

# *Owner's Manual*

**480L**

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**Digital Effects System**

**lexicon**

## Unpacking and Inspection

After unpacking the 480L and LARC, save all packing materials in case you ever need to ship the unit. Thoroughly inspect the 480L, LARC, and packing materials for signs of damage in shipment. Report any damage to the carrier at once.

## Precautions

The Lexicon 480L is a rugged device with extensive electronic protection. However, reasonable precautions applicable to any piece of audio equipment should be observed.

- Always use the correct AC line voltage. Refer to Chapter 1 of this manual for power requirements.
- Do not install the 480L in an unventilated rack, or directly above heat-producing equipment. Maximum ambient operating temperature is 35°C (95°F).
- Never attach audio power amplifier outputs or other power sources directly to any of the 480L's connectors.
- To prevent fire or shock hazard, do not expose the 480L to rain or moisture.

## Notices

In the interest of continued product development, Lexicon reserves the right to make improvements in this manual and the product it describes at any time and without notice or obligation.

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3 Oak Park  
Bedford, MA 01730 USA  
Tel: 781-280-0300  
Fax: 781-280-0490

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### Warranty Notice

The Nonvolatile Memory Cartridge supplied with this unit carries a 30-day limited warranty.

# Table of Contents

<b>Program Directory</b>	<b>iv</b>		
<b>Introduction</b>	<b>v</b>		
<b>1. Installing the 480L</b>	<b>1-1</b>	<b>4. Bank 5: the Effects Programs</b>	<b>4-1</b>
About the Rear Panel	1-2	About the Effects Algorithm	4-2
About the Front Panel	1-3	About the Effects Parameters	4-3
Behind the Front Panel	1-4	Bank 5 - Effects	4-5
About the LARC	1-5		
How to Mount the 480L	1-6	<b>5. Bank 6: the Twin Delays Programs</b>	<b>5-1</b>
Power Requirements	1-6	About the Twin Delays Parameters	5-2
How to Interface the LARC	1-7	Bank 6 - Twin Delays	5-4
Audio Connections	1-9		
How to Float the Analog Ground	1-9	<b>6. Bank 7: the Sampler Programs</b>	<b>6-1</b>
		Introduction	6-2
<b>2. Basic Operation</b>	<b>2-1</b>	Bank 7 Samplers	6-3
Glossary	2-2	Bank 7 Samplers — SME Only	6-3
Operating Modes	2-3	How to Use the Samplers	6-4
Checking Your System's Status	2-5	About the Sampling Parameters	6-9
Selecting a Configuration	2-5		
Using Two LARCs to Control		<b>7. Bank 8: the Pitch and Doppler Programs</b>	<b>7-1</b>
a Single 480L	2-7	About the Pitch Shift Parameters	7-2
Controlling a 224XL from		Bank 8 - Pitch Shift	7-4
a 480L and LARC	2-7	About the Doppler Parameters	7-5
Selecting Input Type	2-7	Bank 8 - Doppler	7-6
How to Load Programs	2-8		
Level Calibration	2-9	<b>8. Bank 9: the Mastering Programs</b>	<b>8-1</b>
Setting Analog Output Levels	2-10	About the Stereo Adjust Parameters	8-2
Setting Analog Input Levels	2-11	PONS Adjust	8-5
Levels in the Digital Domain	2-12	Digital Parametric EQ	8-7
Using Digital Signals	2-13	Panorama (Binaural)	8-10
How to Edit Parameters	2-14		
How to Use Registers	2-14	<b>9. Bank 10: the Compressor Programs</b>	<b>9-1</b>
Storing and Naming Programs	2-15	About the Compressor Parameters	9-2
Loading Registers	2-15	Bank 10 - Compressor/Expander	9-3
Protecting Against Loss of			
Register Contents	2-15	<b>10. Banks 11-12: the Random Halls and Spaces Programs</b>	<b>10-1</b>
Moving Registers with the		About the Random Reverb Parameters	10-2
Register Transporter	2-16	Bank 11 - Random Hall	10-5
Clearing Register Contents	2-16	Bank 12 - Random Rooms	10-7
MIDI SysEx Program Dumps	2-27		
		<b>11. Banks 13-14: the Ambience Programs</b>	<b>11-1</b>
<b>3. Banks 1-4: the Reverb Programs</b>	<b>3-1</b>	About the Ambience Parameters	11-2
About the Reverberation Algorithm	3-2	Bank 13 - Ambience	11-5
About the Reverberation Parameters	3-4	Bank 14 - Post Ambience	11-7
Bank 1 - Halls	3-8		
Bank 2 - Rooms	3-10		
Bank 3 - Wild Spaces	3-12		
Bank 4 - Plates	3-14		

<b>12. Bank 15: the Prime Time III Programs</b>	<b>12-1</b>	<b>Appendix</b>
About the Prime Time III Parameters	12-2	
Bank 12 - Prime Time III	12-4	<b>A. MIDI and the 480L</b>
<b>13. Bank 16: the Freq. Stuff Programs</b>	<b>13-1</b>	<b>B. Solving Problems</b>
About the Frequency Dynamics Parameters	13-2	<b>C. Specifications</b>
Bank 16 - Frequency Dynamics	13-3	<b>D. Voltage Changeover and Optional Transformers</b>
About the Distresson Parameters	13-4	<b>E. Control Mode Reference</b>
Bank 16 -Distresson	13-6	
<b>14. Bank 17: the Test and Reference Programs</b>	<b>14-1</b>	
About the Test & Reference Parameters	14-2	
Bank 17 - Test & Reference	14-3	

## Program Directory - Software Version 4.0

Bank	Program Name									
<b>Bank 1</b> 480 Halls	1.1 Large Hall	1.2 Large + Stage	1.3 Medium Hall	1.4 Medium + Stage	1.5 Small Hall	1.6 Small + Stage	1.7 Large Church	1.8 Small Church	1.9 Jazz Hall	1.0 Auto Park
<b>Bank 2</b> 480 Rooms	2.1 Music Club	2.2 Large Room	2.3 Medium Room	2.4 Small Room	2.5 Very Small Room	2.6 Large Wood Rm	2.7 Small Wood Rm	2.8 Large Chamber	2.9 Small Chamber	2.0 Small & Bright
<b>Bank 3</b> Wild Spaces	3.1 Brick Wall	3.2 Buckram	3.3 Big Bottom	3.4 10W-40	3.5 20W-50	3.6 Metallica	3.7 Silica Beads	3.8 Inside Out	3.9 Ricochet	3.0 Varoom
<b>Bank 4</b> Plates	4.1 A Plate	4.2 Snare Plate	4.3 Small Plate	4.4 Thin Plate	4.5 Fat Plate					
<b>Bank 5</b> Effects	5.1 Illusion	5.2 Surfin'	5.3 Vocal Whispers	5.4 Doublor	5.5 Back Slap	5.6 Rebound	5.7 Git It Wet	5.8 Sudden Stop	5.9 In the Past	5.0 Tremolo L & R
<b>Bank 6</b> Twin Delays	6.1 4-Voice Double	6.2 Double Delay	6.3 4-Bounce Delay	6.4 Pitter Patter	6.5 X-Pan Double	6.6 Delay Cave	6.7 Circles	6.8 There & Back	6.9 Soft Roller	6.0 On and On
<b>Bank 7</b> Samplers*	7.1 Stereo 3 Sec.	7.2 Mono 6 Sec.	7.3 Mono 3 Sec.	7.4 Dual Rate Change	7.5 Mono Fwd & Rev	7.6 Stereo 3S Drum	7.7 Dual Rate Chg Drum	7.8 Mono Fwd Rev Drum	7.9 Stereo 10S Rate Chg	7.0 Mono 20S Rate Chg
<b>Bank 8</b> Pitch/Doppler	8.1 Pitch Change	8.2 Pitch Chorus	8.3 1% Up 1% Down	8.4 Barber Pole	8.5 Half Steps	8.6 Stair Case	8.7 XPres Subway	8.8 Indy Corner	8.9 Airport Land	8.0 Airport Tkof
<b>Bank 9</b> Mastering	9.1 The In-Out	9.2 Stereo Adjust	9.3 PONS Adjust	9.4 M/S Decode	9.5 Invert L-R	9.6 Channel Swap	9.7 Stereo Param EQ	9.8 Stereo 60Hz Ntch	9.9 Mono Param EQ	9.0 Panorama
<b>Bank 10</b> Compressor	10.1 2:1 Comp	10.2 6.24:1 Comp	10.3 18.28:1 Comp	10.4 Transfer	10.5 Vox Easy	10.6 FM	10.7 Nailed	10.8 Noise Gate	10.9 2-Slp-Mach A	10.0 2-Slp-Mach B
<b>Bank 11</b> Random Hall	11.1 Large R Hall	11.2 Large R + Stage	11.3 Medium R Hall	11.4 Medium R + Stage	11.5 Small R Hall	11.6 Small R + Stage	11.7 Large R Church	11.8 Small R Church	11.9 Jazz R Hall	11.0 Auto Park R
<b>Bank 12</b> Random Spaces	12.1 Music Club R	12.2 Large Room R	12.3 Medium Room R	12.4 Small Room R	12.5 Very Small R	12.6 Large Chamber R	12.7 Small Chamber R	12.8 Small & Bright R	12.9 Chorus Room	12.0 Wet & Tacky
<b>Bank 13</b> Ambience	13.1 Very Large Ambience	13.2 Large Ambience	13.3 Medium Ambience	13.4 Small Ambience	13.5 Strong Ambience	13.6 Heavy Ambience	13.7 Ambient Hall	13.8 Announcer	13.9 Closet	13.0 Gated Ambience
<b>Bank 14</b> Post Ambience	14.1 Car Interior	14.2 Living Room	14.3 Bathroom	14.4 Kitchen Ambience	14.5 Kellars Cell	14.6 Small Foley	14.7 Warehouse	14.8 Airhead	14.9 Dial It Up	14.0 Reverb Tail
<b>Bank 15</b> Prime Time III	15.1 Prime Chorus	15.2 Slap Chorus	15.3 Bounce Glide	15.4 Swirls	15.5 Heavy Detune	15.6 Shake It Up	15.7 Wowza	15.8 Wowza 2	15.9 Vocalz	15.0 XFlange
<b>Bank 16</b> Freq. Stuff	16.1 Mix Finish	16.2 BG Vocals	16.3 Drums	16.4 AC Guitars	16.5 Bass Thump	16.6 Carbon Mic	16.7 Saxophone	16.8 Horn Blast	16.9 Softener	16.0 Some Fuzz
<b>Bank 17</b> Test & Reference	17.1 100 Hz -12dB	17.2 500 Hz -17dB	17.3 500 Hz -12dB	17.4 1 kHz -17dB	17.5 1 kHz -12dB	17.6 10 kHz -12dB	17.7 30 Hz Slate	17.8 A-440 Tuner	17.9 Pink Noise	17.0 Binaural Simulator

## Program Directory - 480L Classic Cart\*\*

Bank	Program Name									
<b>Bank 18</b> Concert Hall	18.1 Concert Hall	18.2 Medium Hall	18.3 Small Hall	18.4 Gold Hall	18.5 Guitar Hall	18.6 Reggae Hall	18.7 Reflex Hall	18.8 Bright Hall	18.9 Dark Hall	18.0 Oliver's Hall
<b>Bank 19</b> Rich Plate	19.1 Large Plate	19.2 Drum Plate	19.3 Vocal Plate	19.4 Vox Plate #2	19.5 Slap Plate	19.6 Guitar Plate	19.7 Short Plate	19.8 Horn Plate	19.9 Echo Plate	19.0 Silver Plate
<b>Bank 20</b> 6-Voice Chorus	20.1 Woo-Woo	20.2 6 Vc Korus	20.3 3-Voice Chorus	20.4 4-Voice Chorus	20.5 Ekoz & Korus	20.6 Vocal Chorus	20.7 Soft Echoes	20.8 Hard Echorus	20.9 Canyon Korus	20.0 Tuff Stuff
<b>Bank 21</b> Multiband Delays	21.1 Closet	21.2 Telephone	21.3 Phartage	21.4 Stadium	21.5 Downstairs	21.6 Bandsweep	21.7 BassEchoes	21.8 BandBounce	21.9 Whispers	21.0 On Stage

\* Programs 7.9 and 7.0 require the Sampling Memory Expansion option. See your Lexicon dealer for details.

\*\* Programs in Banks 18-21 require the Classic Cart option. See your Lexicon dealer for details.

## Introduction

You are about to begin using the most advanced digital effects system available—the Lexicon 480L. The 480L is engineered for the all-digital production environment. Digital audio places strict requirements on every link in the signal chain, and the 480L meets those requirements. With its unique 18 bit A/D and D/A converters, the 480L produces a dynamic range of 98 dB in the wet signal path. It is probably the only effects system available that doesn't raise the noise floor of a digital master. And the PCM 1610/1630 compatible digital I/O interface lets you add true stereo ambience and effects without leaving the digital domain.

The 480L doesn't just sound better—sheer computational power allows it to perform multiple audio tasks at the same time. And what tasks! In the current glut of throwaway digital devices with ever-cheaper versions of the same sounds, the 480L offers remarkable new effects and reverb sounds.

Its innovative reverb algorithms reflect a more accurate and natural model of the acoustic and psychoacoustic phenomena of reverb and ambience. Put the 480L up against any other reverberator—you'll be amazed at the difference.

Reverb is only part of the story. The 480L produces astonishing effects you haven't even begun to dream about yet. And its sampling programs offer a variety of useful and unique features.

The present software is powerful and comprehensive, a dramatic step forward in digital signal processing technology. Yet it hasn't explored the limit of the 480L's architecture, which is itself configured for future hardware expansion.

If you are familiar with the venerable 224XL, you'll feel right at home with the LARC used to control the 480L. However, there are enough differences in the way the two units operate that we strongly suggest that you read this manual as soon as possible. In it, you'll discover that the 480L's two high speed processors can operate in a variety of configurations. Samples can be processed with reverb or effects, all in the digital domain. Two 480Ls can be connected through their digital I/O ports for even wider creative horizons. The 480L can even be connected to a 224XL and both units operated from a single LARC. And that's just the beginning of what you'll discover--when you read this manual!

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# Installation and Audio Connections

This chapter contains overviews of all of the 480L controls, as well as information on mounting, installation and audio connection.

## 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 Outputs (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.

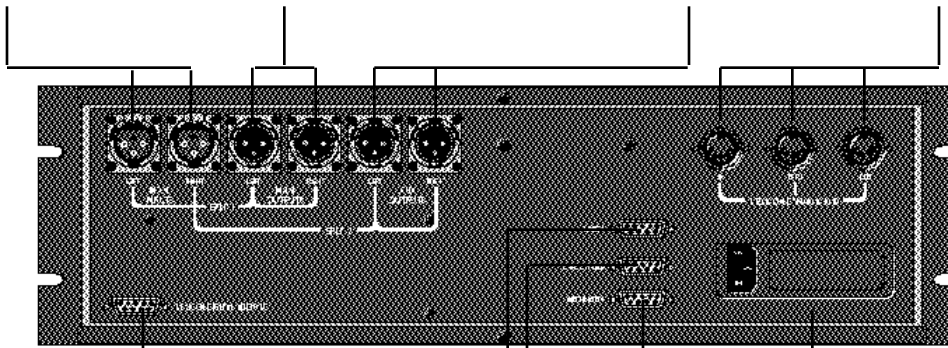
**Important.** Reversing polarity on either input or output connectors can produce audible phase inversion effects. Improper phasing in the stereo path can create a weak or thin mix. Ensure that inputs and outputs are wired consistently.

### 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 used to transmit Automation data.



### 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.

### 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 D for voltage changeover information.

### LARC 2 (Thru) Connector

This DE9 connector allows connection of a second LARC. 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 available from Lexicon for this application.



## About the Front Panel

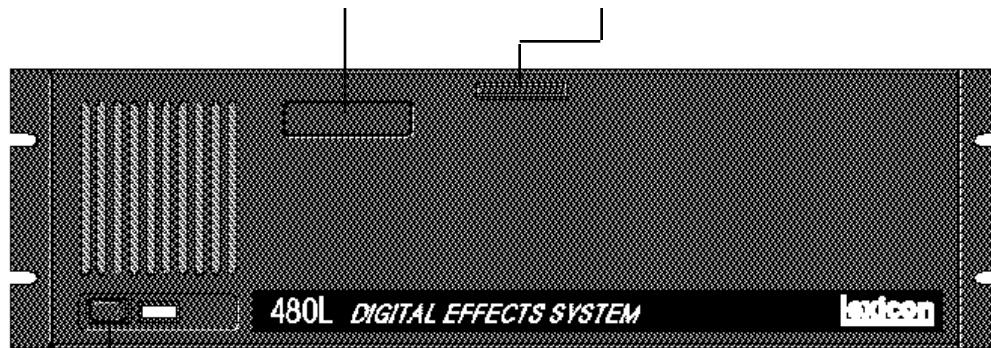
### Cartridge Slot

The Cartridge Slot accepts both ROM-based program cartridges, and Nonvolatile RAM cartridges for additional memory storage.

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.

### 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.

## 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

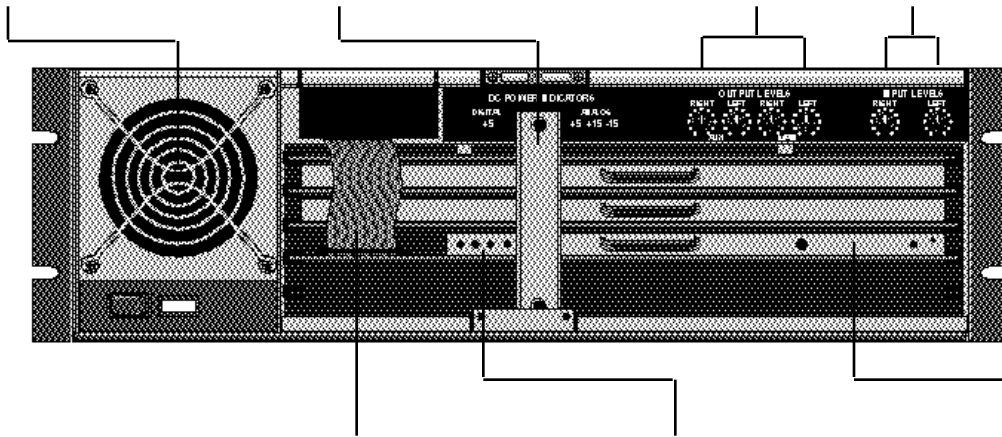
The card retainer ensures that the HSP and host processor cards remain firmly seated.

### 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.

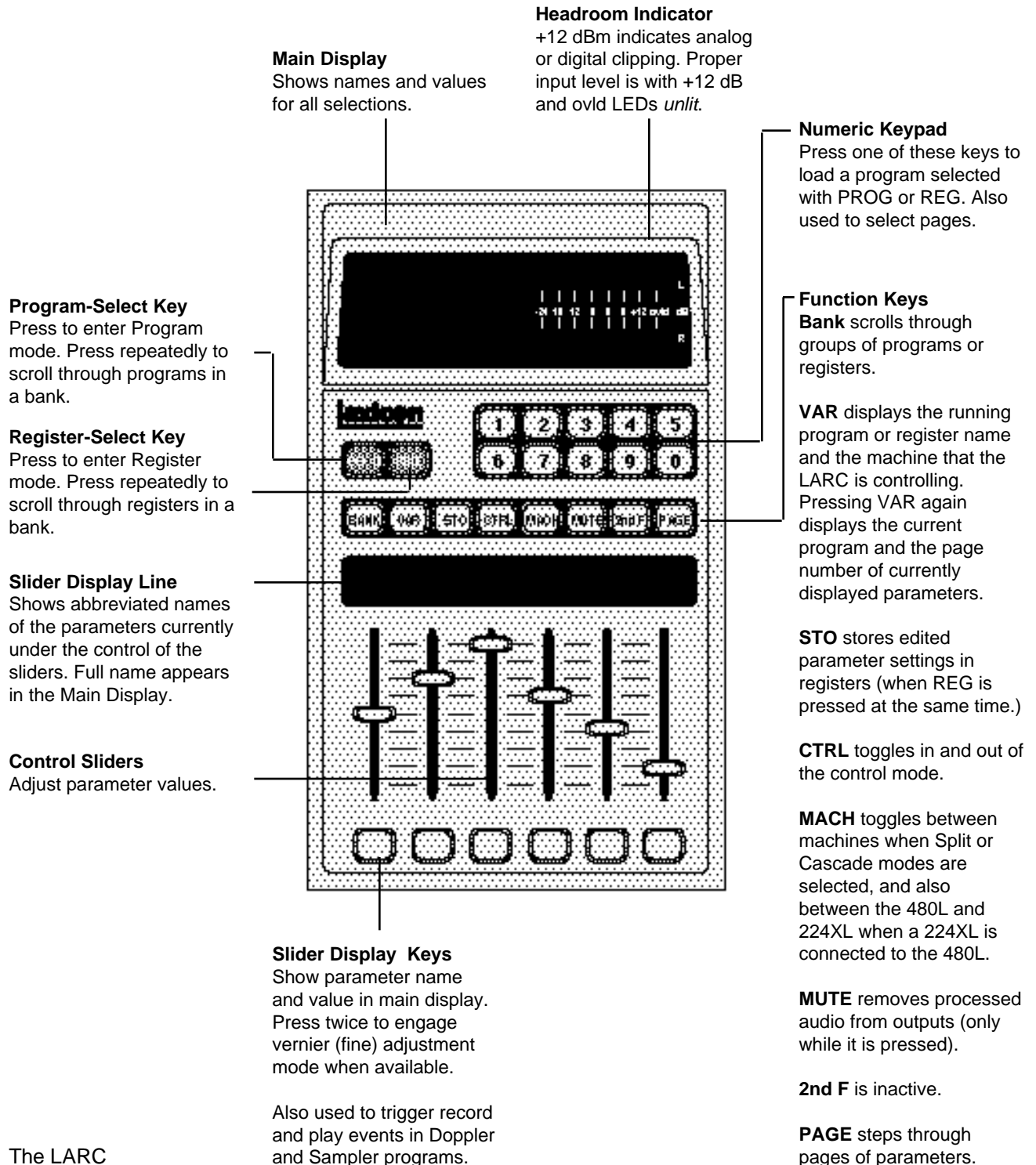
### Diagnostic Indicators

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.

# About the LARC



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.

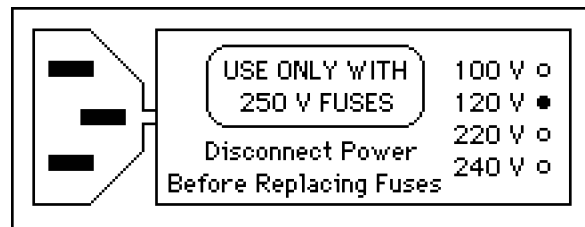
If the 480L is mounted in a rack or road case, we recommend that you provide support for the rear of the chassis during transport to prevent possible damage from severe mechanical shock.

## 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 D.

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



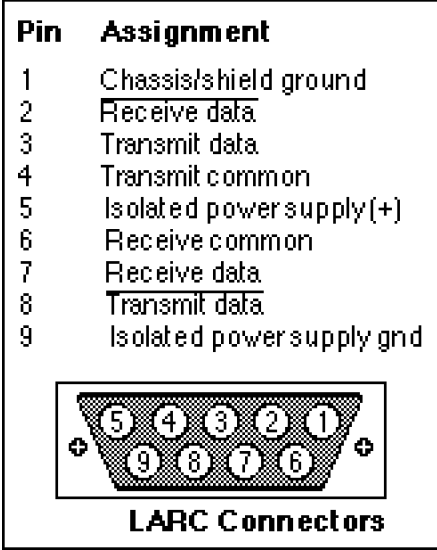
Voltage selector set for 120 V.

## How to Interface the LARC

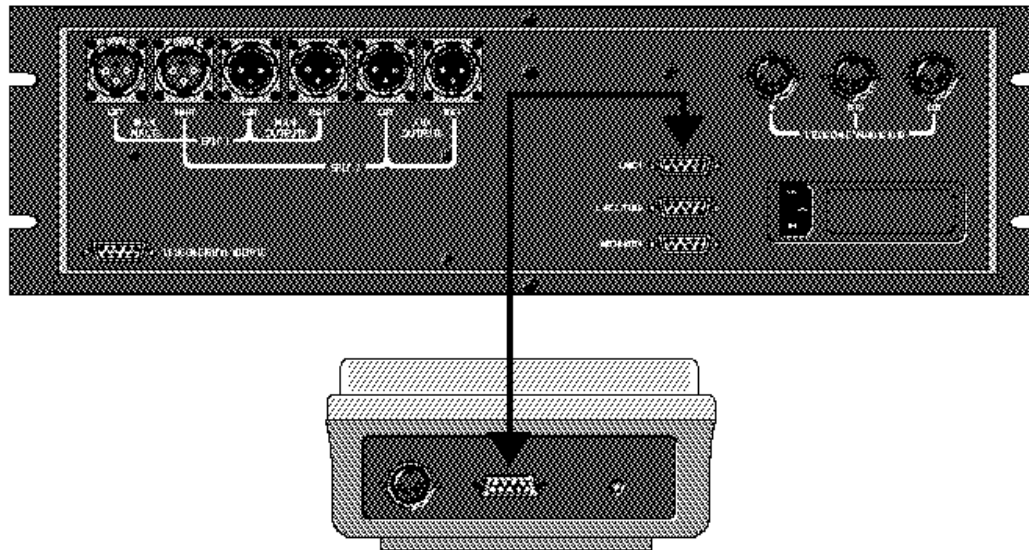
The LARC 1 connector interfaces the mainframe to the Lexicon Alphanumeric Remote Control (LARC) via a flexible 50-ft cable (supplied). If your system is equipped with a single LARC, this is the connector you should use.

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. It can also be used to connect the 480L to a 224XL with both units under the control of a single LARC. A 10' cable is available from Lexicon for this connection. (Lexicon Part No. 680-04734) The pin assignments for the LARC connectors are shown to the right.

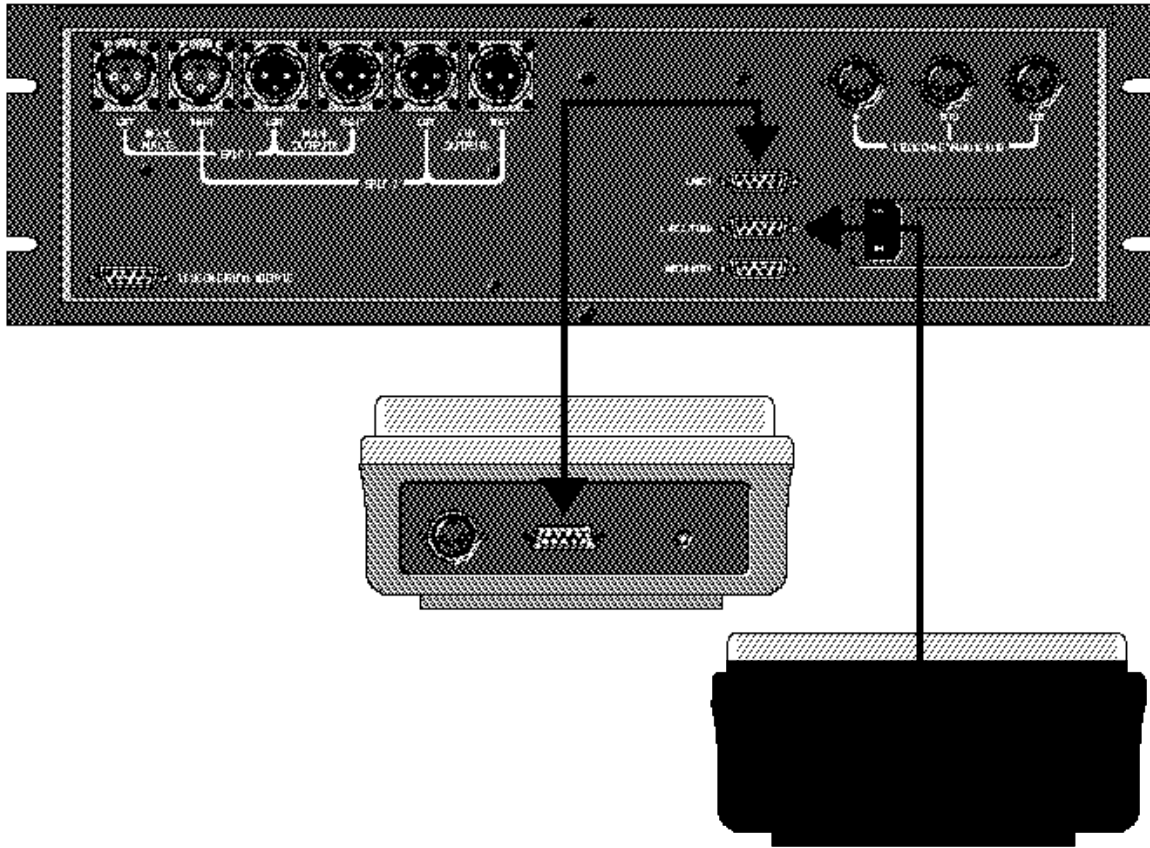
**Important:** Never connect a LARC to the automation connector. Doing so may blow the internal automation connector fuse.



