the drivers. In some cases Windows will not search the correct drive, and you must manually specify the path to the A:\ drive (if installing with floppies) or the D:\ drive if installing from CD-ROM by clicking the **Browse** button. Once you have done this, click **Finish**. All of the drivers will be copied from the proper drive.

**Problems operating the Lexicon Core-32 Diagnostics**

The Core-32 can be an ASIO device for only one audio application at a time. You cannot run Cubase VST (or any ASIO application) simultaneously with the Core-32 Diagnostics without receiving errors.

**Problems with Lexicon Studio Installation after removal of an Event Gina card**

The following error messages may appear when booting VST after removal of the Event Gina card and installation of the Lexicon Studio card and drivers: 'Could not create native audio engine', 'Could not load ASIO Lexicon Studio'.

The Event Gina card uses the same VENDOR ID number as Lexicon Studio. This VEN ID number is entered in the Windows Registry when Gina is installed and remains in the Windows registry after the Gina card is removed. When the Core32 is installed, the registry sees it as a previously installed Gina card and attempts to configure it as such. To reconfigure for Studio:

1. Run Regedit - Click the Start button, select RUN and type regedit, click OK

2. Click the “+” sign next to HKEY\_LOCAL\_MACHINE to open
3. Click the “+” sign next to Enum to open
4. Click the “+” sign next to PCI to open
5. Highlight the VEN\_1057&DEV\_1801 entry(s) and delete them.
6. Close regedit and reboot the computer.

Windows will automatically reconfigure the card and the registry entry upon reboot. If there is a GINA card installed it too will reconfigure.

#### **Studio sharing an IRQ with ‘IRQ Holder for PCI Steering’ in Device Manager**

This is normal. An IRQ Holder for PCI Steering may be displayed when you view the System Devices branch of Device Manager. An IRQ Holder for PCI Steering indicates that an IRQ has been programmed to PCI mode and is unavailable for ISA devices, even if no PCI devices are currently using the IRQ.

#### **Various problems if you have incorrectly updated your Lexicon Studio software**

The following problems may appear if you incorrectly update Lexicon Studio’s drivers and PC-90 plug-ins from V3.02 to a later version:

1. Clicking on the Launch Lxpanel icon in the System Tray causes the following message “Could not bring up Control panel” (with OK button)
2. IDS folder (this is renamed LxStudio Folder in later versions of software) was not deleted.
3. No driver file appear in the Lxstudio Folder

Reason:

Update Driver was performed incorrectly. Selecting NO, for ‘Do you want windows to search for a new driver?’ causes the Device Driver Wizard to use the old .inf file from the previous version installed. To remedy this problem:

Run the Device Driver Wizard again as follows:

1. From the Device manager select Sound, Video and Game controllers.
2. Double click on Lexicon IDS System.
3. Select the Driver tab
4. Select Update driver
5. Select YES (recommended) when the update Device Driver Wizard window opens.
6. Continue normally with the driver update

### **Audio Errors**

#### **Could not create native audio engine error message appears.**

Cubase VST may be looking for the AWE64 as a default card (if you've preinstalled one). If it can not be configured correctly, VST will not boot and you will not be able to select ASIO Lexicon Studio as the ASIO device. Set the ASIO Multimedia Setup to **Advanced options**.

Sync Reference: Sample Position - Output

Card Options

Activated: Full-Duplex, Start Input First

De-Activated: Open all Devices before Start

Audio Buffers: 5

Buffer Size: 5512

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Notes:

1. With these settings, the card will perform audio recordings with 16-bit resolution in full-duplex operation, while the output will give you 8-bit resolution. For a playback with 16-bit resolution, you must de-activate the input in Cubase VST in the Audio menu's "Inputs" window.
2. If you use an AWE 64 or the WaveSynth upgrade for other Sound Blaster cards, you must de-activate the Wavesynth/WG (Creative Music Synthesizer) MIDI output port in the Setup MME program which comes with Cubase VST. It may also be necessary to disable the Creative Instrument Mapper. With a SoundBlaster card, you cannot use the WaveSynth and audio recording capabilities simultaneously.
3. Certain combinations of System BIOS and Sound Blaster Drivers may produce a system freeze when Full Duplex mode is selected. Deselecting Full Duplex mode will usually make this problem disappear. Contact Steinberg or Creative Labs for the best driver version for your machine.

