Owner's Manual

480L

Digital Effects System

lexicon

Lexicon Part #070 - 06404 Rev. 2.3 11/87
This manual accompanies Software Version 2.00

1

Installing the 480L

Installing the 480L involves six basic steps:

1. Familiarizing yourself with basic controls and functions.

2. Mounting.

3. Connecting the LARC(s) to the 480L mainframe.

4. Applying AC mains power.

5. Connecting the 480L to your console.

6. Setting audio levels.

Each of these subjects is discussed in this chapter.

About the Rear Panel

Main Inputs (L & R)
The left and right Inputs accept 3-pin male XLR connectors. They are electronically balanced and (optionally) transformer isolated. Either pin 2 or pin 3 can be used as high, but to maintain polarity when transferring data to the digital domain, pin 2 should be high. Pin 1 and either pin 2 or pin 3 of each input *must* be grounded for unbalanced operation. Input impedance is 30 kilohms in parallel with 100 pF. Inputs accept input levels from +6 to +24 dBm.

Main Outputs (L & R)
The left and right Main Outputs accept 3-pin female XLR connectors. They are electronically balanced and (optionally) transformer isolated. Either pin 2 or pin 3 can be used as high, but to maintain polarity when transferring data to the digital domain, pin 2 should be high. Pin 1 and either pin 2 or pin 3 of each output *must* be grounded for unbalanced operation. Output impedance is 33 ohms, and levels up to +24 dBm are possible.

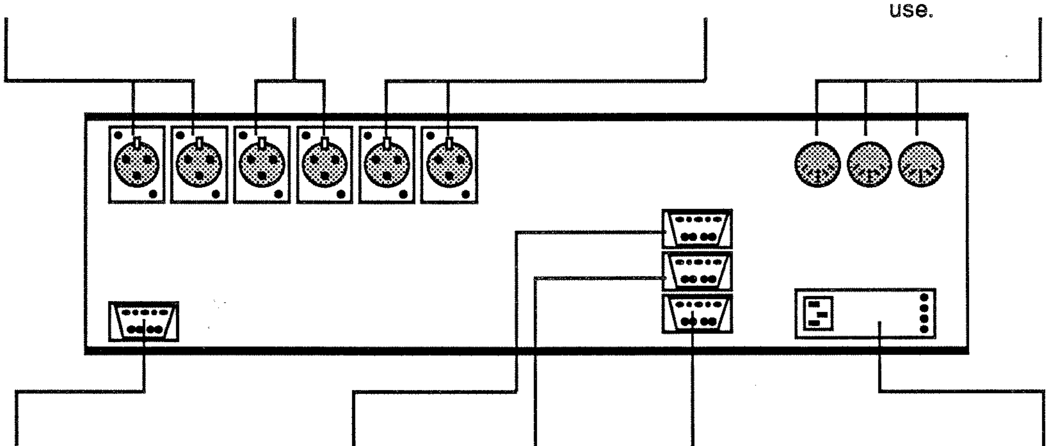
Aux Ouputs (L & R)
The left and right aux outputs are identical to the Main Outputs, except that they are used as secondary outputs when split or cascade modes are selected.

Important. Reversing polarity on either input or output connectors can produce audible phase inversion effects. Improper phasing in the stereo path can create a weak or thin mix. Ensure that inputs and outputs are wired consistently.

MIDI Connectors
MIDI IN receives MIDI information from other MIDI-equipped devices.

MIDI Thru retransmits MIDI information received at the MIDI In connector, without any change.

MIDI Out is currently not in use.



Lexicon Digital Audio I/O Connector
This DE9 connector provides PCM 1610-compatible digital I/O. It has 18-bit word length capability and can be slaved to a 48 kHz, 44.1 kHz or 44.056 kHz external word clock.

LARC 1 Connector
This DE9 connector connects the mainframe to the Lexicon Alphanumeric Remote Control (LARC) via a flexible 50-ft cable (supplied)

Automation Connector
The Automation Connector is provided for future computer control and automation features.

Important: Never connect a LARC to this connector.

LARC 2 (Thru) Connector
This DE9 connector allows connection of a second LARC to the mainframe. It also allows the 480L to be connected to a 224XL, with both units under control of a single LARC. A 10 ft cable is provided for this application.

Power Connector and Fuse Holder Cartridge
The Power Connector is a standard 3-pin IEC power connector. The Fuse Holder Cartridge contains the AC mains fuse(s). The voltage changeover card is also contained in this compartment. Read Appendix A for voltage changeover information.

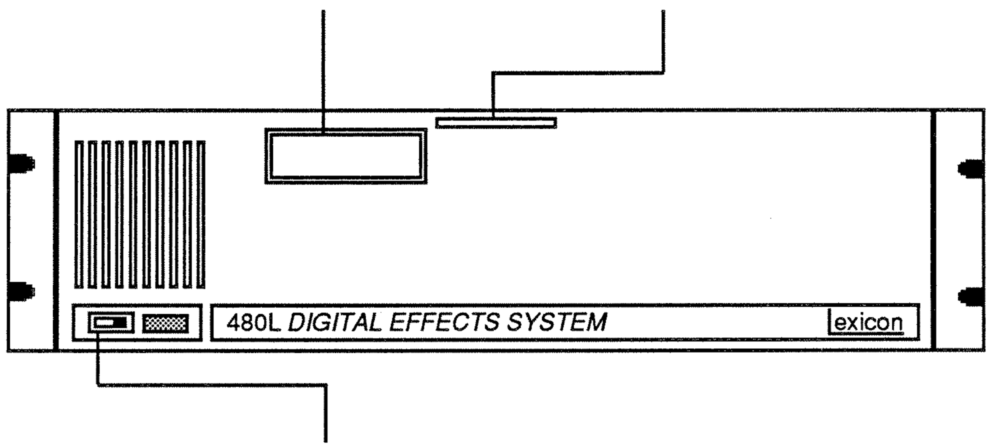
Figure 1.1. 480L Rear Panel.

About the Front Panel

Nonvolatile Memory Cartridge Slot
The 480L is shipped with one Nonvolatile Memory Cartridge, providing five banks of portable register storage. A write-protect switch prevents accidental erasure of contents.

Note: Cartridges may be shipped with the write-protect switch in the *ON* position.

Front Panel Latch
The front panel is hinged at the bottom; pull on the handle to open. Keep the front panel closed during normal operation to maintain dust filtration.



Power Switch and Indicator
The Power Switch turns the 480L on and off; the indicator lights when the unit is on. A lithium battery retains the data memory when power is off or disconnected.

Figure 1.2. 480L Front Panel.

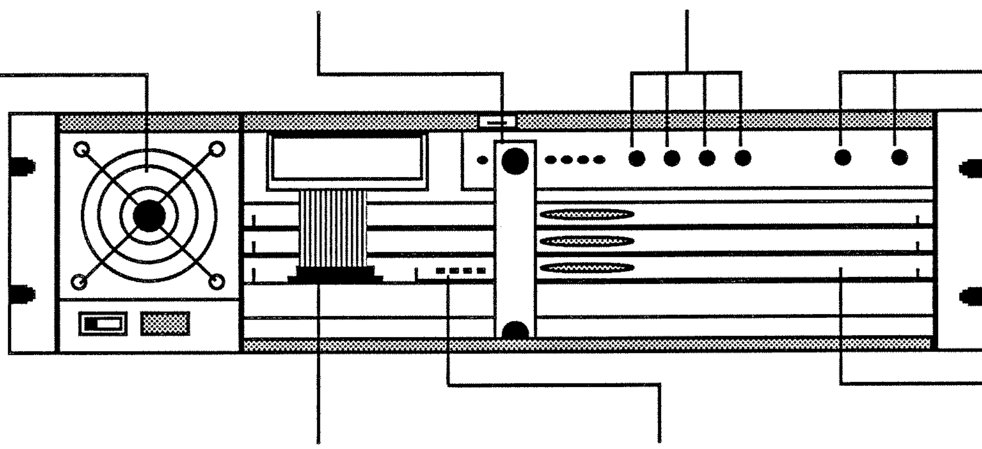
Behind the Front Panel

Cooling Fan
The cooling fan provides filtered forced air (the front panel vent is an air intake). The filter is removable and should be cleaned periodically with mild detergent and warm water.

Card Retainer
The card retainer ensures that the HSP and host processor cards remain firmly seated.

Output Level Controls
The output levels for the Main and Aux analog outputs may be adjusted independently over a range from +6 to +24 dBm (into 600 ohms) with these controls.

Input Level Controls
The input sensitivity for the left and right analog inputs may be adjusted independently to match inputs over a range of +8 to +28 dBm with these controls.



Nonvolatile Cartridge Cable
This ribbon cable connects the cartridge slot to the host processor card via a locking connector on the host processor card.

> **Caution:** Use of excessive force when inserting cards into the 480L can result in serious damage. Always make sure that the connectors are lined up properly before applying seating force.

Diagnostic Indicators
The four diagnostic indicators on the host processor card flash briefly upon powerup.

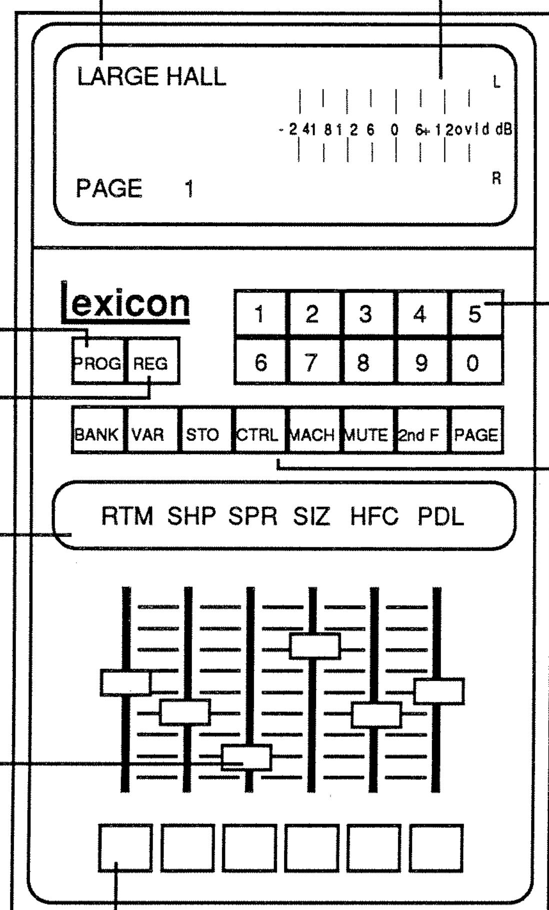
Removable Modules
The 480L is completely modular. Every subassembly in the mainframe can be unplugged and removed for service or exchange. The standard complement for a 480L is two HSP cards and a host processor card. The cards can be plugged into any slot in the mainframe, but for best noise performance, the HSP cards should be installed in the two top slots, and the host processor card directly beneath them. The empty bottom slot is provided for the optional SME card.

Figure 1.3. Behind the Front Panel.

About the LARC

Main Display
Shows names and values for all selections.

Headroom Indicator
+12 dBm indicates analog or digital clipping. Proper input level is with +12 dB and ovld LEDs *unlit.*



Program-Select Key
Press repeatedly to scroll through programs in a bank.

Register-Select Key
Press repeatedly to scroll through registers in a bank.

Slider Display Line
Shows abbreviated names of the parameters currently under the control of the sliders. Full name appears in the Main Display.

Control Sliders
Adjust parameter values.

Numeric Keypad
Press one of these keys to load a program selected with PROG or REG. Also used to select pages.

Function Keys Bank
scrolls through groups of programs or registers.

VAR displays the running program or register name and the machine that the LARC is controlling.

STO stores edited parameter settings in registers (when REG is pressed at the same time.

CTRL toggles in and out of the control mode.

MACH toggles between machines when Split or Cascade modes are selected, and also between the 480L and 224XL when a 224XL is connected to the 480L.

MUTE removes processed audio from outputs (only while it is pressed).

2nd F is inactive.

PAGE steps through pages of parameters.

Slider Display Keys
Show parameter name and value in main display. Press twice to engage vernier (fine) adjustment mode.

Also used to trigger record and play events in Doppler and Sampler programs.

Figure 1.4. The LARC.

How to Mount the 480L

The 480L measures 19" wide x 5.25" high x 14.5" deep (483 x 133 x 368 mm). It can rest on any flat surface, or it can be mounted in a standard 19-in. (483 mm) relay rack. Do not install the 480L directly above equipment which produces significant amounts of heat (such as power amplifiers); maximum ambient operating temperature is 40°C (104°F). Do not obstruct the ventilation exhaust ports on the right side panel, or the air intake on the front panel.

If the 480L is mounted in a rack or road case, we recommend that you provide support for the rear of the chassis during transport to prevent possible damage from severe mechanical shock.

About the 480L's Power Requirements

The 480L is equipped with a three-pin IEC connector and detachable power cord, providing chassis grounding to the ac mains line. It can be operated at either 100/120 Vac or 220/240 Vac, depending on the fuses installed and the setting of the voltage changeover board.

Note: Voltage changeover is described in Appendix A.

The nominal operating voltage set at the Lexicon factory is indicated by a small protruding pin on the power connector/fuse holder (see Figure 1.5). Check this voltage setting before applying power to the unit! Power consumption is 70W typical, 180W maximum.

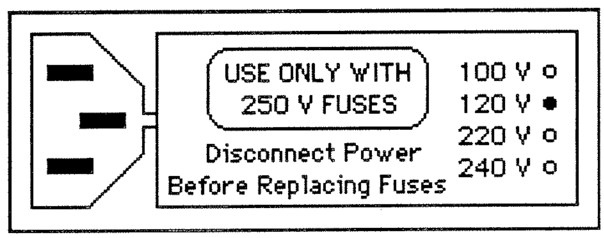


Figure 1.5. Voltage Selector Set for 120 V.

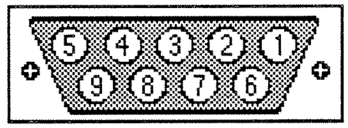
How to Interface the LARC

The LARC 1 connector interfaces the mainframe to the Lexicon Alphanumeric Remote Control (LARC) via a flexible 50-ft cable (supplied). If your system is equipped with a single LARC, this is the connector you should use.

The LARC 2 connector has two functions. It allows connection of a second LARC to the mainframe for applications where use of two LARCs is required (see Figure 1.8). It can also be used to connect the 480L to a 224XL (using the supplied 10' cable) with both units under the control of a single LARC (see Figure 1.9). The pin assignments for the LARC connectors are shown in Figure 1.7.

> ***Important:*** Never connect a LARC to the automation connector. Doing so may blow the internal automation connector fuse.

Pin	Assignment
1	Chassis/shield ground
2	Receive data
3	Transmit data
4	Transmit common
5	Isolated power supply (+)
6	Receive common
7	Receive data
8	Transmit data
9	Isolated power supply gnd



LARC Connectors

Figure 1.7. Wiring diagram for the LARC mainframe connectors.

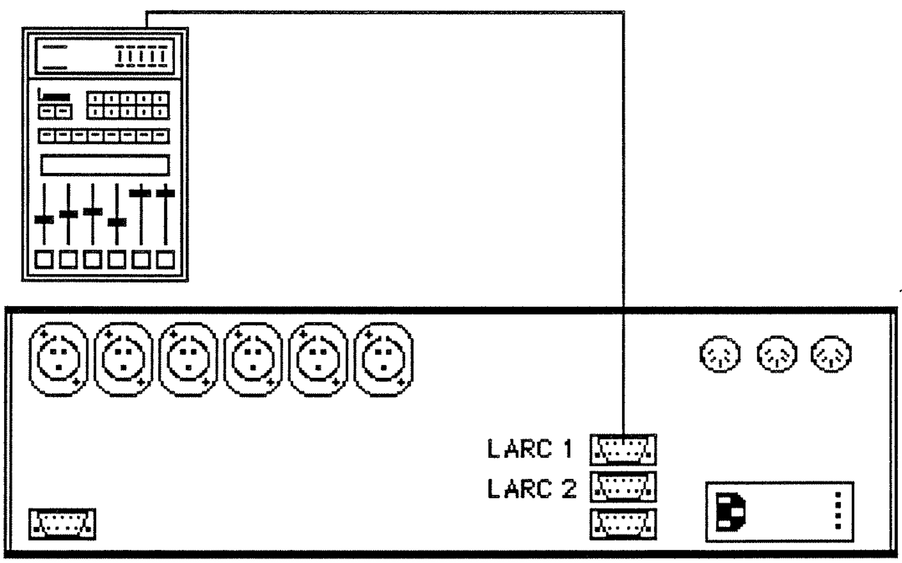


Figure 1.6. Connections for 480L with one LARC.

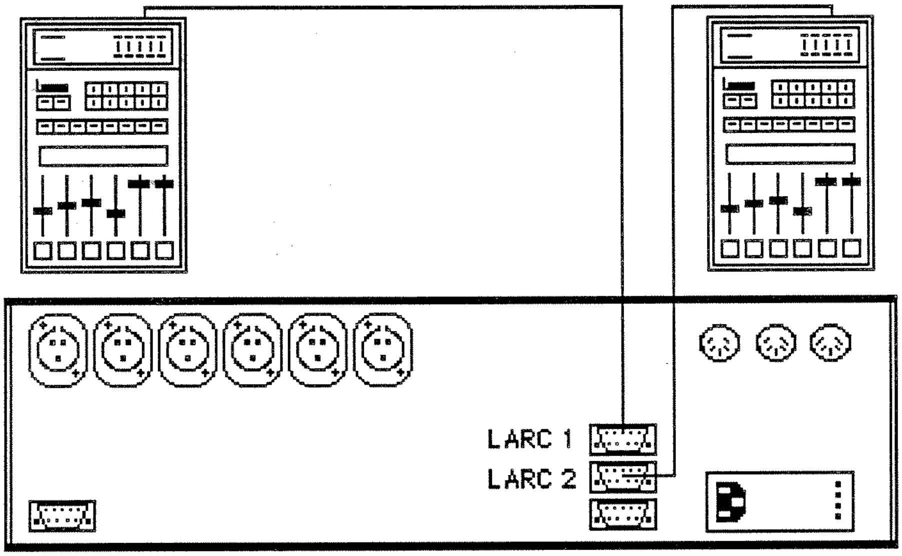


Figure 1.8. Connections for 480L with two LARCs.

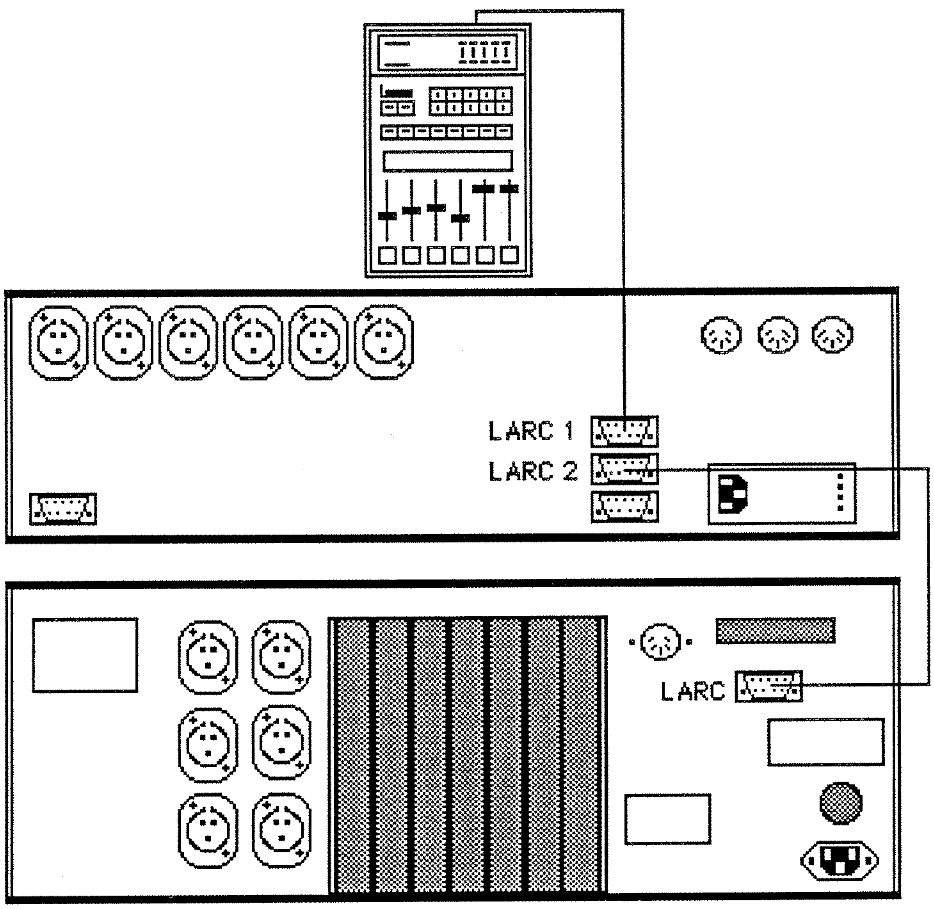


Figure 1.9. Connections for 480L and 224XL with one LARC.

Audio Connections

See page 1-2 for details about audio interfacing (pinouts, impedance, etc.)

The 480L is designed to take advantage of the flexibility of a mixing console. Figure 1.10 shows a typical configuration. For maximum utility, use independent sends that can be assigned as either prefader or postfader. You can use the console's effects returns if they are pannable or assignable, but for greater creative control, you may wish to connect the 480L outputs to regular input channels.

We recommend experimentation to arrive at the best configuration for your own system. Actual connections should always be checked carefully for proper impedance, polarity, and levels.

When using mono signal sources, either connect the left and right inputs in parallel, or use the mono split configuration (described in Chapter 2).