Wiring diagram for the LARC mainframe connectors.



Connections for 480L with one LARC.



Connections for 480L with two LARCs.

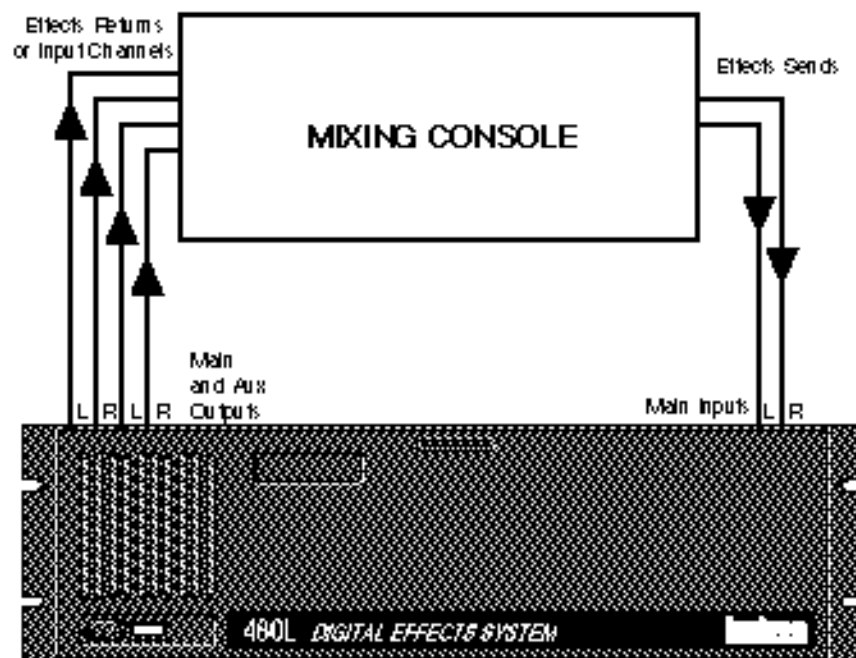
## Analog Audio Connections

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. The figure below shows a typical configuration. For maximum utility, use independent sends that can be assigned as either pre-fader or post-fader. 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).



Typical Audio Connections.

## 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.

## 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 accommodated 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.

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.

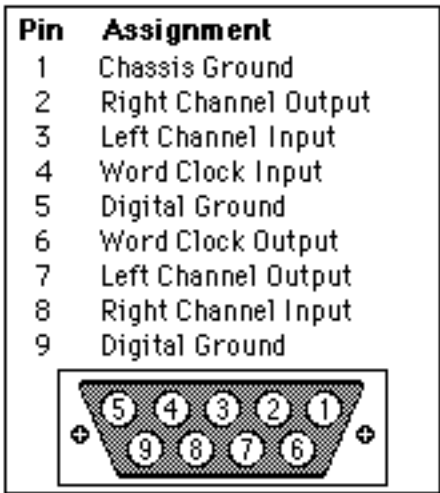
**Important!** Do not power up with external word clock present at the 480L's digital I/O port. Doing so may prevent the unit from finishing its power up sequence.

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

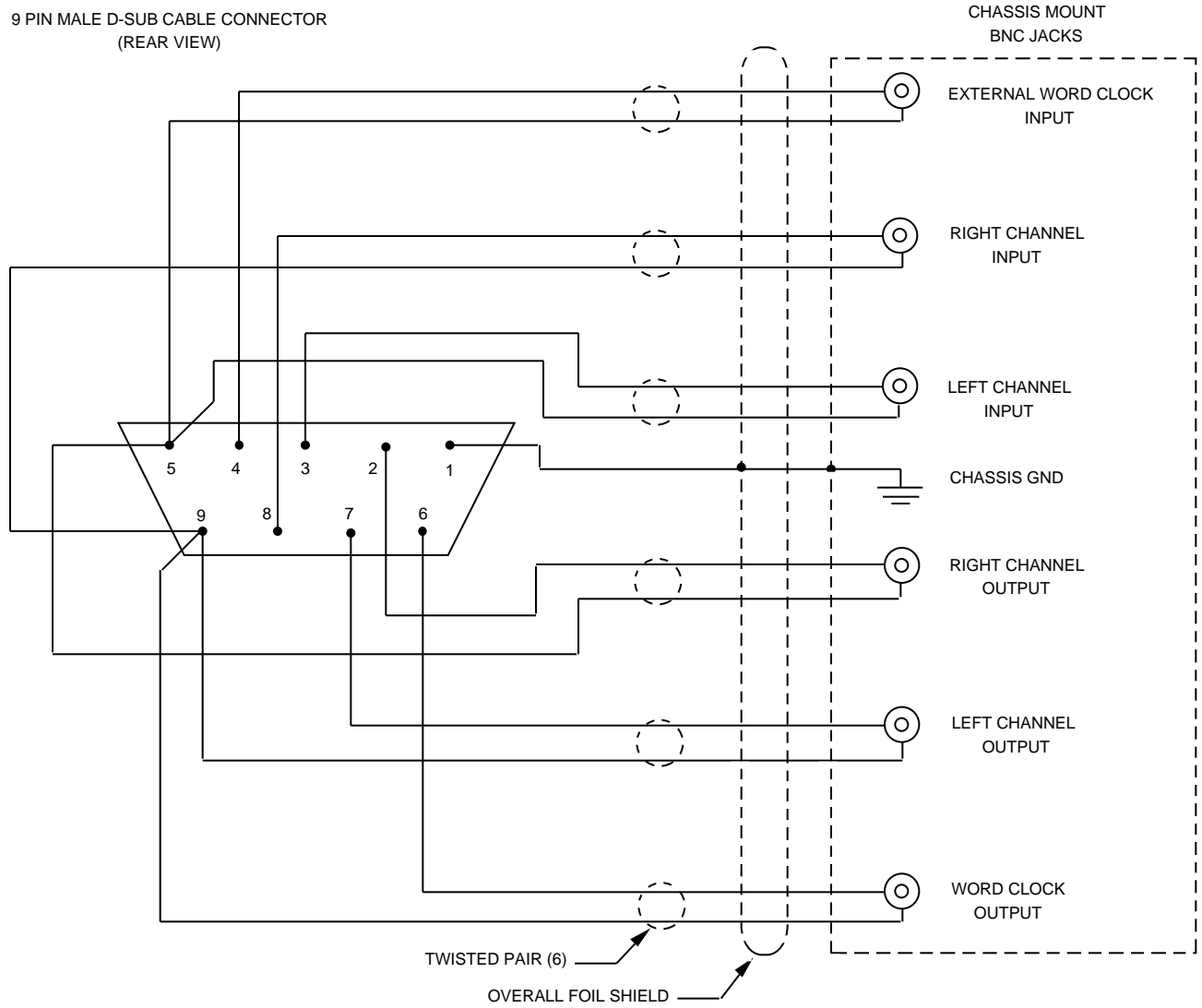
**Note:** A digital audio I/O interface box which can greatly simplify interfacing the 480L is available as an option. Contact your Lexicon Representative for details.

### Conversion to AES/EBU

The 480L's 1610/1630 compatible signal can be translated to AES/EBU to allow interfacing with devices using this digital I/O format. To accomplish this, an external format converter such as Lexicon's LFI-10 is required. The following figure details digital interconnection using a format converter.

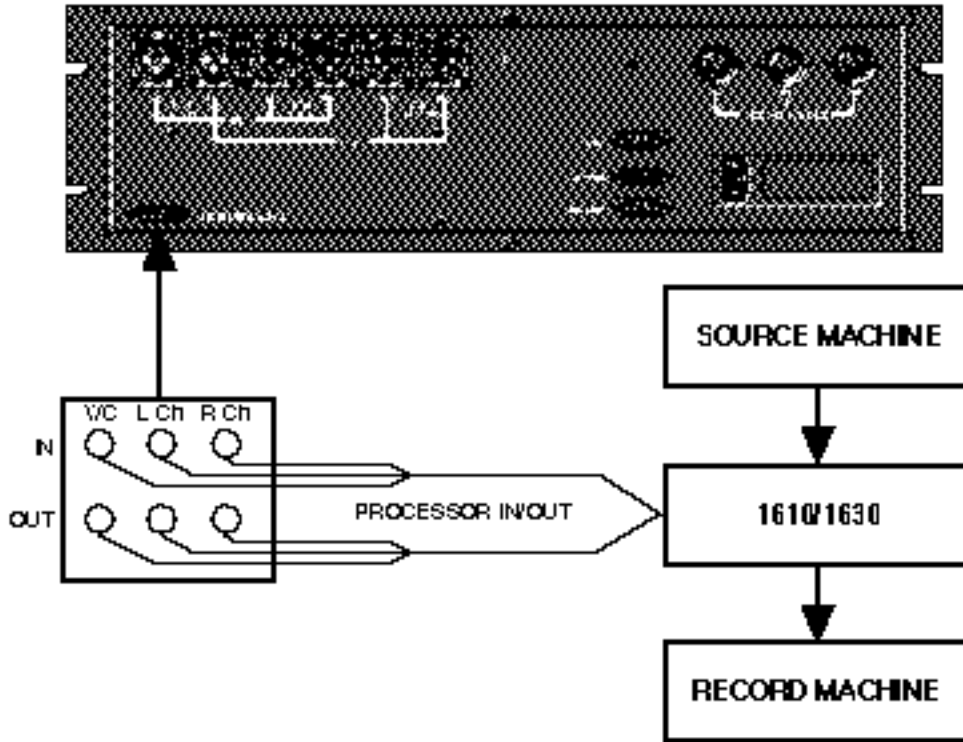




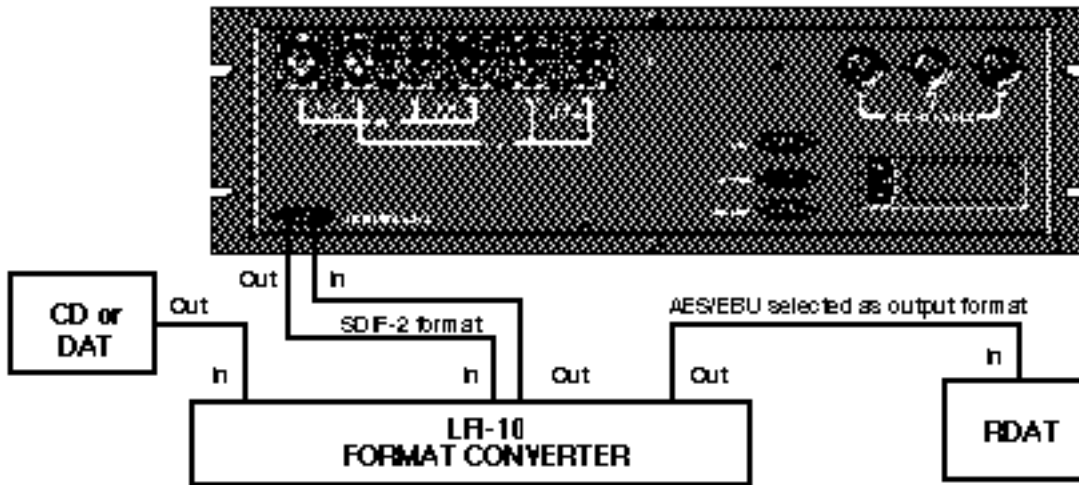


USE BELDEN #S 9682, 9506, OR EQUIVALENT SHIELDED,  
LOW CAPACITANCE CABLE WITH 6 TWISTED PAIRS.

Suggested interface for Digital I/O Connector.



Typical 1610/1630 Connection



Converting Digital Signals to AES/EBU Format.

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## Basic Operation

This chapter describes the operations necessary to properly calibrate the 480L in your facility.

These include:

- Learning the operating modes
- Selecting machine configuration
- Switching machines from the LARC
- Selecting analog or digital inputs
- Loading programs
- Calibrating levels

## Glossary

- **Mainframe** *Mainframe* refers to a functional 480L operating with one or more *machines*.
- **Machine** Each 480L HSP board is addressed in the mainframe as an independent *machine*, or signal processing engine. In addition, the 480L can recognize the Lexicon 224XL and address it as another machine.
- **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. Each Machine in the 480L can process an algorithm independently from the other machine.
- **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 as separate programs in the 480L.
- **Preset** A group of specific parameter settings you create by editing presets and storing the new set of values.
- **Register** Nonvolatile RAM memory locations in the mainframe, or in a removable nonvolatile memory cartridge, for storing presets.
- **Bank** A *bank* is a collection of as many as ten programs or registers. Program banks contain a collection of similar programs derived from one or more algorithm. For example, the *Halls* bank contains reverberation programs that simulate large spaces, while the *Mastering* bank contains programs for level adjustment and digital equalization.
- **Pages** The LARC can display and provide slider control for six parameters at a time. Because most algorithms have more than six parameters, they are grouped in blocks of six called *pages*. The PAGE button provides access to each group of parameters.
- **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 the CTRL button on the LARC.
- **Configuration** The 480L machines can be *configured* to operate independently, or they can be combined to function as a single machine. The configuration is changed from the control mode. *The 480L is shipped in the single configuration.*
- **dBFS** A digital signal at full amplitude, or registering to the most significant bit.

## Operating Modes

The 480L is always in one of the following four operating modes:

Machine Operation Mode  
Program Preview Mode  
Register Preview Mode  
Control Mode

The 480L always powers up in Machine Operation Mode.

Each mode, and the paths for entering and exiting it are described below.

### Machine Operation Mode

Press the PROG or the REG button, then press VAR.

This mode allows you to:

- View the machine currently being addressed by the LARC.
- Display the program or preset currently running in the machine.
- View and/or alter parameters.

### Program Preview Mode

Press the PROG button.

Press the BANK button to view program banks.

Press the PROG button to view programs in the bank.

This mode allows you to:

- Select presets.

### Register Preview Mode

Press the REG button.

Press the BANK button to view register banks.

Press the REG button to view programs in the bank.

This mode allows you to:

- Select, store, and recall programs from registers.
- Address a cartridge in the mainframe cartridge slot.

### Control Mode

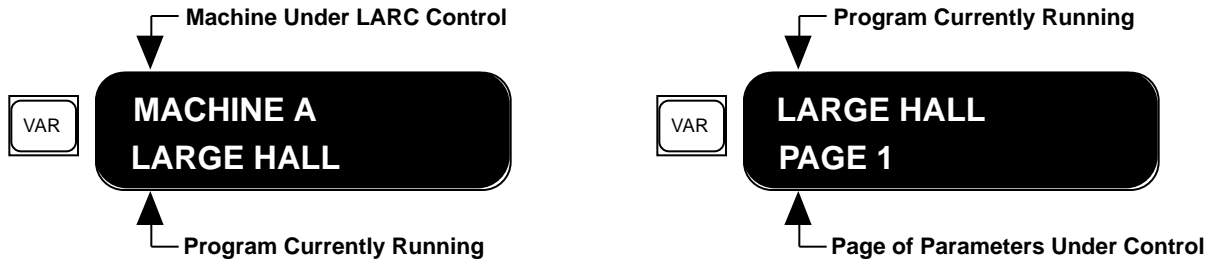
Press the CTRL button. To exit, press CTRL, then press PROG, then press REG.

This mode allows you to:

- View system status.
- Set machine and input configuration.
- Alter input and set digital operation.
- Address the register transporter.
- Name programs.
- Create Dynamic MIDI™ patches.
- Transmit MIDI Program Change messages.
- Enable MIDI SysEx Automation
- \* Transmit MIDI Bulk Dumps

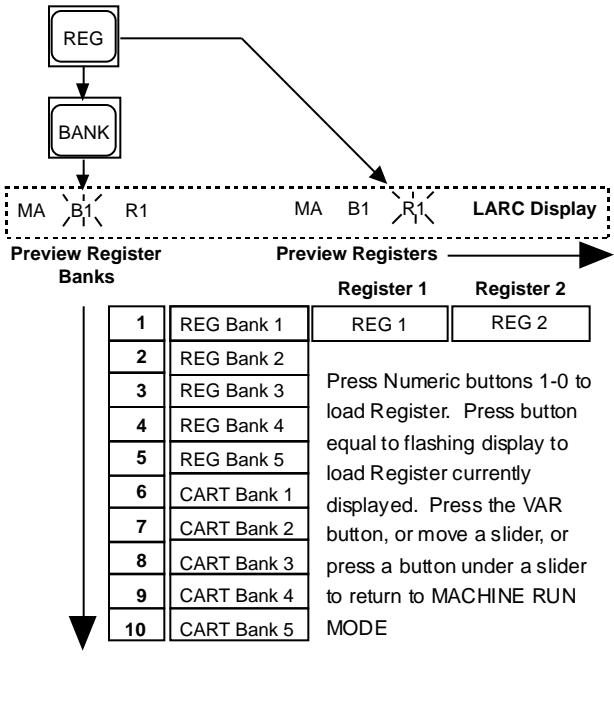
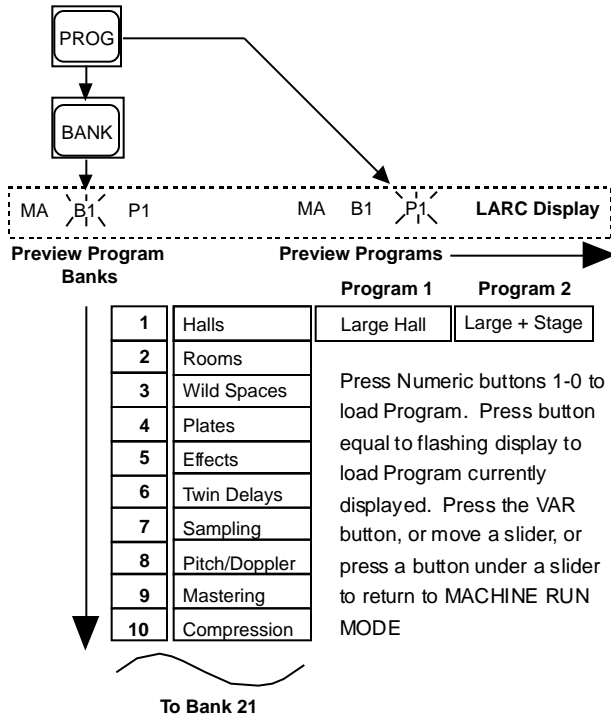
## Machine Operation Mode

On power up, the 480L restores the configuration, routing, and programs loaded before power was shut off. The LARC will display the last machine selected.



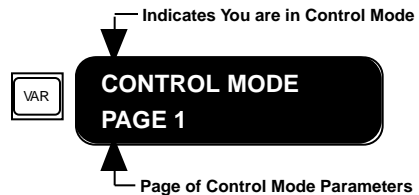
## Program Preview Mode

## Register Preview Mode



## Control Mode

Press **CTRL** to enter or exit CONTROL MODE



Page #	Page Desc.	Slider 1	Slider 2	Slider 3	Slider 4	Slider 5	Slider 6
1	Configuration	Status	Configuration		Sampling Rate	Clock	Input
2	Copy Tools	Copy	From Source	To Destination		Power On Default	REG Protect
3	Name Preset	Select Char.					Change Char.
4	MIDI Patching	Select Patch	Srcce. Controller	Parameter Dest		Scale	Param. Value
5	MIDI Configuration	SysEx Auto	SysEc Channel	Pgm Change #	Reg Table	Pgm Change Mode	MIDI Channel
6	SysEx Functions	SysEx Function	Source to Transmit	SysEx Destination			

## Checking Your System's Status

Press CTRL to enter Control Mode. Press PAGE, then press 1. 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)

### 480L Power-Up State

Normally if power is lost, or turned off, and then restored, the 480L will return to its last previous state. You can also configure the 480L to automatically load pre-selected programs upon power up. To do this, go to Control Mode, Page two, slider 5. This slider, labeled DEF for default, is set at the factory to "off". Advancing the slider will scroll through all of the 480L programs and select one as the new default state. The upper display will indicate the machine you are addressing, followed by the Bank number, Program number and Program name. In order to select a default load for the second machine, press the MACH button and advance the slider to the desired program.

**Control configuration is always remembered regardless of power-up default setting.**

## Selecting a Configuration

System configuration, input type and system clocking are selected in Control Mode. To enter or exit Control Mode, press CTRL.

Note: Pressing VAR will display the operating mode of the mainframe.

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

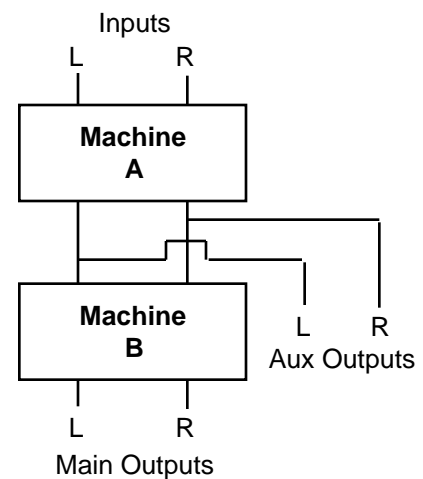
- Cascade
- Stereo Split
- Mono Split
- Single

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

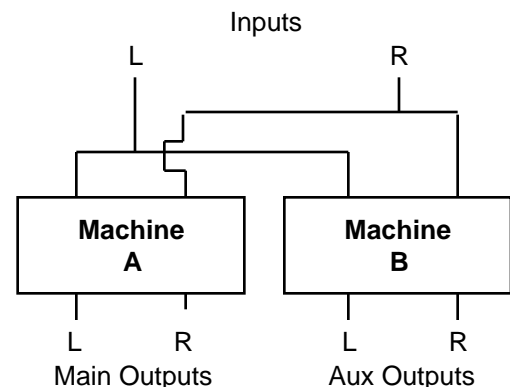
The Cascade configuration feeds the output of one program (Machine A) directly into the input of the second program (Machine B). This allows you to process a stereo signal with two entirely different effects—without ever leaving the digital domain. The Main outputs are connected to Machine B, and contain 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 most programs becomes very important, because it is the only method you have of controlling the mix between the two programs.



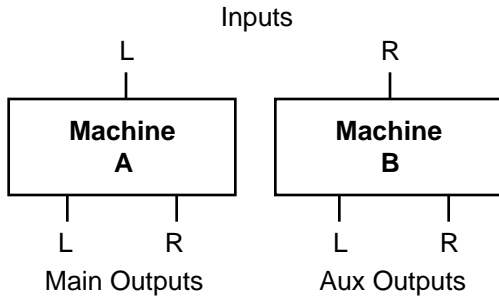
### The Stereo Split Configuration

The Stereo Split configuration also uses the 480L as two independent signal processors. 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.



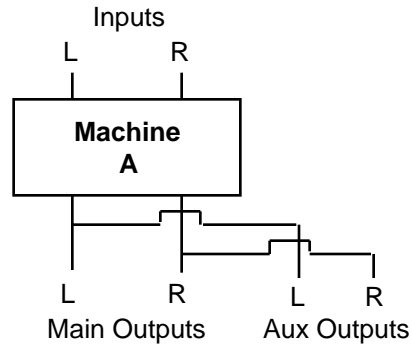
### The Mono Split Configuration

The Mono Split configuration uses the 480L as two independent signal processors. 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.



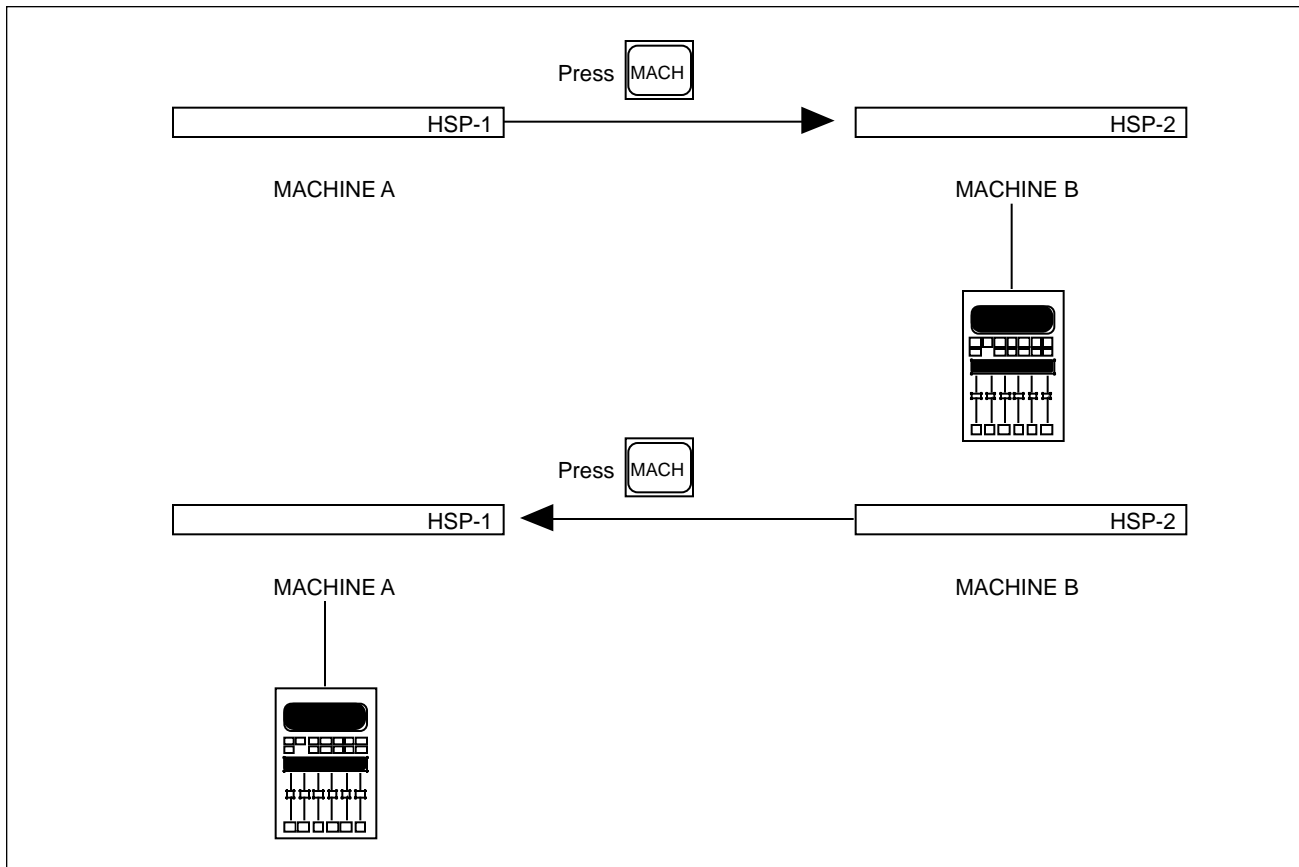
### The Single Configuration

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.



### Selecting a Machine

Once a configuration has been selected, press CTRL to exit Control Mode. Now, pressing MACH switches LARC control from one machine to the other.





## 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, 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.

## Selecting Input Type

Slider 6 on Page 1 of Control Mode allows selection of either analog or digital input. Both analog and digital outputs are always active.

Before selecting digital input, be sure that proper connections have been made to the 480L Digital I/O. (See Chapter 1.)

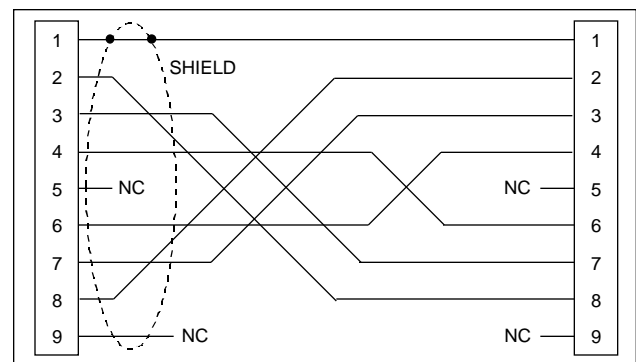
Use the Clock slider on Page 1 of Control Mode to select EXTERNAL CLOCK. Set digital clock and sampling rate to match your application according to the table below.