**White noise with enabled inputs in Cubase VST through Lexicon Studio**

Cubase Version 3.55r2 has input problems with Lexicon Studio hardware. Obtain the 3.55r3 updater from Steinberg's web site. <ftp://ftp.steinberg.net/dist/product updates/>.

**Core-32 busy timeout problems**

There have been reports of an error that displays "Core32 busy timeout" within a dialog box. There are several ways to troubleshoot this problem.

If your audio application boots and seems to function properly, check the following:

1. From the Lexicon Studio Control Panel, disable Timecode reader if you aren't sending timecode. If you are sending SMPTE into the timecode input connector, set time code source to **LDI 12T LTC**. If this source is set to **LDI12T ADAT** and no ADAT SYNC signal is connected and running, this error message will appear.
2. Set sync source to **Internal**.

If your audio application does not boot, check the following:

1. Is the card present under the Device Manager? To check, go to the Window's Control Panel and double click on SYSTEM. This will bring up the System Properties window. Click on the Device Manager tab and find the Sound, Video and Game controller icon (or Multimedia). Expand the category and look for Lexicon IDS Sound System. A yellow exclamation point next to Lexicon Studio Sound System indicates your driver has been improperly installed.
2. Is an IRQ assigned? Is memory I/O assigned? Highlight the Lexicon IDS Sound System and click on the **Properties** button. This brings up the Lexicon IDS Sound System Properties page. Click on the Resources tab. You will see a box called Resource Settings. If an IRQ and Memory are assigned, they will be displayed here. Also look below at the Conflicting device box to see if any conflicts are listed.
3. Is the PC-90 software installed correctly? Our installer copies files to the Cubase VST plugins directory. Score does not use this directory. Also, other programs that support VST plug-ins will need to have PC-90 properly installed for them. Make sure the PC-90 installer is pointed to the correct 'vstplugins' folder. If you have more than one application that supports VST plug-ins on your system, make sure you install PC-90 properly for each of them.

### **Aliasing noise heard with any inputs**

Aliasing or ring modulation noise associated with sample rate truncation heard when monitoring any of the Lexicon Studio inputs through the Cubase VST mixer has been reported on some systems. If you hear this noise on your audio, this noise can get recorded into the files to disk.

Qualifying the problem: This may be an interaction between the computer's PCI bus and the coredsp.pci code in Lexicon Studio version 3.02. This is more likely to happen on non-Intel based computers and slower Intel based computers.

1. Play back the demo song/audio files that came with your audio application. Do these previously recorded 44.1 or 48k .wav files play back normally through Lexicon Studio?
2. Is the audio quality good when monitoring with Lexicon Studio Punch enable and Global disable? (Not monitoring through the VST mixer)



3. Is the audio quality bad when monitoring through the VST mixer with punch disabled and Record or Tape Type monitoring enabled?

Solution: If you have answered 'yes' to the questions above, obtain new drivers for Lexicon Studio. Newer versions of the Coredsp.pci contained in version 3.10 have been optimized for slower and non-Intel based PCI chip sets.

### **PC-90 Problems**

#### **PC-90 is not selectable in Cubase VST as a plug-in.**

If the PC-90 machine 1 and PC-90 machine 2 are not selectable as an effect, the .dlls have probably not been installed into the proper 'vstplugins' folder. Default installation folders for Steinberg are:

For Cubase AudioVST: C:\Program Files\Steinberg\Cubase Audio VST\vstplugins

-or-

C:\Program Files\Steinberg\Cubase VST\vstplugins

For Score: C:\Program Files\Steinberg\Cubase Score VST\vstplugins

For Score Demo: C:\Program Files\Steinberg\Cubase Score VST Demo\vstplugins

#### **PC-90 reverb is assignable from the VST Effects window but can not be heard:**

Check bus-pop up on the PC-90 Rack Xpander. If this is set to manual, change it to an enabled Master output, such as PC90 1. PC90 1 or PC90 2 must be enabled in your Master section. If reverb is still not available, check the routing from input to bus to reverb I/O to source to destination.