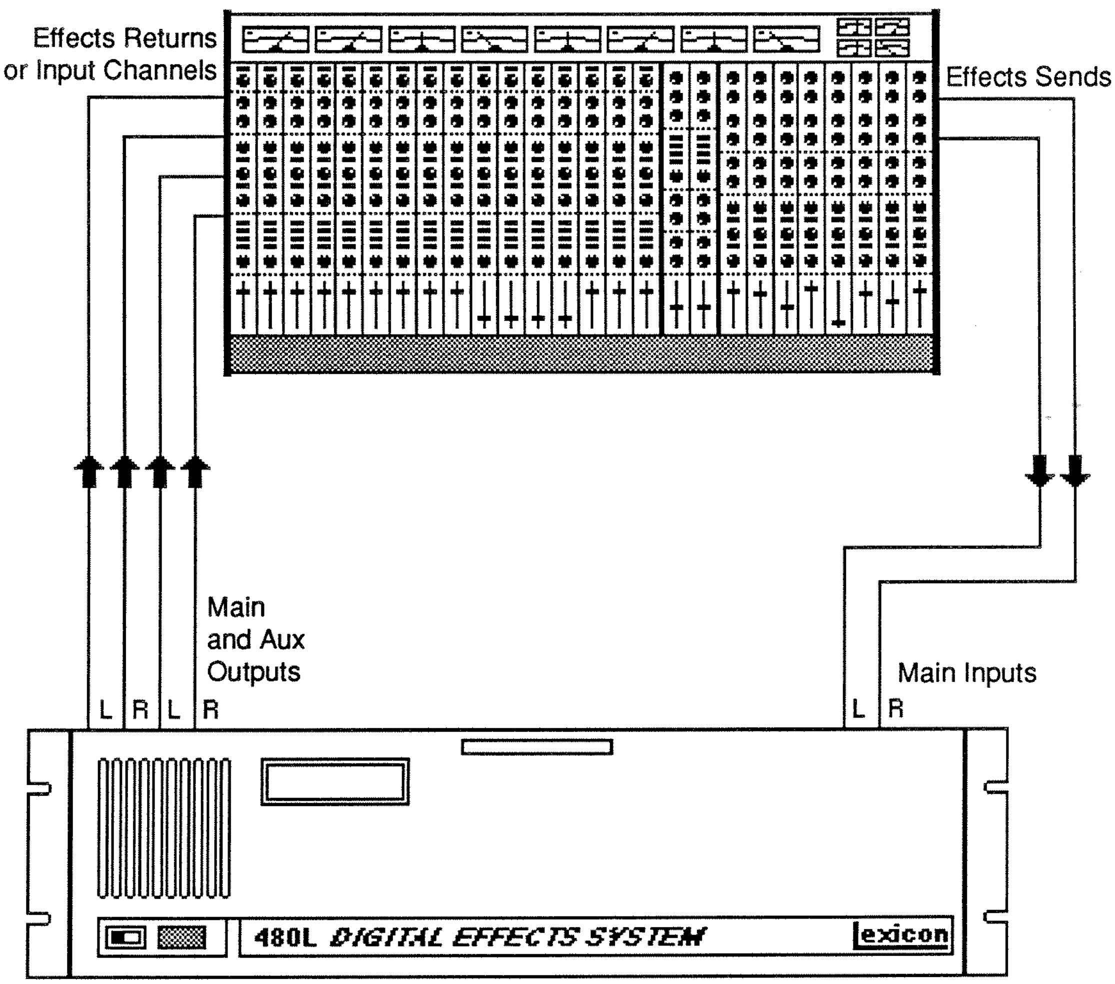


Figure 1.10. Typical Audio Connections.

How to Make Level Adjustments

To obtain the best possible performance from the 480L, input and output levels should be set with care. The factory settings give unity gain (0 dB) with a maximum output level of +18 dBm.

Set the Input Levels

1. Put the 480L in the SINGLE configuration.

2. Load THE IN_OUT program located in Bank 9. This provides 0 dB gain through the 480L from the inputs through the outputs.

3. Feed a musical source or a 1-kHz test tone at the maximum peak level that you use in your system into the left and right audio inputs of the 480L.

4. While watching the Headroom display on the LARC, adjust the INPUT LEVEL controls (found behind the front panel) so that the peak input level comes just short of lighting the +12 dB LEDs (for both left and right inputs).

> **Important:** Do not overdrive the 480L. Its clipping characteristic, like that of other digital audio equipment, is very abrupt.

Set the Output Levels

Set the four OUTPUT LEVEL controls (found behind the front panel) to provide output levels appropriate for the console's inputs.

We recommend running the 480L at +16 to +24 dB. If you must use lower levels (+6 or lower) some loss of dynamic range will be noticed. To rectify this problem, set the 480L output to a higher level (such as +18) and use a resistive pad or a step-down transformer (3:1 or 2:1) to reduce the level to match your equipment.

Floating the Analog Ground

In some applications it may be desirable to float the 480L's analog circuitry from the chassis ground. This can be accomplished by simply removing the blue jumper block located on the top side of the main circuit board near the two main input connectors. Store the jumper block on one of the posts in case you ever need to reinstall it.

When the jumper block is removed, the analog signal grounds are floated from the chassis at DC, but are tied to the chassis for protection through a 1000 pF bypass capacitor and a 180 V metal oxide varistor.

How to Use Digital I/O

In addition to its analog inputs and outputs, the 480L accepts digital audio I/O. Refer to Chapter 6, *The 480L and Digital I/O* for more details.

2

Introducing the Controls

This chapter describes the controls and functions used to make the 480L do what you want it to do.

There are several terms to understand before we go any further.

• ***Algorithm***. The 480L contains several *algorithms*. An algorithm is a set of instructions that tells the 480L's microprocessors how to process the input signal. One algorithm produces effects, another reverberation, another sampling, etc. Algorithms are stored inside the 480L on ROM chips.

• ***Parameter***. Each algorithm has a set of *parameters* (controls) that uniquely characterize it. The settings of the parameters can be changed to create radically different sounds from a single algorithm.

• ***Program.*** A group of specific parameter settings permanently stored in the 480L is referred to as a *program*.

• ***Register***. New parameter settings you create by editing programs can be stored in *registers* for later recall. Registers can be stored within the 480L, or in a removable nonvolatile memory cartridge.

• ***Bank***. A *bank* is a collection of several similar programs or registers. For example, the Rooms bank contains reverberation programs that simulate real spaces, while the Effects bank contains programs which produce a variety of audio effects.

• ***Pages***. Because the programs have more than the six parameters which the LARC can display at one time, parameters are grouped into several *pages*. You move between pages by pressing PAGE.

• ***Control Mode***.. The *control mode* contains several pages of utility parameters and functions which are not directly related to a single algorithm, such as sampling rate, register transporter, program name function, etc. The control mode is entered and exited by pressing CTRL.

• ***Configuration.*** The 480L can run two programs simultaneously. How the two programs relate to each other is called the *configuration*. They can be used with independent inputs and outputs (mono split configuration), they can share the same stereo input signal (stereo split configuration), or the outputs of one program can be fed into the input of another (cascade configuration) The configuration is changed from the control mode. *The 480L is shipped in the stereo split configuration.*

• ***Machine***. *Machine* refers to two separate but related concepts. When two programs are operating simultaneously, they are referred to as Machine A and Machine B. When a 480L and a 224XL are under the control of a single LARC, these are referred to as Machine 1A and 1B, and Machine 2. You can toggle between programs and units by pressing the MACH key.

How to Set the 480L Configuration

The 480L can run two of its programs simultaneously. It can, for example, run a sampling program in one half of the machine while a reverb program is running in the other half. The two programs can be used entirely independently (with separate inputs and outputs) or they can be connected together internally in any of several flexible configurations. You select which program is currently under control of the LARC by pressing MACH.

Choosing a Configuration

Go into the control mode by pressing CTRL. Configurations are selected with Slider 2 on page 1 of the control mode. There are four internal configurations available:

- Cascade
- Stereo Split
- Mono Split
- Single

Because the Configuration slider redefines the internal architecture of the 480L, the display takes a bit longer to update after you move the slider than other parameters. Let's take a closer look at the four configurations.

The Cascade Configuration

The Cascade configuration feeds the output of one program (Machine A) directly into the input of the second program (Machine B). See Figure 2.1. This allows you to process a stereo signal with two entirely different effects--without ever leaving the digital domain. The Main outputs are connected to Machine B, and contain the processed signal from both Machine A and Machine B. The Aux outputs contain only the signal from Machine A.

In the Cascade configuration, the MIX control found in the reverb and effects programs becomes very important, because it is the only method you have of controlling the mix between the two programs.

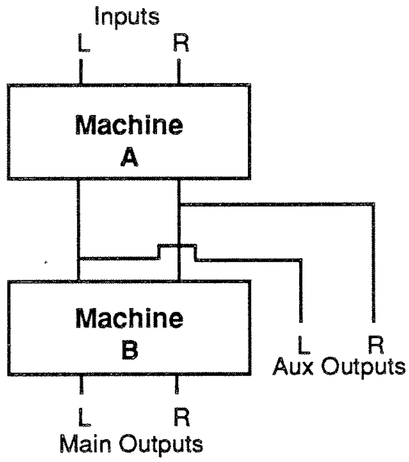


Figure 2.1. 480L in Cascade Configuration.

The Mono Split Configuration

The Mono Split configuration uses the 480L as two independent signal processors. See Figure 2.2. Each program has an independent mono input and an independent stereo output. The Left input always goes to the first program (Machine A), and the Right input always goes to the second program (Machine B). The Main Outputs produce stereo output from Machine A, and the Aux Outputs produce stereo output from Machine B.

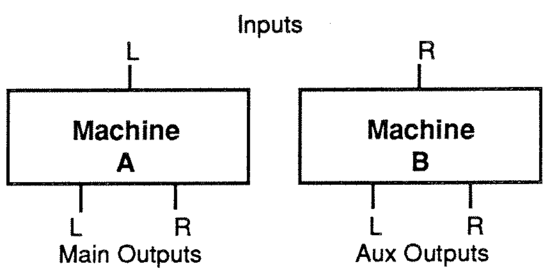


Figure 2.2. 480L in Mono Split Configuration.

The Single Configuration

A few programs (like Stereo Sampler) require all of the 480L's processing power, and cannot be run at the same time as other programs. The Single configuration is provided for these programs. In the Single configuration, the outputs of the program are available at both the Main and Aux Outputs.

The Stereo Split Configuration

The Stereo Split configuration also uses the 480L as two independent signal processors. See Figure 2.3. It differs from the Mono Split in that both inputs are sent to both programs; in other words, Machine A and Machine B receive the same stereo input signal. The Main outputs are used for Machine A, and the Aux outputs are used for Machine B.

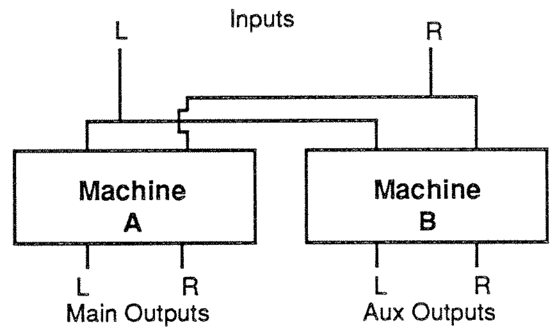


Figure 2.3. 480L in Stereo Split Configuration.

Checking Your System's Status

Move slider one on page one of the control program for a quick display of the following information:

- Configuration
- Sampling rate selected
- Clock source (internal or external)
- Input source (analog or digital)
- External Word Clock present/not present
- Register protection status
- Cartridge Status (formatted or unformatted, present or not present)

Using Two LARCs to Control a Single 480L

If you frequently use your 480L in the Split or Cascade modes, you may wish to consider purchasing a second LARC. Having two LARCs allows you to control two programs simultaneously, without switching back and forth with MACH. Two LARCs are also useful if the 480L is to be shared between two different rooms.

In addition to controlling two programs at once, the second LARC can be used to display two pages of parameters for a single program.

The second LARC should be connected to the LARC 2 (Thru) connector on the rear panel of the 480L. Refer to Chapter 1 for details.

Controlling a 224XL from a 480L and LARC

In facilities equipped with both a 480L and a 224XL, it may be useful to control both systems from a single LARC. To do this, connect the LARC 2 (Thru) connector to the 224XL LARC connector with the supplied cable, as shown in Chapter 1. Use the MACH key on the LARC to switch the LARC between the 224XL and the 480L. If you are running two programs on the 480L at the same time, there will be three choices to step through each time you press MACH.

Connecting a 480L and a 224XL together simply allows you to control the 224XL as you always have--none of the 480L's new capabilities are added to it. For example, the 224XL cannot access the register mover or other 480L control mode functions. Also, the 224XL cannot be accessed by the LARC while the 480L is in the control mode. If you press MACH while in the control mode, the 224XL will not appear in the display. As soon as you exit the control mode, the 224XL can be selected.

How to Load Programs

The 480L is shipped with approximately 43 programs installed (the exact number varies, depending on the version of the software supplied with your unit). This section describes selecting and loading programs. Detailed information about *using* the programs is found in later chapters.

The 480L's programs are stored in banks. A bank is a collection of several similar programs.

Select a Bank

Before selecting a program, you must first select the bank that the program is stored in. There are two ways to select banks:

1. Press BANK repeatedly. The 480L displays a bank name each time you press BANK. *Example*: BANK, BANK, BANK scrolls through three banks. This is a good method to use when you are just "sight-seeing" to find out what programs are available.

2. Press BANK, and then one of the numeric keys. *Example:* press BANK, 2 to select the second bank of programs in the 480L. This is a good method to use once you are familiar with the 480L, and know exactly which bank you need.

While you are selecting a bank, the bank number on the LARC display flashes, and the number of the current bank is displayed:

BK2

Select a Program

After selecting a bank, you can scroll through the programs in that bank without loading them, by pressing PROGRAM repeatedly. While you are selecting a program, the program number on the LARC display flashes, and the number of the currently-selected program is displayed.

Load a Program

1. After finding a program you want to load, note its program number (the flashing number on the LARC display). For example:

PGM4

2. Load program 4 by pressing numeric-select key 4. The program that was running halts and the new program begins processing audio after a brief pause.

3. When you become familiar with the system of banks and programs, you'll find that you can move from program to program very quickly. For example, to go from bank 2, program 3 to bank 1, program 6, you just press BANK, 1, PROG, 6.

After you load a program, the first page of variable parameters appears.

Cue Up the Next Program

You can cue up the next program you will need by selecting it with the BANK and PROG or REG keys. Then, when you are ready for the new program, just press PROG or REG and the numeric-select key.

If the 480L is set up with a split or cascade configuration (see page 2-2), use the following procedure to load a program into machine B:

1. Press MACH to switch control to Machine B.

2. Load the program or register you want into Machine B. The same program or register can be loaded into both machines, or they can be entirely unrelated.

Toggling Between Machines

Once two programs have been loaded, they both process audio continuously and simultaneously. However, the LARC can only actively *control* one program at a time. Press MACH to select the machine you wish to control. Each time you press MACH, the LARC switches control to the other machine, and briefly indicates the machine it is controlling.

How to Edit Parameters

The sounds of the programs supplied with the 480L cover an astounding range of possibilities, but sooner or later you will want to alter the sounds of the programs to more perfectly fit your requirements. Each program in the 480L contains a set of parameters that can be edited to create a sound uniquely your own.

Just Move the LARC's Sliders

After loading a program, you can edit its parameters by moving the LARC's sliders. Most parameters can be edited in real time to alter an effect. However, a few parameters (like SIZE) have such a radical effect on the 480L's algorithms that the effects signal is muted briefly when they are edited.

To indicate the parameter that a slider controls, an abbreviated code appears in the display window above each active slider. You can display a more descriptive title and the current value for each parameter by pressing the keys directly below each slider. Moving a slider also displays this information.

In many cases, pressing a display key twice will engage a vernier (fine) adjustment mode that allows very precise adjustment. The display blinks to indicate that the vernier mode is active.

Change Pages to Access More Parameters

Because the programs in the 480L have more than the six parameters which the LARC can display at one time, parameters are grouped in several pages. Each page contains up to six parameters.

You can use either of two methods to move between pages:

1. Press PAGE repeatedly to step through the pages sequentially.

2. Press PAGE and then a numeric-select key to go directly to the page you want.

> **Important!** When a new program is loaded or another page is selected, each slider is deactivated (i.e., the display does not change) until the slider is moved *through* its preset value.

When changes have been made on a page, and you move to a new page, the previous edits remain intact. However, when a new program is loaded, the edits you made disappear forever (unless you stored the edits in a register).

How to Use Registers

The ability to edit parameters would be of little value if there were no way for the 480L to store the edits. Not to worry--the 480L has 100 registers available to store edited versions of the preset programs. Registers are organized into banks, selected, and loaded exactly like the preset programs. You can also edit parameters in a register, and store the results in the same register or another register.

There are five banks of ten registers in internal memory. Another five banks of ten registers can be stored in a nonvolatile memory cartridge. One cartridge is supplied with the unit, and additional cartridges may be purchased.

> **Important!** Cartridges are equipped with a write protect switch. When the switch is ON, it prevents the 480L from writing to the cartridge, regardless of the register protection selected in the 480L. Cartridges may be shipped with the write protect switch in the ON position. Figure 2.4 shows the location of this switch.

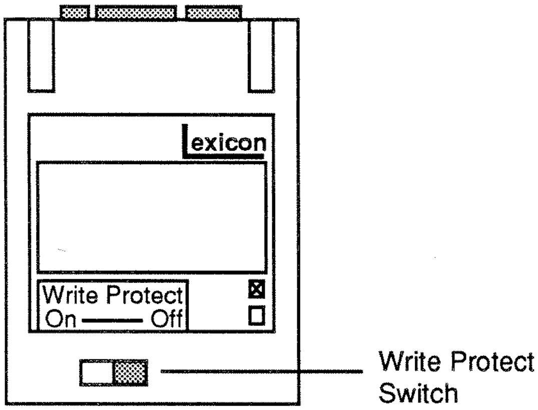


Figure 2.1. Location of Write Protect Switch.

Storing a Modified Program in a Register

After you have made the changes you want to a program's parameters, you can store the changed version in a register:

1. Press REG once to enter the register mode.

2. Press BANK repeatedly to locate the bank you wish to store the register in. Banks 1 through 5 are internal registers, and banks 6 through 10 are stored in the nonvolatile memory cartridge.

> **Note:** If you have difficulty using a cartridge, it may not have been formatted. Also, cartridges formatted with earlier versions of software may not work with later versions until they are reformatted. See Chapter 8, *Solving Problems* for instructions on how to format the cartridge.

3. Press REG repeatedly to locate an "Unused" register, or a register you don't mind erasing.

4. With the register number that you want to use flashing on the display, hold down STO and press REG. The LARC display flashes

SETUP STORED

This lets you know that the register was stored correctly.

Loading Registers

Registers are organized into banks, selected and loaded in exactly the same manner as programs. However, you press REG to switch from program to register mode, and press REG instead of PROG when selecting, storing, and loading registers.

Naming Registers

When you store a register, the edited program still has the same name as the original program. To avoid confusion, you can assign names to registers. To rename a register:

1. Recall the register you want to rename and press CTRL to enter the control mode.

2. Press PAGE, 3 to go to page three.

3. Press the key under the slider marked SEL to activate the select function. The current name of the program appears in the lower display.

4. Move the SEL slider. Note that different characters within the name are selected by a pair of brackets < > as you move the slider. Select the first character in the program name.

5. Use the CHG slider to change the character. Note that a blank space is available at the bottom of the slider's range, as are several symbols.

6. Repeat steps 4 and 5 until all the characters in the new name have been entered successfully.

7. Press CTRL to exit the Control Mode.

8. The new name will be lost unless you store the register again. Press REG once. Then hold down STO and press REG to store the register with its new name.

> **Note:** In this example, we asked you to first store an edited program, change its name, and then store it again. Actually, there is no reason not to change the name before storing the edited program for the first time.

Protecting Against Loss of Register Contents

Setting up a large number of registers to meet your personal requirements can represent a considerable investment of time and effort. To reduce the possibility of accidental loss of the contents of these registers, the 480L has a memory protection feature. When memory protection is on, the 480L does not allow anyone to erase the contents of a register by overwriting it. However, unused registers can be written to.

The 480L has four protection levels:

- PROTECT INT AND CART
- PROTECT CART
- PROTECT INT
- PROTECT OFF

PROTECT INT protects just the internal registers, but allows registers stored in the cartridge to be overwritten. PROTECT CART protects the cartridge, but allows internal registers to be overwritten. PROTECT INT AND CART protects both internal and cartridge registers. To activate memory protection:

1. Press CTRL to enter the control mode.

2. Press PAGE, 2 to go to page 2.

3. Move slider six to select one of the four protection modes.

4. Press CTRL to exit the control mode.

Once activated, memory protection remains in effect until it is turned off again.

Protecting Your Registers Against Another Kind of Loss

After creating a collection of registers, some users may not wish to let others access their "trademark" sounds. If this concerns you, copy any internal registers that you create to a nonvolatile memory cartridge at the end of each session (using the register transporter in the control mode). Then use the register clear function (also found in the control mode) to remove the registers from internal memory. Take the cartridge with you when you leave the facility.

Use of the register transporter and register clear functions are described below.

Moving Registers Around with the Register Transporter

The register transporter has four functions:

- Copy single registers from one location to another
- Move single registers from one location to another
- Copy all internal registers to a cartridge
- Copy all cartridge registers to internal memory

When registers are copied, the original register source remains intact. When registers are moved, the original register source is cleared.

> **Important!** The register protect function found on page two of the control mode must be set to OFF if any moves or copies are to overwrite existing registers.

To copy entire register contents between internal and cartridge memory:

1. Press CTRL to enter the control mode.

2. Press PAGE, 2 to go to page 2.

3. Use slider one to select CPY CART TO INTERNAL or CPY INTERNAL TO CART.

4. Hold down STO and press REG to complete the copy.

> **Note**: When either of these two modes are selected, the SRC and DST sliders are inactive.

To move or copy single registers:

1. Press CTRL to enter the control mode.

2. Press PAGE, 2 to go to page 2.

3. Use slider one to select MOVE or COPY.

4. Use slider two to select the source.

5. Use slider three to select the destination.

6. Hold down STO and press REG to complete the copy or move.

Clearing Register Contents

Page two of the control mode has a CLEAR control that allows complete removal of register contents. CLEAR has three functions:

- Clear a single register
- Clear all internal registers
- Clear all cartridge registers

To clear a single register:

1. Press CTRL to enter the control mode.

2. Press PAGE, 2 to go to page two.

3. Use slider one to select CLR SETUP.

4. Use slider two to select the register that you wish to clear.

5. Hold down STO and press REG to clear the selected register.

To clear all cartridge or internal registers:

1. Press CTRL to enter the control mode.

2. Press PAGE, 2 to go to page two.

3. Use slider two to select CLR ALL INT or CLR ALL CART.

4. Hold down STO and press REG to clear the selected registers.

Note: When either CLR ALL INT or CLR ALL CART are selected, the BANK and REG sliders are inactive.

Control Mode - Reference Section

The following list contains a brief description of every parameter in the control mode, listed in order by page number.

Page One

STA (Status)

Moving the STA slider displays the current status of a variety of different controls on the 480L. This control doesn't permit you to change any settings—it simply allows you to quickly check out the status of several 480L controls.

CFG (Configuration)

The 480L can run any two of its programs simultaneously. The two programs can be used independently, or they can be connected together internally in any of several configurations. The CFG control is used to choose one of these configurations.