Analog		Digital	
Clock	Internal	Clock	External
Sampling Rate	44.1 or 48kHz	Sampling Rate	to match Digital Input Rate
Input	Analog	Input	Digital

To determine whether the 480L is correctly receiving external word clock, move the Status slider (Slider 1, Page 1 of Control Mode) to display external word clock status.

**IMPORTANT**

**DO NOT POWER UP THE 480L WITH EXTERNAL WORD CLOCK PRESENT AT THE 480L'S DIGITAL I/O PORT. DOING SO MAY PREVENT THE UNIT FROM COMPLETING ITS NORMAL POWER UP ROUTINE.**



224XL Mainframe Interconnect Cable Wiring

## How to Load Programs

In order to complete system setup, you will have to load programs into the 480L machines.

Remember that programs are variations of algorithms with parameters that have been set at the factory. Programs are stored in Banks, with a collection of as many as 10 similar programs stored in each bank.

### Select a Bank

In order to select a program, you must first select the bank in which the program is stored. There are two ways to select banks:

1. Press PROG, then press BANK. This puts you in Program Preview Mode. The bank number flashes to indicate banks are being previewed, and the display shows the current bank name and number.

Press BANK repeatedly. The LARC will scroll through the banks and display their names.

2. Press PROG, then press BANK, then press one of the numeric keys (two for double digit bank numbers) to navigate directly to a specific bank.

### Select a Program

Once a bank has been selected, you can view the programs in that bank without loading them. To do this, press PROG. The bank display on the LARC will stop flashing and the program display will begin flashing. The current program name and number will be displayed.

Press PROG repeatedly. The LARC will scroll through the programs in the selected bank.

## Load a Program

480L programs are loaded by pressing the numeric button that matches the flashing LARC display.

Press PROG, and then press a numeric button (1-0). The LARC will flash the message "SETUP LOADED."

You will find that you can navigate between programs very quickly with this method. For example, enter the following keystrokes:

PROG	BANK	1	PROG	1	=	Large Hall
PROG	BANK	9	PROG	7	=	Stereo Parametric EQ
PROG	BANK	7	PROG	4	=	Dual Rate Chg Sampler
PROG	BANK	8	PROG	8	=	Indy Corner
PROG	BANK	3	PROG	1	=	Brick Wall

Once a program is loaded, the 480L will switch from Program Preview Mode to Machine Operation Mode. The new program name will be displayed, and the first page of variable parameters will appear on the LARC.

Detailed information on program parameters is found in later chapters of this manual.

## Level Calibration

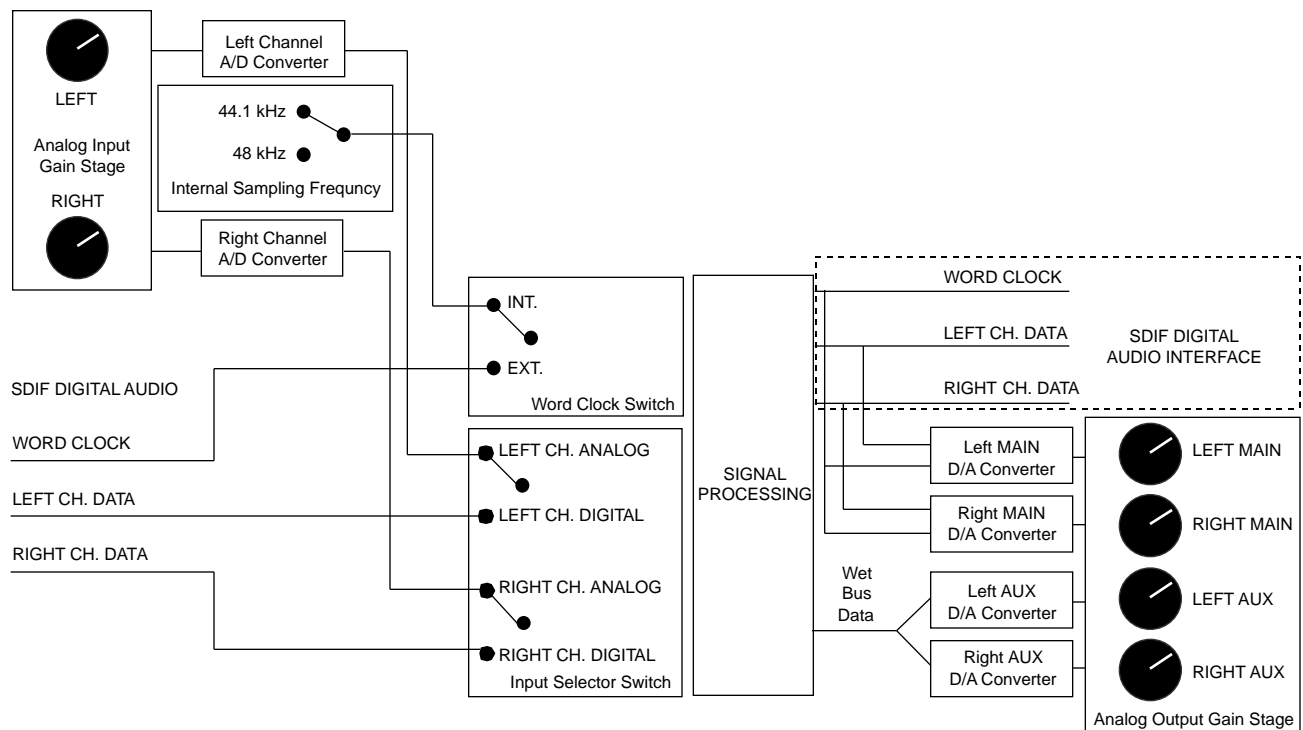
Analog Input and Output levels should be set with care to obtain the best possible performance from the 480L. The diagram below illustrates the signal flow through the mainframe. In addition, several programs are available in Bank 17 *Test & Reference* to help you optimize the performance of your system.

### Set Test Configuration

Use the Single Machine configuration:

1. Press CTRL, press PAGE, then press 1.
2. Move Slider 2 to select SINGLE.
3. Press CTRL or PROG to exit Control Mode and enter Machine Operation Mode.

**IMPORTANT  
MUTE CONTROL ROOM MONITORS  
BEFORE PROCEEDING.**



Signal Flow through the 480L Mainframe

## Setting Analog Output Levels

The 480L has digital oscillator programs to aid in setting system output levels.

Set nominal output level to design center of receiving equipment as follows:

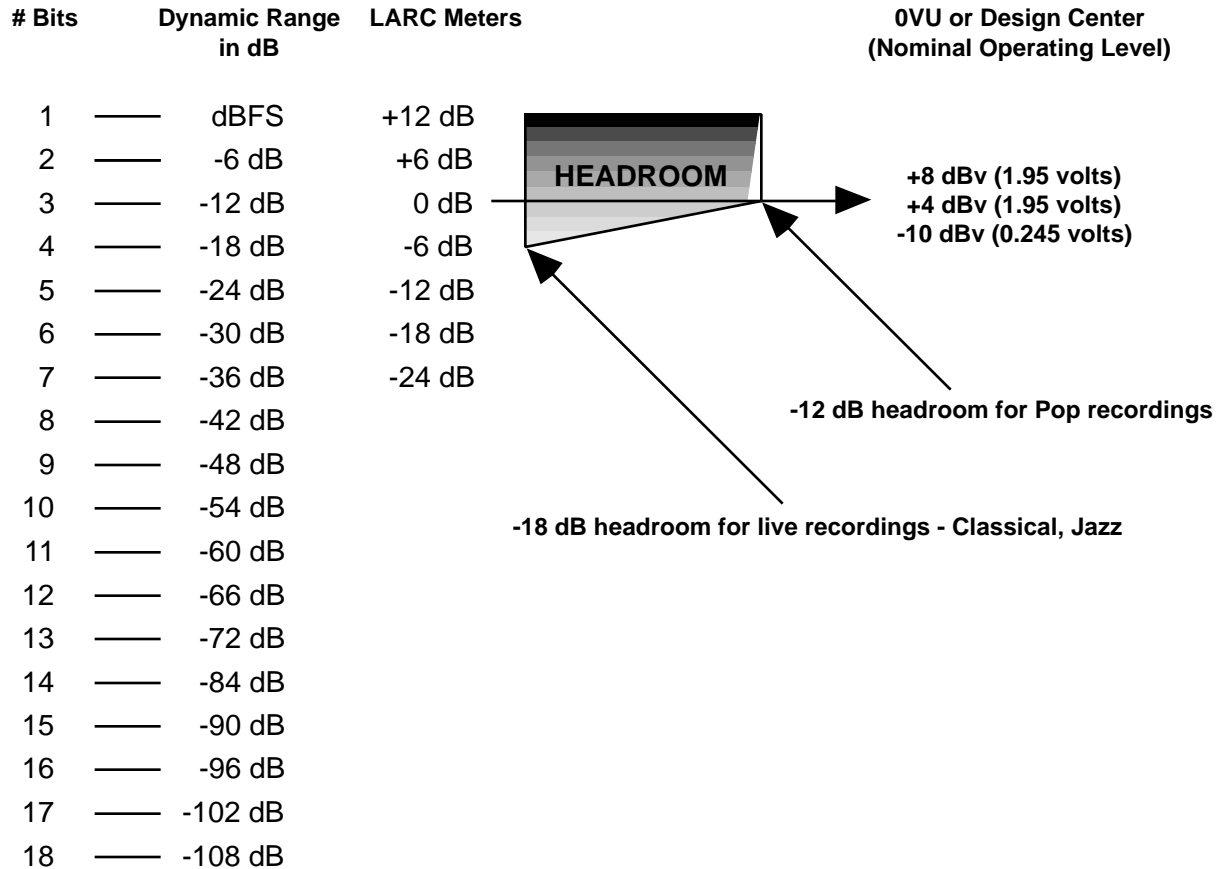
1. Load program 17.5 1kHz -12dB as follows:

- Press PROG
- Press BANK
- Press 1, then press 7
- Press PROG
- Press 5

If additional headroom is desired, load program 17-4 (1kHz -17dB).

2. A digitally-generated 1kHz should appear at all outputs (both analog and digital).

Set the analog output level controls for both Main and Aux Out to register nominal operating level on the device receiving this signal.



### Dynamic Range and Headroom Calibration

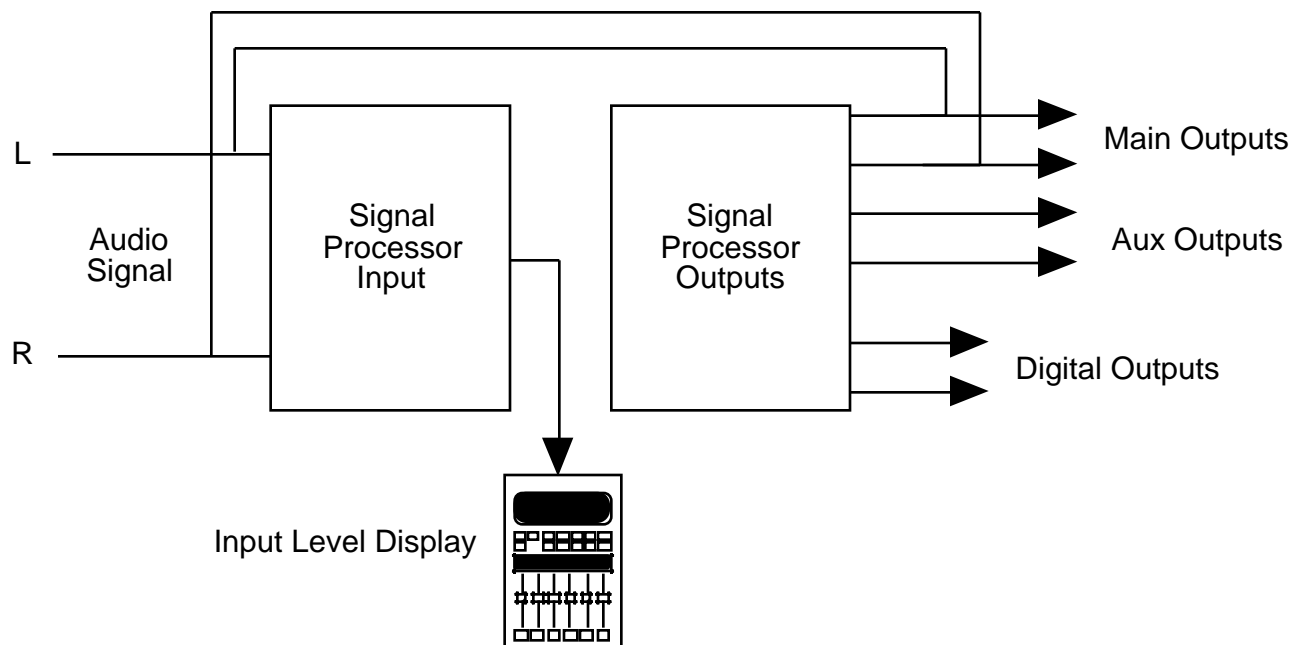
## Setting Analog Input Levels

Once output levels have been calibrated as directed in the previous section, you can proceed to the input levels.

**Before proceeding, make sure that the Oscillator program is loaded. Severe feedback can occur if any other program is running.**

1. Connect Main outputs L&R to the main inputs L&R as shown below.
2. Press the button below the ILVL slider. The LARC will display input level over a 90dB range from dBFS.
3. Adjust analog input gain controls until the LARC LED display level reads 0dB.
4. Move the WAVE slider until 2 is displayed.
5. Press the button below ILEV to display input level.
6. Fine trim analog input gain controls until 78.0dB is displayed for both left and right channels.

Analog signals are calibrated for Unity Gain.



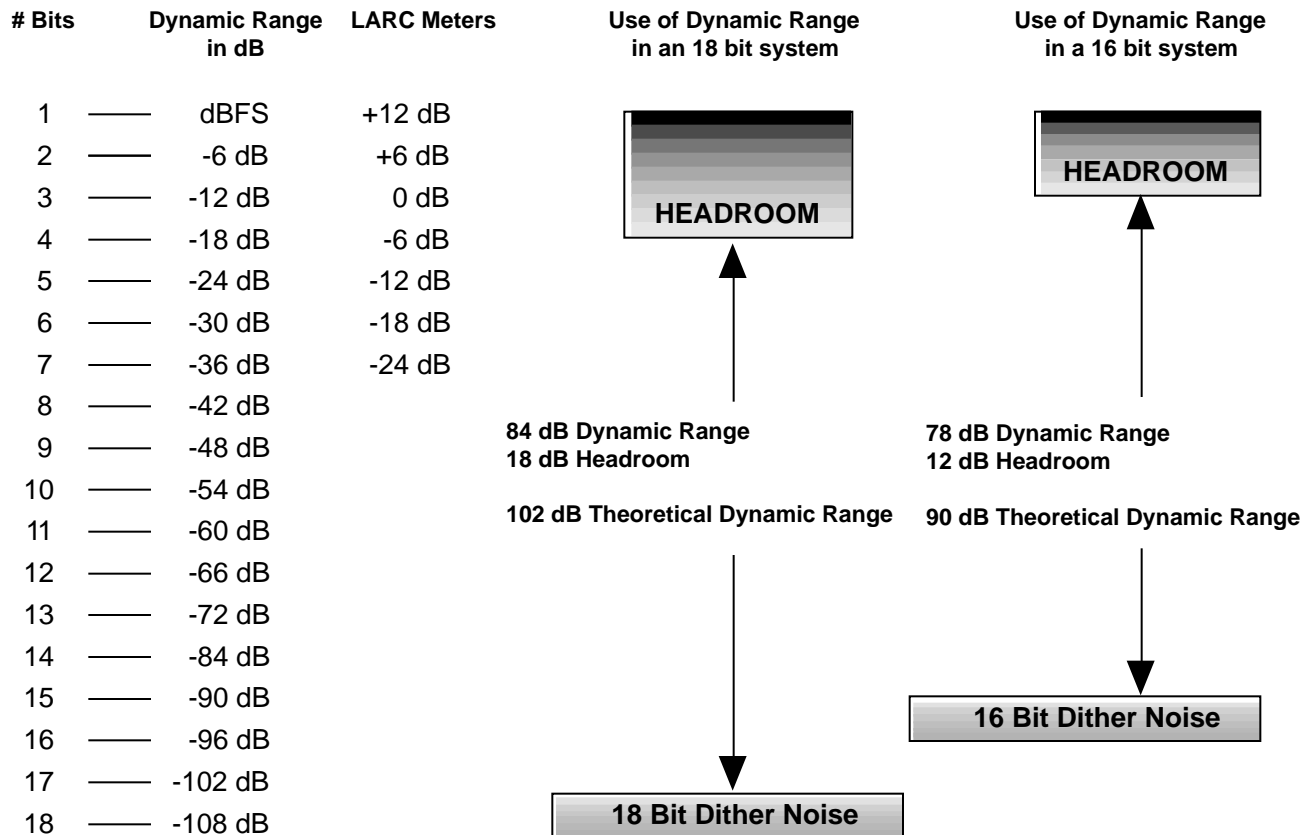
Signal Flow of Oscillator Program — Single Configuration

## Levels in the Digital Domain

Analog signals, by nature, have an infinite range of level. During conversion to digital, levels are quantized to absolute values, and expressed as a number of bits.

The 480L provides 18-bit resolution in both the analog and digital paths. In the digital domain, it is important to provide adequate headroom so that peak amplitude does not exceed dBFS. The difference between the headroom provided and the dither noise, or least significant bit, is the usable dynamic range of the system.

As you can see in the figure below, the 480L provides signal-to-noise and dynamic range that exceed many popular digital recording media.



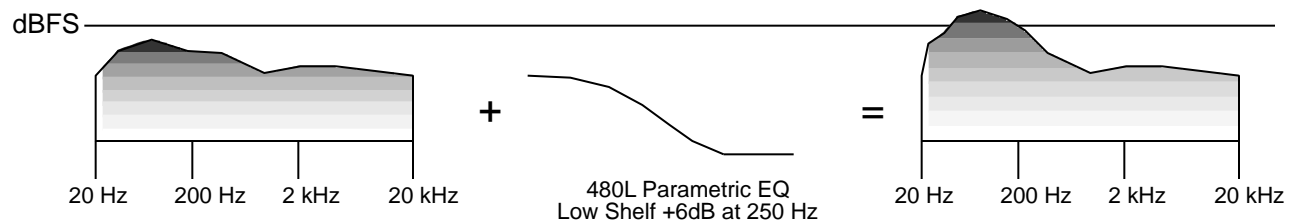
Dynamic Range of Digital Signal Path

## Using Digital Signals

Several programs in the 480L allow you to increase the level of the digital signal beyond dBFS. For example, if the Parametric EQ program is loaded, and a low shelf filter is boosted +6dB at 250Hz, the potential headroom to dBFS is reduced accordingly. (See the figure below.)

If the audio material fed through this program contains a large amount of energy in the region of the shelf boost, an overload in the digital domain may occur. When this happens, the last segments of the LARC display will illuminate.

Many programs, such as Parametric EQ, have a master level control which can offset the overall gain to prevent this situation. If you consistently encounter such situations, contact Lexicon Customer Service.



## 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).

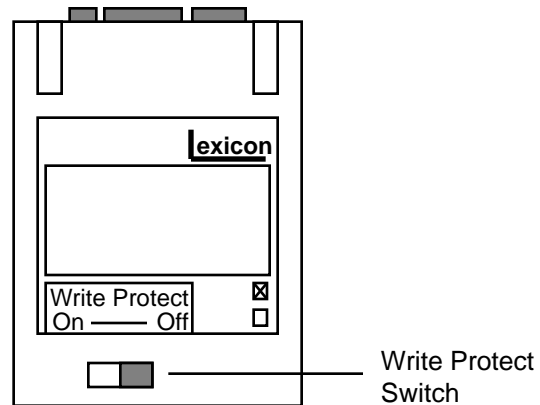
## How to Use Registers

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 programs. Registers are organized into banks, selected, and loaded exactly like the 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.**





## Storing and Naming Programs

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

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. Press CTRL to enter the control mode.
2. Press PAGE, then press 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.

To store your newly-named program in a register:

1. Press REG once to exit Control mode, and 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 Appendix B 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 program 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.

## 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.**

## MIDI SysEx Program Dumps

Programs and presets can also be transmitted and received as MIDI SysEx data. Parameters that enable MIDI SysEx dumps are found on Page 6 in Control mode.

### SysEx Bulk Dump

Slider 1 (SFN) selects a SysEx Bulk Dump function. Slider 2 (SRC) is dependent on the SFN setting, but, in general, it selects the source of the dump function when required. The following SFN settings are available:

- SETUP
- BANK
- ACTIVE
- ALL INT
- ALL CART
- MIDI MAP
- CONTROLS

### Individual Program or Preset Dumps

The SETUP function will bulk dump a single program or preset. When in this mode the SRC slider selects the particular setup to dump. Moving the slider scrolls through all of the available programs and registers containing presets. After selecting a particular program or preset, push and hold the STO button. While holding STO, press REG. This will enable a dump of the selected program or preset from either SFN or SRC mode. The format of the setup bulk dump message is described in Appendix A *MIDI and the 480L*.

### Dumping Banks

The BANK function will bulk dump a whole bank of programs or presets. The SRC slider selects from among the program banks and all ten register banks (five internal and five external cartridge banks). The bank selected is then dumped by pushing STO and REG as in individual setup dumps. A bank is dumped as a contiguous group of ten programs or registers. Each program or register within the bank, is dumped as a separate SysEx message, using the same format as that for individual dumps. When the dump is activated, the LARC displays "DUMPING BANK." When the dump is complete, the LARC displays "BANK DUMPED."

### Dump Active Algorithm(s)

The ACTIVE function dumps one or both of the active algorithms and their current parameter values instead of a register or preset. When the machine is in SINGLE configuration, only Machine A can be dumped, otherwise Machine A, Machine B, or both A and B can be selected with the SRC slider. As with all dumps described earlier, the dump is activated by the combination STO/REG button push. The format of the dump message is described in Appendix A *MIDI and the 480L*.

If the SRC slider is set to "MACHINE A+B" each machine will be sent as an individual SysEx message, just as though you selected "MACHINE A" and did a dump, then repeated the process for "MACHINE B". When the dump operation is activated, the LARC displays "DUMPING ACT A" for Machine A only. The LARC displays "DUMPING ACT B" for Machine B only, or "DUMPING A+B" when both machines are selected. When each Machine is dumped, the LARC displays "ACT A(or B) DUMPED"

### Dump All Internal Registers

To dump all internal registers to MIDI set SFN to "ALL INT". In this mode the SRC slider is not required. The "ALL INT" dumps register banks from 1-5 to MIDI. Each register is dumped as a single SysEx message described in Appendix A *MIDI and the 480L*. The presets are dumped in order of appearance in the register banks. As other functions, the dump action is initiated by holding STO and pressing REG. Once the dump of all internal registers is started, the LARC displays "DUMPING INT." When the dump is completed, the LARC displays "INT DUMPED."

### Dump All Cartridge Registers

The dump all cartridge function is very similar to the Dump All Internal Registers function with Banks 6-0 dumped rather than Banks 1-5. The LARC displays "DUMPING CART." When the dump is completed the LARC displays "CART DUMPED."

### Dump MIDI Program Table Map

The 480L has a program map that translates MIDI program change numbers to 480L program or register numbers. There is a map for each Machine, A and B. As in the ACTIVE function, the SRC slider selects from three possibilities: Machine A, Machine B or Machine A +B. The MIDI MAP bulk dump function dumps the selected map(s) to the MIDI port. Dump action is started holding STO and pressing REG. The format for the MIDI map is documented in Appendix A *MIDI and the 480L*. The LARC displays "DUMPING MAP A," "DUMPING MAP B," or "DUMPING MAPS," depending on selection. When the dump is complete, the LARC displays "MAP (A, B or A+B) DUMPED."

### Dump Controls

The SRC slider is not needed for this function. After moving the slider to CONTROLS and pushing STO/REG, a copy of the current parameters available in Control Mode are dumped. The following is a list of the controls dumped:

- Configuration
- Sample Rate
- Clock Source
- Input Source
- Audio Mute
- Reg Protect
- Sysex Channel Device ID
- Sysex Automation Mode
- Sysex Receive Mode

When dumping is activated, the LARC displays "DUMPING CTRLS." When complete, the LARC displays "CTRLS DUMPED."

### SysEx Restore Bulk Dump

Slider 3, on Page 6, labeled RCV, selects the destination of MIDI bulk dumps loaded back into the 480L. The simplest destination is the default, "ORIGINAL SRC". This function restores presets to wherever they originated. An internal register is written back over the current register at that location. See the message format documentation to discover the source for each register. Programs or presets can also be restored to the active Machines. If the configuration is SINGLE, then only Machine A can be restored. Otherwise, any program or register loaded to the 480L can be sent to either Machine A or Machine B with "ACTIVE A" or "ACTIVE B" selected under RCV.

The bank modes are more complicated. If any of the "INT BNK" or "CART BNK" selections are made, any preset will be sent to the first register in the selected bank. The 480L will attempt to place subsequent presets sequentially into the next registers in the bank selected. This should only be used to restore dumped banks.

### Restore Individual Presets

Loading a previously dumped preset will replace the preset at the location with the MIDI SysEx version when the RCV slider is set to "ORIGINAL SRC". Other settings in the RCV slider cause other actions to occur: "ACTIVE A" or "ACTIVE B" sends the preset to the appropriate machine as a running program

Selecting one of the banks, loads the preset into the first register of the bank, or into a subsequent register if it was dumped in sequence with a previously restored preset.

**Any attempt to restore a program to its original source will be ignored.** The error message "PROG RCVD" will be displayed.

When presets are successfully restored, the LARC will display an appropriate message for each type:

"ACT A LOADED" or "ACT B LOADED" indicates that an active algorithm was restored.

If a preset was received as an Internal Register, "1 REG LOADED" is displayed.

If a preset is received as a Cartridge register, "C REG LOADED" is displayed.

### Restore Banks

A dumped register bank will be restored to its original location if "ORIGINAL SRC" is selected. If the bank was a program bank, restoring to original source will not change anything in the 480L.

If the RCV slider is set to an active Machine, all of the programs or presets in the bank will load one at a time into the selected Machine until the last one, which will be left running when the process is complete.

When completed, the LARC will display "ACT A LOADED," or "ACT B LOADED," depending on the Machine selected.

If a new bank is selected, it will be written to the new location. The LARC will display either "1 REG LOADED," or "C REG LOADED," depending on its destination.

### Restore Active Machines

Any dump of an active Machine will restore as that active machine if the RCV slider is set to "ORIGINAL SRC." If the RCV slider is set to "MACHINE A" or "MACHINE B," the load will be to that machine, despite any destination stored in the message itself.

When RCV is set to restore to a particular bank, then the bulk dump data will replace the first register in the bank unless the data for machine A and B are sent sequentially, in which case the data will replace register 1 and 2 in the selected bank.

The LARC displays either "1 REG LOADED," or "C REG LOADED," depending on its destination.

### Restore All Internal Registers

If all of the dumped internal registers are restored with the RCV slider set to "ORIGINAL SRC," each register will be overwritten. Any other setting of RCV is not recommended.

The LARC will display "1 REG LOADED."

### Restore All Cartridge Registers

Restoring all cart registers works identically to the restore all internal registers except that in "ORIGINAL SRC" mode the data is sent to banks 6-0 instead of to 1-5. The LARC displays "C REG LOADED." All other operations are the same.

### Restore MIDI Program Table Map

Selection of destination is not an option when restoring MIDI program table map. The data is restored to the active Machine where it originated.

### Restore Controls

Selection of destination is not an option when restoring MIDI Controls. The data is restored to the control parameters of the active Machine. See Dump Controls for a list of the controls dumped and restored.

### MIDI Request Data

The request data function is a MIDI SysEx message that triggers an output of the requested bulk dump on the MIDI port. The format for the request function is defined in Appendix A *MIDI and the 480L*. The possible requests are as follows:

- Request All Stored Presets (Internal and Cartridge Registers)
- Request Individual Program
- Request Individual Preset
- Request Controls
- Request Active Machine A
- Request Active Machine B
- Request MIDI Program Map for Machine A
- Request MIDI Program Map for Machine B

The data will be sent after the 480L receives and decodes the message. If automation transmit mode is on, the data sent will also contain automation messages. Internally, the 480L decodes the request message and uses the normal event passing mechanism to transmit the response. This activates the automation mode if enabled. To get just the data requested, turn off the automation transmit mode on the control page, or by transmitting a SysEx message before sending the request.

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## Banks 1-4: the Reverb Programs

In this chapter we'll discuss the original Reverberation and Plate programs created for the 480L.