**PC90 does not record during the Create File process in Cubase VST**

Unlike software-based plug-ins, the PC90 is hardware-based and does not get recorded during VST's Create File process. Much like effects from any outboard gear, you must record the effects as digital audio tracks within the Song.

Work around:

1. Return reverbs to aux. ret L&R
2. Use aux. ret L&R as input sources.
3. Record these sources onto new tracks, then use the Create File function.

**Sync****Failure to sync when using ASIO Multimedia driver in Cubase VST**

On slower machines, Cubase may have difficulty syncing when the sample reference is set to 'Sample Position'. Set the sample reference to 'DMA Block - Output'. Also, you must detect the buffer size for both 44.1kHz and 48kHz sample rates for sync to work properly. Once you have completed these steps, you may hit the 'Store' button and save this as a profile within Cubase.

**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. Turn 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**

## ***Troubleshooting for the Macintosh***

### ***Installation Troubleshooting***

#### ***Software***

The Lexicon Studio Installer will install all necessary components into their default locations. If you want to install Lexicon Studio on a different hard drive, or to a particular location of your hard drive, you must browse to this in the installer.

Note also that you must install Lexicon Studio software for *each* application. For example, if you have installed Cubase VST, VisionDSP, and Logic Audio, you will need to install Lexicon Studio for each application.

#### ***Hardware Recognition***

There are two status LEDs mounted on the top of the card, right next to the PC-90 connector. When a supported audio application is booted and the drivers are loaded, one LED will light and the other will blink repeatedly.

### ***System Troubleshooting***

#### ***Power Up Problems***

If the computer does not power up after installing a Core-32, the Core-32 or another expansion card may be partially dislodged from the expansion slot in the PCI bus. Remove the Core-32 card and verify that the system boots without it installed. Verify that all other expansion cards are seated securely in the slots. Remove all peripheral devices from the Core-32 card, including the PC-90 card and cables. Reinstall the stripped Core-32 card and try re-booting to try to isolate the PC-90 and ca/or cabling as the source of the problem.

### ***Disk with Question Mark during Boot Cycle***

This is a Macintosh error that is telling you that the Macintosh cannot find the boot drive. Power the machine down and make sure that you have not loosened any cables to the hard drives.

## ***Optimizing MacOS for Audio Applications***

Following are some suggestions that may improve your computer's performance as a dedicated digital audio workstation. Some of these suggestions disable some MacOS optimization features and may slow the overall performance of your system when using non-audio applications.

### ***Add Memory***

Adding RAM to your computer will improve the performance of your audio applications.

### ***Streamline your Extensions***

With fewer Extensions and Control Panels active within your system, you will not only have more memory available for your operating system, but less chance of software conflicts. Use the Extensions Manager Control Panel to select only the Extensions and Control Panels that are absolutely necessary for you do do your audio work.

### ***Disable Virtual Memory***

Most audio applications require disabling virtual memory.

### ***Disable Speed Doubler***

Lexicon Studio is not compatible with Speed Doubler.

### ***Scan and Defragment your Disk Regularly***

Defragment your drives on a regular basis. Run Apple Disk Tools or Norton Utilities regularly to repair directory damage to your disk. These software tools can detect and fix problems before they become fatal. Disk Tools is on your original MacOS CD-ROM. Norton Utilities is available from your local Macintosh dealer.

Before defragmenting your drive, make sure that the Mac OS version you are running supports your application. Crashing while defragmenting your disk can lead to irreparable damage.