SMP (Sampling Rate)

SMP selects between 44.1 kHz and 48 kHz sampling rate. Use the higher 48 kHz sampling rate for maximum audio performance. However, when using digital I/O, the 44.1 kHz rate may be required to match an external device.

CLK (Clock Source)

The 480L can generate its own word clock, or it can be slaved to 48 kHz, 44.1 kHz, and 44.056 kHz external word clocks (through the digital I/O connector). For most applications using the 480L's analog inputs, CLK should be set to INTERNAL. For most digital I/O applications, CLK should be set to EXTERNAL. If EXTERNAL is selected, but an external word clock is not present at the digital I/O connector, the 480L will continue to use its internal word clock.

To determine if the 480L is correctly receiving an external word clock, move the STATUS slider (slider one, page one) to display External Word Clock Status.

> **Important!** Do not send external word clock to the 480L until *after* it is powered up.

INP (Input Source)

INP chooses between analog audio input via the main inputs and digital audio input via the digital I/O port.

Page 2

Page 2 contains controls related to copying and moving registers. It is sometimes referred to as the register transporter page.

FUN (Function Setup)

The FUN slider has eight functions:

- COPY SETUP
- INT TO CART
- CART TO INT
- MOVE SETUP
- CLR SETUP
- CLR ALL INT
- CLR ALL CART
- FORMAT CART

COPY SETUP copies a program or register to a specified register location

INT TO CART copies all the registers in internal memory to the cartridge

CART TO INT copies all the registers in the cartridge to internal memory

MOVE SETUP copies a register to a specified register location, and deletes the original.

CLR SETUP deletes the specified register

CLR ALL INT clears all internal registers

CLR ALL CART clears all registers in the cartridge

FORMAT CART formats the cartridge

SRC (Source)

SRC selects the source register or program for clearing, moves or copies.

DST (Destination)

DST selects the destination register for copies or moves.

PRO (Register Protect)

PRO has four options:

- OFF (no protection)
- INTERNAL (internal registers are protected)
- CARTRIDGE (cartridge registers are protected)
- INT & CART (both internal and cartridge registers are protected)

When registers are protected, they cannot be copied to, moved to, cleared, or otherwise erased. Blank registers can still be copied or moved to.

Page Three

This page is used to change the name of the current program.

SEL (Character Select)

SEL selects the character to change.

CHG (Character Change)

CHG changes the selected character. Symbols are at the bottom of the range, numerals in the middle, and characters at the top.

Page Four

Page four contains controls which allow you to set up 10 MIDI patches. Each register can have a unique set of 10 patches.

SEL (Patch Select)

SEL chooses which of the 10 patches will be edited.

SRC (Source)

SRC selects the MIDI controller or event that will be patched to the 480L parameter or event.

DST (Destination)

DST chooses the 480L parameter or event to be controlled by the MIDI controller or event selected with SRC.

SCL/LOW (Scaling Factor/Low Note)

When SRC is set to a MIDI controller, last note, or last velocity, this slider sets the scaling. Scaling determines the relationship between settings of the MIDI controller and the parameter which is under its control. Scaling ranges from -200% through +200%.

In the sampler and doppler programs, when SRC is set to NOTE EVENT, this control sets the LOW NOTE.

PRM/HIGH (Parameter/High Note)

When SRC is set to a MIDI controller, last note, or last velocity, this slider allows control of the parameter selected with DST. This is particularly useful when trying to set the correct SCL value.

In the sampler and doppler programs, when SRC is set to NOTE EVENT, this control sets the HIGH NOTE.

Page Five

Page Five contains the corresponding register table and the MIDI channel selection control.

PGM (MIDI Program Change Number)

PGM has a range of 0 - 127, and sets the MIDI program change number for the corresponding register table.

TBL (Corresponding Register Table)

TBL chooses the 480L program or register to link to the MIDI program change number selected with PGM.

PGM (Program Change Mode)

Pgm determinese what the 480L will do with incoming MIDI program changes. PGM has three options:

- IGNORE (Ignore incoming program changes)
- FIXED (incoming program changes 0 to 99 are mapped directly to register numbers; 100 to 127 are mapped to the first 28 programs)
- TABLE (Uses the corresponding register table created with PGM and TBL)

CHL (MIDI Channel)

CHL sets the MIDI channel for program changes and patches.

> **Note:** Remember that the MIDI channel and corresponding register table are set separately for each machine when the 480L is in Split or Cascade modes.

3

Using the Reverb Programs

In this chapter we'll discuss the Reverb programs and parameters.

Note: There are now two different reverb/room simulation algorithms available. The primary difference between the two is the density of the reverberation. This is most noticable at large sizes and with long reverb times. The denser algorithm has two pre-echo voices, while the other algorithm has six.

The presets are not organized by algorithm. One bank may contain programs made with both algorithms. To discover which algorithm is used by a particular program, go to page three to see how many pre-echo voices are available. If there are two voices, the program uses the dense algorithm. If there are six, it uses the other one. When creating a new reverb sound, make sure you start with a program that uses the algorithm that you want.

Before we jump into detailed descriptions of the programs and parameters, let's take a look at the philosophy behind the reverberation algorithm's radical new structure.

About the Reverberation Algorithm

The 480L incorporates the results of a great deal of research into acoustics and reverberation. It produces four general classes of sounds: ambience, room simulations, plates, and gated sounds.

In Search of Ambience

Ambience is the use of reverberation or reflected sound energy to give recorded music a sense of being performed in a real acoustic location. Ideally, ambience gives warmth, spaciousness and depth to a performance without coloring the direct sound at all.

Recent research into ambience has shown that this phenomenon depends most critically on the shape of the initial reverberation build-up and decay. Ambience is perceived and has benefit while the music is running (which is most of the time). But once the reverberation has decayed 15 dB it is no longer audible in the presence of the direct sound. So the time it takes for the sound to build up and decay 15 dB determines the perceived reverb time, regardless of what the decay time to -60 dB is. Some very good halls for recording have a rather uneven initial build-up and decay, giving a much longer effective reverb time than their -60 dB reverb time might suggest.

It has become common practice to use predelay in an attempt to emulate the sound of these halls. Adding delay to the reverb sends definitely increases the effective audible reverb time and the apparent size of the hall, but the result sounds unnatural.

If we make echograms of real halls, we find that there is usually a gradual buildup of energy between the arrival of the direct sound and the time at which the reverberation reaches maximum loudness. The sharp attack of added predelay in most reverberation devices sounds entirely different.

In the 480L, the SIZE, SPREAD and SHAPE controls allow adjustment of the buildup and decay of the initial part of the reverberation envelope. SHAPE controls the shape of the envelope, while SPREAD and SIZE set the time over which this shape is active.

In the hall and room programs, SIZE acts as a master control for the apparent size of the space being created by the 480L. Both SPREAD and RT MID vary *linearly* with the setting of SIZE. Thus maximum reverb time and spread 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.

When SHAPE is at minimum, the reverberation envelope builds up very quickly to a maximum amplitude, and then dies away quickly at a smooth rate. This envelope is characteristic of small reverberation chambers and reverberation plates. There are few (if any) size cues in this envelope, so it is ineffective in creating ambience. With this SHAPE setting, SPREAD has no effect. The density is set by the size control, and the rate of decay is set by RTMID. This reverberation envelope is typical of many of the popular digital reverberators of the last few years.

As SHAPE is raised to 32 (about 1/8th of the way up) 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 RTMID. SPREAD has little or no effect on this shape.

When SHAPE is at 64 (1/4 of the way up) buildup is even slower and the sustain is longer. Now SPREAD affects the length of both the buildup and the 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 at a SHAPE value of about 128 (1/2 way up).

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 a inverse sound.

Note that none of these shape effects are audible unless RTMID is set short enough. Generally, RTMID 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 the 480L to produce superior ambience—a sound which is spacious and has great depth—without the long RT60 of a church.

Creating a Realistic Ambient Sound

When you set out to create an ambient 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 RTMID 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 actual RT MID values. The setting of the BASS MULTIPLY is also critical in matching the sound of an existing hall. An ideal concert hall would have a BASS MULTIPLY setting of 1.2. It is rare when actual physical spaces exceed 1.5. Many (if not most) good recording environments have values of BASS MULTIPLY of 1.0 or less, and a value of 0.8 should be tried when attempting to match an existing hall.

There are two additional controls to deal with. SHAPE and SPREAD adjust the effective reverb time when the music is running. Higher values of SHAPE and SPREAD produce a longer effective reverb time. Longer effective reverb times give greater spaciousness to the sound.

The Early Reflection Myth

The importance of early reflections in reverberation has become accepted as indisputable fact. We call it a myth. Much of the myth of early reflections is a result of attempts to emulate the sound of discrete reflections from the floor, stage area, and ceiling of a real hall. This sounds reasonable in theory, but it has been our experience that the resulting preechoes are much different from the early reflections present in real halls, and recorded music is often better off without them.

The reason for the difference is not difficult to discover. Early reflections in artificial reverberation are usually discrete--simply a delayed version of the original sound. Transients such as clicks or drums are clearly heard as discrete reflections, resulting in a coarse, grainy sound. But the reflective surfaces of real halls are complicated in shape, and the reflections they produce are smoothed or diffused. Their time and frequency responses are altered, making them much more interesting. In a very good hall, discrete reflections are hard to identify as such.

Another major disadvantage of discrete early reflections is that the same reflection pattern is applied to every instrument which is fed into the reverberation unit, and each instrument has its timbre altered in exactly the same way. In a real hall, every instrument has a different set of early reflections, and each instrument will have its timbre altered in a different way.

Some engineers find any type of early reflection undesireable. In classical music, many recordings are now made with the orchestra in the middle of the hall, with the specific intention of avoiding early reflections. Too much early reflected energy makes the sound muddy, and does not add to richness or spaciousness. This is in part because reflections and reverberation also exist in the playback room.

The 480L reverberation algorithm still offers the option of adding early reflections (preechoes) but we have made them diffused clusters of preechoes. The density of the cluster is set by the DIFFUSION control. We recommend that these preechoes be used with caution, unless you are trying to match the sound of the reverberation to a particular location where such reflections are strong.

When creating new reverberation sounds of your own, don't forget that an Effects program can be put in series with the reverberation (using the Cascade configuration described in Chapter 2). The result can be extremely interesting. Also, try using the Effects program to give high frequencies a different envelope from low frequencies.

About the Reverberation Parameters

Page One						Page Two					
RT Mid	Shape	Spread	Size	HF Cutoff	Pre-Delay	Bass Mult.	Cross-over	RT HF Cut	Dif-fusion	Decay Opt.	Mix

Page Three	Page Four
Preecho Levels	Preecho Delays

Figure 3.1. Reverberation Parameters.

Page One

RT MID (Mid-Frequency Reverb Time)

RT MID sets the reverb time for mid-frequency signals *when the signal stops.* Because low-frequency reverb time (BASS MULT) is a multiplier of RT MID, RT MID acts as a master control for the stopped reverb time. When DECAY OPT is set to Reverb mode, the actual value set for RT MID varies with the setting of SIZE. SIZE should be adjusted before RTMID. This interaction is deactivated when DECAY OPT is set to EFFECTS mode.

SHAPE

SHAPE and SPREAD work together to control the overall ambience of the reverberation created by the 480L. SHAPE determines the contour of the reverberation envelope. With SHAPE all the way down, reverberation builds explosively, and decays quickly.

> **Note**: SPREAD only functions when SHAPE is set higher than eight.

As SHAPE is advanced, reverberation builds up more slowly and sustains for the time set by SPREAD. With SHAPE in the middle, the buildup and sustain of the reverberation envelope emulates a large concert hall (assuming that SPREAD is at least halfway up, and that SIZE is suitably large—30 meters or larger.)

SPREAD

SPREAD works together with SHAPE to control the contour of the overall ambience of the sound created by the 480L. SPREAD controls the duration of the initial contour of the reverberation envelope (SHAPE controls the envelope). 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.

SPREAD and SHAPE control the rate at which reverberation builds up, and how the reverberation sustains as it begins to decay. When DECAY OPT is in Reverb mode, SPREAD is linked to SIZE, and the actual value for SPREAD depends on the selected SIZE.

SIZE

SIZE sets the rate of buildup of diffusion after the initial period (which is controlled by DIFFUSION). It also acts as a master control for RT MID and SPREAD. For this reason, the SIZE control can be used to vary a reverb sound from very large to very small. Generally, you should set the SIZE control to approximate the size of the acoustic space you are trying to create. The size in meters is roughly equal to the longest dimension of the space. Moving SIZE while a signal is present momentarily mutes the reverb signal.

The apparent size of the space created is actually a combination of the settings of the SIZE, SHAPE, and SPREAD controls. Small acoustic spaces are characterized by a rapid buildup of diffusion. However, both small and large spaces frequently have an uneven buildup of initial reverberation. This uneven buildup is what is controlled by the SPREAD and SHAPE controls.

HF CUTOFF

HF CUTOFF sets the frequency above which a 6 dB/octave low-pass filter attenuates the processed signal. It attenuates both preechoes and reverberant sound. High frequencies are often rolled off with this parameter, resulting in more natural sounding reverberation.

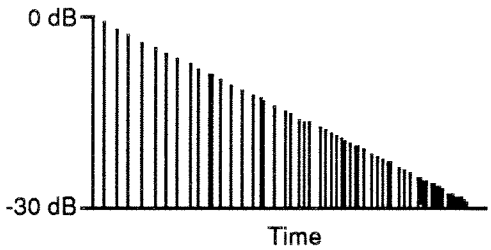


Figure 3.2. SHAPE Set All the Way Down.

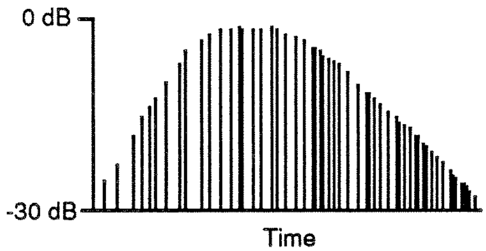


Figure 3.3. SHAPE at 64 - 1/4 of the Way Up.

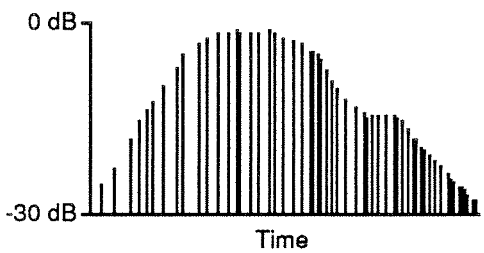


Figure 3.4. SHAPE at 128 - 1/2 of the Way Up.

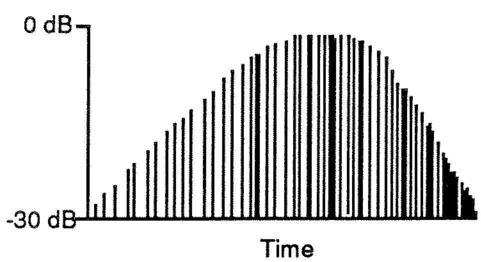


Figure 3.5. SHAPE at 255 - All the Way Up. Short RT MID Setting.

PREDELAY

PREDELAY sets the amount of time which elapses between input of signal and the onset of reverberation. Under natural conditions, the amount of predelay depends on the size and construction of the acoustic space and the relative position of the sound source and the listener(s). PREDELAY attempts to duplicate this phenomenon and is used to create a sense of distance and volume within an acoustic space. Relatively long PREDELAY settings place the reverberant field behind rather than on top of the input material. Extremely long PREDELAY settings produce unnatural sounds that often prove interesting.

A sense of continuity between source and reverb is maintained up to around 40 ms of predelay, after which the sound begins to separate into distinct patterns; however, large values of PREDELAY can effectively give the impression of large size if early reflections are used to fill in the spaces between input and the delayed reverberation.

Much of the effect of PREDELAY can be better achieved by using medium values of SHAPE, and setting the desired apparent predelay with SPREAD. Setting these parameters should be done by ear, since the values don't relate directly to ms.

Note: Very high values of PREDELAY limit the amount of SPREAD available. The display does not reflect this, however.

Page Two

BASS MULTIPLY

BASS MULTIPLY sets the reverb time for low-frequency signals, as a multiplier of the RT MID parameter. For example, if BASS MULTIPLY is set to 2X, and RT MID is set to two seconds, the low frequency reverb time will be four seconds. For a natural-sounding hall ambience, we recommend values of 1.5X or less.

CROSSOVER

CROSSOVER sets the frequency at which the transition from LF RT to RT MID takes place. CROSSOVER should be set at least two octaves higher than the low frequency you want to boost. For example, to boost a signal at 100 Hz, set the CROSSOVER to 400 Hz (This setting works well for classical music). CROSSOVER works best around 500 for boosting low frequencies, and around 1.5 kHz for cutting low frequencies.

RT HF CUT

RT HF CUT sets the frequency above which sounds decay at a progressively faster rate. It filters all the sound except the preechoes. When set relatively low, it gives a darker tone to the reverberation, simulating the effect of air absorption in a real hall. This also helps keep the ambience generated by the program from muddying the direct sound.

DIFFUSION

DIFFUSION controls the degree to which initial echo density increases over time. High settings of DIFFUSION result in high initial buildup of echo density, and low settings cause low initial buildup. After the initial period (in which echo buildup is controlled by DIFFUSION) density continues to change at a rate determined by SIZE. To enhance percussion, use high settings of diffusion. For clearer and more natural vocals, mixes, and piano music, use low or moderate settings of diffusion. The plate presets and some of the room presets use an algorithm with higher inherent diffusion. If high diffusion is desired, start with one of these presets. They are easily identifiable because they have only two preechoes.

DECAY OPT (Decay Optimization)

DECAY OPT alters program characteristics in response to changes in input level, to make reverberation decay sound more natural. DECAY OPT should normally be set to REVERB 7.

To make it easy to create "wild spaces" DECAY OPT has a second mode--EFFECTS. In the EFFECTS mode, the numbers 0 - 9 have the same effect as they do in the REVERB mode. However, in the EFFECTS mode the SPREAD control is not linked to the SIZE control, making it possible to use high values of SPREAD with low values of SIZE. These settings can result in some interesting, but unnatural sounds.

Note: On certain types of program material (such as soft low-frequency tones from a synthesizer) side effects may be audible during level changes. If these are heard, set DECAY OPT to REVERB 0 or EFFECTS 0.

WET/DRY MIX

WET/DRY MIX controls the ratio of direct vs. effect signal in the output from a program. When the 480L is patched into a console, this control should almost always be set to 100% wet. When an instrument is plugged directly into a 480L, or when the Cascade configuration is in use, a setting between 45 and 60% is a good starting point for experimentation with this parameter.

WET/DRY MIX is a sine/cosine fade. Practically speaking, this means that MIX can be adjusted over its range with little or no change in output level. When you control mix at the console, adding effect to the dry signal increases overall level.

Page Three

PREECHO LEVEL

Preechoes can best be understood by visualizing a stage where the early reflections are the sounds emanating from the rear and side stage walls directly after the sound from the stage. Usually the rear stage wall reflection is earlier and louder than those from the two side walls. The preechoes are actually clusters of echoes, with the density of the cluster set by DIFFUSION.

The preecho reflection parameters change the perceived locations of reflecting surfaces surrounding the source. PREECHO LEVEL adjusts the loudness of the reflection.

Note: Some of the presets use an algorithm with six preechoes, and others only have two. If you need more than two when creating a sound, be sure to start with a preset that has six.

Page Four

PREECHO DELAY TIME

For each of the PREECHO LEVEL parameters, there is a corresponding PREECHO DELAY TIME parameter. PREECHO DELAY TIME sets the delay time in ms for one of the preechoes. PREECHO DELAY TIME is not affected by PREDELAY, so preechoes can be placed to occur before the reverberation starts.

Notes

Bank 1 - Halls

Program #	Name	Page One: RT Mid	Shape	Spread	Size	HF Cutoff	Pre-Delay	Page Two: Bass Mult.	Cross-Over	RT HF Cut	Dif-fusion	Decay Opt. Mix
1	Large Hall	2.19 s	126	179	37 m	2.862	24 ms	1.2 x	752 Hz	4.186	99	R 7 All Fx
2	Large+Stage	2.19 s	126	179	37 m	2.862	24 ms	1.2 x	752 Hz	4.186	99	R 7 All Fx
3	Medium Hall	1.74 s	126	128	25 m	4.395	24 ms	1.2 x	752 Hz	3.982	99	R 7 All Fx
4	Medium+Stage	1.74 s	126	128	25 m	4.395	24 ms	1.2 x	752 Hz	3.982	99	R 7 All Fx
5	Small Hall	1.13 s	96	50	25 m	4.395	24 ms	1.0 x	752 Hz	3.784	99	R 7 All Fx
6	Small+Stage	1.13 s	96	50	25 m	4.395	24 ms	1.0 x	752 Hz	3.784	99	R 7 All Fx
7	Large Church	4.04 s	85	247	38 m	2.523	30 ms	1.5 x	1.02 Hz	2.691	80	R 7 All Fx
8	Small Church	2.42 s	65	106	31 m	3.402	0 ms	1.0 x	752 Hz	3.591	70	R 7 All Fx
9	Jazz Hall	1.26 s	34	98	23 m	12.177	0 ms	1.2 x	752 Hz	5.538	80	R 7 All Fx
0	Auto Park	5.29 s	149	247	38 m	7.818	24 ms	1.0 x	752 Hz	5.538	99	R 0 All Fx

Figure 3.6. Halls Bank - Programs and Parameters.

General Description

The programs in the Halls bank are reverberation programs designed to emulate real concert halls.

While the Halls are useful for a wide variety of tasks, they are especially good with traditional and classical music. For popular music, they can be used to give multitrack recordings the sense of belonging to the same performance, by putting the whole mix in the context of a real-sounding acoustic space.

Large Hall - Program 1

Large Hall provides the sense of space and ambience of a large concert hall to music which has already been mixed.

Acoustically, the sound of this program resembles a large, relatively square concert hall. The musicians are not placed in a stage area at one end, but in the middle of the hall, away from nearby walls and other surfaces that produce reflections. The reverberant pickups are located between the sound source and the walls, and are directed away from the musicians, so they pick up little or no direct energy.

The resulting reverberation has the space and ambience of a large hall, but does not color or muddy the direct sound of the recording. Because of the large SPREAD value used, the sound of the Large Hall is most effective when relatively small amounts of it are mixed with the direct signal. If the reverberation sounds obtrusive or tends to reduce clarity, you are using too much of it!

BASS MULT, RT HF CUT, and HF CUTOFF have been set to values typical of good concert halls. SIZE is set at maximum to provide reverberation with medium density and low color. If higher density is required (for material such as closely-miked percussion) try reducing SIZE to about 25.