Two different algorithms are used to create these programs. The primary difference between the two is the density of the reverberation. The algorithm with greater density is used primarily for Room and Plate programs. To discover which algorithm is used by a particular program, go to page three to see how many pre-echo voices are available — the dense algorithm has only two voices.

Both algorithms have fixed, or "static" reverberation characteristics. A new algorithm, with random motion characteristics is described in *Banks11 -12: Random Halls and Rooms*. The Random Hall algorithm provides a smoother reverberant characteristic, and is better suited for material which requires large space emulation, or longer reverb time.

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 an 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 RTMID 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 undesirable. 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.



<b>Page One</b>					
RTM Mid Reverb Time	SHP Shape	SPR Spread	SIZ Size	HFC High Freq Cutoff	PDL Predelay
<b>Page Two</b>					
BAS Bass Multiply	XOV Crossover	RTC RT Hi Freq. Cutoff	DIF Diffusion	DCO Decay Optimization	MIX Wet/Dry Mix
<b>Page Three</b>					
[ Echo Level 1 L>L	PRE- Echo Level 2 R>R	ECHO Echo Level 3 R>L	-LEV Echo Level 4 L>R	ELS- Echo Level 5 L>L	] Echo Level 6 R>R
<b>Page Four</b>					
[ Echo Delay 1	PRE- Echo Delay 2	ECHO Echo Delay 3	-DEL Echo Delay 4	AY5 Echo Delay 5	] Echo Delay 6

## About the Reverberation Parameters

### Page One

#### RTM (Mid-Frequency Reverb Time)

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

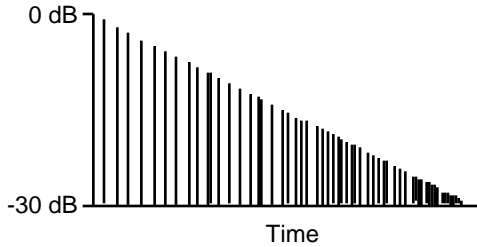
#### SPREAD

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.

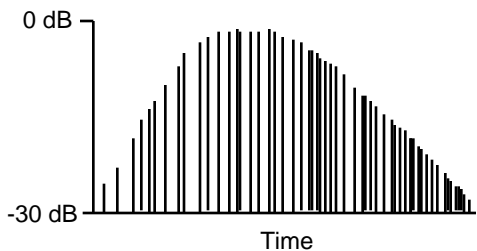
SPREAD and SHAPE control the rate at which reverberation builds up, and how the reverberation sustains as it begins to decay. When DECAY OPT is in Reverb mode, SPREAD is linked to SIZE, and the actual value for SPREAD depends on the selected SIZE.

#### SIZE

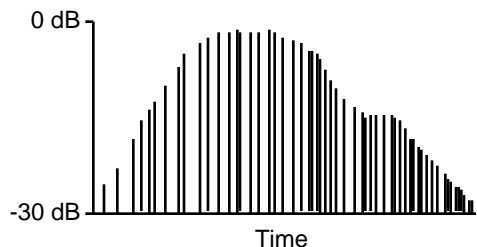
SIZE sets the rate of buildup of diffusion after the initial period (which is controlled by DIFFUSION). It also acts as a master control for RT MID and SPREAD. For this reason, the SIZE control can be used to vary a reverb sound from very large to very small. Generally, you should set the SIZE control 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.



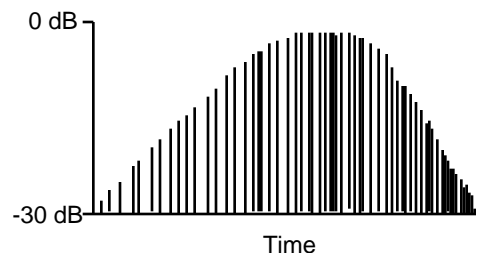
SHAPE Set All the Way Down.



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



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



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

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.

### 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.

## Programs — Bank 1: Halls

### 1 Large Hall

RTM	SHP	SPR	SIZ	HFC	PDL
2.19 s	126	179	37 m	2.862	24 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.2 x	752 Hz	4.186	99	R 7	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 2 Large+Stage

RTM	SHP	SPR	SIZ	HFC	PDL
2.19 s	126	179	37 m	2.862	24 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.2 x	752 Hz	4.186	99	R 7	All Fx
Preecho Levels					
-12 dB	-10 dB	-8 dB	-8 dB	-9 dB	-9 dB
Preecho Delays					
16 ms	22 ms	64 ms	56 ms	112 ms	102ms

### 3 Medium Hall

RTM	SHP	SPR	SIZ	HFC	PDL
1.74 s	126	128	25 m	4.395	24 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.2 x	752 Hz	3.982	99	R 7	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 4 Medium+Stage

RTM	SHP	SPR	SIZ	HFC	PDL
1.74 s	126	128	25 m	4.395	24 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.2 x	752 Hz	3.982	99	R 7	All Fx
Preecho Levels					
-14 dB	-12 dB	-10 dB	-10 dB	-12 dB	-12 dB
Preecho Delays					
16 ms	22 ms	44 ms	38 ms	80 ms	76ms

### 5 Small Hall

RTM	SHP	SPR	SIZ	HFC	PDL
1.13 s	96	50	25 m	4.395	24 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	3.784	99	R 7	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 6 Small+Stage

RTM	SHP	SPR	SIZ	HFC	PDL
1.13 s	96	50	25 m	4.395	24 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	3.784	99	R 7	All Fx
Preecho Levels					
-12 dB	-12 dB	-10 dB	-10 dB	-14 dB	-14 dB
Preecho Delays					
12 ms	18 ms	44 ms	36 ms	72 ms	52ms

### 7 Large Church

RTM	SHP	SPR	SIZ	HFC	PDL
4.04 s	85	247	38 m	2.523	30 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.5 x	1.02 Hz	2.691	80	R 7	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 8 Small Church

RTM	SHP	SPR	SIZ	HFC	PDL
2.42 s	65	106	31 m	3.402	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	3.591	70	R 7	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 9 Jazz Hall

RTM	SHP	SPR	SIZ	HFC	PDL
1.26 s	34	98	23 m	12.177	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.2 x	752 Hz	5.538	80	R 7	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 0 Auto Park

RTM	SHP	SPR	SIZ	HFC	PDL
5.29 s	149	247	38 m	7.818	24 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	5.538	99	R 0	All Fx
Preecho Levels					
-6 dB	-6 dB	-9 dB	-9 dB	-12 dB	-12 dB
Preecho Delays					
22 ms	12 ms	44 ms	66 ms	164 ms	136ms

## **Program Descriptions**

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.

### **1 Large Hall**

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.

### **2 Large + Stage**

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 proscenium arch.

### **3 Medium Hall**

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

### **4 Medium + Stage**

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

### **5 Small Hall**

Small Hall is a smaller version of Medium Hall.

### **6 Small + Stage**

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

### **7 Large Church**

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

### **8 Small Church**

Small Church is a smaller version of program 7.

### **9 Jazz Hall**

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.

### **0 Auto Park**

Auto Park reproduces the sound of an underground parking garage.

## Programs — Bank 2: Rooms

### 1 Music Club

RTM	SHP	SPR	SIZ	HFC	PDL
1.03 s	40	55	25 m	7.181	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	3.784	78	R 7	All Fx
PDE	PDE				
Off	Off				
PDL	PDL				
0 ms	0 ms				

### 2 Large Room

RTM	SHP	SPR	SIZ	HFC	PDL
0.70 s	52	82	19 m	6.593	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0x	752 Hz	3.784	65	R 7	All Fx
PDE	PDE				
Off	Off				
PDL	PDL				
0 ms	0 ms				

### 3 Medium Room

RTM	SHP	SPR	SIZ	HFC	PDL
0.50 s	22	10	19 m	7.181	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	3.784	65	R 7	All Fx
PDE	PDE				
Off	Off				
PDL	PDL				
0 ms	0 ms				

### 4 Small Room

RTM	SHP	SPR	SIZ	HFC	PDL
0.31 s	16	0	10 m	7.181	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	3.784	60	R 6	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 5 Very Small

RTM	SHP	SPR	SIZ	HFC	PDL
0.13 s	8	0	4 m	7.181	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	3.784	55	R 0	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 6 Lg Wood Rm

RTM	SHP	SPR	SIZ	HFC	PDL
1.33 s	73	34	23 m	8.513	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
0.8 x	1.158 Hz	5.538	82	R 7	All Fx
PDE	PDE				
Off	Off				
PDL	PDL				
0 ms	0 ms				

### 7 Sm Wood Rm

RTM	SHP	SPR	SIZ	HFC	PDL
0.71 s	45	19	13 m	8.513	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
0.8 x	1.158 Hz	5.538	82	R 7	All Fx
PDE	PDE				
Off	Off				
PDL	PDL				
0 ms	0 ms				

### 8 Lg Chamber

RTM	SHP	SPR	SIZ	HFC	PDL
0.88 s	3	0	20 m	7.181	10 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	6.047	99	R 6	All Fx
PDE	PDE				
Off	Off				
PDL	PDL				
0 ms	0 ms				

### 9 Sm Chamber

RTM	SHP	SPR	SIZ	HFC	PDL
0.36 s	16	0	10 m	7.181	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	752 Hz	3.784	70	R 6	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 0 Small & Bright

RTM	SHP	SPR	SIZ	HFC	PDL
0.65 s	40	39	9 m	10.591	6 ms
BAS	XOV	RTC	DIF	DCO	MIX
0.8 x	621 Hz	7.493	81	R 7	All Fx
Preecho Levels					
-14 dB	-14 dB	-14 dB	-14 dB	Off	Off
Preecho Delays					
14 ms	10 ms	28 ms	44 ms	0 ms	0 ms

## **Program Descriptions**

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. If you want to closely match the ambient characteristics of a space, try using the programs found in the *Ambience* bank. The Ambience algorithm was designed for this application.

### **1 Music Club**

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

### **2 Large Room**

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

### **3 Medium Room**

Medium Room is a smaller version of Large Room.

### **4 Small Room**

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

### **5 Very Small Room**

Very Small Room has the intimate, close feel of a bedroom or den.

### **6 Large Wood Room**

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

### **7 Small Wood Room**

Small Wood Room is a smaller version of program 6.

### **8 Large Chamber**

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.

### **9 Small Chamber**

Small Chamber is a smaller version of program 8.

### **0 Small and Bright**

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



## Programs — Bank 3: Wild Spaces

### 1 Brick Wall

RTM	SHP	SPR	SIZ	HFC	PDL
0.24 s	0	254	26 m	10.591	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.5 x	1.886 Hz	Full R.	88	E 7	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 2 Buckram

RTM	SHP	SPR	SIZ	HFC	PDL
0.24 s	94	61	24 m	6.882	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.5 x	1.886 Hz	Full R.	98	E 7	All Fx
PDE	PDE				
-6 dB	-6 dB				
PDL	PDL				
64 ms	40 ms				

### 3 Big Bottom

RTM	SHP	SPR	SIZ	HFC	PDL
0.89 s	66	210	31 m	11.084	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
4.0 x	243 Hz	Full R.	88	E 7	All Fx
Preecho Levels					
Off	Off	Off	Off	Off	Off
Preecho Delays					
0 ms	0 ms	0 ms	0 ms	0 ms	0 ms

### 4 10W-40

RTM	SHP	SPR	SIZ	HFC	PDL
0.78 s	10	88	19 m	Full R.	4 ms
BAS	XOV	RTC	DIF	DCO	MIX
4.0 x	885 Hz	1.886	99	E 7	All Fx
Preecho Levels					
Full Up	Full Up	-5 dB	-3 dB	Off	Off
Preecho Delays					
0 ms	0 ms	26 ms	46 ms	0 ms	0 ms

### 5 20W-50

RTM	SHP	SPR	SIZ	HFC	PDL
1.01 s	152	94	23 m	11.084	4 ms
BAS	XOV	RTC	DIF	DCO	MIX
4.0 x	621 Hz	621	99	E 7	All Fx
Preecho Levels					
Full Up	Full Up	-5 dB	-3 dB	Off	Off
Preecho Delays					
0 ms	0 ms	50 ms	64 ms	0 ms	0 ms

### 6 Metallica

RTM	SHP	SPR	SIZ	HFC	PDL
0.97 s	57	187	28 m	14.986	14 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.5 x	1.020 Hz	7.493	90	E 7	All Fx
Preecho Levels					
-7 dB	-7 dB	-18 dB	-12 dB	-18 dB	-20 dB
Preecho Delays					
70 ms	88 ms	136 ms	156 ms	284 ms	276

### 7 Silica Beads

RTM	SHP	SPR	SIZ	HFC	PDL
5.46 s	126	252	37m	9.278	24 ms
BAS	XOV	RTC	DIF	DCO	MIX
0.2 x	4.395 Hz	Full R.	80	E7	All Fx
PDE	PDE				
Off	Off				
PDL	PDL				
64 ms	40 ms				

### 8 Inside Out

RTM	SHP	SPR	SIZ	HFC	PDL
1.36 s	243	112	20 m	10.591	22 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.2 x	752 Hz	4.611	99	E 7	All Fx
PDE	PDE				
-14 dB	-14 dB				
PDL	PDL				
20 ms	22 ms				

### 9 Ricochet

RTM	SHP	SPR	SIZ	HFC	PDL
1.56 s	0	0	34m	14.986	18 ms
BAS	XOV	RTC	DIF	DCO	MIX
0.6 x	1.735 Hz	10.127	90	E 7	All Fx
PDE	PDE				
-12 dB	-10 dB	Off	Off	Off	Off
PDL	PDL				
378 ms	322 ms				

### 0 Varoom

RTM	SHP	SPR	SIZ	HFC	PDL
0.78 s	255	216	28m	12.177	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
2.0 x	621 Hz	12.177	98	E7	All Fx
PDE	PDE				
Off	Off	Off	Off	Off	Off
PDL	PDL				
14 ms	18 ms				

## **Program Descriptions**

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.

### **1 Brick Wall**

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.

### **2 Buckram**

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.

### **3 Big Bottom**

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.

### **4 10W-40**

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.

### **5 20W-50**

A more aggressive oil drum.

### **6 Metallica**

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

### **7 Silica Beads**

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.

### **8 Inside Out**

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

### **9 Ricochet**

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

### **0Varoom**

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

## Programs — Bank 4: Plates

### 1 A Plate

RTM	SHP	SPR	SIZ	HFC	PDL
2.00 s	0	0	20 m	8.513	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
0.6 x	752 Hz	Full R.	97	R 0	All Fx
PDE	PDE				
-8 dB	-9 dB				
PDL	PDL				
14 ms	18 ms				

### 2 Snare Plate

RTM	SHP	SPR	SIZ	HFC	PDL
1.84 s	1	0	16 m	Full R.	60 ms
BAS	XOV	RTC	DIF	DCO	MIX
0.6 x	120 Hz	Full R.	95	R 0	All Fx
PDE	PDE				
-9 dB	-12 dB				
PDL	PDL				
110 ms	152ms				

### 3 Small Plate

RTM	SHP	SPR	SIZ	HFC	PDL
1.65 s	0	6	18 m	15.886	2 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	885 Hz	10.127	99	R 0	All Fx
PDE	PDE				
-2 dB	-4 dB				
PDL	PDL				
10 ms	6 ms				

### 4 Thin Plate

RTM	SHP	SPR	SIZ	HFC	PDL
1.59 s	0	0	15 m	Full R.	0 ms
BAS	XOV	RTC	DIF	DCO	MIX
0.6 x	752 Hz	15.886	85	R 0	All Fx
PDE	PDE				
-6 dB	-6 dB				
PDL	PDL				
14 ms	18 ms				

### 5 Fat Plate

RTM	SHP	SPR	SIZ	HFC	PDL
1.98 s	97	130	34 m	9.278	2 ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0 x	1.586 Hz	21.181	75	R 0	All Fx
PDE	PDE				
-6 dB	-9 dB				
PDL	PDL				
30 ms	30 ms				

## Program Descriptions

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. Programs for additional plates are found in the Classic Cart software option.

### 1 A Plate

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

### 2 Snare Plate

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.

### 3 Small Plate

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

### 4 Thin Plate

Another variation on the plate theme.

### 5 Fat Plate

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

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## **Bank 5: the Effects Programs**

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

<b>Page One</b>					
SPN Spin	SLP Slope	LNG Length	WAN Wander	NUM Number	PDL Predelay
<b>Page Two</b>					
MON Input Blend	FBL Feedback Level	FBD Feedback Delay	DIF Diffusion	IND Input Delay	MIX Wet/Dry Mix
<b>Page Three</b>					
HPL High Pass L	HPR High Pass R	SGN Signs			

## About the Effects Program

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-varying 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.

The unique way in which the 40-voice effects algorithm processes these voices provides a chorus that does not change pitch. This is extremely useful on material such as grand piano, where detuning from standard chorusing yields unacceptable results.

Delay times can be combined in phase, or out-of-phase, to change the timbre of the overall effect.

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 accommodate 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-varying 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.

## 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.

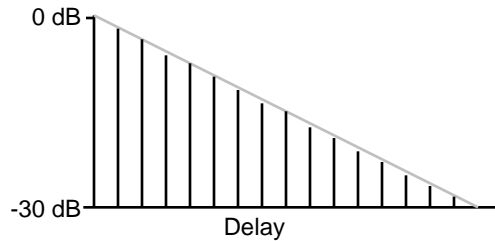


Figure 4.2. SLOPE All the Way Down.

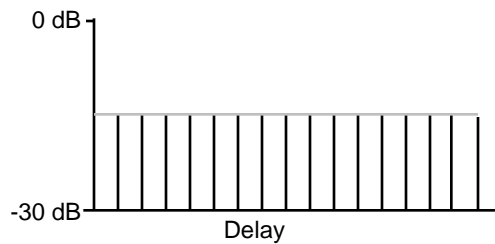


Figure 4.3. SLOPE Midway Up.

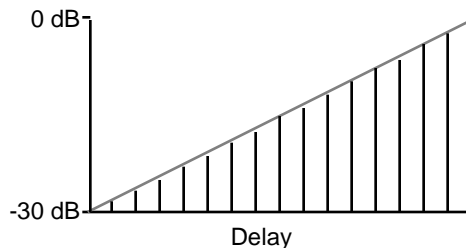


Figure 4.4. SLOPE All the Way Up.

### **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.

### **SIGNS**

When the Signs parameter is set to 1, a significant increase in output gain can occur.

## Programs — Bank 5: Effects

### 1 Illusion

SPN	SLP	LNG	WAN	NUM	PDL
30	154	0 ms	42 us	40	20 ms
MON	FBL	FBD	DIF	IND	MIX
Stereo	Off	0 ms	64	52 ms	All Fx
HPL	HPR	SGN			
0 Hz	0 Hz	0			

### 2 Surfin

SPN	SLP	LNG	WAN	NUM	PDL
44	247	0 ms	882 us	40	20 ms
MON	FBL	FBD	DIF	IND	MIX
Stereo	Off	0 ms	64	0 ms	All Fx
HPL	HPR	SGN			
120 Hz	120 Hz	0			

### 3 Voc. Whisper

SPN	SLP	LNG	WAN	NUM	PDL
48	106	38 ms	23 ms	28	8 ms
MON	FBL	FBD	DIF	IND	MIX
Stereo	-16 dB	412 ms	44	244 ms	76%
HPL	HPR	SGN			
0	0	0			

### 4 Doubler

SPN	SLP	LNG	WAN	NUM	PDL
44	170	16 ms	26 ms	16	48 ms
MON	FBL	FBD	DIF	IND	MIX
Stereo	-18 dB	10 ms	82	0 ms	All Fx
HPL	HPR	SGN			
0	0	0			

### 5 Back Slap

SPN	SLP	LNG	WAN	NUM	PDL
43	219	86 ms	10 ms	40	38 ms
MON	FBL	FBD	DIF	IND	MIX
Stereo	Off	0 ms	52	0 ms	All Fx
HPL	HPR	SGN			
0	0	0			

### 6 Rebound

SPN	SLP	LNG	WAN	NUM	PDL
48	254	444 ms	35 ms	40	254 ms
MON	FBL	FBD	DIF	IND	MIX
Stereo	-9 dB	0 ms	0	122 ms	64%
HPL	HPR	SGN			
0	0	0			

### 7 Git It Wet

SPN	SLP	LNG	WAN	NUM	PDL
17	187	80 ms	35 ms	8	80 ms
MON	FBL	FBD	DIF	IND	MIX
Stereo	-12 dB	44 ms	59	0 ms	All Fx
HPL	HPR	SGN			
0 Hz	0 Hz	1			

### 8 Sudden Stop

SPN	SLP	LNG	WAN	NUM	PDL
48	130	216	37 ms	40	52 ms
MON	FBL	FBD	DIF	IND	MIX
Stereo	-18 dB	48 ms	99	0 ms	All Fx
HPL	HPR	SGN			
0	0	0			

### 9 In the Past

SPN	SLP	LNG	WAN	NUM	PDL
45	247	500 ms	13 ms	40	0 ms
MON	FBL	FBD	DIF	IND	MIX
Stereo	Off	0 ms	87	504 ms	52%
HPL	HPR	SGN			
0	0	0			

### 0 Tremolo L & R

SPN	SLP	LNG	WAN	NUM	PDL
30	135	0 ms	0 us	4	0 ms
MON	FBL	FBD	DIF	IND	MIX
Mono B	Off	0 ms	0	0 ms	All Fx
HPL	HPR	SGN			
0	0	0			



## Program Descriptions

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.

### 1 Illusion

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.

### 2 Surfin'

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

### 3 Vocal Whispers

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

### 4 Doubler

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.

### 5 Back Slap

A strong fast slapback effect.

### 6 Rebound

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

### 7 Git It Wet

It's Saturday afternoon in the guitar section of a large music store. — Just add the metal guitar.

### 8 Sudden Stop

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.

### 9 In the Past

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.

### 0 Tremolo L and R

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 tremolos 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.

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## **Bank 6: the Twin Delays Programs**

This chapter describes the Twin Delays programs located in Bank 6.

<b>Page One</b>					
DRY L Ch Dry Level	DRY R Ch Dry Level	ROL L DLY1 Rolloff	ROL R DLY1 Rolloff	PAN L Channel Pan	PAN R Channel Pan
<b>Page Two</b>					
DL1 L DLY1 Value	LV1 L DLY1 Level	FB1 L DLY1 Feedback	DL2 R DLY2 Value	LV2 R DLY 2 Level	FB2 R DLY2 Feedback
<b>Page Three</b>					
DL3 L DLY3 Value	LV3 L DLY3 Level	FB3 L DLY3 Feedback	DL4 R DLY4 Value	LV4 R DLY4 Level	FB4 R DLY4 Feedback
<b>Page Four</b>					
FIN L Fine Delay	FIN R Fine Delay	MST Delay Multiplier			

## The Twin Delays Program

The Twin Delays program located in Bank 6 is a four voice delay line with independently-adjustable level, feedback, and delay time for each voice. Feedback can be positive or negative. Feedback for Delays 3 and 4 is cross-panned.

Independent low pass filters, adjustable between 120Hz and full bandwidth, are provided for the first and second delay voices (Delay 1 Left Channel and Delay 2 Right Channel) and their respective feedback paths. These voices also have independent panning controls.

### About the Twin Delays Parameters

#### Page One

##### **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.

##### **ROL (Rolloff)**

Rolloff is a low pass filter that can be adjusted independently for LDLY1 and RDLY2 voices. Values for Rolloff are adjustable between 120Hz and full range.

##### **L PAN (Left Channel Pan)**

L PAN sets the panning of the DL1 and FB1 signal to the left and right outputs.

##### **R PAN (Right Channel Pan)**

R PAN sets the panning of the DL2 and FB2 signal to the left and right outputs.

#### Page Two

##### **DL1 (LDLY1 Val)**

Sets the delay time for the first (left channel) delay voice.

##### **LV1 (LDLY1 Lvl)**

Adjusts the level (amplitude) of DL1.

##### **FB1 (LDL1 Fbk)**

Adjusts the amount of feedback (positive or negative) around DL1.

##### **DL2 (RDLY2 Val)**

Sets the delay time for the second (right channel) delay voice.

##### **LV2 (RDLY2 Lvl)**

Adjusts the level (amplitude) of level for DL2.

##### **FB2 (RDL2 Fbk)**

Adjusts the amount of feedback (positive or negative) around DL2.

## Page Three

### **DL3**

Sets the delay time for the third (left channel) delay voice.

### **LV3**

Adjusts the level (amplitude) of DL3.

### **FB3**

Adjusts the level of a cross-panned (L-R) feedback line. Feedback can be positive or negative.

### **DL4**

Sets the delay time for the fourth (right channel) delay voice.

### **LV4**

Adjusts the level (amplitude) of DL4.

### **FB4**

Adjusts the level of a cross-panned (R-L) feedback line. Feedback can be positive or negative.

## Page Four

### **FIN L**

Sets the delay value of the left channel fine delay in samples.

### **FIN R**

Sets the delay value of the right channel fine delay in samples.

### **MASTER**

MASTER is a delay multiplier for all delay voices. It multiplies all delay values (DL1-4).

## Program Descriptions

### **1 4-Voice Double**

Delay voices are doubled in stereo. When added to dry signal, it's crisp, wide, and uncluttered.

### **2 Double Delay**

Two voices produce a double effect. The other two provide a longer delay synced with the double. Cross panned feedback ices the cake.

### **3 4-Bounce Delay**

Between left and right channels, Very clean.

### **4 Pitter Patter**

Delays are widely spaced with reiterative and cross panned feedback.

### **5 X-Pan Double**

Two voices are cross panned through delays. Great for stereo background vocals.

### **6 Delay Cave**

Need we say more?

### **7 Circles**

Long delays with cross panned feedback that creates a "circular" effect.

### **8 There & Back**

Delay starts on one channel, slaps to the other, and returns.

### **9 Soft Roller**

Stereo echo with high frequency cut.

### **0 On and On**

Long echoes that pan across center.

## Programs — Bank 6: Twin Delays

### 1 4-Vce Double

DRY	DRY	ROL	ROL	PAN	PAN
Off	Of	Full	Full	Left	Right
DLY1	LV1	FB1	DLY2	LV2	FB2
32.03ms	0dB	0%	21.04ms	0dB	0%
DLY3	LV3	FB3	DLY4	LV4	FB4
60.01ms	-6dB	0%	42.09ms	0dB	0%
FIN	FIN	MST			
0	0	1.00			

### 2 Double Delay

DRY	DRY	ROL	ROL	PAN	PAN
Off	Off	Full	Full	Left	Right
DLY1	LV1	FB1	DLY2	LV2	FB2
26.04ms	0dB	+25%	32.03ms	0dB	+25%
DLY3	LV3	FB3	DLY4	LV4	FB4
300.0ms	-4dB	+30%	600ms	-4dB	+30%
FIN	FIN	MST			
0	0	1.00			

### 3 4-Bounce Dly

DRY	DRY	ROL	ROL	PAN	PAN
Off	Off	Full	Full	Left	Right
DLY1	LV1	FB1	DLY2	LV2	FB2
100ms	0dB	0%	200ms	0dB	0%
DLY3	LV3	FB3	DLY4	LV4	FB4
300.0ms	-2dB	-30%	400.0ms	-2dB	+30%
FIN	FIN	MST			
0	0	1.00			

### 4 Pitter Patter

DRY	DRY	ROL	ROL	PAN	PAN
Off	Off	1158Hz	7181Hz	Left	Right
DLY1	LV1	FB1	DLY2	LV2	FB2
112.07ms	-6dB	0%	198.04ms	-6dB	0%
DLY3	LV3	FB3	DLY4	LV4	FB4
300.00ms	-15dB	+73%	954.0ms	-15dB	-75%
FIN	FIN	MST			
0	0	1.00			

### 5 X-Pan Double

DRY	DRY	ROL	ROL	PAN	PAN
Off	Off	Full	Full	Right	Left
DLY1	LV1	FB1	DLY2	LV2	FB2
30.03ms	0dB	0%	60.01ms	0dB	0%
DLY3	LV3	FB3	DLY4	LV4	FB4
0ms	Off	0%	0ms	Off	0%
FIN	FIN	MST			
0	0	1.00			

### 6 Delay Cave

DRY	DRY	ROL	ROL	PAN	PAN
Off	Off	8158Hz	8158Hz	42	58
DLY1	LV1	FB1	DLY2	LV2	FB2
21.04ms	0dB	0%	42.84ms	0dB	0%
DLY3	LV3	FB3	DLY4	LV4	FB4
100.0ms	0dB	+37%	80.03ms	0dB	+37%
FIN	FIN	MST			
0	0	1.00			

### 7 Circles

DRY	DRY	ROL	ROL	PAN	PAN
Off	Off	Full	Full	Left	Right
DLY1	LV1	FB1	DLY2	LV2	FB2
0ms	Off	0%	0ms	Off	0%
DLY3	LV3	FB3	DLY4	LV4	FB4
300.10ms	0dB	+40%	150ms	0ms	+40%
FIN	FIN	MST			
0	0	.50			

### 8 There & Back

DRY	DRY	ROL	ROL	PAN	PAN
Off	Off	9691Hz	9691Hz	Left	Right
DLY1	LV1	FB1	DLY2	LV2	FB2
102.0ms	-6dB	0%	198.0ms	-6dB	0%
DLY3	LV3	FB3	DLY4	LV4	FB4
300.1ms	-15dB	+73%	1000ms	-15dB	-75%
FIN	FIN	MST			
0	0	1.00			

### 9 Soft Roller

DRY	DRY	ROL	ROL	PAN	PAN
Off	Off	6882Hz	6882Hz	Left	Right
DLY1	LV1	FB1	DLY2	LV2	FB2
300.1ms	0dB	+26%	300.1ms	0dB	+26%
DLY3	LV3	FB3	DLY4	LV4	FB4
310.8ms	-4dB	-16%	310.8ms	-4dB	-16%
FIN	FIN	MST			
0	0	1.00			

### 0 On and On

DRY	DRY	ROL	ROL	PAN	PAN
Off	Off	5.788Hz	5.788Hz	Left	Right
DLY1	LV1	FB1	DLY2	LV2	FB2
600.8ms	0dB	+24%	600.8ms	0dB	+24%
DLY3	LV3	FB3	DLY4	LV4	FB4
1200ms	-6dB	+33%	1200ms	-6dB	-33%
FIN	FIN	MST			
0	0	1.00			

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## Bank 7: the Sampler Programs

The Sampler programs located in Bank 7 include a variety of useful features for recording and production work. The programs 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. The samplers can also be triggered and controlled via MIDI.