### ***Verify the Performance of your Disk Drive***

Make sure that your disk drive is fast enough to handle audio (a seek time of 10ms or less is good performance). Also, some audio applications may not support Apple HFS+ format. Please consult the documentation for your audio software if you plan to upgrade your machine to HFS+.

### ***Use a Dedicated Drive for Audio***

Having a separate drive for audio will help prevent bandwidth problems with disk transfer. It is best to have two hard drives, one for your operating system, and one dedicated to audio recording.

### ***Disable Screen Savers***

Disabling any program that runs in the background, such as screen savers will help system performance.

### ***Disable Virus Protection Software***

This is most important while installing software, but virus-scan software typically runs in the background of the computer, taking up processing time of the CPU. This should be disabled when running audio software, but used periodically to check for viruses.

### ***Dedicate your Computer to Audio Work***

Computers are complex machines. The software that runs them is equally complex. Dedicating a system to certain applications will minimize conflicts and other system problems. Try to dedicate your machine solely to audio purposes, and do your word processing, web browsing and financing on other machines.

**TIP: Back up your system on a regular basis.**

## *Common Problems and Solutions*

### ***Installation and Hardware Recognition***

#### **Core-32 hardware not present dialog box**

If you encounter this dialog box while launching your audio software, your Core-32 card may not be seated fully in its PCI slot. Also, check your Lexicon Studio cabling.

#### **ASIO Device could not be created error**

Make sure your Lexicon Studio software is installed properly. The Core-32 extension must be present in the Extensions folder. The ASIO Core-32 driver must be present in the ASIO Drivers folder.

#### **Problems operating the Lexicon Core-32 Diagnostics**

The Core-32 can be an ASIO device for only one audio application at a time. You cannot run Cubase VST (or any ASIO application) simultaneously with the Core-32 Diagnostics without encountering errors.

### ***PC-90 Problems***

#### **PC-90 is not selectable in Cubase VST as a plug-in.**

If the PC-90 machine 1 and PC-90 machine 2 are not selectable as an effect, the PC-90 plug-ins have probably not been installed into the proper 'VstPlugins' folder. Default installation folders for Cubase are shown below. Actual program names will differ depending on your version.

**For Cubase VST:** Macintosh HD:Cubase VST:vstplugins

**For Score:** Macintosh HD:Cubase Score VST:vstplugins

**For Score Demo:** Macintosh HD:Cubase Score VST Demo:vstplugins

#### **PC-90 reverb is assignable from the VST Effects window but can not be heard:**

Check bus pop-up on the PC-90 Rack Xpander. If this is set to manual, change it to an enabled Master output, such as PC90 1. PC90 1 or PC90 2. If reverb is still not available, check the routing from input to bus to reverb I/O to source to destination.



### **PC90 does not record during the Create File process in Cubase VST**

Unlike software-based plug-ins, the PC90 is hardware-based and does not get recorded during VST's Create File process. Much like effects from any outboard gear, you must record the effects as digital audio tracks within the Song.

Work around:

1. Return reverbs to aux. ret L&R
2. Use aux. ret L&R as input sources.
3. Record these sources onto new tracks, then use the Create File function.

### ***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.
- Add memory. (Cubase VST recommends at least 64MB.)
- Reduce enabled audio streams in Advanced Settings Panel.

### ***Pops and clicks***

- Minimize any unnecessary windows that are constantly redrawing
- 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.
- Adjust disk block buffer settings in your particular audio application.