Large + Stage - Program 2

Large + Stage is similar to Large Hall, except that the musicians are located at one end of the hall, and several preechoes simulate the effects of a procenium arch.

Page Three						Page Four							
Preecho Levels						Preecho Delays							
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Large Hall	1
-12 dB	-10 dB	-8 dB	-8 dB	-9 dB	-9 dB	16 ms	22 ms	64 ms	56 ms	112 ms	102	Large+Stage	2
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Medium Hall	3
-14 dB	-12 dB	-10 dB	-10 dB	-12 dB	-12 dB	16 ms	22 ms	44 ms	38 ms	80 ms	76	Medium+Stage	4
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Small Hall	5
-12 dB	-12 dB	-10 dB	-10 dB	-14 dB	-14 dB	12 ms	18 ms	44 ms	36 ms	72 ms	52	Small+Stage	6
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Large Church	7
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Small Church	8
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Jazz Hall	9
-6 dB	-6 dB	-9 dB	-9 dB	-12 dB	-12 dB	22 ms	12 ms	44 ms	66 ms	164 ms	136	Auto Park	0

Medium Hall - Program 3

Medium Hall is very similar to Large Hall, but smaller.

Medium + Stage - Program 4

Medium + Stage is very similar to Large + Stage, but smaller.

Small Hall - Program 5

Small Hall is a smaller version of Medium Hall.

Small + Stage - Program 6

Small +Stage is a smaller version of Medium + Stage..

Large Church - Program 7

Large Church is a big space with the musicians centrally located, and a comparatively long RT MID.

Small Church - Program 8

Small Church is a smaller version of program 7.

Jazz Hall - Program 9

Jazz Hall is a relatively small space with hard bright walls and a short RT MID. It emulates a hall full of people, without the noise they make. It has high diffusion, and sounds good with jazz or pop material.

Auto Park - Program 0

Auto Park reproduces the sound of an underground parking garage.

Bank 2 - Rooms

		Page One						Page Two				
Program #	Name	RT Mid	Shape	Spread	Size	HF Cutoff	Pre-Delay	Bass Mult.	Cross-Over	RT HF Cut	Dif-fusion	Decay Opt. Mix
1	Music Club	1.03 s	40	55	25 m	7.181	0 ms	1.0 x	752 Hz	3.784	78	R 7 All Fx
2	Large Room	0.70 s	52	82	19 m	6.593	0 ms	1.0x	752 Hz	3.784	65	R 7 All Fx
3	Medium Room	0.50 s	22	10	19 m	7.181	0 ms	1.0 x	752 Hz	3.784	65	R 7 All Fx
4	Small Room	0.31 s	16	0	10 m	7.181	0 ms	1.0 x	752 Hz	3.784	60	R 6 All Fx
5	Very Small	0.13 s	8	0	4 m	7.181	0 ms	1.0 x	752 Hz	3.784	55	R 0 All Fx
6	Lg Wood Rm	1.33 s	73	34	23 m	8.513	0 ms	0.8 x	1.158 Hz	5.538	82	R 7 All Fx
7	Sm Wood Rm	0.71 s	45	19	13 m	8.513	0 ms	0.8 x	1.158 Hz	2.691	80	R 7 All Fx
8	Lg Chamber	0.88 s	3	0	20 m	7.181	10 ms	1.0 x	752 Hz	6.047	99	R 6 All Fx
9	Sm Chamber	0.36 s	16	0	10 m	7.181	0 ms	1.0 x	752 Hz	3.784	70	R 6 All Fx
0	Small & Bright	0.65 s	40	39	9 m	10.591	6 ms	0.8 x	621 Hz	7.493	81	R 7 All Fx

Figure 3.7. Rooms Bank - Programs and Parameters.

General Description

The room programs are similar to the Hall programs, but the spaces they emulate are smaller and somewhat more colored. The rooms are useful for film and video production, as well as classical and popular music recording.

Music Club - Program 1

Music Club is similar to Jazz Hall, but is smaller and less reverberant--especially at high frequencies.

Large Room - Program 2

Large Room resembles a good-sized lecture room. It is smaller than Music Club, and more colored, with comb filtering and slap echoes.

Medium Room - Program 3

Medium Room is a smaller version of Large Room.

Small Room - Program 4

Small Room is much smaller and less reverberant than the Large and Medium Rooms. It resembles a typical American living room.

Very Small Room - Program 5

Very Small Room has the intimate, close feel of a bed-room or den.

Page Three						Page Four							
Preecho Levels						Preecho Delays							
Off	Off					0 ms	0 ms					Music Club	1
Off	Off					0 ms	0 ms					Large Room	2
Off	Off					0 ms	0 ms					Medium Room	3
Off	Off					0 ms	0 ms					Small Room	4
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Very Small	5
Off	Off					0 ms	0 ms					Lg Wood Rm	6
Off	Off					0 ms	0 ms					Sm Wood Rm	7
Off	Off					0 ms	0 ms					Lg Chamber	8
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Sm Chamber	9
-14 dB	-14 dB	-14 dB	-14 dB	Off	Off	14 ms	10 ms	28 ms	44 ms	0 ms	0 ms	Small & Bright	0

Large Wood Room - Program 6

Large Wood Room is similar to Large Room, but has a lower BASS MULT, simulating a room with thin wooden paneling, or a cheaply made warehouse or auditorium.

Small Wood Room - Program 7

Small Wood Room is a smaller version of program 6.

Large Chamber - Program 8

Large Chamber has few size cues. It produces a sound similar to a good live chamber with nonparallel walls and hard surfaces. Large Chamber can be used wherever a plate would normally be used, but with a more subtle acoustic sound.

Small Chamber - Program 9

Small Chamber is a smaller version of program 8.

Small and Bright - Program 0

Small and Bright adds presence to a sound without adding a lot of obvious reverberation.

Bank 3- Wild Spaces

	Page One						Page Two				
Program # Name	RT Mid	Shape	Spread	Size	HF Cutoff	Pre-Delay	Bass Mult.	Cross-Over	RT HF Cut	Dif-fusion	Decay Opt. Mix
1 Brick Wall	0.24 s	0	254	26 m	10.591	0 ms	1.5 x	1.886 Hz	Full R.	88	E 7 All Fx
2 Buckram	0.24 s	94	61	24 m	6.882	0 ms	1.5 x	1.886 Hz	Full R.	98	E 7 All Fx
3 Big Bottom	0.89 s	66	210	31 m	11.084	0 ms	4.0 x	243 Hz	Full R.	88	E 7 All Fx
4 10W-40	0.78 s	10	88	19 m	Full R.	4 ms	4.0 x	885 Hz	1.886	99	E 7 All Fx
5 20W-50	1.01 s	152	94	23 m	11.084	4 ms	4.0 x	621 Hz	621	99	E 7 All Fx
6 Metallica	0.97 s	57	187	28 m	14.986	14 ms	1.5 x	1.020 Hz	7.493	90	E 7 All Fx
7 Silica Beads	5.46 s	126	252	37m	9.278	24 ms	0.2 x	4.395 Hz	Full R.	80	E7 All Fx
8 Inside Out	1.36 s	243	112	20 m	10.591	22 ms	1.2 x	752 Hz	4.611	99	E 7 All Fx
9 Ricochet	1.56 s	0	0	34m	14.986	18 ms	0.6 x	1.735 Hz	10.127	90	E 7 All Fx
0 Varoom	0.78 s	255	216	28m	12.177	0 ms	2.0 x	621 Hz	12.177	98	E7 All Fx

Figure 3.8. Wild Spaces Bank - Programs and Parameters.

General Description

The programs in the Wild Spaces bank can best be described as reverberation effects. They produce reverberation, but their sounds bear little resemblance to anything found in nature. These programs are specifically intended for use in popular music production, and have no known applications in traditional or classical music.

Brick Wall - Program 1

Brick Wall, as in running into, rather than sounding similar to. This program can best be described as a subtle gated inverse room, but it's really much more. Unlike most gated reverb effects, this one's usefulness extends well beyond drum sounds. Try it on a wide variety of material.

Buckram - Program 2

Buckram is a variation of Brick Wall. The difference is that Buckram doesn't sound as dense as the Brick Wall, and has a longer reverb tail.

Big Bottom - Program 3

Big Bottom has a relatively short RT MID and a much longer bass reverb time. This produces a big boom from low-frequency material, while leaving the high end more or less untouched. This is useful for adding a big bass and tom drum sound to an existing mix, or to a drum machine with premixed stereo outputs.

10W-40 - Program 4

10W-40 emulates the sound of an oil drum. If your facility lacks an oil drum wired for sound, you will be pleased to discover that Lexicon has supplied one—before you even knew you needed it.

20W-50 - Program 5

A more aggressive oil drum.

Page Three						Page Four							
Preecho Levels						Preecho Delays							
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Brick Wall	1
-6 dB	-6 dB					64 ms	40 ms					Buckram	2
Off	Off	Off	Off	Off	Off	0 ms	0 ms	0 ms	0 ms	0 ms	0 ms	Big Bottom	3
Full Up	Full Up	-5 dB	-3 dB	Off	Off	0 ms	0 ms	26 ms	46 ms	0 ms	0 ms	10W-40	4
Full Up	Full Up	-5 dB	-3 dB	Off	Off	0 ms	0 ms	50 ms	64 ms	0 ms	0 ms	20W-50	5
-7 dB	-7 dB	-18 dB	-12 dB	-18 dB	-20 dB	70 ms	88 ms	136 ms	156 ms	284 ms	276	Metallica	6
Off	Off					64 ms	40 ms					Silica Beads	7
-14 dB	-14 dB					20 ms	22 ms					Inside Out	8
-12 dB	-10 dB					378 ms	322 ms					Ricochet	9
Off	Off					14 ms	18 ms					Varoom	0

Metallica - Program 6

Metallica produces dense, metallic reverberation with lots of hard echoes. Designed for heavy metal.

Silica Beads - Program 7

Put a small monitor upside down on top of a snare drum, pour a few thousand beads on top of the drum, and hit the monitor with a couple hundred watts. The result? Not nearly as interesting as the Silica Beads program.

Inside Out - Program 8

Inside Out produces a big echo with a big difference—it's turned inside out. Listen closely to percussive material.

Ricochet - Program 9

Ricochet emulates a fairly large space with a dangerous slapback echo.

Varoom - Program 0

Varoom is a room with no resemblance to any known acoustic space; the sound accelerates as it goes by.

Bank 4- Plates

Program #	Name	RT Mid (Page One)	Shape	Spread	Size	HF Cutoff	Pre-Delay	Bass Mult. (Page Two)	Cross-Over	RT HF Cut	Dif-fusion	Decay Opt.	Mix
1	A Plate	2.00 s	0	0	20 m	8.513	0 ms	0.6 x	752 Hz	Full R.	97	R 0	All Fx
2	Snare Plate	1.84 s	1	0	16 m	Full R.	60 ms	0.6 x	120 Hz	Full R.	95	R 0	All Fx
3	Small Plate	1.65 s	0	6	18 m	15.886	2 ms	1.0 x	885 Hz	10.127	99	R 0	All Fx
4	Thin Plate	1.59 s	0	0	15 m	Full R.	0 ms	0.6 x	752 Hz	15.886	85	R 0	All Fx
5	Fat Plate	1.98 s	97	130	34 m	9.278	2 ms	1.0 x	1.586 Hz	21.181	75	R 0	All Fx

Figure 3.9. Plates Bank - Programs and Parameters.

General Description

The Plate programs mimic the sounds of metal plates, with high initial diffusion and a relatively bright, colored sound. For this reason, they are good choices for percussion. They are designed to be heard as part of the music, mellowing and thickening the initial sound itself. The Plate sound is what most people associate with the word reverb, and it is useful for all popular music.

A Plate - Program 1

A Plate is a basic plate program with a very clear sound; you'll find it useful for everything from vocals to percussion.

Snare Plate - Program 2

Snare Plate has its HFC and RT HFC parameters set to full range, resulting in a rapid buildup in high-frequency information. As its name implies, it has been tuned for optimal results with snare drum.

Page Three	Page Four		
Preecho Levels	Preecho Delays		
-8 dB -9 dB	14 ms 18 ms	A Plate	1
-9 dB -12 dB	110 ms 152ms	Snare Plate	2
-2 dB -4 dB	10 ms 6 ms	Small Plate	3
-6 dB -6 dB	14 ms 18 ms	Thin Plate	4
-6 dB -9 dB	30 ms 30 ms	Fat Plate	5

Small Plate - Program 3

Another plate variation. As its name implies, this program produces the sound of a smaller plate.

Thin Plate - Program 4

Another variation on the plate theme.

Fat Plate - Program 5

Fat Plate produces the sound of a very large, highly-colored plate.

4

Using the Effects Programs

This chapter describes the Effects programs and their parameters. The Effects programs are located in Bank 5.

About the Effects Algorithm

The effects in the 480L are based on *randomly varying* time delays. Within this general class a great variety of sounds are possible. Of greatest interest are the natural acoustical effects, such as the effect of a forest on sound, a drum cage, or reflections from audiences, walls, and rooms. Most of these natural effects are quite complex, and are difficult or impossible to obtain using a delay line with fixed taps. The effects of slightly moving sources, or several musicians, cannot be achieved with fixed time delays and only one input. Simple clusters of delays (which produce an interesting sound when first heard) become annoying when the timbre they create applies in exactly the same way to every sound run through the box.

In the 480L, the delay pattern and the resulting timbre is never constant long enough to become boring. Making the taps randomly vary in time solves many of these problems, and allows the creation of more interesting sounds.

Perhaps the oldest time-varing effect is simple chorusing, where a single input is delayed with a number of taps, and the time delay of each tap randomly varies in time. Such a program makes a chorus out of a single voice. In the 480L, chorusing uses up to 40 voices, 20 on each input channel.

For many effects 40 voices is not enough--we might want much more than that to simulate the irregular surfaces of a drum cage, many trees in a forest, or many cars in a parking lot. To accomodate this, we have added a diffusion control, so that each of the 40 voices may be expanded into a dense cluster of reflections.

Some reflective surfaces, such as people or music stands, reflect high frequencies primarily. To allow emulation of these, we have added a high-pass filter with 12dB/octave slopes.

Natural effects are not the only ones possible. The time-varing taps may be adjusted so they lie on top of each other, creating phasing and flanging which is quite interesting and unique. This phasing can be delayed with the PREDELAY, and then made into echoes with FEEDBACK, creating ghostly sounds which bounce and repeat.

In addition, by using the INPUT DELAY control, the effect can be made to precede the sound which created it; thus a high frequency brilliant edge can be added to a cymbal crash before the crash is struck, and the amount of the edge, and its tone quality, will be different every strike.

All these sounds are made available through a few simple controls.

Page One						**Page Two**					
Spin	Slope	Length	Wan-der	Num-ber	Pre-Delay	Input Blend	Fdbk Level	Fdbk Dly	Dif-fusion	Input Delay	Mix

Page Three			**Page Four**
High Pass L	High Pass R	Not Used	Not Used

Figure 4.1. Effects Parameters.

About the Effects Parameters

Page One

SPIN

SPIN sets the rate of WANDER. SPIN is a log control with a period of 8. In English, this means that if you increase SPIN by eight units, the amount of audible spin increases by a factor of two. There is always some spin--even with SPIN at 0.

There is a trade-off between NUMBER and SPIN; lower NUMBERs increase spin speed.

> **Note:** After changing SPIN or LENGTH, the voices take a while to stabilize. Faster SPIN settings stabilize faster.

SLOPE

SLOPE controls the amplitude of the delays over time. Figure 4.2 shows the decay characteristic with SLOPE all the way down. The variation in level is linear on a log scale as shown. Overall level is adjusted to keep the loudness constant. Figure 4.3 shows the decay characteristic with SLOPE midway up. Figure 4.4 shows the decay characteristic with SLOPE all the way up.

LENGTH

The delay of each voice is equal to the LENGTH setting divided by the number of voices set with NUMBER.

WANDER

With WANDER set to 0, the voices are absolutely fixed to their constant ratio apart. An impulse put in without wander will sound like a single delay line with feedback. As you add wander, delays go backwards and forwards randomly in respect to each other.

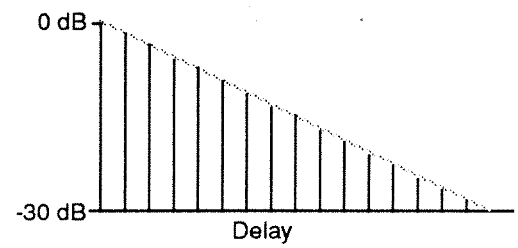


Figure 4.2. SLOPE All the Way Down.

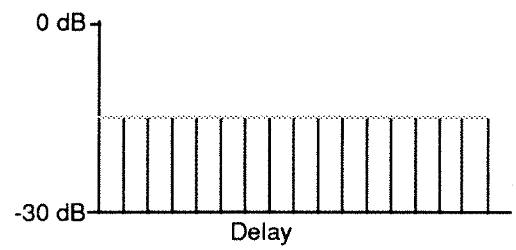


Figure 4.3. SLOPE Midway Up.

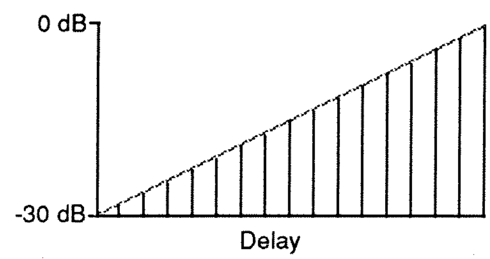


Figure 4.4. SLOPE All the Way Up.

WANDER
WANDER sets the amount of wander in each direction that the delay will move.

NUMBER
NUMBER sets the number of voices used.

PREDELAY
PREDELAY sets the delay before the effect begins.

Page Two

INPUT BLEND (labeled "MON")
INPUT BLEND allows manipulation of the input configuration, from normal stereo through mono, to reverse stereo. The Effects algorithm operates in true stereo. When INPUT BLEND is set to stereo, the left output is derived only from the left input, and the right output is derived only from the right input. So if you are trying to create an effect with sound movement from one output to the other, INPUT BLEND should be set to mono.

FEEDBACK LEVEL
FEEDBACK LEVEL controls the level of signals recirculated back to the input. Increasing the amount of feedback can create interesting resonant effects.

FEEDBACK DELAY
FEEDBACK delay sets the delay that occurs between signal input and the onset of feedback. Try setting FEEDBACK DELAY to the same value as LENGTH for interesting effects.

DIFFUSION
DIFFUSION spreads out the input signal over time, turning sharp transients such as clicks into swishing sounds.

INPUT DELAY
INPUT DELAY adds delay only to the dry signal path—it has no effect on the wet signal path. This effectively allows you to "live in the past," since by delaying the input you can add an effect that happens before the dry signal is heard. This only works if you use WET/DRY MIX to mix the effect with the dry signal. Using the console to mix will negate the effectiveness of the INPUT DELAY.

WET/DRY MIX
WET/DRY MIX controls the ratio of direct vs. effect signal in the output from a program. When the 480L is patched into a console, this control should almost always be set to 100% wet. When an instrument is plugged directly into a 480L, or when the Cascade configuration is in use, a setting between 45 and 60% is a good starting point for experimentation with this parameter.

WET/DRY MIX is a sine/cosine fade. Practically speaking, this means that MIX can be adjusted over its range with little or no change in output level. When you control mix at the console, adding effect to the dry signal increases overall level.

Use of INPUT DELAY can produce effects that actually happen before the dry signal. When producing these effects, you must use WET/DRY MIX--controlling the mix at the console will negate the effect of the input delay. (See INPUT DELAY for more details).

Page Three

HIGH PASS (Left and Right)
High PASS adjusts a 12 dB/octave filter on each input channel to attenuate low frequencies.

Bank 5 - Effects

#	Program Name	Page One: Spin	Slope	Length	Wander	Number	Pre-Delay	Page Two: Input Blend	Fdbk Level	Fdbk Delay	Diffusion	Input Delay	Mix
1	Illusion	30	154	0 ms	42 us	40	20 ms	Stereo	Off	0 ms	64	52 ms	All Fx
2	Surfin	44	247	0 ms	882 us	40	20 ms	Stereo	Off	0 ms	64	0 ms	All Fx
3	Voc. Whisper	48	106	38 ms	23 ms	28	8 ms	Stereo	-16 dB	412 ms	44	244 ms	76%
4	Doubler	44	170	16 ms	26 ms	16	48 ms	Stereo	-18 dB	10 ms	82	0 ms	All Fx
5	Back Slap	43	219	86 ms	10 ms	40	38 ms	Stereo	Off	0 ms	57	0 ms	All Fx
6	Rebound	48	254	444 ms	35 ms	40	254 ms	Stereo	-9 dB	0 ms	0	122 ms	64%
7	Elinar	46	254	4 ms	32 ms	40	18 ms	Stereo	-12 dB	10 ms	96	6 ms	50%
8	Sudden Stop	48	130	216	37 ms	40	52 ms	Stereo	-18 dB	48 ms	99	0 ms	All Fx
9	In the Past	45	247	500 ms	13 ms	40	0 ms	Stereo	Off	0 ms	87	504 ms	52%
0	Tremelo L & R	30	135	0 ms	0 us	4	0 ms	Mono B	Off	0 ms	0	0 ms	All Fx

Figure 4.5. Effects Bank - Programs and Parameters.

General Description

The effects produced by these programs can be very difficult to describe. They range from subtle to outrageous, depending largely on the type of source material used, and how much of the effect is added to the mix. These effects are powerful and complex, and we encourage you to spend a great deal of time listening to them.

Illusion - Program 1

Illusion (when added to the mix in relatively small amounts) is a subtle effect that can enhance a sound without a listener even knowing it is there. One often doesn't notice that it is in use until it is taken away. Illusion is also useful for stereo synthesis.

Illusion is effective on complete mixes and on single tracks.

When greater amounts of Illusion are added to the mix, the effect becomes more obvious, and some interesting phasing and panning are audible. The phasing is strong enough that spatial panning results, with some of the sound swirling around and even behind the listener.

Surfin' - Program 2

Surfin' produces flanging when fed with percussive material. Try it on everything from guitars to vocals and percussion.

Vocal Whispers - Program 3

Vocal Whispers is a delay-based effect designed to enhance vocals.

Doubler - Program 4

Doubler is a doubler with a difference—the diffusion used on the delay lines thickens percussive sounds considerably. This is a good choice for fattening up uninteresting sounds.

Back Slap - Program 5

A strong fast slapback effect.