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 a number of sampling programs, each with different features and capabilities. The optional Sampling Memory Expander (SME) board can be inserted into the fourth card slot in the 480L, providing 10.9 seconds of true phase-locked stereo sampling time, or 21.8 seconds of mono sampling time at a 48kHz sampling rate.

As with all 480L programs, any sampling program can be loaded into either machine. This leaves the other machine free to process signals independently, or in conjunction with the sampling program, as in the following illustrations.

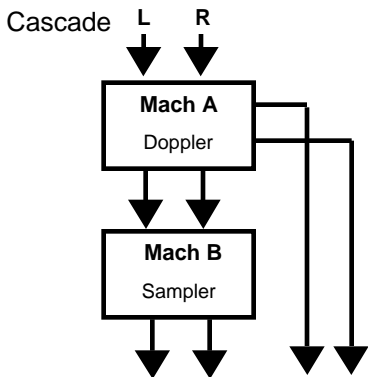
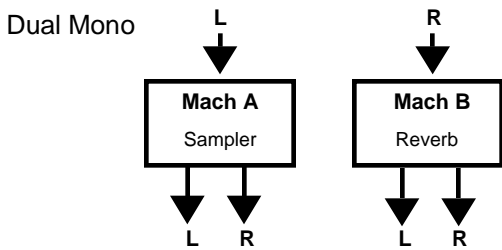
Programs 1-8 in Bank 7 do not require the SME option. They are:

Program #	Program Name	Sampling Time @ 48kHz
1	Stereo 3S	2.7 Sec.
2	Mono 6S	5.4 Sec.
3	Mono 3S	2.7 Sec.
4	Dual Rate Change	2 x 1.4 Sec.
5	Mono Fwd & Rev	1.4 Sec.
6	Stereo 3Sec Drum	2.7 Sec.
7	Dual Rate Change Drum	2 x 1.4 Sec.
8	Mono Fwd/Rev Drum	1.4 Sec.

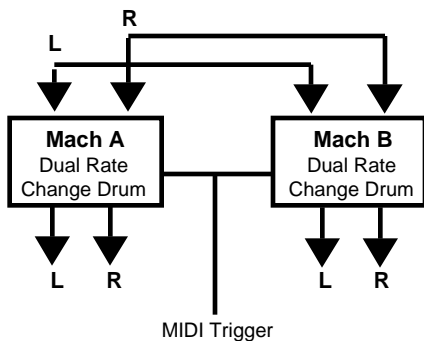
Programs 9 and 0 in Bank 7 require the SME card.

9	Stereo Rate Change 10S	10.9 Sec.
0	Mono Rate Change 20S	21.8 Sec.

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



After capturing Doppler in Sampler, program can be changed and used independently of Sampler with recorded effect.



4-Voice Drum Replace with MIDI playback trigger. Note value changes playback rate and pitch.

## 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 program 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 simultaneously. 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.

## Bank 7 Samplers – SME Only

### Stereo Rate Change/Overdub - Bank 7, Program 9

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 7, Program 0

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.



## 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, press 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 milliseconds) 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. Be sure 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 disable 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 directly 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.

Stereo 3S/Mono 6S, and Stereo Drum

<b>Page One</b>					
	REC Record	MARK Mark Capture	MODE Capture Mode	CAP Manual Trigger	CHK Check Sample
<b>Page Two</b>					
DUB Overdub	REC Record	FAD Fade Type	MODE Capture Mode	CAP Manual Trigger 2	CHK Check Sample
<b>Page Three</b>					
HED Head Trim	TIM Forward Time			FAD Fade Type	CUE Cue Sample
<b>Page Four</b>					
PLAY Play	RP Repeat	TLV Trigger	HOLD Retrigger Hold		CUE Cue Sample

Dual Rate Change

<b>Page One</b>					
VX Sample Voice Select	REC Record	MARK Mark Capture	MODE Capture Mode	CAP Manual Trigger	CHK Check Sample
<b>Page Two</b>					
HEAD Head Trim	TIM Sample1 Fwd Time		FAD Sample1 Fade Type	RAT Sample1 Plybk Rate	CUE Cue Sample1
<b>Page Three</b>					
HEAD Head Trim 2	TRM Sample2 Fwd Time		FAD Sample2 Fade Type	RAT Sample2 Plybk Rate	CUE Cue Sample2
<b>Page Four</b>					
PLAY Play Sample1	RP Repeat Sample1	TLV Trigger Sample1	PLAY Play Sample2	RP Repeat Samp2	TLV Trigger Sample2
<b>Page Five</b>					
MID MIDI Play Samp1	REF MIDI Note Sample1	RAT Samp1 Plybk Rate	MID MIDI Play Samp2	REF MIDI Note Samp2	RAT Samp2 Plybk Rate

Foward and Reverse

<b>Page One</b>					
	REC Record	MARK Mark Capture	MODE Capture Mode	CAP Manual Trigger	CHK Check Sample
<b>Page Two</b>					
HEAD Head Trim	TIM Sample1 Fwd Time	<> Play Order	TAIL Tail Trim	TIM Reverse Time	CUE Cue Sample
<b>Page Three</b>					
LVL Fwd Plybk Level	PAN Fwd Samp Panning	LVL Rev Plybk Level	PAN Rev Samp Panning	FAD Fade Type	CUE Cue Sample
<b>Page Four</b>					
PLAY Play Sample	RP Repeat	TLV Trigger Level			CUE Cue Sample

Stereo Rate Change 10 - Mono Rate Change 20

Page One					
DUB Overdub	REC Record	MARK Mark Capture	MODE Capture Mode	CAP Manual Trigger	CHK Check Sample
Page Two					
HED Head Trim	TAL Tail Trim	EDIT Edit	FAD Fade Type	RAT Rate	CUE Cue Sample
Page Three					
PLAY Play Sample	RP Repeat	TLV Trigger	RAT Rate	HOLD Retrigger Hold	CUE Cue Sample
Page Four					
MID MIDI Play	LVL Ref MIDI Velocity	REF Ref MIDI Note	MIR Pitch Mirror	RAT Rate	CUE Cue Sample

## 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.

A fine mode is provided for precise trim. Press the button under the Head Trim slider until the display flashes. To exit, press the button again and hold it until the display stops flashing.

### **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 semi-tone pitch intervals.

### **MIR**

Pitch Mirror modifies the behavior of MIDI Play by decreasing the rate for increasing note values. 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.

A fine mode is provided for precise trim. Press the button under the Head Trim slider until the display flashes. To exit, press the button again and hold it until the display stops flashing.

**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.



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## Bank 8: the Pitch Change and Doppler Programs

The Pitch Change program is a true stereo pitch shifter which displays pitch change in musical intervals. It can also operate as two independent mono programs in each channel. In addition, a delay line with independent feedback paths is available to create useful effects. Pitch parameters can be quickly linked to MIDI Note values; a mirror feature is also provided.

The Doppler program realistically separates the sound of a large object passing at high speed past a stationary listener. It is designed primarily for use in film and video post-production environments.

<b>Page One</b>					
MOD Stereo/Mono	SNC Play Sync	PCH Pitch L	FIN Fine Pitch L	PCH Pitch R	FIN Fine Pitch R
<b>Page Two</b>					
PDL L Predelay	PDR R Predelay	FBL L Feedback	FBR R Feedback		MIX Wet/Dry Mix
<b>Page Three</b>					
MIDI MIDI Pitch L		REF MIDI Ref Note	MIR Pitch Mirror	GLL Glide L	SPL Splice Time
<b>Page Four</b>					
MIDI MIDI Pitch R		REF MIDI Ref Note	MIR Pitch Mirror	GLR Glide R	

Figure 7.1. Pitch Change Parameters.

## The Pitch Change Programs

The Pitch Change program located in Bank 8 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.

##### 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  $\pm 99\%$ . Some very useful effects can be obtained by combining pitch shift, delay, and feedback.

## 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 current 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  $\pm$  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 mono mode.

MIDI (MIDI Pitch Event Right)

REF (Reference MIDI Note Right)

MIR (Pitch Mirror Right)

GLR (Glide Right)

## Program Descriptions

### **1 Pitch Change**

This program has pitch values set to 0 and is set for stereo shift.

### **2 Pitch Chorus**

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.

### **3 1% Up 1% Down**

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.

### **4 Barber Pole**

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.

### **5 Half Steps**

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.

### **6 Stair Case**

This is a dual mono variation of Half Steps with shorter delays and larger intervals.

## Programs — Bank 8: Pitch Shift

### 1 Pitch Change

MOD	SNC	PCH	FIN	PCH	FIN
Stereo		+null	+null	+null	+null
PDL	PDR	FBL	FBR	MIX	
0ms	0ms	0%	0%	All Fx	
MIDI		REF	MIR	GLL	SPL
+null		48c2	+Off	+null	16
MIDI		REF	MIR	GLR	
+null		72c4	+Off	+null	

### 2 Pitch Chorus

MOD	SNC	PCH	FIN	PCH	FIN
Mono		-3c	-3c	+6c	+6c
PDL	PDR	FBL	FBR	MIX	
6.48 ms	9.81 ms	+23%	+13%	All Fx	
MIDI		REF	MIR	GLL	SPL
-3c		48c2	+Off	-3c	16
MIDI		REF	MIR	GLR	
+6c		72c4	+Off	+6c	

### 3 1% Up 1% Dwn

MOD	SNC	PCH	FIN	PCH	FIN
Mono		+10c	+10c	-10c	-10c
PDL	PDR	FBL	FBR	MIX	
39.27 ms	32.69 ms	+10%	+10%	All Fx	
MIDI		REF	MIR	GLL	SPL
+10c		48c2	+Off	+10c	16
MIDI		REF	MIR	GLR	
-10c		72c4	+Off	-10c	

### 4 Barber Pole

MOD	SNC	PCH	FIN	PCH	FIN
Stereo		-3c	-3c	-3c	-3c
PDL	PDR	FBL	FBR	MIX	
0.0 ms	0.0 ms	-41%	+41%	All Fx	
MIDI		REF	MIR	GLL	SPL
-3c		48c2	+Off	-3c	16
MIDI		REF	MIR	GLR	
-3c		72c4	+Off	-3c	

### 5 Half Steps

MOD	SNC	PCH	FIN	PCH	FIN
Stereo		-m2nd0c	-m2nd0c	-m2nd0c	-m2nd0c
PDL	PDR	FBL	FBR	MIX	
600.03ms	600.03ms	+44%	+44%	All Fx	
MIDI		REF	MIR	GLL	SPL
-m2nd0c		48c2	+Off	-m2nd0c	16
MIDI		REF	MIR	GLR	
-m2nd0c		72	+Off	-m2nd0c	

### 6 Stair Case

MOD	SNC	PCH	FIN	PCH	FIN
Mono		+m3rd+47c	+m3rd+47c	+M3rd+2c	+M3rd+2c
PDL	PDR	FBL	FBR	MIX	
29.03	30.95	+55%	-60%	20%Fx	
MIDI		REF	MIR	GLL	SPL
+m3rd+2c		48c2	+Off	+m3rd+47c	16
MIDI		REF	MIR	GLR	
+M3rd+2c		72c4	+Off	+M3rd+2c	

Page One					
PLAY Play Trigger	TYP Play Type	FG Amp Q/Pch Q	SPD Speed	TIM Time	DST Distance
Page Two					
PLAY Play Doppler					TRIG Trigger Doppler

## The Doppler Program

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. The Cascade mode allows audio to be captured and looped in a sampling program in Machine A, and a Doppler program to be loaded into Machine B that will trigger the effect. 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 parabola, 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.

**Programs — Bank 8: Doppler**

**7 Xpres Subway**

PLAY	TYP	FG	SPD	TIME	DST
	Zoom	89	122	6.2	35.9
PLAY					TRIG

**8 Indy Corner**

PLAY	TYP	FG	SPD	TIME	DST
	Zoom	226	339	1.2	5.3
PLAY					TRIG

**9 Airport Land**

PLAY	TYP	FG	SPD	TIME	DST
	Norm	153	70	12.0	119.2
PLAY					TRIG

**0 Airport Tkof**

PLAY	TYP	FG	SPD	TIME	DST
	Norm	213	184	6.8	268.1
PLAY					TRIG

**Program Descriptions**

**7 Xpres Subway**

Imagine standing at the back of the station platform as the express rushes by.

**8 Indy Corner**

With Speed set at over 300 k.p.h., this is a very high speed turn! High amplitude and pitch change near center create a pronounced "whoosh". If distance is increased, things get tamer — as though you had a seat on the straightaway.

**9 Airport Land**

A long and gradual Doppler shift with a mild amplitude change.

**0 Airport Tkof**

Takeoff is shorter, with a gradual pitch change, and a more aggressive amplitude change.

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## **Bank 9: the Mastering Programs**

This chapter describes the Mastering programs located in Bank 9. These include The In/Out, Stereo Adjust, PONS Adjust, Stereo and Mono Parametric EQ, and Panorama.

## The In/Out Program

The In/Out program is a bypass program that feeds input signal unaltered through processors to outputs. This program has no adjustable parameters.

<b>Page One</b>					
LVL Stereo Level	FIN Fine Level	BAL Balance	ROT Rotate	BAS Bass EQ	TRB Treble EQ
<b>Page Two</b>					
XOV Bas Crossover	HFC Stereo Treb X-over	HFC Ind Treb X-over	SPC Spatial EQ	TBL Treble L	TBR Treble R
<b>Page Three</b>					
DLY Delay	EMP De-emphasis	INV Flip & Swap		SHL Shelf	SHUF Shuffler
<b>Page Four</b>					
AUTO Auto DC Cut	DCL DC L Offset	DCR DC R Offset			

## The Stereo Adjust Program

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.



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 XOY (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**

### **XOY (Bass Crossover)**

XOY 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 XOY.

### **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 low-pass filter whose frequency is set with XOY.

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 circuit 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 XOY setting, 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!

**Shuffler**

Shuffler is designed to be used with Binaural recordings, or with any recording technique involving only omnidirectional microphones placed at close distances (such as true Binaural or directional microphones spaced to equal ear-to-ear distance and aimed forward). It adds a significant amount of L-R low frequency information. If widely spaced omnidirectional microphones, or accent microphones, are mixed with the Binaural pair, Shuffler should be used before the microphone signals are mixed to avoid a severe boost in L-R information.

**Shelf**

Adjusts the +3dB frequency of the Shuffler shelf.

**Flip & Swap (Right Channel Phase Invert)**

Enables either a change of audio channels, or a polarity inversion in one or both channels. This change is made in the digital domain. The available settings are:

	Channel Position	Polarity
+L +R	NORM	NORM
+L-R	NORM	INV
+R +L	INV	NORM
+R -L	INV	INV

**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  $\pm 4$  bits of DC error to  $>48$ dB down for each channel. It maintains a slight positive offset near zero to avoid toggling the MSB 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.

<b>Page One</b>					
LVL Level	FIN Fine Level	BAL LDB Balance R	ROT Rotate	TBL Treble Left	TBR Treble R
<b>Page Two</b>					
XOV Bass Xover	HFC Treble Xover	SPC Spatial EQ	BAS Bass Left	BAS Bass Right	FLP Flip & Swap
<b>Page Three</b>					
DIT Dither	PONS Noise Shaping	SHL Shelf	SHUF Shuffler		

## The PONS Adjust Program

PONS Adjust is similar to the Stereo Adjust Program — A two-channel digital mixer with precision level adjustment, low and high frequency shelving EQ, and precision stereo balance controls. Additional parameters include precision binaural rotation, spatial EQ (L-R shelving boost or cut), and Blumlein shuffler.

PONS Adjust incorporates psychoacoustically optimized noise shaping that can increase the signal-to-noise performance of a 16-bit digital recording medium. This performance increase is attained by processing a digital recording made at 18-bit (or higher) bitwidth, through PONS, then transferring digitally to a 16-bit medium, and converting back to analog through an 18-bit converter.

## About the PONS 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 +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.

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.

#### TBL

TBL is a 6dB/octave high shelving EQ that operates on the left channel. It moves in .5dB increments from +6 to -6dB, and in 1dB increments below -6dB to Full Cut.

#### TBR

TBR is identical, but operates on the right channel.

**Page Two**

**XOV (Bass Crossover)**

XOV sets the crossover point for BAS and SPC . When BAS is set to full cut, the level is -3 dB at the frequency set with XOY.

**HFC (Stereo Treble Crossover)**

HFC sets the crossover point for TBL and TBR. When TBL or TBR 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 low-pass filter whose frequency is set with XOY.

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 circuit 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 XOY setting, this compensation will be quite accurate as long as SPC is set to less than 3 dB boost.

**BAS (Bass Left and Right)**

The BAS controls are 6dB per octave low shelving EQs which operate on the left and right channels. Each moves in .5dB increments from +6 to -6dB, and in 1dB increments below -6dB to full cut.

**Flip & Swap (Right Channel Phase Invert)**

Enables either a change of audio channels, or a polarity inversion in one or both channels. This change is made in the digital domain. The available settings are:

	Channel Position	Polarity
+L +R	NORM	NORM
+L-R	NORM	INV
+R +L	INV	NORM
+R -L	INV	INV

**Page Three**

**DIT (Dither)**

Enables or disables dither noise set with the PONS control as follows:

- 0 = Dither OFF
- 1 = Dither ON

**PONS**

Dither is a low-level pseudo-random signal which is added to digital audio to reduce quantization noise, in effect, by replacing it with a "nicer-sounding" noise. It is possible to filter the quantization noise in such a way that almost all of the noise within the frequency range where human hearing is most acute is shifted to higher frequencies. This is called psychoacoustically optimized noise shaping (PONS) because it takes advantage of the variations in the ear's sensitivity to noise within the 20 Hz - 30 kHz range. This control has been optimized. Do not adjust.

**Shelf**

Adjusts the +3dB frequency of the Shuffler shelf.

**Shuffler**

Shuffler is designed to be used with Binaural recordings, or with any recording technique involving only omnidirectional microphones placed at close distances (such as true Binaural). It adds a significant amount of L-R low frequency information. If widely spaced omnidirectional microphones, or accent microphones, are mixed with the Binaural pair, Shuffler should be used before the microphone signals are mixed to avoid a severe boost in L-R information.

<b>Page One</b>					
LEV Stereo Level	FIN Fine Level	BAL LDB Balance R	LNK Stereo Link	FIN Fine Frequency L	FIN Fine Frequency R
<b>Page Two</b>					
FR1L Filter 1 Left	Q	LEV Filter Level	FR1R Filter 1 Right	Q	LEV Filter Level
<b>Page Three</b>					
FR2L Filter 2 Left	Q	LEV Filter Level	FR2R Filter 2 Right	Q	LEV Filter Level

## The Parametric EQ Programs

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 monoaural parametric equalizer. Both programs provide frequency adjustment on each band between 30 Hz and 17 kHz, boost/cut  $\pm 12$  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 480L 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 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.

<b>Page One</b>					
LVL Level	FIN Fine Level			FIN Fine Freq. 1	FIN Fine Freq. 2
<b>Page Two</b>					
FR-LF Low Frequency	Q	LEV Level	FR-LM Low-Mid Freq.	Q	LEV Level
<b>Page Three</b>					
FR-HM High-Mid Freq.	Q	LEV Level	FR-HF High Freq.	Q	LEV Level

**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.

**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**

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.

Page One					
CH1 Left Channel In	CH2 Right Channel In	EFF Effect	ANG Speaker Angle	LFB L-R Boost	MIX Dry Input

## The Panorama (Binaural) Program

Panorama is a crosstalk elimination program. It increases the apparent width of stereo sounds by eliminating the crosstalk around the ears of the listener. One of the major uses of this program in the 480L is to spread the sound of a reverb or effect outside the bounds of the loudspeakers. This is easily done with a mixing board by putting the 480L in cascade mode with Ambience or a reverb program in machine A, and Panorama in machine B. The mix control in machine A should be ALL EFFECTS. In machine B the Effect control and the Low frequency width are used to increase the width of the reverb, and the Dry Input Mix control or an external mixer can be used to add the unexpanded original sound back into the expanded reverb.

In making expanded recordings the listener is assumed to be exactly between the two loudspeakers, and the speakers should be at the angle set in the program. For this reason, this expansion is best done at the time of playback, using a home processor. However, there may be good reasons for doing some expansion in a recording. Low frequencies are spread by the program through the whole listening room — they do not require precise adjustment. Even under non-ideal playback conditions, some improvement may be noted. Recordings made with ORTF, or closely spaced omni microphone techniques, will probably greatly benefit from some LF L-R boost, which can be adjusted independently from the crosstalk cancellation controlled by the EFFECT control.

This program may also be of use to professionals using Binaural recordings to study acoustic or noise environments. Binaural recordings played with Panorama through a carefully set up loudspeaker system give nearly ideal reproduction of the original recorded sound field, at least in the front hemisphere.

Setting up the program for this application is best done with a noise source at the input. Use CH1 to turn off the left input level, and move your head to find a position which gives a sharp null in the left ear. Now raise CH1 and lower CH2. The null should now be in the right ear. If it is not, adjust ANG and the listening position until a sharp null occurs for each ear.

### About the parameters

#### CH1, CH2 (Left In, Right In)

These controls are simple linear attenuators on the left and right inputs. They can be used to set up the program, or they can be used as input balance controls. They can also be used to trim the balance into the L-R boost control.

#### EFF (Effect)

EFF sets the amount of crosstalk cancellation in the mid frequencies, and acts as a width control. If EFF is set to 0, and L-R boost is also set to 0, the program has no effect on the sound.

#### ANG (Speaker Angle)

ANG sets the speaker angle expected in the listening environment. This is the total angle between the speakers. Lower settings of the speaker angle control (40° or so) are more effective when the actual speaker positions to be used by the listener are unknown. This is because most listeners tend to use smaller speaker angles, and because the effective listening area is greater when ANG is low.

#### LFB (LF-LR BOOST)

LFB is a L-R bass boost with a maximum boost of 20dB. It is similar to the Spatial Eq control in Stereo Adjust, but works differently. It is a pure boost in the difference channel only. It does not affect the mono channel. If Panorama is being used to process a binaural recording for loudspeaker reproduction, LFB should be at least half-way up. This control is independent of EFF, so the low frequency width can be increased without changing the mid-frequency width.



LFB may also be useful in film mixing in a matrixed surround system. It will tend to send low frequencies into the surround. For this application, be sure to set EFF to 0.

### MIX (Dry Input MIX)

Dry Input Mix is *not* like the other mix controls in the 480. It mixes the *stereo* input to the 480L into the output of Panorama, regardless of the configuration selected for the 480L. In Single mode this is equivalent to the usual MIX. In Split modes the results will be not useful. In Cascade, however, when Panorama is in machine B, this control allows you put a reverb or effect in machine A, spread the reverb only, and then mix the dry input with the spread out reverb.

#### 1 The In/Out

NO LARC DATA					
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#### 2 Stereo Adjust

LVL	FIN	BAL	ROT	BAS	TRB
0	0	Center	Center	Flat	Flat
XOV	HFC	HFC	SPC	TBL	TBR
493Hz	5.538kHz	5.538kHz	Flat	Flat	Flat
DLY	EMP	INV		SHL	SHUF
0	0	+L/+R		0	0
AUTO	DCL	DCR			
-	0	0			

#### 3 PONS Adjust

LVL	FIN	BAL	ROT	TBL	TBR
0	0	Center	Center	Flat	Flat
XOV	HFC	SPC	BAS	BAS	FLP
367Hz	2040Hz	Flat	Flat	Flat	+L/+R
DIT	PONS	SHL	SHUF		
30	-	0	0		

#### 4 MS Decode

LVL	FIN	BAL	ROT	BAS	TRB
0	0	Center	2:9 - 72.1	Flat	Flat
XOV	HFC	HFC	SPC	TBL	TBR
493Hz	5.538kHz	5.538kHz	Flat	Flat	Flat
DLY	EMP	INV		SHL	SHUF
0	0	+L/+R		0	0
AUTO	DCL	DCR			
-	0	0			

#### 5 Invert L-R

LVL	FIN	BAL	ROT	BAS	TRB
0	0	Center	Center	Flat	Flat
XOV	HFC	HFC	SPC	TBL	TBR
493Hz	5.538kHz	5.538kHz	Flat	Flat	Flat
DLY	EMP	INV		SHL	SHUF
0	0	+L/-R		0	0
AUTO	DCL	DCR			
-	0	0			

#### 6 Channel Swap

LVL	FIN	BAL	ROT	BAS	TRB
0	0	Center	Center	Flat	Flat
XOV	HFC	HFC	SPC	TBL	TBR
493Hz	5.538kHz	5.538kHz	Flat	Flat	Flat
DLY	EMP	INV		SHL	SHUF
0	0	+L/+R		0	0
AUTO	DCL	DCR			
-	0	0			

#### 7 Stereo Parametric EQ

LVL	FIN	BAL	LNK	FIN	FIN
0	0	Center	On	1.0	1.0
FR-1L	Q	LEV	FR-1R	Q	LEV
1.0	1.5	0	Use Left	Use Left	Use Left
FR-2L	Q	LEV	FR-2R	Q	LEV
1.0	1.5	0	Use Left	Use Left	Use Left

#### 8 60Hz Notch

LVL	FIN	BAL	LNK	FIN	FIN
0	0	Center	On	60	60
FR-1L	Q	LEV	FR-1R	Q	LEV
60	32	-36	Use Left	Use Left	Use Left
FR-2L	Q	LEV	FR-2R	Q	LEV
1.0	1.5	0	Use Left	Use Left	Use Left

#### 9 Mono Parametric EQ

LVL	FIN			FIN	FIN
0	0	Center	On	60	60
FR-1L	Q	LEV	FR-1R	Q	LEV
1.0	1.5	0	1.0	1.5	0
FR-2L	Q	LEV	FR-2R	Q	LEV
1.0	1.5	0	1.0	1.5	0

#### 0 Panorama

CH1	CH2	EFF	ANG	LFB	MIX
254	254	254	56 deg	0	All Fx

## Program Descriptions

### 1 The In/Out

A bypass program that feeds an input signal unaltered through the processors. This program has no adjustable parameters.

### 2 Stereo Adjust

A digital mixing program designed for mastering applications. The program loads with parameters nulled.

### 3 PONS Adjust

A variation of Stereo Adjust which incorporates psychoacoustically-optimized noise shaping (PONS) dither.

### 4 MS Decode

A variation of Stereo Adjust that is preset to decode MS recordings. The mid signal is fed to the left input, and side is fed to the right. The Flip & Swap parameter can reverse the channels if a digital source is used. The BAL control provides the mix ratio of M and S signals.

### 5 Invert L-R

A variation of Stereo Adjust that has the Flip & Swap parameter preset with a digital phase inversion.