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# Appendix D

## Specifications

### *LDI-10T Interface*

#### Analog Audio Interface

##### Audio Inputs (8)

<b>Connectors:</b>	Balanced TRS Phone Jacks
<b>Impedance:</b>	50k $\Omega$ balanced; 25k $\Omega$ unbalanced
<b>Levels:</b>	Balanced +4 mode, +20dBu full scale; Unbalanced -10 mode, +6dBV full scale

##### A/D Performance

<b>Frequency Response:</b>	20Hz to 20kHz $\pm$ 0.5dB, ref 1kHz
<b>Crosstalk:</b>	-96dB @ 1kHz
<b>THD:</b>	<0.005%, 20Hz to 20kHz
<b>Dynamic Range:</b>	102dB typical, A-weighted
<b>Resolution:</b>	24 Bits Delta-Sigma Conversion

##### Audio Outputs (8)

<b>Connectors:</b>	Balanced TRS Phone Jacks
<b>Impedance:</b>	600 $\Omega$ each side, balanced
<b>Levels:</b>	+26dBu full scale, balanced

##### D/A Performance

<b>Frequency Response:</b>	20Hz to 20kHz $\pm$ 0.5dB, ref 1kHz
<b>Crosstalk:</b>	-96dB @ 1kHz
<b>THD:</b>	<0.005%, 20Hz to 20kHz
<b>Dynamic Range:</b>	105dB typical, A-weighted
<b>Resolution:</b>	24 Bits Delta-Sigma Conversion

**Digital Audio Interface**

**Input Connector:** Coaxial RCA type (1)  
**Output Connector:** Coaxial RCA type (1)  
**Format:** S/PDIF (IEC-958, CP-340) consumer audio interface

**System Sample Rates**

44.1kHz, 48kHz

**Control Interfaces**

**Time Code Input:** Balanced TRS Phone Jack, EIA-422; Formats supported: 24, 25, 30 drop/non-drop

**Power Requirements**

100-240 VAC, 50-60Hz, 15W, 3-pin IEC connector

**RFI/ESD**

Conforms to FCC class B and European EMC Directive 89/336/ECC in accordance with EN55103 and EN 55103-2 for category E2 environments

**Dimensions**

19.0"x1.75"x6.6" WHD (483x45x165mm) standard 19-inch 1U rack mount

**Weight**

4.5 lbs

**Environment**

**Operating Temp:** 32° to 104°F (0° to 40°C)  
**Storage Temp:** -22° to 167°F (-30° to 70°C)  
**Humidity:** Max 95% non-condensing

## **LDI-12T Interface**

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

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

### **A/D Performance**

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

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

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

### **D/A Performance**

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

### **Digital Audio Interface**

<b>Input Connectors:</b>	Coaxial RCA type (1) Optical TOSLINK (1), assignable as S/PDIF or TOSLINK
<b>Output Connectors:</b>	Coaxial RCA type (1) Optical TOSLINK (1), assignable as S/PDIF or TOSLINK
<b>Format:</b>	S/PDIF (IEC-958, CP-340) consumer audio interface

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<b>System Sample Rates</b>	44.1kHz, 48kHz
<b>Control Interface</b>	
<b>Timecode Input:</b>	Balanced, female XLR-3 type connector, EIA-422, Formats supported @±15%: SMPTE non-drop/drop frame
<b>External Word Clock Input:</b>	BNC connector, TTL compatible, unterminated
<b>Alesis ADAT Sync:</b>	9-pin D-sub female connectors, input/output
<b>Machine Control:</b>	9-pin D-sub female connector
<b>Host Computer:</b>	68-pin high-density connector
<b>General Dimensions</b>	19.0" x 1.75" x 4.0" (WHD) (438 x 45 x 102mm), 19" rack mount standard, 1U high
<b>Power Requirements</b>	9 VAC, 1A wall transformer provided
<b>RFI/ESD</b>	Conforms to FCC Class B; EN55022 Class B (CE), IEC 801-2, IEC 801-3
<b>Environment</b>	
<b>Operating Temperature:</b>	32° to 104°F (0° to 40°C)
<b>Storage Temperature:</b>	-22° to 167°F (-30° to 70°C)
<b>Humidity:</b>	Max 95% non-condensing

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## **Core-32**

<b>External Interfaces:</b>	60-pin proprietary multi-pin connector and/or 50-pin proprietary multi-pin connector
<b>Internal Interfaces:</b>	PCI Interface TBUS daughterboard interface (not on all models) PC-90 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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