Page Three

High Pass L	High Pass R
0 Hz	0 Hz
120 Hz	120 Hz
120 Hz	120 Hz
120 Hz	120 Hz
120 Hz	120 Hz
120 Hz	120 Hz
120 Hz	120 Hz
120 Hz	120 Hz
120 Hz	120 Hz
120 Hz	120 Hz

Rebound - Program 6

Throw something at this one and it comes rippling right back at you. Try it on vocals with short, explosive syllables.

Elinar - Program 7

This effect has just about everything.

Sudden Stop - Program 8

Sudden Stop produces a sound like a grainy inverse gated room. It's rather interesting on snare, high toms and cymbals. However, it is not intended for use on low frequency material. Avoid low toms, kick drums, and bass guitar.

In the Past - Program 9

In the Past is unique in that the dry signal is set to 504 ms so that it appears *after* the build-up of the effects signal. It should be used with program content being mixed *through* the 480L; in other words, keep the source fader down and send audio to the 480L pre-fade. In the Past uses 40 well-diffused voices. The length of the delay is set to 500 ms with a build-up slope of 247.

Tremolo L and R - Program 0

Tremolo L and R uses four undiffused voices with the delay line and WANDER set to 0. SPIN controls the rate at which the mono blended signal tremelos between the left and right outputs. Tremolo depends for its effect on having the delay lines slightly out of sync. If you load the program and the effect seems to lack depth, load it again.

5

Using the Sampler Programs

The Sampler programs located in Banks 6 and 7 include a variety of useful features for recording and production work. Features allow slip syncing, copying, and time shifting of segments. Accurate triggering (response time is under 300 microseconds) allows replacement of drum or cymbal sounds (with full decay times) when tracking or overdubbing. Capture mode and editing features allow precise manipulation of sampled data.

This chapter describes the samplers, their capabilities, and the parameters used to control them.

The variety of samplers available in the 480L can be a bit overwhelming to the uninitiated. To get started, we recommend loading each sampler, and experimenting with the controls. If you are uncertain about the effect of a parameter, look up its description at the end of this chapter.

Introduction

The 480L has two banks of sampling programs, each with slightly different features and capabilities. Bank 6 (Sampling) contains programs for use with the optional Sampling Memory Expander (SME) card. The SME card occupies the fourth card slot in the 480L mainframe, providing 10.9 seconds of true phase-locked stereo or 21.8 of mono at the 48 kHz sampling rate. At 44.1 kHz, the total sampling time is extended to 11.8 seconds of stereo and 23.7 seconds of mono.

The SME card's extra data capacity permits the sampling programs in Bank 6 to use just one 480L Machine—even at maximum length. This frees up the other Machine to enhance sampled sounds with the 480L's full repertoire of effects processing. Furthermore, since the SME handles all the audio data in one place, the other processors have more time to perform control functions. The controls for rate changing and recording in the SME sampling programs are more powerful than those in the Bank 7 samplers. For example, SME samplers play backwards at variable rates.

The sampling programs use some LARC slider/key sets differently from the standard parameter adjustments used in reverb and effects programs. Normally, a parameter is adjusted by its slider; the key is used to display the current setting, or to switch the slider to fine adjustment. For certain sampler functions, analogous to tape recorder controls, however, some LARC keys trigger events. Moving their sliders describes those events.

Starting with software V2.00, all samplers have been enhanced to include Time Variant Recording. In earlier software, the 480L recorded in what we now refer to as the IMMEDIATE mode. It started recording the instant you pressed REC, and continued until it ran out of memory. Although IMMEDIATE recording is still available, it is now just one of three *capturing* modes. The two new modes offer a significant improvement in precision of control and artistic flexibility.

The preset mode of the 480L samplers is now -24 dB level-triggered capturing. This means that they record continuously after you press REC. This continues more or less indefinitely until capturing is triggered by audio appearing at the left input at a higher level than the preset -24 dB point. The samplers then record until audio memory is filled. Under most conditions, you need do nothing more than press REC, wait for capturing to complete, and play back the sample. Because of the accuracy of the level detection, trimming the sample with the HEAD control to delete "dead air" should no longer be necessary.

Playback is similar for all samplers. Level, manual, and MIDI triggering of sample playback are available for all programs, as are Manual and MIDI retriggering. Level retriggering is available for all programs except Mono Forward/Reverse. HOLDOFF (the time before level retriggering is activated) is adjustable for all programs except Dual Rate Change.

Overdubbing is available for the SME samplers and some non-SME samplers. When overdubbing, the 480L uses the IMMEDIATE recoding mode while playing the existing sample at X1 forward speed.

The programs in bank 6 are:

1. Stereo Rate Change/Overdub(10.9 seconds @ 48 kHz)
2. Mono Rate Change/Overdub (21.8 seconds @ 48 kHz)

The programs in bank 6 require the SME card.

Bank 7 contains samplers which do not require the optional Sampling Memory Expander:

1. Stereo 3S (2.7 seconds @ 48 kHz)
2. Mono 6S (5.4 seconds @ 48 kHz)
3. Mono 3S (2.7 seconds @ 48 kHz)
4. Dual Rate Change (2x 1.4 seconds @ 48 kHz)
5. Mono Forward/Reverse (1.4 seconds @ 48 kHz)

The following programs in bank 7 are adjusted for percussion sampling:

6. Stereo 3S Drum (2.7 seconds @ 48 kHz)
7. Dual Rate Change Drum (2x 1.4 seconds @ 48 kHz)
8. Mono Forward/Reverse Drum (1.4 seconds @ 48 kHz)

Note: All samplers record slightly longer times when the 480L sampling rate is set to 44.1 kHz.

- ░ = **Parameter**
- ☐ = **Event**

Page One						**Page Two**					
Over dub	Rec	Mark	Cap Mode	Cap- ture	Chk	Head Trim	Tail Trim	Edit	Fade Type	Rate	Cue
Page Three						**Page Four**					
Play	Rpt	Trig Level	Retrig Hold	Rate	Cue	MIDI Play		Ref Note	Pitch Mirror	Rate	Cue

Figure 5.1. **SME** Stereo and Mono Rate Change Overdubbing Samplers (Bank 6).

Page One						**Page Two**					
	Rec	Mark	Cap Mode	Cap- ture	Chk	Over- Dub	Rec	Fade Type	Cap Mode	Cap- ture	Chk
Page Three						**Page Four**					
Head Trim	Fwd Time			Fade Type	Cue	Play	Rpt	Trig Level	Retrig Hold		Cue

Figure 5.2. Stereo 3S, Mono 6S, Mono 3S, Stereo Drum (Bank 7).

Page One						**Page Two - Left Channel Voice 1**					
Rec Voice	Rec Key	Mark	Cap Mode	Cap	Chk	Head Trim	Fwd Time		Fade Type	Rate Chg	Cue
Page Three - Right Channel Voice 2						**Page Four**					
Head Trim	Fwd Time		Fade Type	Rate Chg	Cue	V1 Play	V1 Rpt	Trig Lvl 1	V2 Play	V2 Rpt	Trig Lvl 2
Page Five											
MIDI Play 1	Ref Note 1	V1 Rate	MIDI Play 2	Ref Note 2	V2 Rate						

Figure 5.3. Dual Rate Change Sampler (Bank 7).

Page One						**Page Two**					
	Rec	Mark	Cap Mode	Cap- ture	Chk	Head Trim	Fwd Time	Dir	Tail Trim	Rev Time	Cue
Page Three						**Page Four**					
Level	Pan	Level	Pan	Fade Type	Cue	Play	Rpt	Trig Level			Cue

Figure 5.4. Forward/Reverse Sampler (Bank 7).

Bank 6 Samplers – SME Only

Stereo Rate Change/Overdub - Bank 6, Program 1

The Stereo SME sampler records and overdubs 10.9 seconds of phase-locked stereo audio at 48 kHz from the left and right inputs, and plays back through the corresponding output channels. It provides level-triggered capturing from the left input only, referenced to the LARC level indicators. MARK is set to save two ms of pre-trigger audio for a fast fade-up.

Playback can be varied continuously from -100% (X1 reverse) through 0 (stopped) to +199% (just under X2 forward). The rate can be varied continuously during playback. Due to computational limits, some reverse rates introduce audible clicks during playback. -100% and the rates near it are free of noise; all positive rates are free of noise.

In the SME samplers the FORWARD TIME parameter slider is replaced by TAIL TRIM. The user marks the HEAD and TAIL points of the sample; the 480L calculates the play time. These two edit points are interchanged for reverse playback. The 480L will play back the edit accurately regardless of rate variations unless the rate crosses the forward/reverse boundary at 0%.

Mono Rate Change/Overdub - Bank 6, Program 2

The Mono SME sampler records and overdubs 21.8 seconds of audio at 48 kHz from the left input, and plays it back through both left and right outputs. Capturing is set identically to the stereo programs. Playback can be varied continuously from -200% (X2 reverse) through 0 (stopped) to +199% (just under X2 forward). The rate can be varied continuously during playback. All rates are free of any clicks.

Bank 7 Samplers

There are three types of non-SME samplers. They all record in multiples of 1.36 seconds at 48 kHz (1.48 seconds at 44.1 kHz). The presets provide level-triggered capturing from the left input only, referenced to -24 dB on the LARC level indicators. MARK is set to save five ms of pre-trigger audio for a fast fade-up, except for the Drum percussion samplers, where it is set to 0, or "MARK THE HEAD."

Stereo 3S - Bank 7, Program 1

This progam records and overdubs 2.7 seconds of phase-locked stereo at 48 kHz. It features overdubbing of the entire sample. The first capture may use any record mode. The overdub always uses the IMMEDIATE mode. This program must be run in the 480L's SINGLE configuration.

Mono 6S - Bank 7, Program 2

This program records and overdubs 5.4 seconds at 48 kHz, from the left input only. It features overdubbing of the entire sample. The first capture may use any record mode. The overdub always uses the IMMEDIATE mode. This program must be run in the 480L's SINGLE configuration.

Mono 3S - Bank 7, Program 3

This is similar to program 2, except that it is small enough to run in any of the 480L's configurations (not just SINGLE).

Dual Rate Change - Bank 7, Program 4

This program features two independent rate-changing samplers. You may vary the pitch while playing, but avoid trying to cross the pitch up/pitch down boundary. There are two independent editing pages to design the samples, and play pages that permit you to trigger both samples simultaneously. Level retriggering is always active, after a short fixed holdoff.

Mono Forward/Reverse - Bank 7, Program 5

This program can play a sample reversed, either alone, or with forward play. It records and triggers from the left input only, but pans the two playback voices between left and right. The playback timing relationship may be shifted so that either voice precedes the other, or they may play simulaneously. Level retriggering is not available in this program.

Drum Samplers

Programs 6, 7, and 8 in Bank 7 are identical to the samplers described above, except that MARK and FADE TYPE have been set for percussion instead of FAST FADE UP. This ensures that the initial attack of a percussive sample is not dulled in a fade up.

How to Use the Samplers

These general instructions apply to all samplers. (Variations for SME samplers are in parentheses):

Recording

1. Single channel mono samplers record only from the left channel. Dual and Stereo Samplers record from both channels.

2. Adjust audio input levels (as shown on the LARC Headroom display) for +6 dB on peaks.

3. Press REC to begin recording and notice the line of "*******" on the LARC under the label "RECORDING". The 480L is now recording audio to memory. All sampling presets will CAPTURE and preserve the sound when the left channel audio exceeds -24dB on the LARC. You may also, at any time, trigger capturing manually with the CAP key. When triggered, the label switches to "CAPTURING" and the "*******" indicate the remaining recording time. When the last "*" is gone, p.ress CHK to audition the sample. If you don't like the sample, record again as many times as necessary.

Editing

4. When you have a satisfactory sample, use PAGE to go to the editing page.

5. Use HEAD TRIM to remove excess material from the beginning of the sample. *Removing all silence from the beginning of a sound is absolutely necessary for accurate triggering with the audio play trigger.* Always trim only a little bit at a time, and use CUE (EDIT) frequently to audition the results.

6. Use FORWARD TIME (TAIL TRIM) to remove excess material from the end of the sample.

7. Use FADE TYPE to select a hard cut for percussive material, or the normal 5 ms fast fade up.

Playing

8. When you have trimmed your sample, go to the play page to select multiple play and triggering options. The sampler's PLAY key is preset just like the CUE key; press PLAY to manually trigger one play of the edited sample. Press PLAY before the sample finishes to manually retrigger the sample.

9. Use the RP, repeat, slider for more than one play per trigger. The value is not used until the next time you trigger PLAY, either manually, by MIDI, or with audio level, so you may program it in advance. If you start a large number of repeats and need to stop playback, just press CUE.

10. Use the TLV, trigger, slider to set the trigger mode. The samplers are preset with the slider all the way up to respond to a manual trigger from the PLAY key or a MIDI patch. Pulling the slider down to CONTINUOUSLY enables continuous repeats of the sample. Adjusting TLV in dB selects a LARC level for audio triggering from the left input.

11. You must press PLAY to arm the level trigger or to start CONTINUOUS playing. If you wish to disarm level triggering or stop continuous play, press CUE..

12. When the input signal level to the 480L reaches the level you set with PLAY TRIGGER, the sample is played back. For most of the samplers, playback starts 80 us after audio is detected. The sample is fully faded-in 5 ms later. The Forward/Reverse Sampler starts 115 microseconds after audio is detected. The level trigger rearms automatically when each play is complete.

13. To prevent the sampler from accidentally retriggering off incoming audio before the entire sample has been played, all samplers with a RETRIG HOLD slider have been preset to NO RETRIGGER.

14. To enable retriggering, set the RETRIGGER HOLD somewhere *above* NO RETRIGGER. Select a time that is short enough to allow retriggering as often as you desire it, yet long enough to prevent retriggering before you want it. *Setting too short a RETRIGGER HOLD can result in multiple retriggers from a single sound.*

Optimizing Level Triggered Playback

These tips will enable you to obtain tight level triggering for percussion replacement and other critical applications.

14. Make sure there is no "dead air" at the head of your sample. The non-percussion samplers are preset to MARK and preserve a few milliseconds of pre-trigger audio for the FAST FADE UP. You may either MARK THE HEAD (0 milliseonds) or trim this later.

15. Marking the HEAD while using the preset -24dB level-triggered recording should provide a tight enough HEAD trim for most users. You may tighten the recording further by setting MODE to a higher dB level.

16. Besure to set FADE TYPE to PERCUSSION for sounds with tight, clean attacks. If the cue is still too loose, then trim HEAD until just before you hear loss of the attack.

17. Adjust TLV, playback trigger level, for the greatest sensitivity that doesn't give false triggers.

> **Note:** If you are using level retriggering to interrupt and restart sample playback, a combination of too sensitive a trigger level and insufficient retrigger holdoff time can lead to a stuttering, which some hear as a great effect. If it's not for you, remember these general guidelines.
>
> Use a short duration retrigger signal with a sharp attack.
> Raise the level threshold - if this can be done without delaying the initial trigger.
> Make the HOLDOFF time as long as possible

The Bank 7 Dual Rate Change sampler has a fixed retrigger HOLDOFF of about 100 milliseconds. The forward/reverse sampler cannot perform level retriggering, it can only be retriggered manually or via MIDI.

Using Rate Change Samplers

1. Both SME samplers and the Bank 7 sampler, Dual Rate Change, permit you to adjust the playback RATE. Starting with Version 3.00 software, the display now shows percent of normal speed and, over a certain range, musical interval from unison. The pitch interval display corresponds to that of the pitch changer, Bank 8, programs 1 through 6, and shows the range over which you may correct the sampler's pitch. You can cascade these programs for simple time compression.

2. The non-SME Dual Rate Change sampler allows you to record, edit and playback two independent samples. Use the VX, RECORD VOICE slider to select a voice for the Page 1 Record controls. Record and check a sample as above.

3. There are two independent editing pages, and independent play controls for each voice. Voice 1 comes out the left side, Voice 2 out the right.

Using the non-SME FORWARD/REVERSE Samplers

1. The sampler is preset to play only the forward voice. To activate the reverse voice, adjust REV TIME (slider 5) to some large number, and adjust TAIL TRIM to tighten the beginning of the backwards play.

2. To turn off either voice, move its TIME slider to 0.000 SEC.

3. To adjust the relative start times of two active voices, use the "<>" slider, PLAY ORDER. For a different kind of inverse effect, try sliding the forward voice so that it starts just as the reverse is finishing.

4. Adjust the relative gains and panning of each voice on Page 3.

Time Variant Recording

The original (and still available) IMMEDIATE record mode allows control over the start time only. Recording starts when the REC button is pressed, and continues until all audio memory is used. We now use the term *capturing* to describe a sampler that records until a triggering event stops it. Time Variant Recording provides controls that determine exactly what audio is captured (the MARK slider) and how it is captured (the MOD slider, REC key, and CAP key).

The recording side of the sampler has three states: READY, RECORDING, and CAPTURING. In the READY state, all recording is complete, and the 480L is ready for playback or to record again. Tap REC to begin recording.

To determine which of the three states the sampler is currently in, tap CHK or CUE. If the sampler is READY, the LARC displays the name of the key and plays the sample. If the sampler is recording, but not yet triggered, the LARC displays:

RECORDING
****

The asterisks indicate that the sampler is armed and ready to record. If the sampler has been triggered manually (by pressing CAP), by level, or IMMEDIATE, the LARC displays CAPTURING and the asterisks disappear one by one until completion. Pressing CHK or CUE will not disturb recording.

MARK may be adjusted from the HEAD of the sample to the TAIL. If MARK is 0, everything after the trigger is saved. If MARK is TAIL, everything before the trigger is saved.

The most obvious application for the Capture mode is level triggered capture, with MARK time set for zero or just above zero. The response is extremely fast, so the capture will really nail a percussive sound. Set FADE TYPE to PERCUSSION, and set the Trigger Level as low as possible.

Some useful applications are described below:

1. To obtain the original, simple form of recording.

   MOD = IMMEDIATE

2. Percussion capturing. Triggers on first audio, trimmed exactly. Everything after the trigger is saved.

   MOD = Appropriate dB level
   MARK = HEAD

3. Stop when finished. Triggered by user when sound is ended. Everything prior to the trigger is saved.

   MOD = USE CAP KEY
   MARK = TAIL

4. Reaction time manual capturing. User taps REC to arm the 480L, waits for sound to start, and then taps CAP. Sound for the period of time set with MARK prior to tapping CAP is saved.

   MOD = USE CAP KEY
   MARK = An appropriate period of time

5. Level triggered capturing. Same as reaction time manual capturing, but level triggered instead. Useful for sounds that build more slowly than percussion.

   MOD = An appropriate level
   MARK = An appropriate time

Sampling Percussion

When sampling percussion, set FADE TYPE to PERCUSSION. The sample will start with a hard cut instead of the normal FAST FADE. If the sample plays completely, the end is always faded down. From Version 3.00 on, the SME and the overdubbing non-SME samplers have been modified so that a PERCUSSION mode retrigger will cut rather than fade out the running sample. This improves synchronization in fast percussion replacement applications. (It may also result in a click in non-percussive samples, so be careful in choosing which type of fade you want.)

Scrubbing

A feature of the SME samplers is their ability to continuously vary the speed of playback within the limits described in the earlier program descriptions. There are four speed regions defined by these end points: -200%, -100%, 0%, +100% and +199%. You can vary the RATE at will within these regions, but crossing the boundaries may cause glitches whose audibility depends on the source material.

Edits will always be accurate over the entire range of the sampler unless you change direction (cross the 0 boundary) after triggering. There is, therefore, no limitation on auditioning a tight edit at low speed and playing it back fast – just avoid changing direction while playing. Changing direction will cause minor inaccuracies in the play time.

From Version 3.00 on, the SME samplers have a third FADE TYPE called SCRUB MODE, for those applications where play time is less important than the fun you can have moving the RATE slider around. In SCRUB MODE there is no fade out, ever. The sample starts at the HEAD time, then plays continuously through memory and around again.

To kill SCRUB MODE, adjust FADE TYPE to PERCUSSION or FAST FADE UP and press CUE.

Using MIDI to Control a Sampler

Dynamic MIDI® is very helpful when used with the sampling programs. MIDI controllers can be patched to control sampler parameters such as HEAD TRIM, FWD TIME, and FADE TYPE. MIDI Note On events can be patched to control sampler events, such as RECORD, CHECK, PLAY, etc.

A single note can be patched to trigger an event, or a range of notes can control a single event. The following procedure assumes you have made MIDI connections and set the 480L's MIDI channels (as described in MIDI and the 480L).

> **Note:** Because of different processing times for events and parameters in the 480L, you should be careful patching parameters to note event data (LST NOTE, LAST VEL) when the note event is triggering a 480L sampling event like PLAY. The parameter will usually not be updated until after play starts, so the parameter updates will be missed unless the same event is repeated.

In the rate-changing samplers, a special event (MIDI PLAY) has been created that dierctly updates the rate parameter before play starts. This allows triggering samples from a MIDI keyboard at rates that correspond to the MIDI semitone pitch intervals. On the same LARC page are two parameters that control the MIDI PLAY interaction with RATE, Reference MIDI Note, and Pitch MIrror. They are fully described in the next section.

Typically, two patches are needed to control a sampler, assuming that you will handle recording and preliminary editing manually. You will need to patch MIDI source "NOTE EVENT" (over some range of note values) to a 480L sampler "event" (usually a PLAY key) so that playing a synthesizer key will trigger the edited sample.

Having done this, there are several interesting parameters you can vary, including FORWARD TIME (TAIL TRIM in the SME), REPEATS, and RATE. These may be patched to MIDI sources such as PITCH WL, MOD WL and other controllers.

To patch a MIDI controller to a sampler parameter:

1. Press CTRL to enter Control Mode. Go to page 4.
2. Use SEL to select the patch to create (there are 10 possible patches).
3. Use SRC to select the MIDI controller for the patch.
4. Use DST to select the parameter you will control.
5. Use SCL to set the scaling of the MIDI controller to the sampler parameter.
6. Use PRM to set the base setting for the parameter, and audition the effect of the controller.
7. After setting up your patches, be sure to save the new settings in a register.

To patch a MIDI note event to trigger a sampler event:

1. Press CTRL to enter Control Mode. Go to page 4.
2. Use SEL to select the patch to create (only the first four patches can be used for events).
3. Push the SOURCE slider all the way up to NOTE EVENT.
4. Use DST to select the event you wish to control.
5. If you want to use a single key to trigger an event, set LOW NOTE and HIGH NOTE to the same value.
6. If you want a range of keys to trigger an event, use LOW NOTE to set the low end of the range, and HIGH NOTE to set the high end of the range.
7. After setting up your patches, be sure to store the new settings in a register.