### 6 Channel Swap

A variation of Stereo Adjust that has the Flip & Swap parameter preset to swap left and right channel input signals.

### 7 Stereo Parametric EQ

2-Band stereo parametric EQ and mixer with parameters nulled.

### 8 60Hz Notch

A variation of Stereo Parametric EQ preset with a 60 Hz cut.

### 9 Mono Parametric EQ

4-Band mono parametric EQ and mixer with parameters nulled.

### 0 Panorama

Integral crosstalk cancellation program adds width to stereo image. Use in cascade configurations with reverb or ambience in Machine A, and Panorama in Machine B.

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## **Bank 10: the Compressor Programs**

This chapter describes the Compressor programs located in Bank 10.

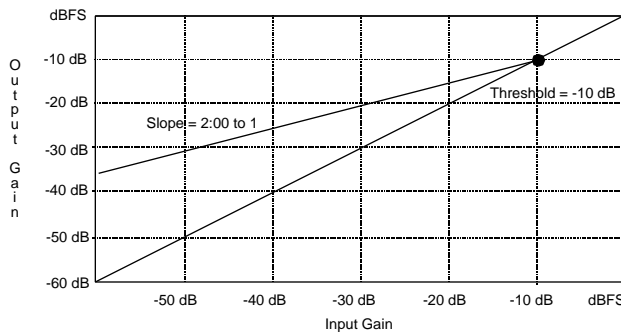
Page One					
DLY Front Delay	SLP Slope	GAIN Maximum Gain	THR Threshold Down	ATC Attack Time Const	RTC Release Time Const
Page Two					
EXS Expander Slope	EXG Expander Gain	EXT Expander Thresh			

## The Compressor/Expander Program

Compressor is a stereo compressor program, with a user interface similar to several mastering compressors. The important thing to keep in mind as you use this program is: *It always has unity gain at high levels.* The compression is achieved by adding gain as the input level decreases.

If you are accustomed to standard compressors, the following may be helpful:

Many compressors currently available allow you to set the threshold — the level above which the output level will be decreased. The threshold in the 480L's compressor defines the level *below which the output level is increased.* Signal level that is above this threshold setting remains at unity gain. This allows you to leave peak values intact while increasing the overall program loudness.



Compressor with a 2:00=1 ratio, with threshold set at -10dB.

## About the parameters

### Page One

#### DEL (Delay)

DEL is a stereo delay line which allows the digital detectors time to adapt to the incoming signal. With Attack Time set to 30ms, input delays of 30ms or so are desirable. Much longer attack times are possible, and the delay can be set up to approximately 200ms. Delay is not a problem in a mastering situation, or when time alignment can be corrected. If delay through the compressor is not tolerable, shorter attack times can be used, but there will be some noise and distortion on transients.

#### SLP (Slope or Compression Ratio)

SLP varies from 1:1 (no compression), up to 99.99:1 (high compression). With high compression, *the output level is constant as the input level decreases.* At 2:1 compression the output level drops 3dB for every 6dB drop in input level.

#### GAIN (Max Gain)

GAIN sets the maximum gain the compressor will have. Typically this would be set between 6 and 12dB. More than 15dB will bring up noise on a low level signal, and noise pumping may be obvious. The maximum allowed is 24dB.

#### THR (Threshold Dn)

THR sets the level above which the compressor always has unity gain. Note that the level detected by the compressor circuitry is the *average* level; the 480L input lights read the *peak* level. Therefore, if the peak to average ratio of the music is 10dB, THR must be set to -10dB or greater, or the average output level can exceed 10dB — and the peak output will overload. This means that THR must be set carefully. As THR is lowered, the compressor maintains unity gain until the input level drops lower. If you are listening to the output, the gain will appear to decrease as this control is raised. Typically, you want to set THR to get the maximum gain possible without overload.

**ATC (Attack Time Constant)**

ATC adjusts the speed with which the compressor responds to an increase in input level. The nominal value is about 30ms. The times on the slider move in factors of 2, from 7ms to about 1 second. Both 7ms and 15ms will cause some distortion and are not recommended unless delay through the program cannot be tolerated. High values of ATC can be used with large delays in a program-leveling application. The filter used for ATC is a non-linear filter. The number of dB/second depends on the rate at which the input is changing. Fast changes will make a quicker attack, causing the gain to decrease quickly.

**RTC (Release Time Constant)**

RTC adjusts the release time constant. This filter maintains a constant number of dBs per second, regardless of the size of the change. Thus, the rate of increase in gain is constant. The value displayed for the release time is something of an approximation, since the ordinary meaning of “time constant” does not apply to this type of control.

**Page Two**

**EXS (Expand Slope)**

EXS adjusts the slope of the expansion. It behaves similarly to the compression ratio, but it is important to keep in mind that the expander and the compressor in this program are not in series. They both respond to the same input level, but in different ways. For example, if some material is compressed with a 2:1 ratio, recorded, and then played back through the expander with a 2:1 expansion ratio, the original dynamics will be restored. The same will occur if two compressor programs are run in cascade. However, if the compressor and expander are *both* set to operate *in the same program at the same time with identical THR points and slopes*, the result will *not* have the original dynamics.

Consider a drop in input level of 6dB. For a 2:1 compression in level the output gain should increase 3dB, giving a net drop in output level of 3dB. However, for a 2:1 expansion, the output gain should drop 6dB, giving a net output of -12dB. If both the compressor and expander are operated at the same time, the net result will be an output gain decrease of 3dB, representing a net expansion of the signal.

**EXT (Expander Threshold)**

EXT sets the level below which expansion starts. When the expansion is intended to apply to the whole program, or perhaps just to the peaks of the program, EXT should be set to about 10 or 12dB, similar to the setting of the THR control of the compressor. If the expander is to be used as a noise gate, EXT should be set much higher, to just below the level of the noise.

**EXM (Expand Maximum Attenuation)**

EXM sets the maximum attenuation reached by the expander. It is similar in operation to the GAIN control of the expander.

**Programs — BANK 10:  
Compressor/Expander**

**1 2.00=1 COMP**

DLY	SLP	GAIN	THR	ATC	RTC
27.45ms	2.00=1	9dB	21dB	15ms	91ms
EXS	EXG	EXT			
2.00=1	10dB	-48dB			

**2 6.24=1**

DLY	SLP	GAIN	THR	ATC	RTC
12.56ms	6.24=1	12dB	16dB	7ms	91ms
EXS	EXG	EXT			
1.88=1	7dB	-40dB			

**3 18.28=1 COMP**

DLY	SLP	GAIN	THR	ATC	RTC
20.38ms	18.28=1	10dB	20dB	15ms	91ms
EXS	EXG	EXT			
2.00=1	10dB	-48dB			

**4 Transfer**

DLY	SLP	GAIN	THR	ATC	RTC
40.26ms	1.33=1	10dB	10dB	30ms	73ms
EXS	EXG	EXT			
1.00=1	0dB	0dB			

**5 Vox Easy**

DLY	SLP	GAIN	THR	ATC	RTC
12.56ms	5.00=1	8dB	15dB	7ms	73ms
EXS	EXG	EXT			
1.88=1	7dB	-40dB			

**6 FM**

DLY	SLP	GAIN	THR	ATC	RTC
15.64ms	14.22=1	7dB	15dB	15ms	58ms
EXS	EXG	EXT			
1.88=1	7dB	-40dB			

**7 Nailed**

DLY	SLP	GAIN	THR	ATC	RTC
39.27ms	21.33=1	6dB	0dB	7ms	286ms
EXS	EXG	EXT			
2.00=1	10dB	-40dB			

**8 Noise Gate**

DLY	SLP	GAIN	THR	ATC	RTC
20.38ms	1.00=1	0dB	0dB	7ms	37ms
EXS	EXG	EXT			
2.98=1	24dB	-30dB			

**9 2-Slp-Mach A**

DLY	SLP	GAIN	THR	ATC	RTC
9.40ms	1.77=1	10dB	-18dB	7ms	181ms
EXS	EXG	EXT			
1.50=1	10dB	50dB			

**0 2-Slp-Mach B**

DLY	SLP	GAIN	THR	ATC	RTC
0ms	2.41=1	4dB	-12dB	7ms	91ms
EXS	EXG	EXT			
1.00=1	10dB	46dB			

**Program Descriptions****1 2.00=1 Compressor**

A two-to-one compressor/expander with time constants optimized for both processes.

**2 6.24=1 Compressor**

A more aggressive compressor/expander that provides more overall loudness than 2.00=1 Compressor.

**3 18.00=1 Compressor**

An even more aggressive compressor/expander — the dynamic range becomes noticeably restricted.

**4 Transfer**

Very linear compressor designed for mastering applications.

**5 Vox Easy**

A compressor/expander that is a good starting point for a vocal track — use it on a mixer's patch point. Peaks remain intact and the loudness of the vocalist(s) in the mix is increased.

**6 FM**

Gain below threshold ramps rapidly and hits a brick wall — feed program material in and compare it with your favorite station.

**7 Nailed**

Provides a compressor/expander that affords little dynamic range on output — true to its name!

**8 Noise Gate**

Compressor is inactive, and the expander time constants have been optimized for gating.

**9 2-Slp-Mach A****0 2-Slp-Mach B**

These programs are designed to be used with the 480L in Cascade configuration. Load the programs as their names describe them:

in Machine A, load 2-Slp-Mach A

In Machine B, load 2-Slp-Mach B

These programs provide a compressor that has a nonlinear gain slope. Machine A provides a slow rise in gain from -18dB up to -12dB. Machine B provides a more aggressive gain slope from -12dB forward. Really sounds great on dynamic pop program material.

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## **Banks 11-12: the Random Halls and Spaces Program**

This chapter describes the Random Hall programs located in Bank 11, and the Random Spaces programs located in Bank 12.

<b>Page One</b>					
RTM Reverb Time	SHP Shape	SPR Spread	SIZ Size	HFC High Freq Cutoff	PDL Predelay
<b>Page Two</b>					
BAS Bass Multiply	XOV Crossover	RTC High Freq Cutoff	DIF Diffusion	MOD Mode	MIX Wet/Dry Mix
<b>Page Three</b>					
PRE L>L	ECHO R>R	LEV R>L	ELS L>R	SPN SPin	WAN Wander
<b>Page Four</b>					
PRE Delay 1	ECHO Delay 2	DEL Delay 3	AYS Delay 4	SHL Shelf	LEV Reverb Level

## The Random Halls and Spaces Program

RHALL is similar to the standard Hall program in the 480L, with the addition of random delay elements. These elements have several effects. First, there is a reduction of long-lived modes in the reverberant decay, which makes the decay less metallic and reduces the apparent reverb time. The apparent reverb time of RHALL is much closer to the value indicated by the display than the standard Hall. For this reason, decay optimization is not needed in this program. However, the controller has been left in place to enable switching between the REVERB and EFFECTS modes of operation. The random elements also improve the steady-state timbre of the program.

The amount of coloration has been substantially reduced from that in the standard Reverb programs — especially at small settings of the spread control. In addition, the steepness of the filter in the TREBLE DECAY control has been increased. — Don't be surprised if you need to set this control higher in this program than in the standard Reverb programs.

The Random Hall and Ambience programs are particularly useful in sound reinforcement where it can improve the existing acoustics of a hall by adding lateral reflections (and possibly also delayed dry signal) from speakers hidden around the listening space. The fact that many of the reflections are time-varying is important to increasing gain before feedback in such a situation. Ambience incorporates both a PRE-DELAY and an INPUT DELAY that can be set to further enhance this application.

## About the parameters

### Page One

#### RTM (Mid-Frequency Reverb Time)

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

When RTM is moved to full scale, infinite reverb is switched on, and is displayed on the LARC. Program material fed to the inputs of the machine running RHall prior to enabling infinite mode will reverberate infinitely! Moving the RTM slider until the LARC displays Reverb time disables infinite mode. The RTM slider will fade the infinite loop if no other input signal is present.

#### 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.

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. Parameters are unlinked in Effect mode.

## **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.

## **PREDELAY**

PREDELAY adjusts an additional time delay between the input of signal and the onset of reverberation. The control is not intended to mimic the time delays in natural spaces. In real rooms the build-up of reverberation is gradual, and the initial time gap is usually relatively short. Natural spaces are best emulated by setting SHAPE at a middle value and adjusting SPREAD for the desired effective pre-delay.

Additional delay added with the PREDELAY control can increase the initial time gap slightly, emulating a situation where reverberant pick-up microphones are located much further from the source than the main microphones. If less than about 30ms of pre-delay is added, this additional delay can add clarity with some music, but it can also sound unnatural. Large pre-delays can be useful for slap-echo effects.

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

## **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 400 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.

## **MODE**

Selects between linked and unlinked modes of operation for RT MID, SHP, SPR, and SIZ parameters. The Reverb mode maintains optimum relational values between these controllers as settings are changed. Effects mode permits independent parameter control.

## **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.

Controls are similar to those available in the standard Reverb programs. However, there are only 4 preechoes. The remaining two sliders on Page 3 control SPIN and WANDER, as in the EFFECTS or AMBIENCE program .

## **SPN (Spin)**

SPN is identical to the spin control in the EFFECTS program. It affects the movement of many of the delay taps in the program. The object of SPN and WAN is to continuously alter the timbre of the reverberant sound. This makes the result more natural. It is not intended to make the position of instruments unstable. SPN should typically be 37 or higher. High values may make the pitch of piano or guitar unstable.

## **WAN (Wander)**

WAN is identical to the wander control in the EFFECTS program. It sets the distance in time that the early reflections will move. For best results WAN should be about 10ms at larger sizes.

## **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.

### **REVERB LEVEL**

Digitally controls the output gain from the processor. This is useful for setting different versions of overall program balance. This parameter may also be controlled via MIDI, which allows the amount of reverb to be easily controlled remotely without affecting the wet/dry mix.

### **SHELF**

A level control that adds pre-high frequency cutoff energy to the reverb output, allowing a double "knee" in the low pass filter.

## Programs — Bank 11: Random Hall

### 1 Large RHall

RTM	SHP	SPR	SIZ	HFC	PDL
2.09s	120	158	38M	3402Hz	0ms
BAS	XOV	HFC	DIF	MOD	MIX
x1.2	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	10ms
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 2 Lg RHall & Stg

RTM	SHP	SPR	SIZ	HFC	PDL
2.61s	120	158	38M	3402Hz	0ms
BAS	XOV	HFC	DIF	MOD	MIX
x1.2	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
-8dB	-8dB	-7dB	-7dB	43	10ms
DLY1	DLY2	DLY3	DLY4	SHL	RLV
14ms	20ms	40ms	48ms	0	160

### 3 Medium RHall

RTM	SHP	SPR	SIZ	HFC	PDL
2.06s	120	127	30M	3402Hz	0ms
BAS	XOV	HFC	DIF	MOD	MIX
x1.2	493Hz	3591Hz	68	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	10ms
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 4 Med RHall & Stage

RTM	SHP	SPR	SIZ	HFC	PDL
2.06s	120	127	30M	3402Hz	0ms
BAS	XOV	HFC	DIF	MOD	MIX
x1.2	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
-8dB	-8dB	-7dB	-7dB	43	10ms
DLY1	DLY2	DLY3	DLY4	SHL	RLV
14ms	20ms	40ms	48ms	0	160

### 5 Small RHall

RTM	SHP	SPR	SIZ	HFC	PDL
1.12s	92	51	25M	4611Hz	0ms
BAS	XOV	HFC	DIF	MOD	MIX
x1.0	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	10ms
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 6 Sm RHall & Stage

RTM	SHP	SPR	SIZ	HFC	PDL
1.12s	92	51	25M	4611Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x1.0	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
-9dB	-9dB	-8dB	-8dB	43	6762µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
8ms	12ms	44ms	36ms	0	160

### 7 Large RChurch

RTM	SHP	SPR	SIZ	HFC	PDL
4.07s	82	228	39M	2523Hz	36ms
BAS	XOV	HFC	DIF	MOD	MIX
x1.5	885Hz	2862Hz	70	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	10ms
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 8 Small RChurch

RTM	SHP	SPR	SIZ	HFC	PDL
2.42s	72	107	31M	3784Hz	16ms
BAS	XOV	HFC	DIF	MOD	MIX
x1.0	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	6762µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 9 Jazz RHall

RTM	SHP	SPR	SIZ	HFC	PDL
1.32s	34	103	24M	12788Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x1.2	752Hz	5538Hz	98	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	38	6762µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 0 Auto Park R

RTM	SHP	SPR	SIZ	HFC	PDL
5.33s	149	248	39M	7181Hz	24ms
BAS	XOV	HFC	DIF	MOD	MIX
x1.0	752Hz	5538Hz	98	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
-8dB	-8dB	-12dB	-12dB	34	9114µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
22ms	16ms	40ms	48ms	0	160

## **The Random Halls Program**

Like the programs in Bank 1 (480L Hall), the Random Hall programs in Bank 11 are designed to emulate real concert halls. The random elements in these programs provide smoother decay, particularly where values of Size and Reverb Time are high.

### **1 Large RHall**

Large RHall 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.

### **2 Lg RHall + Stage**

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

### **3 Medium RHall**

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

### **4 Med RHall + Stage**

Med RHall + Stage is very similar to Lg RHall + Stage, but smaller.

### **5 Small RHall**

Small RHall is a smaller version of Medium RHall.

### **6 Sm RHall+ Stage**

Sm RHall +Stage is a smaller version of Med RHall + Stage.

### **7 Large RChurch**

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

### **8 Small RChurch**

Small RChurch is a smaller version of program 7.

### **9 Jazz RHall**

Jazz RHall 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.

### **0 Auto Park R**

Auto Park R reproduces the sound of an underground parking garage.

## Programs — Bank 12: Random Spaces

### 1 Music Club R

RTM	SHR	SPD	SIZ	HFC	PDL
1.06s	40	57	26M	7181Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x1.0	752Hz	3591Hz	78	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	6762µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 2 Large Room R

RTM	SHP	SPR	SIZ	HFC	PDL
0.71s	52	83	20M	6593Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x1.2	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	10ms
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	-	160

### 3 Medium Room R

RTM	SHP	SPR	SIZ	HFC	PDL
0.50s	22	10	20M	7493Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x1.2	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	5586µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 4 Small Room R

RTM	SHP	SPR	SIZ	HFC	PDL
0.32s	16	0	11M	7493Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x1.0	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	3402µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 5 Very Small R

RTM	SHP	SPR	SIZ	HFC	PDL
0.15s	8	0	5M	7493Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x1.0	493Hz	3591Hz	65	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	3066µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 6 Large Chamber R

RTM	SHP	SPR	SIZ	HFC	PDL
0.90s	3	0	21M	7181Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x1.0	1158Hz	5538Hz	99	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	3066µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 7 Small Chamber R

RTM	SHP	SPR	SIZ	HFC	PDL
0.37s	16	0	11M	7181Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x1.0	1158Hz	3591Hz	70	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	3066µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 8 Small & Bright R

RTM	SHP	SPR	SIZ	HFC	PDL
0.68s	40	41	10M	10127Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x0.8	1158Hz	7493Hz	70	REV0	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
Off	Off	Off	Off	43	3066µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
0	0	0	0	0	160

### 9 Chorus Room

RTM	SHP	SPR	SIZ	HFC	PDL
0.12s	255	12	2M	Full	16
BAS	XOV	HFC	DIF	MOD	MIX
x1.5	2.198Hz	21.18Hz	5	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
-4dB	-4dB	Off	Off	48	1302µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
22ms	12ms	0	0	0	160

### 0 Wet & Tacky

RTM	SHP	SPR	SIZ	HFC	PDL
1.39s	23	36	24M	7493Hz	0
BAS	XOV	HFC	DIF	MOD	MIX
x0.8	1158Hz	4395Hz	70	REV	All Fx
LVL1	LVL2	LVL3	LVL4	SPN	WAN
-14dB	-14dB	-14dB	-14dB	48	9618µs
DLY1	DLY2	DLY3	DLY4	SHL	RLV
14ms	18ms	28ms	34ms	0	160

## **Program Descriptions**

The Random Spaces presets are similar to the Room presets located in Bank 2. Most of the presets simulate the same room sizes as those in the standard Rooms. The random delay elements, however, make the rooms seem more “live”. These elements are very useful when you are attempting to simulate atmospheres that are busy, or that have movement. In two presets, CHORUS ROOM and WET & TACKY, these elements have been optimized to provide a chorusing effect with spatial qualities.

### **1 Music Club R**

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

### **2 Large Room R**

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

### **3 Medium Room R**

Medium Room R is a smaller version of Large Room R.

### **4 Small Room R**

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

### **5 Very Small**

Very Small has the intimate, close feel of a bedroom or den.

### **6 Large Chamber R**

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

### **7 Small Chamber R**

Small Chamber R is a smaller version of program 6.

### **8 Small & Bright R**

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

### **9 Chorus Room 2**

A small room with random delay elements that create a subtle chorus effect. This is useful for horns, strings, and vocals.

### **0 Wet & Tacky**

A larger room with longer reverb time and chorus-like random delay elements. The random delay elements add a shimmer to the reverberant decay.

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## **Banks 13-14: the Ambience Programs**

This chapter describes the Ambience programs located in Bank 13 and the Post-Ambience programs located in Bank 14.

Page One					
RT Reverb Time	RTL Reverb Level	SIZ Size	ROL Rolloff	DIF Diffusion	MIX Wet/Dry Mix
Page Two					
SPN SPin	WAN Wander	PDL Predelay	IND Dry Delay		MIX Wet/Dry Mix

## The Ambience Program

The Lexicon 480L reverberation programs (HALLS and ROOMS Banks) are designed to add a cushion of reverberance to recorded music, while leaving the clarity of the direct sound unaffected. Ambience is different. It is intended to become a part of the direct sound — to give it both better blend and a definite position in space. The difference between the two approaches to reverb is very dramatic, and can best be heard by comparing this program with any Hall or Random Hall program. Review *Using the Reverb Programs* in the Owner’s Manual for more information on Reverb and Ambience.

Ambience is very useful for adding a room sound to recorded music or speech. It is particularly easy to match a studio recording of dialog to a typical room environment. In music recording, using Ambience is an effective way of realistically adding distance to a close-miked signal. If an ensemble has been recorded with close-mikes and pan pots, Ambience can provide the missing blend and depth. The apparent position of the instruments is preserved in the reverb while the apparent distance is increased. Ambience is also useful in matching a closely miked accent microphone to the overall ambience of a recording. This allows a soloist to be increased in level without changing the apparent distance. Ambience can be used in a recording situation any time a close-miked sound is undesirable.

## About the parameters

### Page 1

#### RT (Reverb Time)

This control is not of great importance to the sound. The range of action is limited. Be careful — both long and short reverb times may sound unnatural. If a much longer or shorter reverb time than the preset provides is desired, SIZ should be changed first.

#### RTL (Reverb Level)

RTL controls the level of the reverberant part of the ambient decay. At 0, only the early reflections will be present in the sound, and there is an abrupt end to the sound when these early reflections are gone. Setting RTL to about 70 results in a natural blend of early and late reflections.

#### SIZ (Size)

SIZ allows you to vary the apparent size of the space over a wide range. SIZ is the most dramatic control, and must be selected to match the music or program material. It should be the first control that you adjust to tailor the desired space. SIZ also affects the reverb time, in a similar way to the standard reverb programs.

#### ROL (Rolloff)

ROL controls the -3dB point of a 6dB/octave filter on the output. It sets the effective bandwidth of both the early reflections and the reverberance.

#### DIF (Diffusion)

DIF controls the degree to which initial echo density increases over time. High settings of DIF result in high initial buildup of echo density; low settings cause low initial buildup. After the initial period (controlled by DIF), density continues to change at a rate determined by SIZ. To enhance percussion, use high settings of DIF. For clearer, more natural vocals, mixes, and music, use low or moderate settings.



## Page 2

### **SPN (Spin)**

SPN is identical to the spin control in the EFFECTS algorithm. It affects the movement of several early reflections. The object of SPN (and WAN) is to continuously alter the timbre of the early reflection parts of the ambient sound. This makes the result more natural. It is not intended to make the position of instruments unstable.

### **WAN (Wander)**

WAN is also identical to the wander control in the EFFECTS algorithm. It sets the distance in time that the early reflections will move.

### **PDL (Predelay)**

PDL adds an additional delay to the reflections and to the reverberation. Normally PDL should be set to 0, since the delays in the program will have already been set correctly by the hall synthesis. This control may be useful in a sound reinforcement situation, or for ambience effects.

### **IND (Dry Delay)**

IND controls the amount of delay in the dry signal mixed by the MIX control. Normally, this control should be set to 0. This control may be useful in a sound reinforcement situation, when both delayed dry sound and synthesized reflections are desired.

### **MIX (Identical to Wet/Dry mix above)**

MIX is the same as other standard 480L programs. Keep in mind that, in the Ambience program, MIX closely simulates microphone proximity effect. For convenience, this control has been placed on both Pages 1 and 2.

## **Using the AMBIENCE Program to Match Recorded Ambience**

On Page One, set Reverb Level (Slider 2) to 0. You should now be listening to the ambient build of the space.

### **Matching the High Frequency Contour**

This step involves closely approximating the High Frequency contour of the original sound source (the predominant instrument, vocalist, etc.)

Move the HF CUT control to its highest value. This is likely *not* to be the correct value. Now, set HF CUT to its lowest value. This is also likely to be incorrect. To locate the correct value, adjust HF CUT between these two points while listening carefully. Make this adjustment BY EAR — no by reference to displayed values. Once you think you have determined the optimal point, move the slider above and below the value to confirm your selection.

### **Matching Room Size**

Approximating the ambient build of the space requires adjustment of the SIZE control. Use the same technique as that described for establishing a value for HF CUT.

Set the MIX value to 90%, or listen to the DRY signal from time to time to hear where you are.

Note: The SIZE control does not update in real-time. To simplify the listening process, start at the highest setting of SIZE, and decrease the value by 5 meter increments, listening for:

an extraneous delay when the value is too high, or

a "cramping" of the program material when the value is too low. Use low frequency information to "zero in" on the correct value. Once you have found a value you think is correct, move SIZE two values above and below your selection to confirm your choice.

### **Creating Depth**

The MIX control in the Ambience programs emulates the movement of a coincident pair of microphones from the sound source into the room.

Until now, with MIX set to nearly 100% wet, you have been listening at the rear of the ambient space created by the SIZE control. Slowly moving MIX to 100% dry, simulates moving the microphone pair closer to the sound source.

Listen only to the returns of the 480L. If you have closely approximated the HF contour and size of the space, moving MIX should yield a smooth transition from the rear of the space to the original sound source (the predominant instrument, vocalist, etc.) If your approximations were close, doing this will immediately alert you as to which controls are set incorrectly.

### **Adding Reverberation**

The reverberant contour of the Ambience algorithm is not as complex as that of the Reverb algorithms. Reverb Time is linked with the Size control to provide a close approximation of naturally occurring reverb time for the space determined by SIZE. REVERB TIME can be tailored to taste. Raising the REVERB LEVEL will increase the level of these later reflections without altering the balance of the ambient build.

### **Store your work so far to a register, so that it will not be lost when you try the following examples.**

One reason this program is so powerful is that, when using conventional analog mixing methods (sends from dry signal and 480L100% wet MIX on returns or channels), the console send control emulates a depth control into the room you have created. If the 480L is set up in a Stereo Input configuration, the dry signal can be effectively panned in TWO DIMENSIONS — left to right, and front to rear.

To try this, set up two sources (such as dialog and footsteps, or an instrument) on adjacent channels dual-mono. Each can be panned L-R via the pan control on the console. The two sends control depth into the ambience for L or R. (This works wonderfully with stereo sends!) The independent send level emulates the movement of a single directional microphone from the sound source into the room with the rear of the capsule facing a rear corner of the room.

Front to rear panning is equal send level X2.  
Rear L-R panning is unequal send level.  
Front L-R panning is the console pan control

Now, try changing the configuration of the 480L to Cascade. In MACH A, load the register you stored. In MACH B, load STEREO ADJUST from the Mastering program. Using the ROT control on Page 1 of STEREO ADJUST, you can pan a L+R signal, left to right, against an L-R signal to shift the image of the ambience. The SPATIAL EQ control on Page 2 of STEREO ADJUST can tailor the width of the space. Loading these in reverse order, you can pan a L+R signal left to right against an L-R signal to shift the image feeding the ambience.

Add Predelay as necessary to increase the gap between the direct sound and early reflections. Note the Input Delay, as the effect occurs *before* the dry signal. Use these two controls to precisely position the source when using the 480L in the digital domain.

## Programs — Bank 13: Ambience

### 1 Very Lg Ambience

RTM	RTL	SIZ	ROL	DIF	MIX
3.34s	71	40M	2862Hz	75	All Fx
SPN	WAN	PDL	IND		MIX
34	7098µs	0	0		All Fx

### 2 Large Ambience

RTM	RTL	SIZ	ROL	DIF	MIX
2.75s	71	33M	2862Hz	75	All Fx
SPN	WAN	PDL	IND		MIX
34	7098µs	0	0		All Fx

### 3 Medium Ambience

RTM	RTL	SIZ	ROL	DIF	MIX
1.77s	71	20M	2862Hz	75	All Fx
SPN	WAN	PDL	IND		MIX
34	7098µs	0	0		All Fx

### 4 Small Ambience

RTM	RTL	SIZ	ROL	DIF	MIX
1.10s	71	13M	2862Hz	75	All Fx
SPN	WAN	PDL	IND		MIX
34	7098µs	0	0		All Fx

### 5 Strong Ambience

RTM	RTL	SIZ	ROL	DIF	MIX
0.59s	34	27M	3591Hz	56	All Fx
SPN	WAN	PDL	IND		MIX
38	12ms	10.15ms	0		All Fx

### 6 Heavy Ambience

RTM	RTL	SIZ	ROL	DIF	MIX
1.68s	100	23M	3591Hz	99	All Fx
SPN	WAN	PDL	IND		MIX
34	7098µs	19.96ms	0		All Fx

### 7 Ambient Hall

RTM	RTL	SIZ	ROL	DIF	MIX
3.0s	106	36M	2198Hz	94	All Fx
SPN	WAN	PDL	IND		MIX
29	25ms	27.04ms	0		All Fx

### 8 Announcer

RTM	RTL	SIZ	ROL	DIF	MIX
0.69s	48	10M	2862Hz	88	All Fx
SPN	WAN	PDL	IND		MIX
16	1218µs	0	0		All Fx

### 9 Closet

RTM	RTL	SIZ	ROL	DIF	MIX
0.51s	106	7M	1586Hz	99	All Fx
SPN	WAN	PDL	IND		MIX
0	0	0	0		All Fx

### 0 Gated Ambience

RTM	RTL	SIZ	ROL	DIF	MIX
0.29s	29	20M	6315Hz	51	All Fx
SPN	WAN	PDL	IND		MIX
33	7098µs	0	0		All Fx

## Program Descriptions

The programs listed above for the AMBIENCE program were designed to emulate the real spaces typically required for music, jingle, and post-production work.

### 1 Very Large Ambience

Resembles a very large ambient space (large shopping mall, parking garage, warehouse) that has far more "clutter" than a concert hall or performance environment. Lowering the RT LEVEL reduces the clutter while maintaining the sense of a very large ambient space.

### 2 Large Ambience

Similar to Very Large Ambience, but less spacious. Provides the ambience of a large symmetrical room.

### 3 Medium Ambience

Similar to Large Ambience, but smaller. Imagine a large courtroom, or a lecture room — then load the preset.

### 4 Small Ambience

Similar to Medium Ambience, but smaller — typical lobby, or small lounge.

### 5 Strong Ambience

The room size is larger than Medium Ambience, but the RT LEVEL has been reduced to provide a strong "wash" of ambience with a relatively short decay time.

### 6 Heavy Ambience

Resembles a large rectangular performance space with musicians or performers positioned in the middle of the space. Microphone proximity to the musicians or performers can be simulated by adjusting the MIX control. If the 480L is used with a mixing console and interfaced with stereo sends and returns, the MIX control should be left at ALL FX and the level of the sends will determine proximity effect.

### 7 Ambient Hall

Fast, dense ambient attack. Reverberant characteristics of Large RHall.

### 8 Announcer

Adds a very useful ambient spaciousness to a dry announcer's dialog track.

### 9 Closet

Just as you would imagine — it even feels cramped!

### 0 Gated Ambience

Very strong ambience with fast decay — just add your favorite snare drum!

## The Post Ambience Program

The Post Ambience Program bank (14) contains a group of programs that are optimized to meet the requirements of post production. Several algorithms are used in this bank. Each program description refers to the algorithm used to create it.

### Programs - Bank 14: Post Ambience

#### 1 Car Interior

RTM	SHP	SPR	SIZ	HFC	PDL
0.01ms	26	0	1	12177Hz	0ms
BAS	XOV	RTC	DIF	MOD	MIX
0.6	4611Hz	8886Hz	0	Reverb	All Fx
Preecho Levels				SPN	WAN
Off	Off	Off	Off	20	651µs
Preecho Delays				SHL	LEV
2ms	4ms	4ms	6ms	0	160

#### 2 Living Room

RTM	SHP	SPR	SIZ	HFC	PDL
.24ms	24	245	10	4832Hz	6ms
BAS	XOV	RTC	DIF	DCO	MIX
1.0	752k	5060Hz	38	Effects 4	All Fx
Preecho Levels					
-4dB	-4dB	-4dB	-4dB	-4dB	-4dB
Preecho Delays					
8ms	12ms	10ms	14ms	12ms	8ms

#### 3 Bathroom

RTM	SHP	SPR	SIZ	HFC	PDL
0.32s	18	231	7M	Full	0ms
BAS	XOV	RTC	DIF	DCO	MIX
0.4	120Hz	14986Hz	36	Effects 9	All Fx
Preecho Levels					
Full	Full	-7dB	-7dB	-8dB	-8dB
Preecho Delays					
8ms	4ms	14ms	12ms	6ms	6ms

#### 4 Kitchen Ambience

RT	RTL	SIZ	ROL	DIF	MIX
0.47s	78	3.5M	6.315Hz	36	All Fx
SPN	WAN	PDL	IND	MIX	
18	42	4.65ms	29.45	All Fx	

#### 5 Kellars Cell

RTM	SHP	SPR	SIZ	HFC	PDL
0.20s	37	12	2M	11084Hz	14ms
BAS	XOV	RTC	DIF	MOD	MIX
1.5	367	8.886	0	Reverb	All Fx
Preecho Levels				SPN	WAN
-3dB	-3dB	Off	Off	10	126µs
Preecho Delays				SHL	LEV
16ms	8ms	0ms	0ms	70	160

#### 6 Small Foley

RT	RTL	SIZ	ROL	DIF	MIX
0.18s	87	1.5M	6.315k	17	All Fx
SPN	WAN	PDL	IND	MIX	
10	672	4.65	19.65	All Fx	

#### 7 Warehouse

RT	RTL	SIZ	ROL	DIF	MIX
3.12s	44	40M	3.591k	40	All Fx
SPN	WAN	PDL	IND	MIX	
48	3738µs	41.26ms	0ms	All Fx	

#### 8 Airhead

RTM	SHP	SPR	SIZ	HFC	PDL
0.01s	0	0	2M	9.278k	0ms
BAS	XOV	RTC	DIF	MOD	MIX
1.5	493	6.315k	29	Reverb	All Fx
Preecho Levels				SPN	WAN
Off	Off	Off	Off	37	0µs
Preecho Delays				SHL	LEV
0ms	0ms	0ms	0ms	0	160

#### 9 Dial It Up

DST	EXM	XOV	HFC	XOM	DGN
255	63	654Hz	4.395Hx	All Fx	224
TSC	TBR	TBM	EXP	SUS	DHP
219	20dB	22dB	14	14dB	461Hz
DLY	LSL	LFR	LMG	ATC	RTC
12.31ms	130	-8dB	10dB	1	8
GAT					
99					

#### 0 Reverb Tail

RTM	SHP	SPR	SIZ	HFC	PDL
32.24s	101	246	39M	4.611k	0ms
BAS	XOV	RTC	DIF	MOD	MIX
1.2	243	6.315	51	Effects	All Fx
Preecho Levels			SPN	WAN	
Off	Off	Off	Off	44	19ms
Preecho Delays			SHL	LEV	
0ms	0ms	0ms	0ms	128	160

## Program Descriptions

### 1 Car Interior (Random Hall)

It's a 4-door. Raise the value of Shape to make it a station wagon. Raise pre-echo levels to close the windows!

### 2 Living Room (Random Hall)

The average suburban type.

### 3 Bathroom (480 Hall)

Larger than average.

### 4 Kitchen Ambience (Ambience)

Can you find your disposal sound effect?

### 5 Kellars Cell (Random Hall)

No, it's not the padded cell. Small, deep, and the surfaces are hard.

### 6 Small Foley (Ambience)

This program uses the Ambience algorithm for foley applications. Moving Size from its 1.5M setting, will cause the ambience bloom to "open up." Varying MIX from 100% wet, will present the 19ms dry delay into the audio path.

### 7 Warehouse (Ambience)

It's big — real big!

### 8 Airhead (Random Hall)

Take a pair of headphones, remove the elements and replace the m with diffuser panels spaced 10" from your ears. Now, hold a diffuser panel above your head. This program eliminates the need for you to look as though you are communicating with aliens. Use aggressively!

### 9 Dial It Up (Distression)

Telephone emulation — uses Frequency Distression.

### 0 Reverb Tail (Random Hall)

Not infinite. — A very warm, very long reverberant wash that makes a great fade. Unlike infinite, the inputs are always active.

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## **Bank 15: the Prime Time III Programs**

This chapter describes the PrimeTime III programs located in Bank 15.

<b>Page One</b>					
SPD Speed L	SPD Speed R	MAX Max Pitch Shift L	MAX Max Pitch Shift R	WIN Window L	WIN Window R
<b>Page Two</b>					
PDL Predelay	PDL Predelay	IND Dry Delay L	IND Dry Delay R	TRB Treble Cut	MIX Wet/Dry Mix
<b>Page Three</b>					
FBK Feedback Level L	FBK Feedback Level R	CFB Cross Feedback L	CFB Cross Feedback R	PAN Pan L	PAN Pan R

## The Prime Time III Program

The Prime Time algorithm provides left and right control mechanisms for predelay, high frequency rolloff (6db/octave), max pitch shift, speed, time event length, feedback, x-feedback, input delay, and delay pan. The input signal goes into a stereo delay line which has three modulation controls: MAX L&R, SPEED L&R, and WINDOW L & R. The glide is performed inside the delay lines, thus it is possible to have large amounts of delay but retain control of the overall modulation effect. These three glide parameters interact with each other. Speed is the rate of climb (acceleration slope) to maximum pitch shift; MAX is maximum amount of pitch shift; WIN is the time event length. WIN defines the amount of time over which the entire glide event takes place. The actual time that it takes a cycle of modulation to complete is determined by speed and the MAX pitch shift. The delay will always speed up to MAX and then back down again, which takes a certain amount of time. This time can be lengthened by setting WIN large. This allows long flanges with low pitch shifts to be achieved. Window is specified in samples, a unit of time. Feedback paths are routed outputs to inputs and x-feedbacks are routed outputs to opposite inputs. An input delay is provided to match the pre-delay for “over the top” type flanges while mixing “internally” in the box. The treble filter is functionally placed at the output of the algorithm.

## About the Parameters

### Page 1

#### SPD (Speed )

SPD controls the acceleration rate to reach maximum pitch shift. There are two controls for SPD, one for the left and one for the right.

#### MAX (Maximum Pitch Shift)

MAX sets the maximum amount of equivalent pitch shift. Each step is equivalent to 6 cents of pitch shift. The total maximum pitch shift can be set for 180 cents — almost a full tone. There are two independent controls for left and right.

#### WIN (Window)

Left and Right WIN controls define the minimum amount of time over which the entire glide event takes. WIN is specified in samples, a unit of time.



**Page 2**

**PDL and PDR (Left and Right Predelays)**

PDL and PDR are predelay lines which are independently set for left and right. The entire modulation effect takes place within these two delay lines. All feedbacks wrap back to these delay lines.

**INDL and INDR (Left and Right Input Delays)**

INDL and INDR are dry path delay lines. Any amount of dry mix passes through these two delay lines.

**TREB (Treble Cut)**

TRB controls a low-pass filter which rolls off the processed audio. It is functionally placed at the output of the algorithm.

**MIX**

MIX controls the amount of dry audio (unprocessed) relative to processed audio. Most programs have the mix value set to 100% wet (All FX).

**Page 3**

**FBK (Left and Right Feedback)**

FBL and FBR are feedback paths which route back to their respective inputs.

**CFB (Left and Right Cross Feedback)**

XFBL and XFBR are independent cross-feedbacks which route the outputs of each delay line to the opposite input.

**PAN**

Left and Right PAN controls position the processed audio of each delay line between the left and right outputs.

**Programs — Bank 12: Prime Time III**

**1 Prime Chorus**

SPD	SPD	MAX	MAX	WIN	WIN
2	19	3	3	59smp	31smp
PDL	PDL	IND	IND	TRB	MIX
19.30ms	18.55ms	22.79 ms	22.79 ms	13.4kHz	All Fx
FBK	FBK	CFB	CFB	PAN	PAN
2	-2	-11	-10	Left	Right

**2 Slap Chorus**

SPD	SPD	MAX	MAX	WIN	WIN
121	158	2	3	31smp	125smp
PDL	PDL	IND	IND	TRB	MIX
93.79ms	80.35ms	22.79ms	22.79ms	6.5kHz	All Fx
FBK	FBK	CFB	CFB	PAN	PAN
24	24	-27	-27	Left	Right

**3 Bounce Glide**

SPD	SPD	MAX	MAX	WIN	WIN
67	28	2	2	59smp	62smp
PDL	PDL	IND	IND	TRB	MIX
325.0ms	661.29ms	22.79ms	22.79ms	6.0kHz	All Fx
FBK	FBK	CFB	CFB	PAN	PAN
+6	+6	+70	+35	Right	Left

**4 Swirls**

SPD	SPD	MAX	MAX	WIN	WIN
3	5	2	3	51smp	43smp
PDL	PDL	IND	IND	TRB	MIX
0.0ms	0.0ms	8.30ms	5.62ms	5.62kHz	All Fx
FBK	FBK	CFB	CFB	PAN	PAN
-39	-39	24	-24	Left	Right

**5 Heavy Chorus**

SPD	SPD	MAX	MAX	WIN	WIN
110	85	4	5	78smp	74smp
PDL	PDL	IND	IND	TRB	MIX
13.34ms	8.89ms	5.62ms	5.62ms	6.5kHz	All Fx
FBK	FBK	CFB	CFB	PAN	PAN
-38	15	-15	38	Left	Right

**6 Shake It Up**

SPD	SPD	MAX	MAX	WIN	WIN
204	192	5	7	23smp	31smp
PDL	PDL	IND	IND	TRB	MIX
98.24ms	107.2ms	5.62ms	5.62ms	5.53kHz	All Fx
FBK	FBK	CFB	CFB	PAN	PAN
0	0	20	20	Right	Left

**7 Wowza**

SPD	SPD	MAX	MAX	WIN	WIN
182	260	4	5	78smp	39smp
PDL	PDL	IND	IND	TRB	MIX
0.0ms	0.00	18.30ms	5.62ms	5.5kHz	All Fx
FBK	FBK	CFB	CFB	PAN	PAN
-36	40	40	-29	Left	Right

**8 Wowza 2**

SPD	SPD	MAX	MAX	WIN	WIN
115	115	4	5	59smp	122smp
PDL	PDL	IND	IND	TRB	MIX
10.6ms	9.44ms	5.62ms	5.62ms	7.8kHz	All Fx
FBK	FBK	CFB	CFB	PAN	PAN
-38	21	38	-21	70	30

**9 Vocalz**

SPD	SPD	MAX	MAX	WIN	WIN
39	49	3	4	771smp	850smp
PDL	PDL	IND	IND	TRB	MIX
4.44ms	4.44ms	48.57ms	48.57ms	4.18kHz	80 Fx
FBK	FBK	CFB	CFB	PAN	PAN
-21	21	25	-25	Left	Right

**0 X Flange**

SPD	SPD	MAX	MAX	WIN	WIN
50	25	1	1	200smp	500smp
PDL	PDL	IND	IND	TRB	MIX
0.0ms	0.0ms	24.21ms	24.21ms	19.4kHz	All Fx
FBK	FBK	CFB	CFB	PAN	PAN
-30	-30	-45	-45	Right	Left

**Program Descriptions****1 Prime Chorus**

This is a basic preset which independently wobbles the left and right audio sources. Each channel runs asynchronously (different speeds and windows). However, there is a small amount of cross feedback which runs the output of each channel back into the opposite input.

**2 Slap Chorus**

Similar to Prime Chorus except a longer pre-delay is utilized. Great for chunky electric guitar rhythm pads. Reduce feedbacks and cross feedbacks for less decay.

**3 Bounce Glide**

This preset bounces back and forth. The outputs are panned reversed, so don't drive yourself crazy if you feed in a discrete stereo source. The delay returns are slowly modulated. Pushing MAX to a greater value will result in a heavy, heavy unearthly effect.

**4 Swirls**

This preset is a fast moving modulating effect. Each pre-delay is set to 0ms and all feedbacks and cross feedbacks are used with one of each phase inverted. Any use of this effect which winds up passing through a surround decoder will swirl all around you.

**5 Heavy Detune**

Use with caution. Strong effect for an aggressive "Idol" type lead vocal.

**6 Shake It Up**

Strong slap-fast vibrato effect. For more intensity, increase the values of FBL and FBR to 50. If you slow it down with the speed controls you may want to lessen MAX shift as well. The outputs are panned reversed.

**7 Wowza**

Kinda like a fast Leslie on acid....

**8 Wowza 2**

Four hours later.....

**9 Vocalz**

This preset is somewhat filtered down, but the dry delay set to 48ms with the mix control set to 80% helps create a doubling effect. Definitely for smooth sound sources.

**0 X Flange**

Use this pre-fader. This is an "over the top" flange effect which chases its own tail. Mix the returns to mono for an even stronger effect. In stereo it drives a surround decoder crazy.

---

## **Bank 16: the Freq. Stuff Programs**

This chapter describes the Frequency Dynamics and Distress programs located in Bank 16.

<b>Page One</b>					
TSL Treble Slope	TBR Treble Rotate	TBM Treble Max Gain	XOV Xover Frequency	HFC High Freq Cut	XOM Xover Mix
<b>Page Two</b>					
LSL Lo Freq Comp Ratio	LFR Lo Freq Rotation Pt	LMG Lo Freq Max Gain	ATC Attack Time Const	RTC Release Time Const	DLY Front Delay
<b>Page Three</b>					
GATE Noise Gate					

## The Frequency Dynamics Program

This frequency compression algorithm is a stereo two channel effect which is very similar to the Distression Program except there is no distortion module. There are various presets which create a distinct effect as an enhancement to the original source.

The Frequency Dynamics program utilizes an adjustable cross-over feeding independent bass and treble compressors. There is an adjustable gate which can conceptualized as being at the output of the processor.

**ALL PRESETS SHOULD BE UTILIZED  
"PRE-FADER" FROM THE CONSOLE SOURCE.**

## About the Parameters

### Page 1

#### TSL (Treble Slope)

TSL defines the ratio of the high-frequency compressor. A LARC value of 128 equals a 2:1 compression ratio.

#### TBR (Treble Rotate - threshold)

TBR determines the point at which treble compression starts to occur.

#### TBM (Treble Maximum Gain)

TBM controls the amount of gain introduced below the Rotate value. ie: If Rotate is set to -14, and gain is set at 10, when signals go below -14dB (average) then the signal will be boosted by 10dB.

#### XOV (Crossover Frequency)

XOV is a6dB/octave low frequency, 12dB/octave high frequency sliding filter which determines what frequencies are low frequency and high frequency.

#### HFC (High Frequency Cut)

HFC controls the roll-off characteristics of the treble compressed content of the signal.

#### XOM (Crossover Mix - sine/cosine)

XOM controls the mix of low frequency and high frequency signals. It is the final mix control in the program.

### Page 2

#### LSL (Low Frequency Compression Ratio)

LSL defines the ratio of the low-frequency compressor. A LARC value of 128 equals a 2:1 compression ratio.

#### LFR (Low Frequency Rotation Point)

LFR determines the point at which low-frequency compression starts to occur.

#### LMG (Low Frequency Maximum Gain)

LMG controls the amount of low-frequency gain introduced below the Rotate value. ie: If Rotate is set to -24, and gain is set at 6, when signals go below -24dB (average) then the signal will be boosted by as much as 6dB.

#### ATC (Attack Time Constant)

Sets the attack time.

#### RTC (Release Time Constant)

Sets the release time.

#### DLY (Front Delay)

DLY determines the amount audio delay before the compressor modules. This effectively allows the 480L to perform a "look ahead" function to the compressors.

### Page 3

#### GATE (Noise Gate)

GATE sets a value at which the noise gate starts to occur. Units are described in dB below 0. These values should, typically, be set very high.

## Programs — Bank 16: Frequency Dynamics

### 1 Mix Finish

TSL	TBR	TBM	XOV	HFC	XOM
144	-20	27dB	7.4K	FULL	60%
LSL	LFR	LMG	ATC	RTC	DLY
140	-21	6dB	1	5	8.23
GATE					
94					

### 2 Big Vocals

TSL	TBR	TBM	XOV	HFC	XOM
141	-20	12	654Hz	full	54%
LSL	LFR	LMG	ATC	RTC	DLY
180	-26	5	1	5	5.15ms
GATE					
99					

### 3 Drums

TSL	TBR	TBM	XOV	HFC	XOM
255	20	17	5.5K	12.7K	43%
LSL	LFR	LMG	ATC	RTC	DLY
123	30	8	1	5	6.07
GATE					
87					

### 4 AC Guitars

TSL	TBR	TBM	XOV	HFC	XOM
96	20	27	3.0k	full	60%
LSL	LFR	LMG	ATC	RTC	DLY
128	-21	6	2	5	8.23ms
GATE					
94					

### 5 Bass Thump

TSL	TBR	TBM	XOV	HFC	XOM
166	-15	5dB	90	6.04k	30%
LSL	LFR	LMG	ATC	RTC	DLY
248	-25	14dB	3	7	23.2
GATE					
90					

## Program Descriptions

### 1 Mix Finish

This preset adds a nice compressed top end (>7.4k) to any mix. The low end component is also “pressed” up as well. If too much, lower XOV (crossover) slightly and TBM (max gain).

### 2 BG Vocals

BG Vocals will enhance a background vocal premix before compression. You may need to adjust both TBR(Treble Rotate) and LFR(Low Frequency Rotate).

### 3 Drums

Quick attack and quick release. Great for a drum premix. Strong compression on the top end.

### 4 AC Guitars

Soft Compression above 3.0kHz. 2:1 ratio below 3.0kHz. Try it on a piano track as well.

### 5 Bass Thump

Pumped up hard below 60Hz. Predelay is set to 23ms so as to “look ahead” and catch any low frequency transients. Need more hump, increase XOV one or two “ticks” but you may need to reduce LSL(low frequency slope).

<b>Page One</b>					
DST Distortion Mix	EXM Expansion Mix	XOV Xover Frequency	HFC High Freq Cut	XOM Xover Mix	DGN Distortion Gain
<b>Page Two</b>					
TSL Treble Slope	TBR Treble Rotate	TBM Treble Max Gain	EXP Expansion Slope	SUS Sustain	DHP Distortion Hi Pass
<b>Page Three</b>					
DLY Front Delay	LSL Lo Freq Comp Ratio	LFR Lo Freq Rotation Pt	LMG Lo Freq Max Gain	ATC Attack Time Const	RTC Release Time Const
<b>Page Four</b>					
GATE Noise Gate					

## The Distression Program

The Frequency Distression algorithm is extremely versatile at creating frequency compressed or distorted effects.

Distression is a Mono in (left channel), Dual Mono output program. The right channel input is ignored and not processed. These programs can be run in any 480L configuration.

**ALL PRESETS SHOULD BE UTILIZED  
"PRE-FADER" FROM THE CONSOLE SOURCE.**

This program can be thought of as a low pass/high pass crossover feeding independent compressor modules. The high pass compressor feeds a dedicated distortion module capable of driving various parametric controls. The Frequency Distression programs represent a wide variety of different frequency-compressed and distorted effects from the very subtle to the awesome sublime. Many presets are simply labeled with an instrument name. This suggests the type of instrument which should be fed into the effect input.

## About the Parameters

### Page 1

#### **DST (Distortion Mix - sine/cosine)**

DST controls the amount of signal following treble compression either passing through the distortion module or bypassing around it.

#### **EXM (Expansion Mix - linear)**

EXM controls the amount of signal following treble compression either passing through all of the distortion parameters (including the distortion expansion) or bypassing it. A LARC value of 0 means absolutely no amount of distorted signal is being passed to XOM.

#### **XOV (Crossover Frequency)**

XOV is a 6dB/octave low frequency, 12dB/octave high frequency sliding filter which determines what frequencies are low frequency and high frequency.

#### **HFC (High Frequency Cut)**

HFC controls the roll-off characteristics of the distorted content of the signal.

#### **XOM (Crossover Mix - sine/cosine)**

XOM controls the mix of low frequency and high frequency signals. It is the final mix control in the program.

#### **DGN (Distortion Gain)**

DGN controls the amount of distortion gain up to 6dB.

## Page 2

### **TSL (Treble Slope)**

TSL defines the ratio of the high-frequency compressor. A LARC value of 128 equals a 2:1 compression ratio.

### **TBR (Treble rotate - threshold)**

TBR control determines the point at which compression starts to occur.

### **TBM (Treble Maximum Gain)**

TBM controls the amount of gain introduced below the Rotate value. ie: If Rotate is set to -14, and gain is set at 10, when signals go below -14dB (average) then the signal will be boosted by 10dB.

### **EXP (Expansion Slope)**

EXP determines the amount of expansion after the distortion DSP.

### **SUS (Sustain)**

SUS works in conjunction with the expander to provide additional sustain throughout dynamic transitions.

### **DHP (High Pass on Distortion)**

DHP controls a 6dB/octave filter which is conceptually placed after the distortion module but before the Expander.

## Page 3

### **DLY (Front Delay)**

DLY determines the amount audio delay before the compressor modules. This effectively allows the 480L to perform a "look ahead" function to the compressors.

### **LSL (Low frequency Compression Ratio)**

LSL defines the ratio of the low-frequency compressor. A LARC value of 128 equals a 2:1 compression ratio.

### **LFR (Low Frequency Rotation Point)**

LFR d determines the point at which low-frequency compression starts to occur.

### **LMG (Low Frequency Maximum Gain)**

LMG controls the amount of low-frequency gain introduced below the Rotate value. ie: If Rotate is set to -24, and gain is set at 6, when signals go below -24dB (average) then the signal will be boosted by as much as 6dB.

### **ATC (Attack Time Constant)**

ATC sets the attack time.

### **RTC (Release Time Constant)**

Sets the release time.

## Page 4

### **GATE (Noise Gate)**

GATE sets a value at which the noise gate starts to occur. Units are described in dB below 0. These values should, typically, be set very high.

## Programs — Bank 16: Distresson

### 6 Carbon Mic

DST	EXM	XOV	HFC	XOM	DGN
208	63	181Hz	5.5k	allfx	106
TSL	TBR	TBM	EXP	SUS	DHP
84	-17	17dB	3	94	304Hz
DLY	LSL	LFR	LMG	ATC	RTC
5.82m	122	-19	9dB	2	4
GATE					
99					

### 7 Saxophone

DST	EXM	XOV	HFC	XOM	DGN
225	45	430Hz	5.5k	52%	255
TSL	TBR	TBM	EXP	SUS	DHP
255	14	20	7	68	4.6k
DLY	LSL	LFR	LMG	ATC	RTC
7.32k	196	-19	10	3	7
GATE					
99					

### 8 Horn Blast

DST	EXM	XOV	HFC	XOM	DGN
255	63	60Hz	6.3k	50%	57
TSL	TBR	TBM	EXP	SUS	DHP
226	15	9dB	94	21	90
DLY	LSL	LFR	LMG	ATC	RTC
10.3m	206	-23	9dB	2	4
GATE					
99					

### 9 Softener

DST	EXM	XOV	HFC	XOM	DGN
65	63	151	5.5k	64%	0
TSL	TBR	TBM	EXP	SUS	DHP
190	21	22	115	0	212Hz
DLY	LSL	LFR	LMG	ATC	RTC
14.9m	82	-13	0	2	8
GATE					
77					

### 0 Some Fuzz

DST	EXM	XOV	HFC	XOM	DGN
255	63	752Hz	6.3	64%	66
TSL	TBR	TBM	EXP	SUS	DHP
226	15	21	0	0	30Hz
DLY	LSL	LFR	LMG	ATC	RTC
5.82	206	-23	9	2	4
GATE					
99					

## Program Descriptions

### 6 Carbon Mic

The sound from 1938. This preset sends a soft amount of treble gain into the distressor. The treble component starts at 181Hz. Crossover(XOM) mix is set to all treble. EXM is also set to max at 63. DHP (Distresson High Pass) rolls off the bottom end. If you reduce XOM, more of the low frequency component will be audible. Use this to obtain an old, scratchy sort of sound.