When using MIDI to control the sampler, be careful not to send the 480L a program change command. This will load a new program or register, resulting in the loss of the sample in memory. To prevent this from happening, the PGM CHANGE parameter in the control mode can be set to IGNORE. This causes the 480L to ignore MIDI program changes.

About the Sampling Controls and Parameters

All the parameters available in the sampling programs are listed below, in alphabetical order. No single program has all parameters. Refer to the program descriptions for parameter availability.

CAP Key
CAP interacts with the MARK and CAPTURE MODE parameters. When CAP MODE is set all the way up, CAP triggers the capture event described by MARK. In the SME samplers, CAP may also be used to switch monitoring from playback to source.

Capture MODE
Capture MODE selects the capture mode and trigger level. When the slider is at the bottom of its range, MODE is set to IMMEDIATE, which means that capturing is triggered as soon as recording starts. Moving the slider up selects the audio level for level triggering. When incoming audio exceeds this preset level, capturing begins.

With Capture MODE all the way up at USE CAP KEY. Only the CAP key triggers capturing.

CHK Key
CHK is used to audition the entire sample immediately after it is recorded. The effects of the editing controls (HEAD TRIM, TAIL TRIM, etc.) are not heard when CHK is pressed.

CUE Key
CUE is used to audition edits as they are made.

DUB Key
DUB is essentially a CHK play key that also switches on recording when it starts, permitting overdub (sound-on-sound) recording.

EDIT Key
Making minute adjustments of HEAD TRIM and FORWARD TIME to get a sample sounding just right can be tedious. This is especially true with the longer samples possible with the SME. EDIT speeds up the process by allowing you to listen to just the relevant portions of a sample when editing.

EDIT functions as a CUE key, with one important difference—it only plays two seconds of audio. If HEAD TRIM was the last control used, EDIT plays the first two seconds of the sample. If TAIL TRIM was last, EDIT plays the last two seconds of the sample.

FADE TYPE
FADE TYPE selects between Fast Fade Up and Percussion. Use Fast Fade Up for most sampled material except percussion. Use Percussion for drum sounds and percussion.

The SME samplers have a third mode, SCRUB, to eliminate the fade down that occurs at the end of every play. In SCRUB mode audio begins at the selected point, then plays the entire sample memory continuously, This allows rocking the audio back and forth with the RATE slider without any fade down.

FORWARD TIME
FORWARD TIME selects how much of the recorded sample to play back (in forward play).

HEAD TRIM
Once a sample has been recorded, head trim is used to remove unwanted information at the beginning of the sample, selecting a new start point for playback.

LEVL FWD (Mono Forward/Reverse)
LEVL FWD sets the playback level for forward play.

LEVL REV (Mono Forward/Reverse)
LEVL REV sets the playback level for reverse play.

MARK
MARK adjusts the amount of pre-trigger audio that is finally recorded. If MARK = 0, MARK THE HEAD, then no pre-trigger audio is saved. If MARK is set to MARK THE TAIL, the trigger event is a STOP recording command, and only pre-trigger audio is saved. MARK can be adjusted for a few milliseconds in order to fine tune an attack, or up to 0.5 second (1 second in SME) for other uses.

If MARK is set to TAIL, or a large value, the LARC "*******" display will fill up from left to right when REC is pressed. When all twelve "*"s are lit, all old audio will be erased by the new recording, Remember this when using MARK THE TAIL – if you trigger early, old audio will remain in memory. (This could be useful; you can always trim it out.)

MIDI Play Key
From the LARC, MIDI Play functions exactly like Play (below) and plays the sample at the previous rate. When patched to MIDI NOTE EVENT, before playing the sample it first sets a new rate based on the note value, Reference MIDI Note parameter, and Pitch Mirror (SME only). The rate intervals occur on semitone pitch intervals.

MIR
Pitch Mirror modifies the behavior of MIDI Play by decreasing the rate for increasing note vaues. This is particularly useful when playing mono SME samples in reverse. (It can also be used for stereo samples, but some reverse rates may produce audible clicks.)

PAN FWD (Mono Forward/Reverse)
PAN FWD sets the pan location between the two outputs for forward playback.

PAN REV (Mono Forward/Reverse)
PAN REV sets the pan location between the two outputs for reverse playback.

PLAY Key
PLAY is the manual playback trigger. When pressed, it arms level triggering if active, or immediately triggers playing the sample for the selected REPEAT count.

PLAY ORDER (Mono Forward/Reverse)
PLAY ORDER determines whether the forward or reverse sample is played first. With the slider all the way down, the sample is played in reverse first, and forward second. With the slider centered, the sample is played in forward and reverse at the same time. With the slider all the way up, the sample is played forward first, and in reverse second. A wide range of settings between these three basic points is available.

RATE
RATE changes playback speed, resulting in a changed audio pitch. A setting of 100% gives an unchanged pitch on playback. RATE can be varied at any time, either manually, or by MIDI, within the limitations mentioned in the Bank/Program descriptions.

REC Key
The 480L begins recording the instant REC is pressed. REC may be pressed at any time to restart a recording. Forward/Reverse Sampler, and Mono Samplers record via the left input.

REF
Reference MIDI Note selects the MIDI note value that produces X1 forward playback. It is used to transpose the control region of a MIDI keyboard. When MIR is off, notes higher than REF produce faster rates, lower notes produce slower rates, and notes more than an octave down produce reverse play rates.

REPEAT
REPEAT sets the number of times a sample is played. After selecting the number of play repeats, you must enter the selection by pressing PLAY.

RETRIGGER HOLD
When using audio triggering, RETRIGGER HOLD sets the period of time the sampler will wait before retriggering. When set to NO RETRIGGERING (all the way down) the sample may be level retriggered only when play is completed.

REVERSE TIME (Mono Forward/Reverse)
REVERSE TIME sets how much of the sample to play back (in reverse play).

TAIL TRIM
Once a sample has been recorded, TAIL TRIM is used to remove unwanted information at the end of the sample, and select the start point for reverse playback.

TLV, Play TRIGGER
Play Trigger selects the method for triggering playback. With the control all the way down, playback is continuous. In the middle range are playback trigger levels corresponding to the Headroom display on the LARC. With the slider all the way up, triggering is manual only, via the PLAY key.

After selecting a play trigger method or level, enter the selection by pressing PLAY.

VX, Record VOICE MODE
In the Dual Rate Change Sampler, RECORD VOICE selects which of the two voices to record.

6

Using the Doppler Program

This chapter describes the Doppler programs located in Bank 8.

Page One						Page Two		
Play Trig.	Play Type	Amp Q/ Pitch Q	Speed	Time	Dis- tance	Play Trig.	Not Used	Trig. Mode

Figure 6.1. Doppler Parameters.

General Information

Everyone is familiar with the doppler effect heard when a train or truck goes zooming by. The Doppler program recreates this effect with startling realism by reproducing the panning, amplitude and pitch variations heard as a sound source moves past the listener.

These programs were designed specifically for film and video applications. An audio trigger allows the Doppler to be cued into a mix.

Try very short times (one second or less) and small distances (0.3 meters) to produce an illusion of a sound zooming by your head.

Note that using a combination of high settings for all parameters can result in noise and aliasing becoming audible.

About the Doppler Parameters

Page One

PLAY (Play Trigger)
When triggering the doppler effect manually, press PLAY to trigger. The sound must have started at some time before PLAY is pressed. This is called *memory preload* and is equal to the amount of time it takes the sound to travel from the starting point of the object to the listener. The amount of time required depends on the settings of SPEED and TIME. At maximum speed and time the sound must be started up to 1.3 seconds before PLAY is pressed. At minimum SPEEDs and TIMEs, the time needed to preload the machine's memory is quite short.

If an audio trigger has been set up on page two, pressing PLAY does not start the effect immediately; it arms the effect, and then waits for the appropriate level to start the effect. The memory preload time is built in, so the effect starts shortly after the trigger.

TYPE (Play Type)
TYPE affects the rate of change of the level of the sound. It has two modes--NORMAL and ZOOM. In NORMAL, the level is inversely proportional to the distance from the object to the listener. The object moves in a straight line from one side to the other. In ZOOM, the level is inversely proportional to the distance squared. The sound moves in a parobola, moving rapidly toward the listener and then away.

FG (Fudge Factor)
FG is AMP Q/Pitch Q * 64. The amplitude Q (i.e., the sharpness of the amplitude increase as the object goes by) is set only by DISTANCE Q. When FG is set to 64, the pitch change follows the amplitude change in a theoretically accurate manner. Sometimes it sounds better to have the pitch vary more gradually.

This can be achieved by raising FG. For example, when FG is set to 128, the pitch acts as if the object is twice as far away, while the amplitude remains at the distance set with DISTANCE Q.

SPEED
SPEED sets the total pitch shift that will occur. When SPEED is set to 0, there will be no pitch shift. The pitch shift set with SPEED is quite accurate.

TIME
TIME sets the time between when the device is triggered and when the sound is midway between the two loudspeakers. *The total time of the effect is twice the value set with TIME.* TIME has great impact on perceived speed. Short times and small distances make the object appear to be moving quite fast.

DISTANCE Q
DISTANCE Q sets the sharpness of the effect in both amplitude and frequency. The control displays the distance of closest approach, and the displayed distance depends on the TIME selected. For an accurate emulation of a real event, time should be set first.

Page Two

PLAY (Play Trigger)
Identical to PLAY on page one.

TRIGGER (Trigger Mode)
TRIGGER sets the level of the audio trigger. At the maximum setting, the effect is triggered manually with PLAY or MIDI. When a level has been selected, PLAY arms the effect. It then waits for a signal at the selected level to run. It must be rearmed before running again; the continuous setting has no meaning. When audio level exceeds the trigger level, the 480L waits for the memory preload and then starts the effect.

7

Using the Twin Delay Program

This chapter describes the Twin Delay program located in Bank 9.

Page One						Page Two					
L Dry Level	R Dry Level	L Dly Level	R Dly Level	L Dly Value	R Dly Value	L - L Fdbk	R - R Fdbk	L - L Delay	R - R Delay	L Ch Pan	R Ch Pan

Figure 7.1. Twin Delay Parameters.

General Information

The Twin Delay program located in Bank 9 is a two voice delay line with independently-adjustable feedback, level, and delay time for each voice. The delay sound may be mixed with the input sound. Delay times up to 1.48 seconds in 88 microsecond steps are available at the 44.1 kHz sampling rate. Delay times up to 1.33 seconds in 80 microsecond steps are available at the 48 kHz sampling rate.

There are two additional delay taps for feedback, and the feedback may be panned. These feedback taps are separate from the signal taps. To set up a repeating slap-echo you must adjust both the feedback delay and the signal delay to the same value.

As a basic delay, this program allows the 480L to be used as a disk cutting delay line of exceptional quality. In addition, reverb can be added, or Stereo Adjust (See Chapter 8) can be used to make final level or equalization adjustments at the same time.

Twin Delays is also useful as a simple panning control. To accomplish this, set DRY LEVEL to 0, WET LEVEL to maximum, and WET DELAY to 0. L PAN and R PAN can then be used to set the position of the left and right outputs. The Stereo Adjust program can be used to readjust levels if needed.

About the Twin Delay Parameters

Page One

L DRY and R DRY (L and R Channel Dry Level)
L DRY sets the dry signal level from the left input to the left output. It is not affected by L or R PAN. R DRY sets the dry signal level from the right input to the right output. It is not affected by L or R PAN.

L WET and R WET (L and R Channel Delay Level)
L WET sets the output level of the left delay line. This signal is the input to L PAN. R WET sets the output level of the right delay line. This signal is the input to R PAN.

L DLY and R DLY (L and R Channel Delay Value)
L DLY sets the left channel delay line's delay time. R DLY sets the right channel delay line's delay time.

Page Two

L - L FBK (L Channel to L Channel Feedback)
L- L FBK controls the amount of feedback from a tap in the left channel delay line to the left channel delay line input. The delay of this tap is set by L DLY.

R - R FBK (R Channel to R Channel Feedback)
R - R FBK controls amount of feedback from a tap in the right channel delay line to the right channel delay line input. The delay of this tap is set by R DLY.

L - L DLY (Left Channel to Left Channel Delay)
L - L DLY sets the delay value for the L - L FBK tap. It can be greater than, equal to, or less than the left channel delay value.

R - R DLY (Right Channel to Right Channel Delay)
R - R DLY sets the delay value for the R - R FBK tap. It can be greater than, equal to, or less than the right channel delay value.

L PAN (Left Channel Pan)
L PAN sets the panning of the L WET signal to the left and right outputs.

R PAN (Right Channel Pan)
R PAN sets the panning of the R WET signal to the left and right outputs.

Page Three					Page Four
L-R Fdbk	R - L Fdbk	L - R Delay	R - L Delay	Not Used	Not Used

Page Three

L - R FBK (Left Channel to Right Channel Feedback)
L - R FBK controls the feedback level from a delay tap in the left channel delay line to the right channel delay line input.

R - L FBK (Left Channel to Right Channel Feedback)
R - L FBK controls the feedback level from a delay tap in the right channel delay line to the left channel delay line input.

L - R DLY (Left Channel to Right Channel Delay)
L - R DLY controls the delay time of the left channel to right channel feedback tap.

R - L DLY (Right Channel to Left Channel Delay)
R - L DLY controls the delay time of the right channel to left channel feedback tap.

8

Using the Stereo Adjust Program

This chapter describes the Stereo Adjust program located in Bank 9.

General Information

The Stereo Adjust program permits slight but important adjustments to level and equalization when preparing a compact disc master. It also allows digital fades to true zero at the end of a track. It supplies a stereo digital fader, as well as shelving equalization. The frequencies of the shelving filters can be adjusted. In addition, a SPATIAL EQ control allows adjustment in the digital domain of this important property of recorded sound. SPATIAL EQ (used in conjunction with BAS) increases the stereo width at low frequencies, enhancing the richness, spaciousness, and depth of the recording.

About the Stereo Adjust Parameters

Page One

LVL (Level)
LVL is a stereo level control, with both channels equally attenuated or boosted. From -12 to +12 the slider moves in .25 dB increments. Below -12 it moves in .50 dB increments. Below -60 dB the calibration comes in larger steps, finally dropping to zero output at -72 dB.

FIN (Fine Level)
FIN is identical to the LVL, but has a range of $\pm$3.5 dB the setting of LVL. This allows fine adjustment of level while the mix is proceeding, without fear of over or undershooting the desired setting. If in a mix you want to make a level increase at some point of 4.5 dB, and then drop back to zero, you can set the FIN to the bottom of its range beforehand. Then readjust LVL so that the attenuation is once again zero. Now the FIN control will have a range of 0 to 7 dB of boost.

BAL (Balance)
BAL implements a sine/cosine balance adjustment. Balance is smoothly adjusted over a wide range, with excellent resolution in the critical area around zero. The display indicates the actual channel gains as the control is varied

ROT (Rotate)
ROT is similar to BAL, but it treats stereo information somewhat differently. Any signal panned to the center (mono) will be treated by ROT exactly as it would be treated by BAL. However, if a signal is panned full right and the control is moved toward the left, instead of simply being attenuated (as BAL would do it) the right channel is inverted in phase and added to the left channel. A stereo image appears to rotate when this control is used. Ambient information is preserved, and both channels appear to retain equal loudness.

Page One					
Stereo Level	Fine Level	Bal-ance	Rotate	Bass EQ	Treb EQ

Figure 8.1. Stereo Adjust Parameters.

If stereo material is recorded with a coincident pair of figure-of-eight microphones, moving the ROT slider is exactly equivalent to rotating the microphone pair. Other microphone arrays and multimicrophone setups do not rotate perfectly, but using this control is frequently preferable to simply adjusting balance. The display shows the actual channel gains for a continually panned source.

BAS (Bass EQ)
BAS is a 6 dB/octave shelving EQ control with a range of +6 dB boost and full cut. It moves in .50 dB steps from +6 to -6 dB. The crossover point is adjusted with XOV (on page 2). BAS acts on both stereo channels equally.

TREB (Treble EQ)
TRB is a 6 dB/octave shelving EQ controls with a range of +6 dB boost and full cut. It moves in .50 dB steps from +6 to -6 dB. The crossover point is adjusted with STREB HFC (on page 2). TREB acts on both stereo channels equally.

Page Two

XOV (Bass Crossover)
XOV sets the crossover point for BAS (on page 1)and SPC (on page 2). When BAS is set to full cut, the level is -3 dB at the frequency set with XOV.

HFC (Stereo Treble Crossover)
HFC sets the crossover point for TREB (on page 1). When TREB is set to full cut, the level is -3 dB at the frequency set with HFC.

HFC (Independent Treble Crossover)
HFC sets the crossover point for TREB LEFT and TREB RIGHT (on page 2). When TREB is set to full cut, the level is -3 dB at the frequency set with HFC.

SPC (Spatial EQ)
SPC sets the amount of a crossfeed between channels. The signal first goes through a 6 dB/octave lowpass filter whose frequency is set with XOV.

Page Two						Page Three					
Bass Xover	Streb Xover	Itreb Xover	Spat. EQ	Treb Left	Treb Right	Delay	De-emp	Phase Invert	Auto DC cut	DC L Offset	DC R Offset

When SPC is set positive (above 0) the crossfeed has a negative sign. When SPC is set negative (below 0), the crossfeed has a positive sign. When the control is set to either maximum or minimum, the gain in the crossfeed cicuit is unity.

The result of this control is to change the separation of low frequency stereo signals. When the control is raised low frequencies in the sum (mono) channel are reduced, and low frequencies in the difference (stereo) channel are raised. With the control at maximum, low frequency mono signals are completely removed. This represents an extreme setting which should seldom be needed in practice.

With material which has stereo bass information, or which contains some reverberation, the effect of raising SPC is to increase the sense of spaciousness and depth of the sound. It is particularly useful on material recorded with panpots, or coincident and semi-coincident microphone technique.

When most of the bass in a recording is in the sum (mono) channel, raising SPC may reduce the bass level. This effect can be compensated for by raising the overall bass level with BAS. Since both controls use the same XOV seting, this compensation will be quite accurate as long as SPC is set to less than 3 dB boost.

TBL and TBR (Independent L and R Treble)
These controls allow independent adjustment of right and left treble. They may be used together with the stereo adjustments to create a 12 dB/octave cut or boost. Note that the 3 dB frequencies can be different.

Page Three

DLY (11 usec Correction On/Off)
When this control is on, the left channel is delayed relative to the right by 11 usec. This allows a PCM-F1 tape to be corrected for compact disc.

EMP (De-emphasis On/Off)
When EMP is On, the incoming signal is digitally de-emphasized. This should not be used unless the material has been emphasized in the record process—such as a PCM-F1 tape. *Note that the automatic sensing bit is not turned off at the same time*

When digital de-emphasis is applied, the CD mastering lab must be informed that the tape is not emphasized, and the CD emphasis bit should be manually set to Off. Mastering labs are happy to do this, but they *must* be informed!

INV (Right Channel Phase Invert)
When INV is on the right channel is inverted in phase. This should only be necessary to correct some serious problem in the recording process.

AUTO (Automatic DC Cut)
Enables routines for correcting DC offset from material recorded through analog-to-digital converters that are not properly trimmed for DC.

RESET
RESET disables all DC adjustment.

HOLD
HOLD freezes automatic nulling and enables DCL and DCR for manual individual channel adjustment.

AUTO NULL
AUTO NULL automatically reduces ± 4 bits of DC error to >48dB down for each channel. It maintains a slight positive offset near zero to avoid toggling theMSB D/A converters downstream. DC errors greater than -24dB will not be nulled.

DCL (DC Offset Left) and DCR (DC Offset Right)
Replaces any previously obtained value with the slider value. The display indicates the percent of the correction relative to the -24dB maximum.

9

Using the Pitch Change Program

This chapter describes the Pitch Change program located in Bank 9.

Page One						Page Two					
Stereo /Mono	Play Sync	Pitch Ster./L	Fine Pitch L	Pitch R	Fine Pch R	L Pdly	R Pdly	L Fdbk	R Fdbk		Mix

Page Three						Page Four					
MIDI PitchL		Ref MIDI	Pitch Mirror	Glide Ster/L	Splice Time	MIDI PitchR		Ref MIDI	Pitch Mirror	Glide Right	

Figure 9.1. Twin Delay Parameters.

General Information

The Pitch Change program located in Bank 9 is a stereo or two-channel mono pitch shifter with several useful effects, including predelay, feedback, and glide. These are independently adjustable for each channel.

About the Pitch Change Parameters

Page One

MOD (Stereo/Mono Mode)
MOD selects stereo or mono mode. Move the slider all the way up to select mono, and down to select stereo. In stereo the two channels are linked, pitch shifting by the same amount and splicing at the same time.

SNC (Play Sync)
Normally the channels are in sync, but if PCH, FIN, or GL are moved frequently they can get out of sync. They can be resynchronized by pressing SNC. A small click may be heard when the button is pressed.

PCH (Pitch Interval Stereo/Left)
PCH adjusts the pitch interval of both channels in stereo mode, and the left channel in mono mode. The exact tuning can be altered by the fine pitch control, and the exact pitch shift in intervals and cents is displayed. The fine control must be set to the exact middle of its range if perfect pitch intervals are to be obtained.

FIN (Fine Pitch Stereo/Left)
FIN acts on both channels together in stereo mode, and the left channel in mono mode. It adjusts pitch continuously over a range of a few hundred cents, and is additive to the PCH control. If PCH is set to the middle of its range the FIN control can be used to set very small values of pitch shift, producing a chorusing effect.

PCH (Pitch Interval R)
This control performs the same functions as the other PCH control, except that it is only active in Mono mode, in which it adjusts the right channel.

FIN (Fine Pitch R)
This control performs the same functions as the other FIN control, except that it is only active in Mono mode, in which it adjusts the right channel.

Page Two

PDL (Predelay L)
PDL adjusts the length of a delay line in series with the left input. The range is zero to over 800 ms, with a fine scale available when the button is pushed. Pre-delay also affects the delay of any feedback which is applied. In stereo mode the two predelays must be set to the same value, or the signals will not be in phase.

PDR (Predelay R)
PDR is the same as the PDL, except that it acts upon the right channel.

FBL and FBR (Feedback Left and Right)
FBL and FBR control the amount of feedback from the output of the pitch shift to the input of the predelay line. The control is at zero feedback when centered, and is adjustable from 0 to ±99%. Some very useful efects can be obtained by combining pitch shift, delay, and feedback.

MIX (Wet/Dry Mix)
MIX controls the ratio of processed signal to direct signal from the input to the program. It allows a chorus effect to be set up with at least two voices with delay and feedback, and then have the effect mixed back into the original signal.