### 7 Saxophone

This preset is designed as an alternative to reaching for the EQ knobs. There is a reasonable amount of low-frequency compressed energy. Take the result and send into a Plate program in the Cascade mode machine B. (Mix the Plate program to something other than 100%)

### 8 Horn Blast

This aggressive preset is designed to enhance your horn section. If you want to tone it down, reduce HFC(High Frequency Cut for distresson component) and/or reduce TSL (treble slope).

### 9 Softener

Very soft edge on anything. Try it on a vox track. Subtle effect.

### 0 Some Fuzz

If you need More Fuzz increase DGN(Distresson Gain), lower XOY and increase XOM.



---

## Bank 17: the Test & Reference Programs

This chapter describes the Test & Reference programs located in Bank 17. These are:

- The Oscillator Programs (1-8)
- Pink Noise (9)
- Binaural Simulator (0)

Page One					
LVL Level	DEC Decade	FRQ Frequency	WAV Wave Form	ACC Accuracy	ILEV Input Level

## The Oscillator Program

Unlike standard 480L programs, the Oscillator reconfigures the mainframe so that signal no longer passes from inputs to outputs. This allows the mainframe outputs to be fed to the mainframe inputs to calibrate analog signal levels. Likewise, an external signal generator (from, for instance, a console) can be used to calibrate the mainframe for a studio's design center.

### About the Parameters

#### LVL (Output Level)

LVL controls the oscillator output for both digital and analog signal paths.

#### DEC (Decade)

DEC multiplies/divides the selected frequency in x10, x100, or x1000 increments. The preset value is 2.

Decade	Example: Frequency = 20 Hz
0 = ÷	.2
1 = ÷	2
2 = Norm	20
3 = x 10	200
4 = x 100	2000
5 = x 1000	20000

As you can see, when the oscillator is set for high resolution, .01 Hz increments can be selected with a combination of frequency and decade. For example: If the desired frequency is 1.02 Hz, move DEC to 3, set FRQ to 1020 Hz, then move DEC to 0.

#### FRQ (Frequency)

Sets the frequency of the oscillator, and displays the actual oscillator frequency in Hz. and .01Hz. increments.

#### WAV (Wave Form)

WAV changes the oscillator waveform as follows:

- 0 = Sine Wave
- 1 = Triangle Wave
- 2 = Square Wave

#### ACC (Accuracy)

ACC adjusts the accuracy of the oscillator as follows:

- 0 = .5%
- 1 = .01%

#### ILEV

ILEV displays input level in 0.01dB increments over a 90dB range from dBfs. (dBfs = 90.00dB)

Meter detection is always averaged. A plus sign (+) will appear next to the L or R channel display to indicate long averaged mode. Moving the slider changes averaging time from approximately 1 second to weeks!

## Programs — Bank 17: Oscillator

### 1 100Hz-12dB

LVL	DEC	FRQ	WAV	ACC	ILEV
-12dB	2	100.00	0	1	-

### 2 500Hz-17dB

LVL	DEC	FRQ	WAV	ACC	ILEV
-17dB	2	500.00	0	1	-

### 3 500Hz-12dB

LVL	DEC	FRQ	WAV	ACC	ILEV
-12dB	2	500.00	0	1	-

### 4 1kHz-17dB

LVL	DEC	FRQ	WAV	ACC	ILEV
-17dB	3	1000.0	0	1	-

### 5 1kHz-12dB

LVL	DEC	FRQ	WAV	ACC	ILEV
-12dB	3	1000.0	0	1	-

### 6 10kHz -12dB

LVL	DEC	FRQ	WAV	ACC	ILEV
-12dB	4	10000.0	0	1	-

### 7 30Hz Slate

LVL	DEC	FRQ	WAV	ACC	ILEV
-12dB	2	30.00	0	1	-

### 8 A-440 Tuner

LVL	DEC	FRQ	WAV	ACC	ILEV
-12dB	2	440.00	0	1	-

### Program Descriptions

Programs 1-8 are sinewave high accuracy oscillators with frequency and level settings that match program names.

## Programs — Bank 17: Pink Noise

### 9 Pink Noise

LVL
-12dB

### Program Description

The Pink Noise program produces filtered noise with equal energy per octave. The only parameter provided is the level control, which attenuates from dBFS and displays attenuation in dB. The program loads with a value of -12dB from dBFS.

### About the Binaural Simulator Program

The Binaural Simulator is designed to alter stereo recordings for binaural presentation using headphone monitoring. It includes the ability to add controlled amounts of early reflections and reverberation, as well as the ability to independently alter the binaural width and level of direct sound, reflections, and reverberation.

Although it can be used in all machine configurations, this program is intended for use in a cascade configuration with it loaded into Machine B and a reverb or ambience program loaded into Machine A.

The synthesis is based on a simple model of the human head. Information normally fed to one channel is delayed, low pass filtered, and cross-fed to the opposite channel. By adjusting delay time and frequency of low pass filters, apparent binaural width is realistically varied.

The Binaural Simulator has three independent synthesis circuits: one of direct sound, one for early reflections, and one for reverberation. Reverberation generated in Machine A is fed directly to the reverberation simulator. Direct sound feeding 480L inputs is fed to the direct sound simulator. The program includes six independent echo delays, each with variable time and level control parameters. Their outputs are fed to the early reflections simulator. A precision mixer provides level adjustment for each simulator.

**About the Parameters**

**Page One**

**DRY (Dry Level)**

Controls the level of the direct sound. The output of this parameter is fed to the direct sound simulator.

**WET (Reverb Level)**

Controls the level of the reverberation generated in Machine A. This slider interacts with the Cal parameter on Page 4. The output of this parameter is fed to the reverberation simulator.

**ERL (Early Reflections Level)**

Master level control for the levels found on Page 2. The output of this parameter is fed to the early reflections simulator.

**DWT (Dry Width)**

Sets the interaural time delay for the dry signal. It becomes active when raised above 0.

**WDT (Reverberation Width)**

Sets the interaural time delay for the reverberant signal. It becomes active when raised above 0.

**EWT (Early Reflection Width)**

Sets the interaural time delay for the early reflections. It becomes active when raised above 0.

**Page Two**

**Pre-Echo Levels**

Pre-echoes can best be understood by visualizing a stage where the early reflections are the sound emanating from the rear and side stage walls immediately following the direct sound. An independent level control is provided for each delay. There are a total of six pre-echoes, some which cross-pan. Each pre-echo delay is actually a cluster of early reflections with density controlled by DIF.

**Page Three**

**Pre-Echo Delay Times**

For each of the pre-echo level parameters, there is a corresponding pre-echo delay time parameter.

**Page Four**

**DIF (Diffusion)**

Controls the density of the pre-echoes.

**DOF (Diffusion Off)**

Enables/disables the diffusion set by DIF. This control allows you to quickly reference undiffused early reflections to any value of DIF.

**DWC (Dry Treble Width Control)**

Low pass frequency for the direct sound synthesizer. For maximum width this control should be set between 500Hz and 1.5kHz.

**WET TWC (Reverberation Treble Width Control)**

Low pass frequency for the reverberation synthesizer. For maximum width this control should be set between 500Hz and 1.5kHz.

**EARLY TWC (Early Reflection Treble Width Control)**

Low pass frequency for the early reflection synthesizer. For maximum width this control should be set between 500Hz and 1.5kHz.

**CAL (Reverberation Level Calibration)**

Controls the amount of gain applied to the output of the reverberator or ambience program in Machine A. Set to -7.5 when DRY TWC is off. Set to -4.5 when DRY TWC is 1Hz or higher.

**Page Five**

**ROL**

Sets the overall low pass for the early reflections.

**0 Binaural Simulator**

DRY	WET	ERL	DWT	WDT	CWT
0	Off	Off	30	30	30
[	PRE-	ECHO	-LEV	ELS	]
-5dB	-5dB	-5dB	-5dB	-5dB	-5dB
[	PRE-	ECHO	-DEL	AYS	]
11.39ms	22.38ms	22.79ms	36.19ms	42.43ms	54.99ms
DIF	DOF	DWC	WWC	EWG	CAL
80	0	1020kHz	1020kHz	1020kHz	-4.50
ROL					
Full					

**Program Description**

The Binaural Simulator program has parameters set to provide width for all signals on loading, but only the dry level has been raised. Pre-echo levels and pre-echo delays are set to provide a field of early reflections, but the ERL parameter must be raised in order to turn them on. Likewise, the WET parameter must be raised in order to hear the reverb in Machine A.



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## **MIDI and the 480L**

This section 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

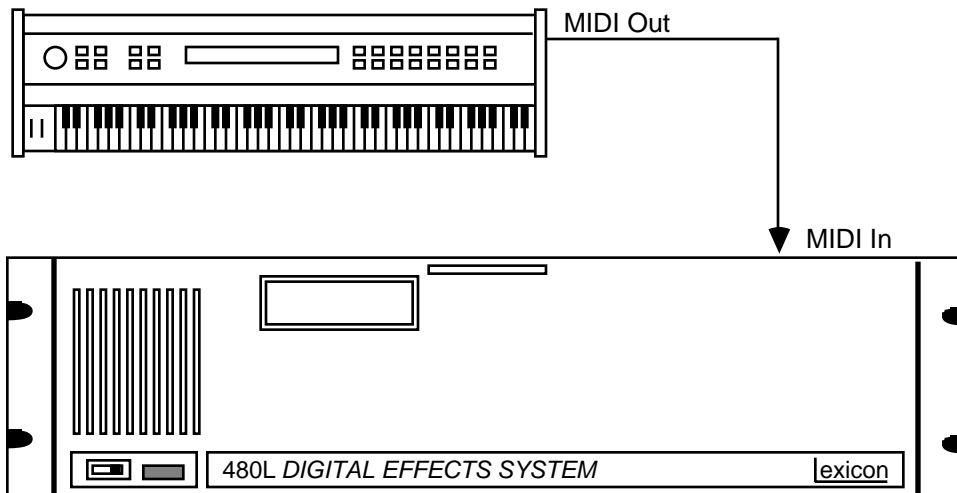
We'll discuss each of these applications in this chapter, but first let's cover some typical MIDI installations.

## MIDI Connections

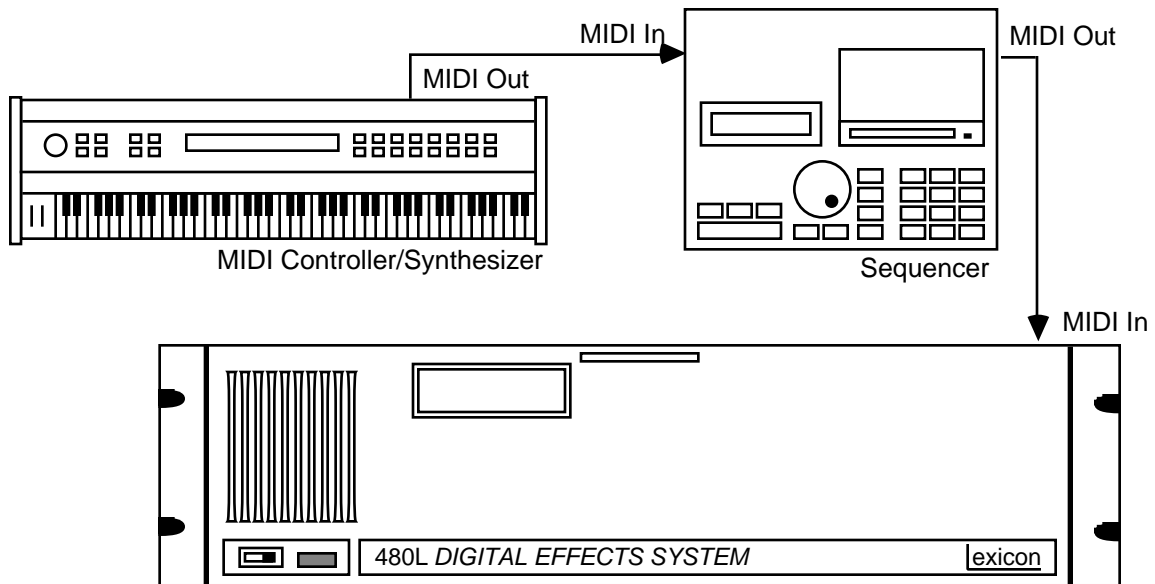
All MIDI connections described in this chapter use the MIDI IN, THRU, and OUT connectors located on the rear panel of the 480L. As with any MIDI connection, use only standard MIDI cables and keep them as short as possible to avoid possible data errors. 15 meters is generally accepted as the longest length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



Connecting a Keyboard and the 480L.



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 and 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.

## MIDI SysEx Data Dumps

### MIDI Bulk Dumps

Bulk Data dumps are handled through the Control mode as a Copy function. Moving the FUN slider will scroll through the current set of Copy functions which include Bulk Data Dumps; Dump Reg, Dump Mach A, Dump Mach B, Dump All Internals, Dump All externals, Dump MIDI Map A, and Dump MIDI Map B.

Selecting Dump Reg allows individual source registers to be selected. After identifying Dump Reg as the MIDI Dump type, simultaneously press STORE and REG to initiate the dump (exactly like Copying individual registers).

### MIDI Bulk Loads

Bulk data loads happen automatically on receipt of a "send" command from the MIDI storage device if the 480L is not be in the Copy Prohibit mode. All Bulk data loads can be selected to load to the internal or external registers (manually selected) or they can be automatically loaded in the original location (for example, internal registers reload to internal).

### MIDI Automation

MIDI Automation via SysEx is enabled on Page 5 in Control Mode. There are two states of AUTO: OFF, and TRANSMIT. If AUTO is set to TRANSMIT, all LARC movements will have a corresponding data string output attached.

## MIDI Implementation

Lexicon 480L

Function		Transmitted	Recognized	Remarks
Basic Channel	Default	X	O	1
	Channel	X	O	1-16
Mode	Default	X	Mode 3	
	Messages	X		
	Altered	X		
Note Number	True Voice	X	O	0-127 Used as controller
		X	O	
Velocity	Note ON	X	O	1-127 Used as controller
	Note OFF	X	X	
After Touch	Keys	X	X	Used as controller
	Channels	X	O	
Pitch Bender		X	O	Used as controller
Control Change	0-95	X	O 0-31 64-95	Controller can be patched to control effect parameters
Program Change	True #	X	O	0-127 table or fixed
		X	X	
System Exclusive		O	O	
System Common	:Song Pos	X	X	
	:Song Sel	X	X	
	:Tune	X	X	
System Real Time	:Clock	X	X	
	:Commands	X	X	
Aux Messages	:Local ON/OFF	X	X	
	:All Notes OFF	X	X	
	:Active Sense	X	X	
	:Reset	X	X	
Notes:				

Mode 1: OMNI ON, POLY  
Mode 3: OMNI OFF, POLYMode

Mode 2: OMNI ON, MONO  
4: OMNI OFF, MONO  
O : Yes  
X : No

## Sysex Message Formats

### 480 Generic System Exclusive Message Format

\$F0	System Exclusive ID
\$06	Lexicon ID
\$01	480 ID
0ccc nnnn	Message Class and Sysex Device ID ccc = Message Class 0-7 nnnn = Channel Device ID 0-15
0sssssss	Message Subclass sssssss = Subclass \$0-\$7F
0xxxxxxx	Data Characters xxxxxxx = Data \$0-\$7F
\$F7	End of System Exclusive

Many messages have checksums associated with them. These checksums are calculated by simply adding all of the bytes of the checksummed data together and truncating to 7 bits. The message formats describe which data bytes are checksummed.

### Automation Format

Automation data is used to transmit and receive sysex automation data

### Parameter Data Format

\$F0	480 System Exclusive Header	
\$06		
\$01		
\$2n	Parameter Data Message Class, Sysex Channel Device ID	
\$0d	Subclass = Control Class	
\$2	Global Controls	
\$3-\$4	Program Controls Machine A and Machine B	
\$5-\$6	Patch Controls Machine A and Machine B	
\$7-\$8	Process Controls Machine A and Machine B	
\$9-\$A	Table Controls Machine A and Machine B	
\$B-\$C	Name Controls Machine A and Machine B	
0nnnnnnn	Parameter Number Index within Control Class 0-127	
\$0v	Nibblized Most Significant	4 bits (out of 20 bits)
\$0v		4 bits
\$0v		4 bits
\$0v		4 bits
\$0v	Nibblized Least Significant	4 bits (out of 20 bits)
\$F7	End of System Exclusive	

**Event Data Format**

\$F0	480 System Exclusive Header		
\$06			
\$01			
\$3n	Parameter Data Message Class, Sysex Channel Device ID		
\$0d	Subclass = Control Class		
\$2	Global Controls		
\$3-\$4	Program Controls Machine A and Machine B		
\$5-\$6	Patch Controls Machine A and Machine B		
\$7-\$8	Process Controls Machine A and Machine B		
\$9-\$A	Table Controls Machine A and Machine B		
\$B-\$C	Name Controls Machine A and Machine B		
0nnnnnnn	Event Number Index within Control Class 0-127		
\$0v	Nibblized Most Significant	4 bits (out of 20 bits)	
\$0v		4 bits	
\$0v		4 bits	
\$0v		4 bits	
\$0v	Nibblized Least Significant	4 bits (out of 20 bits)	
\$F7	End of System Exclusive		

**Stored Bulk Data Format**

\$F0	480 System Exclusive Header		
\$06			
\$01			
\$1n	Parameter Data Message Class, Sysex Channel Device ID		
0sss 0000	Subclass = Setup Class		
\$30	Internal Register		
\$50	Preset		
\$70	External (Cartridge) Register		
\$0n	Nibblized Setup Index	Most Significant Bits	4 bits (out of 12)
\$0n			4 bits (out of 12)
\$0n	Nibblized Setup Index	Least Significant Bits	4 bits (out of 12)
\$0c	Nibblized Data Byte Count	Most Significant Bits	4 bits (out of 16)
\$0c	(Number of Nibblized Pairs + 4)		4 bits (out of 16)
\$0c			4 bits (out of 16)
\$0c		Least Significant Bits	4 bits (out of 16)
\$0x	Nibblized Data Byte Pairs	Most Significant Bits	4 bits (out of 8)
\$0x		Least Significant Bits	4 bits (out of 8)
\$0b	Bulk Data Flags		4 bits (out of 32)
\$09			4 bits (out of 32)
\$06			4 bits (out of 32)
\$0d			4 bits (out of 32)
\$0b			4 bits (out of 32)
\$06			4 bits (out of 32)
\$09			4 bits (out of 32)
\$0d			4 bits (out of 32)
0mmmmmm	Data Checksum calculated by adding each data character to previous sum starting with Nibblized Setup Index		
\$F7	End of System Exclusive		

**Active Bulk Data Format Active Machines A and B**

\$F0	480 System Exclusive Header		
\$06			
\$01			
\$0n	Parameter Data Message Class, Sysex Channel Device ID		
0sss ssss	Subclass = Active Class		
	\$33	Active Machine A	
	\$34	Active Machine B	
\$0n	Nibblized Setup Index	Most Significant Bits	4 bits (out of 12)
\$0n			4 bits (out of 12)
\$0n	Nibblized Setup Index	Least Significant Bits	4 bits (out of 12)
\$0c	Nibblized Data Byte Count	Most Significant Bits	4 bits (out of 16)
\$0c	(Number of Nibblized Pairs + 4)	4 bits (out of 16)	
\$0c			4 bits (out of 16)
\$0c		Least Significant Bits	4 bits (out of 16)
\$0x	Nibblized Data Byte Pairs	Most Significant Bits	4 bits (out of 8)
\$0x		Least Significant Bits	4 bits (out of 8)
\$0b	Bulk Data Flags		4 bits (out of 32)
\$09			4 bits (out of 32)
\$06			4 bits (out of 32)
\$0d			4 bits (out of 32)
\$0b			4 bits (out of 32)
\$06			4 bits (out of 32)
\$09			4 bits (out of 32)
\$0d			4 bits (out of 32)
0mmmmmm	Data Checksum Calculated by adding each data character to previous sum starting with Nibblized Setup Index		
\$F7	End of System Exclusive		

**Active Bulk Data Format Active Global Controls**

\$F0	480 System Exclusive Header		
\$06			
\$01			
\$0n	Parameter Data Message Class, Sysex Channel Device ID		
\$50	Subclass = Active Global Controls Class		
\$0c	Nibblized Data Byte Count	Most Significant Bits	4 bits (out of 16)
\$0c	(Number of Nibblized Parameters)		4 bits (out of 16)
\$0c			4 bits (out of 16)
\$0c		Least Significant Bits	4 bits (out of 16)
\$0x	Nibblized Data Bytes	Most Significant Bits	4 bits (out of 16)
\$0x			4 bits (out of 16)
\$0x			4 bits (out of 16)
\$0x		Least Significant Bits	4 bits (out of 16)
0mmmmmm	Data Checksum Calculated by adding each data character to previous sum starting with Nibblized Data Byte Count		
\$F7	End of System Exclusive		

### MIDI Program Table Map Format

\$F0	480 System Exclusive Header		
\$06			
\$01			
\$7n	Parameter Data Message Class, Sysex Channel Device ID		
\$0p	Subclass = Machine		
	p = 0	= Machine A	
	p = 1	= Machine B	
\$0x	Nibblized Data Bytes	Most Significant Bits	4 bits (out of 12)
\$0x			4 bits ( out of 12)
\$0x		Least Significant Bits	4 bits (out of 12)

(128 total 12 bit words = 384 data byte characters)

0mmmmmm	Data Checksum Calculated by adding each data character to previous sum starting with Nibblized Data Bytes		
\$F7	End of System Exclusive		

### Request Data Format

\$F0	480 System Exclusive Header		
\$06			
\$01			
\$4n	Parameter Data Message Class, Sysex Channel Device ID		
0ssssss	Subclass		
0nnnnnn	Request Opcode		
	\$00	\$03	All Registers (Internal and External)
	\$00	\$05	Preset Setup
	\$00	\$07	Register Setup
	\$00	\$09	Global Controls
	\$03	\$09	Machine A
	\$04	\$09	Machine B
	\$03	\$0A	Midi Program Map Machine A
	\$04	\$0A	Midi Program Map Machine B
\$0x	Nibblized Index		Most Significant Bits 4 bits (out of 16)
\$0x			4 bits ( out of 16)
\$0x			4 bits ( out of 16)
\$0x			Least Significant Bits 4 bits (out of 16)
\$F7	End of System Exclusive		

### Understanding Events and Parameters

The system of events and parameters used by the 480 is directly accessible using the automation mode. These events and parameters have a direct relationship to the events and parameters available from the LARC. Every parameter and event seen on the LARC can be changed using automation.

The order and relative offset of the parameters and events do not correspond with the parameters displayed on the LARC. To determine which parameter number is associated with which LARC parameter, experiment with different parameters and events, record the output with SysEx automation, and examine the results.

As an example, the mono sampling preset has several parameters and events. The LARC displays the following:

MONO 3S

Page 1	-	REC	MARK	MODE	CAP	CHK
Page 2	DUB	REC	FAD	MODE	CAP	CHK
Page 3	HED	TIM	-	-	FAD	CUE
Page 4	PLAY	RP	TLV	HOLD	-	CUE

Parameters and Events are in separate lists from the automation viewpoint. The parameter list for “Mono 3S” is:

voice	Larc: NA	Automation: Parameter: 0
mark	Larc: Page 1 Slider 3	Automation: Parameter: 1
C_trig	Larc: Page 1 Slider 4	Automation: Parameter: 2
shift	Larc: NA	Automation: Parameter: 3
ftime	Larc: Page 3 Slider 2	Automation: Parameter: 4
head	Larc: Page 3 Slider 1	Automation: Parameter: 5
rtime	Larc: NA	Automation: Parameter: 6
tail	Larc: NA	Automation: Parameter: 7
play	Larc: Page 2 Slider 3	Automation: Parameter: 8
repeat	Larc: Page 4 Slider 2	Automation: Parameter: 9
P_trig	Larc: Page 4 Slider 3	Automation: Parameter: 10
holdoff	Larc: Page 4 Slider 4	Automation: Parameter: 12

Some of the parameters are repeated on multiple pages. This list represents just one instance of each parameter.

Events are very similar. As far as the automation is concerned the events are ordered based on an internal representation. The Larc representation is not predictable. The event list for Mono 3S is:

rec	Larc: Page 1 Button 2	Automation: Event: 1
stop	Larc: Page 1 Button 5	Automation: Event: 2
rcue	Larc: Page 1 Button 6	Automation: Event: 3
overdub	Larc: Page 2 Button 1	Automation: Event: 4
repro	Larc: Page 4 Button 1	Automation: Event: 5
cue	Larc: Page 3 Button 6	Automation: Event: 6

Some of the events are repeated in several places. This list is represents just one instance of each parameter. In the case of “Mono 3S” all of the events available from the Larc are available from the Automation mode.



## Understanding Preset and Register Indexes

Presets and Registers share a range of indexes to represent locations within the 480 software. All indexes are in the range of 1-1023. The range 1-923 is reserved for presets. Currently presets 1-210 are available. Indexes between the range of 924 and 973 are internal registers and 974 through 1023. These numbers are used in all automation or bulk dump sysex messages when referring to a stored or preset 480 setup.

The following equations define the locations in the array:

### Presets

$\text{index} = (\text{bankNumber} - 1) * 10 + (\text{programNumber} - 1)$   
where "PGM0" represents  $\text{programNumber} = 10$

### Internal Registers

$\text{index} = (\text{registerBankNumber} - 1) * 10 + (\text{registerNumber} - 1) + 924$   
where "REG0" represents  $\text{registerNumber} = 10$

### Cartridge Registers

$\text{index} = (\text{cartridgeBankNumber} - 1) * 10 + (\text{registerNumber} - 1) + 974$   
where "REG0" represents  $\text{registerNumber} = 10$



---

## **Solving Problems**

This section 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.

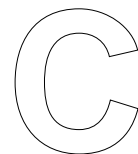
## LARC Diagnostic Programs

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	Inactive
VOLTAGE	Displays LARC power supply voltage - should be stable between 4.8 and 5.2 (048-052). Low voltage could indicate excessive cable power drop and need for remote power pack
MAINFRAME	Returns to normal operation



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# Specifications

This section 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

\*These specifications are for 48 kHz sampling rate setting.

#### IM Distortion Channel Separation

<0.05% SMPTE IM @ limit level  
>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

### 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 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 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)

LARC 5.9"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)



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# **Voltage Changeover and Optional Transformers**

This section 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.

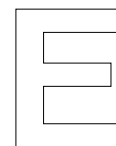
For the 480L outputs, we recommend Jensen JE-123-SLPC transformers.

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.



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# Control Mode Reference

This section contains descriptions of each parameter in the Control Mode.

## 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 Two

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
- DUMP

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

DUMP allows MIDI transmission of Automation data.

To perform any of these functions, hold down STO and press REG.

#### 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.

### **DEF (Default)**

DEF selects the power-up default mode. Advancing the slider selects a program to load automatically when power is interrupted, then restored. Press MACH to select a default program for the other machine. (When shipped, DEF is set to "Off.")

### **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. To enter the name change, hold down STO and press REG.

### **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.

### **AUTO**

Sets the two states of SysEx Automation: Off or On (Transmit). **Do not leave this slider set to TRANSMIT when performing SysEx bulk dumps or loads.**

### **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.

### **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 determines 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)

### **Page Six**

Page Six contains MIDI SysEx controls.

### **SFN (SysEx Function)**

Determines the type of bulk data dump. There are seven choices:

- Setup
- Bank
- Active
- All Internal
- All Cartridge
- MIDI Map
- Control Page Info

### **SRC (Source)**

Selects the SysEx source for the dump. The available selections are determined by the setting of Slider 1 (SFN).

### **RCV (Receiver)**

Determines the destination of all SysEx loads.

### **480L Power-Up State**

Normally if power is lost, or turned off, and then restored, the 480L will return to its last previous state. You can also configure the 480L to automatically load pre-selected programs upon power up. To do this, go to Control Mode, Page two, slider 5. This slider, labeled DEF for default, is set at the factory to "off". Advancing the slider will scroll through all of the 480L programs and select one as the new default state. The upper display will indicate the machine you are addressing, followed by the Bank number, Program number and Program name. In order to select a default load for the second machine, press the MACH button and advance the slider to the desired program.

**Control configuration is always remembered regardless of power-up default setting.**