Page Three

MIDI (MIDI Pitch Event Left)
MIDI Pitch Event provides a fast link between a MIDI Note Event and interval pitch shifting. By establishing a patch with NOTE EVENT as source and PITCH EVENT as destination, the amount of pitch shift can track incoming notes in semitone intervals. Pressing the KEY will tell you the curent pitch shift.

REF (Reference MIDI Note Left)
REF sets the MIDI Note Value for no pitch change, NULL + 0C, when under MIDI control.

MIR (Pitch Mirror Left)
MIR selects the direction of MIDI pitch control. When MIR is ON, pitch is shifted down for increasing note values.

GLL (Glide Stereo/Left)
GLL affects both channels in stereo mode, and the left channel in mono mode. It changes the pitch continuously over a range of ± one octave. This allows a glissando to be performed, either manually with the LARC, or via MIDI. The full resolution of the pitch shift is not available on this control, although vernier (fine tuning) is available by pressing the button under the slider twice.

SPL (Splice Length)
SPL sets the amount of time the splice takes to complete. It is only active at moderate values of pitch shift. Very short splices produce less of a metallic or combing sound in the pitch shift, but can sometimes be audible as a click or a glitch. Longer splices are sometimes less obvious, but can affect the timbre of the sound. High values of pitch shift require short splices, so SPL is deactivated if the pitch shift selected is quite high. The default value of 16 gives good results in most applications.

Page Four

The following controls work on the right channel when the pitch shifter is in momo mode.

MIDI (MIDI Pitch Event Right)
REF (Reference MIDI Note Right)
MIR (Pitch Mirror Right)
GLR (Glide Right)

Pitch Chorus - Program 2
Pitch Chorus is a dual mono program, which means that both the left and right channels can be set for different amounts of pitch shift. The left channel is preset for -3 cents. The right channel is preset for +6 cents. This produces a medium rolling chorus effect with a lush characteristic.

1% Up 1% Down - Program 3
This is another dual mono program. The pitches are set 1% up and 1% down, creating a heavily processed sound. 39.27 ms and 32.69 ms of delay are used on the pitches, reinforcing the overall effect. This program is ideal for guitar and vocals.

Barber Pole - Program 4
This is a true stereo program. Both sides are set for 3 cents of downward pitch shift. No additional delay is used, but -41% feedback is assigned to the left and +41% is assigned to the right. This helps give the smooth, slow downward resonance characteristic of this program. Use the left/stereo Pitch slider for additional amounts of stereo pitch shifting.

Half Steps - Program 5
This stereo program uses 600 ms of delay and 44% feedback on both left and right channels, routed back to a Minor 2nd downward pitch shift. This program is strictly for effects use — try changing the delay settings for even more outrageous effects.

Stair Case - Program 6
This is a dual mono variation of Half Steps with shorter delays and larger intervals.

		Page One						Page Two					
Program #	Name	Stereo/ Mono	Play Sync	Pitch Stereo/L	Fine Pitch L	Pitch R	Fine Pitch R	L Pdly	R Pdly	L Fdbk	R Fdbk	Not Used	Mix
1	Pitch Change	Stereo		+null	+0c	+null	+0c	0	0	0	0		
2	Pitch Chorus	Mono	N/A	-3c	-3c	+6c	+6c	6.48 ms	9.81 ms	+23%	+13%		All Fx
3	1% Up 1% Dwn	Mono	N/A	+10c	+10c	-10c	-10c	39.27 ms	32.69 ms	+10%	+10%		All Fx
4	Barber Pole	Stereo		-3c	-3c	N/A	N/A	0.0 ms	0.0 ms	-41%	+41%		All FX
5	Half Steps	Stereo		-min 2nd	-0c	-min 2nd	-0c	600.03ms	600.03ms	+44%	+44%		All Fx
6	Stair Case	Mono	N/A	+min 3rd	+47c	+maj 3rd	+2c	29.03	30.95	+55%	-60%		All Fx

Figure 9.2. Bank 9 - Pitch Shift Programs and Parameters.

10

Using the Parametric EQ Programs

This chapter describes the Pitch Change program located in Bank 9.

Page One					
Stereo Lvl LEV	Fine Lvl FIN	LDB Bal R BAL	Stereo Link LNK	Fine Freq L FIN	Fine Freq R Fin
Page Two					
FR - 1 L	Q	LEV	FR - 1 R	Q	LEV
Page Three					
FR - 2 L	Q	LEV	FR - 2 R	Q	LEV

Figure 10.1 Stereo Parametric EQ Parameters

General Information

Two Parametric EQ programs are located in Bank 9. The Stereo Parametric EQ program provides a two band stereo, or dual mono, parametric equalizer. The Mono Parametric EQ program provides a 4-Band monaural parametric equalizer. Both programs provide frequency adjustment on each band between 30 Hz and 17 kHz, boost/cut + - 12 dB, and Q adjustable between Shelf and 7. In addition, the low frequency filters provide a Notch (Q=32) with a boost/cut of 36 dB. Coarse and fine level control, panning (stereo only) and fine frequency adjustment are also provided. Both programs operate entirely in the digital domain.

About the Stereo Parametric EQ Parameters

Page One

LVL (Stereo Level)
LVL is a stereo level control, with both channels equally attenuated or boosted. From -12 to +12 the slider moves in .25 dB increments. Below -12 it moves in .50dB increments. Below -60 dB calibration is in larger steps, finally dropping to zero output at -72 dB.

FIN (Fine Level)
FIN is identical to LVL, but has a range of $\pm$3.5 dB in reference to the LVL setting. This allows fine adjustment of the level while the mix is proceeding without fear of over or undershooting the desired setting.

Note: LVL and FIN provide proper scaling for the filters. For example, if you set a 12 dB boost at some frequency, it is possible to exceed the dynamic range of the 480 if a high level signal comes in at the center frequency you have chosen to boost. You can reduce the drive to the filter with the level controls to prevent overload.

BAL (Balance)
BAL implements a sine/cosine balance adjustment. Balance is smoothly adjusted over a wide range, with excellent resolution in the critical area around zero. The display indicates the actual channel gain as the control is varied.

LNK (Stereo Link)
LNK synchronizes the left and right channel settings of the level, Q, and frequency sliders. When LNK is on, only the left channel sliders on Page Two and Page Three are active. When LNK is turned off, independent control of left and right channel settings is established. This is particularly useful in mastering applications.

FIN
FIN FREQ L and FINE FREQ R provide fine frequency adjustment of the Page Two filters by adding a small amount to their respective frequency settings. (They set the frequency *only* for the Page Two filters.) These settings are particularly useful for fine tuning of notch filters.

Page Two

FR-1 L and FR-1R
These controls allow frequency settings to be adjusted independently on either the left or the right channel. Frequency settings fall between 30 Hz and 17 kHz.

Q
Q adjusts the amount of bandwidth affected by the level control. The Q is determined by dividing the center frequency by the bandwidth to be affected (in Hz.). For example, if a frequency of 1000 Hz is selected, and the bandwidth to be boosted or cut is 500 Hz, the Q = 2. The Page Two filters provide a Q adjustment from shelf to Notch (Q =16 or Q =32). The Shelf (SL) on Page Two

Page One					
Level	Fine Lvl		Fine Fr - 1	Fine Fr	2
Page Two					
FR - LF	Q	LEV	FR - LM	Q	LEV
Page Three					
FR - HM	Q	LEV	FR - HF	Q	LEV

Figure 10.2 Mono Parametric EQ Parameters

filters is a low frequency shelf. This means that with a frequency of 500 Hz, a level of +12 dB, and Shelf (SL) selected, frequencies below approximately 300 Hz will be boosted by 12 dB, and 500 Hz will be boosted approximately +9 dB. The Notch feature is found only on the Page Two filters and is optimized for low frequencies.

LEV
LEV provides level boost or cut at the defined frequency and Q settings. LVL is adjustable between -12 to +12 dB when the Q setting falls between shelf and Q = 7. When a Notch Q is selected (Q=16 or Q=32), LVL is adjustable between -36 and +36 dB. This only applies to the Page Two filters.

Page Three

FR-1 L and FR-1R
These controls allow frequency settings to be independently adjusted on either the left or the right channel. Frequency settings fall between 30 Hz and 17 kHz.

Q
Q adjusts the amount of bandwidth affected by the level control. The Q is determined by dividing the center frequency in Hz by the bandwidth to be affected in Hz. For example, if a frequency of 1000 Hz is selected, and the bandwidth to be boosted or cut is 500 Hz, the Q = 2. The Page Three filters provide a Q adjustment from shelf to Q = 7. The filters on Page Three have a high frequency shelf that boosts all frequencies above the set frequency. This is identified by "SH" in the Q display.

LEV
LEV provides level boost or cut at the defined frequency and Q settings. LVL is adjustable between -12 to +12 db.

About the Mono Parametric EQ Parameters

Page One

LVL (Level)
LVL is a mono level control, From -12 to +12 the slider moves in .25 dB increments. Below -12 it moves in .50dB increments. Below -60 dB calibration is in larger steps, finally dropping to zero output at -72 dB.

FIN (Fine Level)
FIN is identical to LVL, but has a range of ± 3.5 dB in reference to the LVL setting. This allows fine adjustment of the level while the mix is proceeding without fear of over or undershooting the desired setting.

Note: LVL and FIN provide proper scaling for the filters. For example, if you set a 12 dB boost at some frequency, it is possible to exceed the dynamic range of the 480 if a high level signal comes in at the center frequency you have chosen to boost. You can reduce the drive to the filter with the level controls to prevent overload.

FIN
FIN FREQ 1 and 2 provide fine frequency adjustment of the Page Two filters by adding a small amount to their respective frequency settings. (They set the frequency *only* for the Page Two filters.) These settings are particularly useful for fine tuning of notch filters.

Page Two

FR-1 LF

This control sets the frequency to be adjusted for the Low Frequency band. Frequency settings fall between 30 Hz and 17 kHz.

Q

Q adjusts the amount of bandwidth affected by the level control. The Q is determined by dividing the center frequency by the bandwidth to be affected (in Hz.). For example, if a frequency of 1000 Hz is selected, and the bandwidth to be boosted or cut is 500 Hz, the Q = 2. The Page Two filters provide a Q adjustment from shelf to Notch (Q=16 or Q=32) The Shelf (SL) on Page Two filters is a low frequency shelf. This means that with a frequency of 500 Hz, a level of +12 dB and Shelf (SL) selected, frequencies below approximately 300 Hz will be boosted by 12 dB and 500 Hz will be boosted approximately +9 dB. The Notch feature is found only on the Page Two filters and is optimized for low frequencies.

LEV

LEV provides level boost or cut at the defined frequency and Q settings. LVL is adjustable between -12 to +12 db when the Q setting falls between shelf and Q = 7. When a Notch Q is selected (Q=16 or Q=32), LVL is adjustable between -36 and +36 dB. This only applies to the Page Two filters.

FR-LM

This control sets the frequency to be adjusted for the Low-Mid Frequency band. Frequency settings fall between 30 Hz and 17 kHz.

Q

Same as above

LEV

Same as above

Page Three

FR-HM

This control sets the frequency to be adjusted for the High-Mid Frequency band. Frequency settings fall between 30 Hz and 17 kHz.

Q

Q adjusts the amount of bandwidth affected by the level control. The Q is determined by dividing the center frequency by the bandwidth to be affected (in Hz.). For example, if a frequency of 1000 Hz is selected, and the bandwidth to be boosted or cut is 500 Hz, the Q = 2. The Page Three filters provide a Q adjustment from shelf to Q = 7. The filters on Page Three have a high frequency shelf that boosts all frequencies above the set frequency. This is identified by "SH" in the Q display.

LEV

LEV provides level boost or cut at the defined frequency and Q settings. LVL is adjustable between -12 to +12 db.

FR-HI

This control sets the frequency to be adjusted for the High Frequency band. Frequency settings fall between 30 Hz and 17 kHz.

Q

Same as above

LEV

Same as above

11

MIDI and the 480L

This chapter describes the use
of MIDI with the 480L.

Introduction

Most uses of MIDI with the 480L fall into one of four basic categories:

• Automatic selection of a 480L program or register when a program is selected on any other MIDI device

• Real time control of 480L parameters from a remote keyboard or controller, using the 480L's Dynamic MIDI®

• Real time triggering of LARC events from a remote keyboard or controller, using the 480L's Dynamic MIDI®

• Automatic program selection and parameter control from a MIDI digital sequence recorder

We'll discuss each of these applications in this chapter, but first let's cover some typical MIDI installations.

MIDI Connections

All MIDI connections described in this chapter use the MIDI IN, THRU, and OUT connectors located on the rear panel of the 480L. As with any MIDI connection, use only standard MIDI cables and keep them as short as possible to avoid possible data errors. 15 meters is generally accepted as the longest length that should be used if absolute data integrity is important.

Basic MIDI Setup

1. Press CTRL to enter the control mode.

2. Press PAGE, 5 to go to page 5.

3. Use slider six to set the MIDI Channel. Many users assign instruments to lower channels, and then jump to the higher channels (14, 15, and 16) for MIDI-controlled effects like the 480L.

4. Use slider five to set the program change mode to FIXED.

5. If you will be using the 480L in Mono Split, Stereo Split, or Cascade modes, press MACH to switch to Machine B. Use slider six to set the MIDI Channel, and slider five to set the program change mode to FIXED for Machine B.

6. In some applications you will want to set Machine A and Machine B to the same MIDI Channel. Generally, it is best to use separate channels.

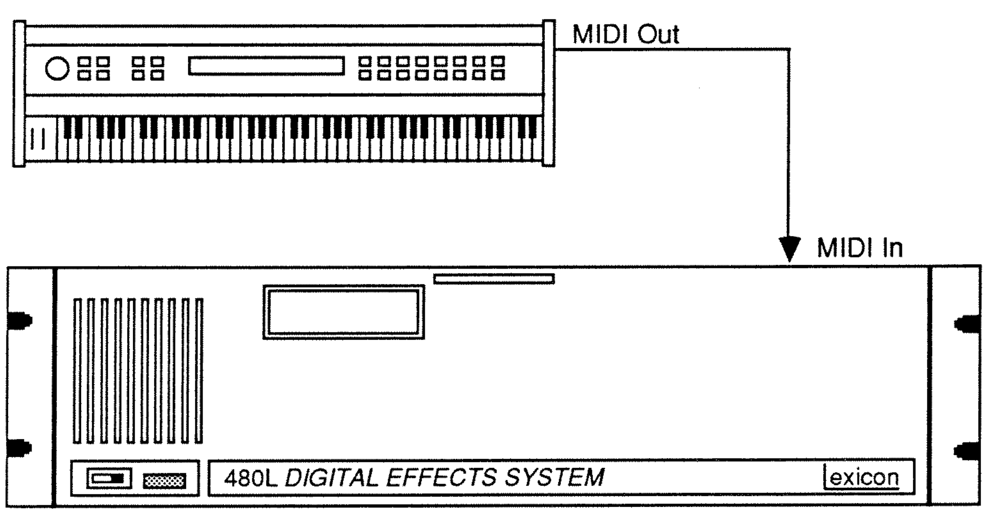


Figure 10.1. Connecting a Keyboard and the 480L.

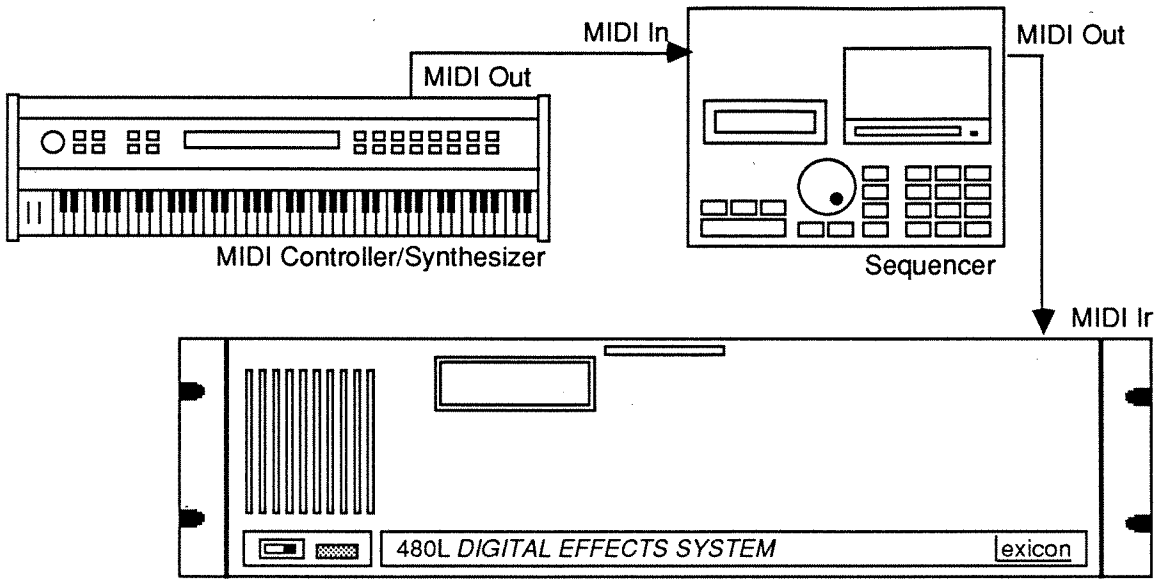


Figure 10.2. Connecting a Keyboard, Sequencer, and the 480L.

Applications

Using a MIDI Keyboard to Control the 480L

You can select 480L registers and control up to ten parameters and/or events simultaneously using the controllers and switches found on MIDI-equipped instruments. Nearly any MIDI-equipped keyboard or synthesizer can be used to remotely select registers on the 480L.

Remote Register Selection

1. Connect the 480L and the MIDI controller as shown in Figure 10.1 or 10.2.

2. Enter the control mode and set slider 5 on page 5 (PGM CHANGE MODE) to FIXED.

3. Exit the control mode.

Select some presets on the synthesizer. The 480L should select and load a different register each time you select a preset on the synthesizer. MIDI program changes from 0 to 49 will load the 480L *internal* registers (0.0 to 4.9). MIDI program changes from 50 to 99 will load 480L *cartridge* registers (5.0 to 9.9)

Note: Some synthesizers and controllers start numbering their presets at one instead of zero. Selecting preset 1 sends MIDI program change 0. A few synthesizers organize their presets into banks of eight, numbered from 1.1 to 1.8, 2.1 to 2.8, and so forth. If you experiment a bit, the relationship between the MIDI controllers preset numbers and the MIDI program changes it actually sends will become clear.

Using Corresponding Registers

You will quickly discover that a fixed relationship between MIDI program changes and the 480L register numbers is not very convenient. Changing the presets in a synth, or the register contents in the 480L is not easy, since you have to arrange everything so that MIDI program changes have the desired effect.

To solve this problem, the 480L has a *corresponding register table* which lets you assign any register or program to any MIDI program change number. Dealing with MIDI program changes becomes much easier, since you can completely rearrange the relationship of MIDI program change numbers to 480L registers and programs in minutes.

The corresponding register table also lets you make more economical use of registers, since several MIDI program change numbers can be assigned to a single 480L register. Finally, it allows you to load any of the preset programs with MIDI program changes--something which is not possible in the fixed mode.

The corresponding register table is found on page 5 of the control mode. To use it:

1. Enter the control mode and go to page 5.

2. Slider three (PGM) selects the MIDI program change number, and slider four (TBL) selects the 480L register or program.

3. Use PGM to select the MIDI program change number you wish to assign to a register or program.

4. Use TBL to select the register or program to assign to the MIDI program change number. Note that the first selection for TBL is IGNORE. This allows the 480L to ignore specific incoming MIDI program change numbers.

5. Repeat this process until all the MIDI program change numbers that you will be using have been assigned to registers or programs.

> **Important!** The 480L actually has two corresponding register tables---one for machine A and one for machine B. Use MACH to toggle between them.

Using Dynamic MIDI®

Some extremely interesting effects can be created when one or more parameters are controlled remotely in real time. Many of the controllers found on a MIDI keyboard or controller (pitch benders, mod wheels, breath controllers, sliders, and switches) can be used to control 480L parameters remotely in real time via MIDI. MIDI events like last note played, last velocity, and aftertouch can also be used.

To use Dynamic MIDI®, you "patch" a parameter to a MIDI controller or event, using the patch parameters found on page 3 of the control mode. There are ten patches for each register, allowing you to control up to 10 parameters remotely at the same time. Because each register has its own unique set of patches, each register can respond to MIDI in a different way.

To get an idea of what patching can do for you try the following example:

1. Connect the 480L as shown in Figure 9.1. Set the 480L and the keyboard to the same MIDI channel.

2. Load one of the sampler programs in Bank 6 or 7.

3. Press CTRL to enter the control mode.

4. Press PAGE, 4 to go to page 4 of the control mode.

5. Use SEL to select the number of the patch you will create.

6. Use SRC to select the MIDI controller or event which will be used to control the 480L. For this example, set SRC to MOD WHEEL. Notice that as the slider is moved up past PATCH OFF, two additional controls become available (SCL and PRM). These controls are discussed below.

7. Use DST to select the 480L parameter which will be controlled via MIDI. For this example set DST to FWD TIME.

8. Use SCL to set the scaling for the controller. SCL sets the range of effectiveness for the MIDI controller. Scaling can be set from -200 to +200%. When SCL is set to +100%, the full range of the MIDI controller will apply to the 480L parameter. Setting a *negative* value of scaling will cause the 480L to reduce the setting of a parameter as the controller increases.

9. The parameter you chose in step 7 appears when you press PRM. Move the mod wheel, and the parameter value should change. Use the PRM slider to set the parameter value to the starting point you wish to use for MIDI control.

Once you have created a patch, be sure to store it in a register. Otherwise it will be lost the next time you load a program or register.

> **Important!** Be extremely careful when creating patches while a MIDI keyboard is connected. If you accidentally send a MIDI program change before saving the patches in a register, they will be lost. To avoid this possibility, you may want to set PGM CHG on page 5 of the control mode to IGNORE.

In the sampler and doppler programs only, moving SRC to the top of its range displays a NOTE EVENT option. When SRC is set to NOTE EVENT, MIDI Note On events can be patched to control sampler and doppler program events like RECORD, CHECK, PLAY, etc.

A single note can be patched to trigger an event, or a range of notes can control a single event. The following procedure assumes you have made MIDI connections and set the 480L's MIDI channels.

To patch a MIDI note event to a sampler event:

1. Press CTRL to enter the Control Mode. Go to page 4.

2. Use SEL to select the patch to create (only the first four patches can be used for events).

3. Push the SRC slider all the way up to NOTE EVENT.

4. Use DST to select the event you will control.

5. If you want to use a single key to trigger an event, set LOW NOTE and HIGH NOTE to the same value.

6. If you want a range of keys to trigger an event, use LOW NOTE to set the low end of the range, and HIGH NOTE to set the high end of the range.

7. After setting up your patches, be sure to save the new settings in a register.

Creating Custom Master Controls

Control over a single parameter at a time is useful, but things really begin to get exciting as you experiment with controlling several parameters simultaneously from a single MIDI controller. In effect, you can create a custom master control for a unique set of parameters. Using this custom master in real time can produce stunning effects never heard before.

The ability to choose different settings of SCALING for two or more parameters controlled by the same controller also raises some interesting possibilities. Don't forget that using negative SCALING for one parameter and positive SCALING for another will cause the first parameter to decrease while the other increases.

A word of caution: not all parameters respond well to real-time control. Due to the current limitations of digital technology, it is simply impossible to alter certain parameters in real time without audible artifacts. This is the case whether you are controlling the parameter remotely via MIDI, or from the unit's front panel. We considered locking out these parameters, but after careful thought we included them, since what is not acceptable in one application may not be a problem in another.

Some Notes On Controllers

Many MIDI synths and keyboards have a very limited number of controllers. Sometimes the pitch and modulation wheels or levers are the only options available for remotely controlling the 480L. However, you may not wish to produce modulation or pitch bending at the same time that you are controlling the 480L. All is not lost. Most synths allow you to shut off the effect of these controllers. So, for example, moving the pitch bender doesn't actually bend pitch.

This is where things get interesting. Usually, when the synth is set to ignore its controllers, *controller data is still sent out the MIDI port.* We have found this to be the case with a variety of different brands and models of synthesizers. Just set the synth to ignore its mod wheel and pitch bender, and then use them to control the 480L. As long as you don't wish to control the 480L *and* bend pitch or add modulation at the same time, these controllers can easily do double duty. Synthesizers which memorize ranges for the mod wheels and pitch benders for each preset program are the best choices for use with the 480L.By using the corresponding register table, you could have some programs use the pitch bender to bend pitch, and not affect the 480L, and other programs could control the 480L, but not bend pitch.

The Yamaha DX7 II D and DX5, the Oberheim Matrix 6, and the Korg DW-8000 are just a few examples of synthesizers that can be used in this manner.

If you plan to do serious work with MIDI, consider purchasing a keyboard (such as the Yamaha KX76 and KX88, or the Kurzweil MIDIBoard) which is specifically designed to function as a MIDI system controller. These keyboards have several programmable controllers, allowing you to control the 480L without sacrificing control over other equipment in your MIDI system.

Controlling a 480L from a Sequencer

Since you can control the 480L in real time with MIDI controllers, it stands to reason that you could record your manipulation of those controllers with a MIDI sequencer, and then repeat the performance automatically. In fact, this works perfectly, and this capability gives the 480L a fairly sophisticated level of automation. If your sequencer can sync to tape, you can even use it to provide automated effects for non-MIDI instruments. For example, you can control the 480L from a keyboard, recording commands onto a sequencer, but the audio the 480L processes might be percussion, guitar, vocals, or even the whole mix. If you perform live with sequencers, there is no reason why you can't sequence several effects processors along with everything else.

When working with sequencers, it is always a good idea to put the 480L on a different MIDI channel than other devices in the system. This avoids the possibility of the 480L responding to commands that aren't really intended for it.

Keep in mind that as of this writing, no sequencers offer "chase" mode for MIDI controllers. This may change in the future, but for now it means that if you attempt to punch in to the middle of a sequence, the 480L's parameters will be in an unknown condition. To avoid problems, always start the sequence at the very beginning when overdubbing or adding new parts. Also, it is a good idea to use the first measure of the sequence to reset all the controllers to 0.

12

Using Digital I/O

This chapter describes the 480L's digital audio I/O capabilities.

The 480L and Digital Audio I/O

In addition to its analog inputs and outputs, the 480L is equipped with a PCM digital I/O connector. One application for digital I/O is processing material from a PCM 1610 or compatible unit. The WET/DRY MIX control in the reverb and effects programs makes it possible for the 480L to add signal processing to a stereo mix; without ever leaving the digital domain.

Another application for digital I/O is to cascade two or more 480Ls together to create complex effects, again, without leaving the digital domain. In this application the first 480L in the chain supplies word clock for the other units. Set the first unit for internal 48 kHz mode, and the second and subsequent units for external 48 kHz mode.

Drive levels and data format are compatible with the PCM 1610. Sync, preemphasis and flag bits are derived from the input bit stream. The 480L may also be interfaced with the Sony 3324 digital multitrack recorder. The 3324 uses a balanced 1610 format, but this is easily accomodated by grounding the low side of each signal line at the 480L interface connector.

Input and Output Configuration

The digital audio outputs can be used at the same time as the analog outputs, and they are always available at the Digital Audio I/O connector on the rear panel. The digital audio outputs receive the same material as the Main Outputs. The Aux Outputs are not available at the Digital I/O connector.

Pin	Assignment
1	Chassis Ground
2	Right Channel Output
3	Left Channel Input
4	Word Clock Input
5	Digital Ground
6	Word Clock Output
7	Left Channel Output
8	Right Channel Input
9	Digital Ground

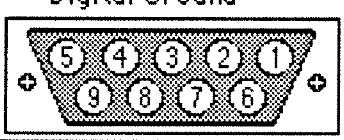


Figure 11.1. Digital I/O Connector Pinout.

The digital audio inputs cannot be used at the same time as the analog inputs. When the digital inputs are in use, the analog inputs are disabled.

How to Select Sampling Rate

The 480L can generate its own word clock, or it can be slaved to 48, 44.1, or 44.056 kHz external word clocks. In most digital I/O applications, the external word clock (slave) mode should be used.

1. Press CTRL to enter the control mode.

2. Use the SMP slider on page one to select sampling rate.

3. Use the CLK slider on page one to select internal or external clock

4. Use the INP slider on page one to choose between analog or digital input.

5. If external clock is selected, but external word clock is not present at the digital I/O connector, the 480L will continue to use its internal word clock.

To determine if the 480L is correctly receiving an external word clock, move the STATUS slider (slider one, page one) to display External Word Clock Status.

Important! Do not power up with external word clock present at the 480L's digital I/O port. Doing so may prevent the unit from finishing its power up sequence.

If you encounter problems with distortion or loss of signal, the PLL circuitry may have become misaligned in shipment. See Chapter 12, *Solving Problems* for details.

Note: A digital audio I/O interface box which can greatly simplify interfacing the 480L is available as an option. Contact your Lexicon Representative for details.

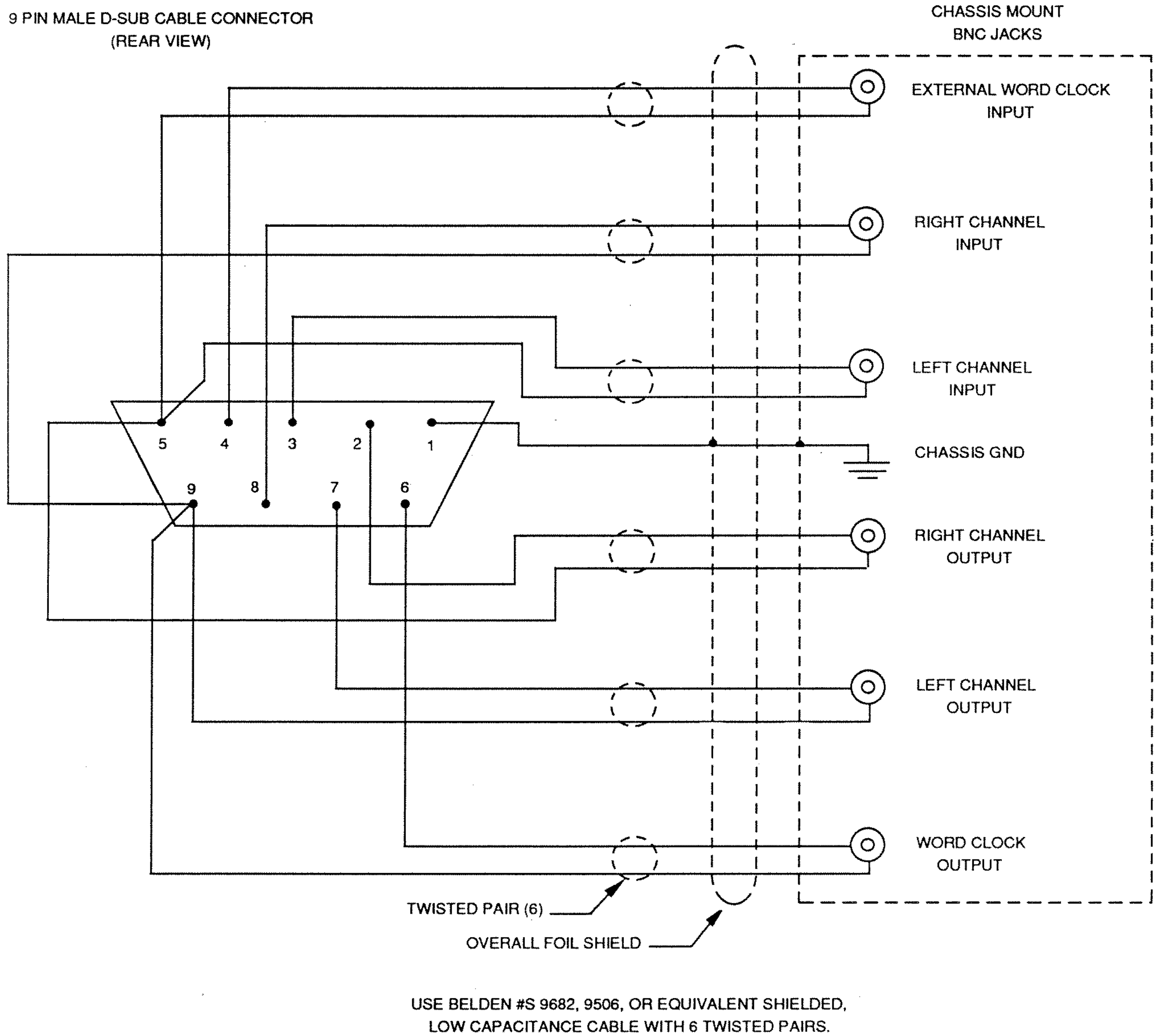


Figure 11.2. Suggested Interface for Digital I/O Connector.

13

Solving Problems

This chapter describes some common problems and their solutions.

Solving Problems

When I try to use a cartridge, the 480L tells me that the cartridge is not formatted.

Before a cartridge is used for the first time, it must be formatted. While this is generally done at the Lexicon factory or by your dealer, it is possible that you have obtained an unformatted cartridge.

To format a cartridge:

1. Press CTRL to enter the control mode.

2. Press PAGE, 2 to go to page 2.

3. Use the FUN slider to select the format function.

4. Hold down STO and press REG to format the cartridge.

> **Important!** A cartridge cannot be formatted if the register protect function is activated, or if the cartridge's memory protect switch is ON.

My 480L was just upgraded to a new software version, and my cartridges no longer work.

Some (but not all) software improvements may be incompatible with cartridges formatted with older software. To make a cartridge usable again, it must be reformatted as described above. This will erase the old contents of the cartridge.

When connected to the effects loop on my console, turning up the console's effects send and returns just makes the dry signal louder--I don't hear any effects from the 480L.

Assuming that your system is wired correctly, the problem is probably that the Wet/Dry Mix control on the registers you are using has been set to 100% dry. Set it to 100% wet and try again.

LARC displays "Mainframe Link Failed" message.

When you first power up, all four indicators on the Host card should light up momentarily, and then go off. If they don't, contact Lexicon Customer Service. If ROMs have been changed recently, make sure all pins are inserted in their sockets correctly. Make sure that Host board is correctly seated in its socket. Make sure the LARC cable is correctly connected. Check line voltage. Inadequate line voltage will prevent the 480L from powering up normally. Finally, check to see if a LARC port has been accidentally connected to the automation port. Connection of a LARC port to the automation port will blow an internal power supply fuse and shut down the 480L.

My 480L loses register contents.

The most likely cause is dead batteries on the Host card. Contact Lexicon Customer Service for information. Another possibility is a blown fuse. The LARC fuse is accessed by removing all of the cards. There are two fuses on the backplane. The lower fuse is the automation fuse and the upper fuse is the LARC fuse. The automation fuse is currently unused, and can be used as a spare LARC fuse. The power supply fuses are inside the unit and can only be accessed by removing the top cover. The +5V power supply fuse is on the left side of unit, and the +15V power supply fuses are near the rear.

My 480L cannot run more than one program at a time. Whenever I try to control Machine B, the message "Not enough HSPs" appears.

There are four possibilities. Your machine is missing one HSP board (two are required to run two programs simultaneously); one of the two HSP boards is not seated correctly; or one of the two HSP boards is malfunctioning. Finally, you may be attempting to run a program (such as the stereo sampler) which can only run when the 480L is in the SINGLE configuration.

I tried to use Digital I/O, and it is extremely noisy and distorted. Sometimes it doesn't work at all.

In the course of shipment the phase locked loop (PLL) may have become misaligned. With external word clock connected and selected, use a small insulated screwdriver to adjust the trimmer capacitor on the far right hand side of the Host Card until undistorted audio passes. Find the extremes where the PLL goes out of lock, and then set the pot for the center of this range.

If this fails to correct the problem, but the status slider indicates that external word clock is present, you may have excessive noise in the digital interface. This problem should be referred to a qualified service technician for diagnosis and correction.

LARC Diagnostic Programs

To enter the LARC diagnostic test mode, after the 480L has powered up and resumed normal operation, press PAGE and, while holding it down, press PROG. To scroll through the menu, press PAGE; to load a displayed program, press PROG.

The following table shows how the diagnostic program is organized, how it is loaded and how each of the programs function.

To enter any diagnostic program, press PROG. To exit, press PROG again.

EXIT	Returns to normal operation
SLIDER	Tests slider action through all positions; each slider should pass without interruption through 256 positions (0 to 255)
BUTTON	Tests button functionality; position of last button pushed and last button released is displayed
DISPLAY	Lights all LED's ; pressing PAGE steps through three displays
TAPEOUT	Does not affect the 480L
DROPOUT	Does not affect the 480L
SERIAL	Transmits a series of bytes to the mainframe and displays returned results
VOLTAGE	Displays LARC power supply voltage - should be stable between 4.8 and 5.2 (048-052). Low voltage could indi cate excessive cable power drop and need for remote power pack
MAINFRAME	Returns to normal operation

14

Specifications

This chapter contains the specifications for the 480L.

Specifications

The following specifications are subject to change without notice.

Audio

Audio Inputs (Two)

Levels	+6 to +28 dBm; electronically balanced +6 to +28 dBm; unbalanced
Impedance	30 kilohms in parallel with 100 pF
Common Mode Rejection Ratio	>40 dB, 20 Hz to 20 kHz
Connectors	Female XLR
Transformer Option	User-installable; Jensen JE-11P-1

Audio Outputs (Four)

Levels	+6 to +24 dBm transformerless balanced (600 ohms) +6 to +20 dBm unbalanced (600 ohms) Minimum load impedance 150 ohms
Impedance	33 ohms
Common Mode Rejection Ratio	>35 dB, 20 Hz to 20 kHz
Connectors	Male XLR
Transformer Option	User-installable; Jensen JE-123-SLPC

Frequency Response*	20 Hz to 20 kHz, +0.5 dB, -1 dB
Dynamic Range*	98 dB typical over temp. range, 22.4 kHz unweighted noise bandwidth
Total Harmonic Distortion and Noise	<0.015% @ 1 kHz limit level (+18 dBm unity gain) <0.05% 20 Hz to 20 kHz @ 20 dB below limit level
IM Distortion	<0.05% SMPTE IM @ limit level
Channel Separation	>75 dB @ 1 kHz or >70 dB, 20 Hz to 20 kHz
Encoding	18 bit equivalent linear PCM
Sampling Rate	48.0 kHz/44.1 kHz – selectable

*These specifications are for 48 kHz sampling rate setting.

LARC (Lexicon Alphanumeric Remote Control)

Controls	Four mode-select buttons (BANK, PROG, VAR, REG) used with ten numeric select buttons (1 to 0); a page select button (PAGE); a control program key (CTRL); a machine-select key (MACH); two auxiliary control buttons (MUTE, STO); six sliders for smooth control of up to 128 parameters per program with associated display-select buttons
Display	Two lines of 12 alphanumeric LEDs for interactive display; additional line of 24 alphanumeric LEDs (six groups of four for each slider); dual 16-position LED headroom indicator (calibrated -24 to +12 dBm with overload warning)
Connector Type	DE9
Cable	50-ft extra-flexible cable; cables can be linked
Operating Distance	Up to 100 feet when powered from mainframe; up to 1000 feet possible with optional remote power source for LARC

Interface

Digital Audio Interface

Interface	PCM 1610-compatible digital I/O; 18-bit word length capability; slaveable to 48 kHz, 44.1 kHz, or 44.056 kHz external word clock
Connector Type	Female DE9
LARC Connector	Female DE9 (2) -- Dual LARC control
Mainframe Controls and Indicators	Power switch and indicator light; Left and Right input level controls, four output level controls; four LEDs for internal DC power supplies
Automation Port	Female DE9 -- for future expansion
MIDI Interface	In, Thru, Out (Standard 5-pin female DIN)

Power Requirements

Mainframe

Nominal	100, 120, 220, 240 Vac (+5, -10%) Switch-selectable; 50-60 Hz, 180 W maximum, 70 W typical
Protection	All secondaries fused; voltage transient suppression; overvoltage and short circuit protection on logic supply
Mains Fuse	100/120 Vac: 3AG 3 A SLO-BLO 220/240 Vac: 5x20 mm 1.6 A SLO-BLO; dual-fused
Connector	Standard 3-pin IEC power connector with rear-panel accessible mains fuse and voltage selector
LARC	10 to 24 Vdc or 10 to 18 Vac, 6.25 W; Normally powered by 480L mainframe; miniature jack accepts optional remote power supply (for operation at distances greater than 100 feet from mainframe)

Miscellaneous

Nonvolatile Memory Cartridge	CMOS static RAM with built-in lithium battery provides storage for registers; write-protect switch prevents accidental erasure of contents
Serviceability	Each major assembly is modular and can be replaced in the field; hinged front panel allows access to plug-in boards, fan filter, and LARC fuse
Diagnostic Programs	Control and display with LARC
Muting	Audio outputs are muted during power-up, power-down, power failure, or power supply failure
RFI Shielding	Ac power connector, audio connectors, and LARC cables are RFI-shielded; unit complies with FCC Class A computer equipment requirements

Environment

Operating Range	0 to 40°C (32 to 104°F)
Max. Storage	-30 to 70°C (-22 to 158°F)
Humidity	95% maximum without condensation
Cooling	Filtered forced air with ultra-quiet fan; filter is removable for cleaning

Dimensions

Mainframe	Standard 19" rack mount 19"w x 5.25"h x 14.5"d(483 x 133 x 368 mm)
	LARC5.9"w x 9.5"h x 3.2"d (150 x 242 x 82 mm)

Weight

Mainframe	24 lbs (10.89 kg)
LARC	1.9 lbs (0.9 kg)

A

Voltage Changeover and Optional Transformers

This chapter describes voltage changeover and installation of optional transformers.

Voltage Changeover

Voltage changeover is a fast and easy process:

1. Remove the power cable from the 480L.

2. Insert a small flat-bladed screwdriver or an IC puller into the slot next to the fuseholder cartridge, which is located just to the right of the power connector. Pry the cartridge out so that it drops out of the chassis. Set the fuseholder cartridge aside.

3. The voltage changeover board is mounted vertically in a small compartment which is normally covered by the fuseholder cartridge (which you removed in step 2). Remove the board with a pair of needle-nosed pliers or tweezers.

4. The four sides of the board are marked with the four voltages at which the 480L can be operated (100, 120, 200, 240). Slide the voltage changeover pin around until it fits in the notch opposite the side marked with the operating voltage you require.

5. With the pin facing out, replace the board in the chassis. Press it until it snaps into place or fits firmly in its socket.

6. The fuseholder cartridge is supplied with two sets of fuses—a single 3 A, 3AG Slo-blo fuse for 100/120 V, and two 1.5 A 20-mm Slo-blo fuses for 220/240 V. To change the fuses over to 220/240 V, remove the small Phillips-head screw on the cartridge, and turn over the board. Reinstall the screw. The European 20-mm fuses should now be visible.

Reverse the process to change from 220/240 V to 100/120 V operation.

7. Reinstall the fuseholder cartridge. Check the pin indicator to verify that you selected the correct voltage. If none of the holes in the fuseholder line up with the pin on the voltage changeover board, you installed the board upside down. Reinstall it correctly and try the fuseholder again.

8. This completes the voltage changeover.

Installing the Optional Transformers

Some applications require that the 480L operate under adverse electrical conditions. In these situations, it may be beneficial to transformer-couple the 480L's audio inputs and outputs. The 480L allows easy installation of audio transformers inside the unit. Please note that transformers are not available from Lexicon. They can be purchased directly from their manufacturer or a pro audio dealer.

If you choose to install transformers, follow these instructions carefully.

For the 480L *inputs*, we recommend Jensen JE-11P-1 transformers.

1. On the main circuit board (component side) of the 480L, cut the etch between E4 and E7, E5 and E8, E10 and E13, E11 and E14.

2. Install a 510 pF 2.5% polypropylene capacitor at C143.

3. Install a 510 pF 2.5% polypropylene capacitor at C183.

4. Install a 15 kilohm, 1% resistor at R152.

5. Install a 15 kilohm, 1% resistor at R187.

6. Connect a Jensen JE-11P-1 transformer as follows for the left channel:

480L	Transformer
E3	White
E4	Brown
E5	Red
E6	Black
E7	Orange
E8	Yellow

7. Connect a Jensen JE-11P-1 transformer as follows for the right channel:

480L	Transformer
E9	White
E10	Brown
E11	Red
E12	Black
E13	Orange
E14	Yellow

8. After wiring, mount the transformers to the side of the chassis, using the brackets supplied with the transformers.

For the 480L outputs, we recommend Jensen JE-123-SLPC transformers.

1. Remove R45, 46, 78, 79, 80, 82, 88, 89, 90, 91, 92, 128, 129, 130,131, and 132.

2. Install 33 ohm, 1/4 Watt, 1% resistors at R47, 81, 83, 93, 94, 133, 134, 135.

3. Move jumpers W5, W6, W7, W8, W9, W10, W11, and W12 from position 2-3 to position 1-2.

4. Solder the four transformers into place. Note that the transformers can be installed in either direction with no change in performance.