

# ALESIS QuadraVerb2

## Reference Manual

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## Introduction

Thank you for purchasing the Alesis QuadraVerb 2 simultaneous effects processor. To take full advantage of the QuadraVerb 2's functions, and to enjoy long and trouble free use, please read this user's manual carefully.

## How To Use This Manual

This manual is divided into the following sections describing the various modes of the QuadraVerb 2. Though we recommend you take time to read through the entire manual once carefully, those having general knowledge about effects devices should use the table of contents and index to reference specific functions while using this device. If you are planning to use the Alesis optical digital I/O, read chapter 7 carefully.

**Chapter 1: Setting Up.** Deals with the necessary preparation before using, including connections to other components, such as instruments, mixing consoles, patchbays, multitrack recorders, as well as digital connections to ADAT.

**Chapter 2: Your First Session with the QuadraVerb 2.** A basic introduction to getting the unit up and running, auditioning the factory Programs, adjusting levels, comparing and storing edited Programs.

**Chapter 3: Overview.** A detailed look at the signal processing capabilities of the QuadraVerb 2, and the concepts of multi-effect programming.

**Chapter 4: Making Your Own Patches.** A guided tour for programming typical single and multi-effect applications.

**Chapter 5: Description of Controls.** A "dictionary" of all parameters, buttons and connectors, including Global mode parameters.

**Chapter 6: Advanced Applications.** Advanced uses of the QuadraVerb 2, such as MIDI functions, footswitches and using "tap tempo" to control delay times.

**Chapter 7: Alesis Optical Interface.** How the QuadraVerb 2 fits into an ADAT-compatible studio.

**Chapter 8: Trouble-Shooting.** Contains the Trouble-shooting Index, maintenance and service information, and MIDI implementation chart.

## Conventions

The buttons, knobs, and rear panel connectors are referred to in this manual just as their names appear on the QuadraVerb 2, using all capital letters and in brackets (*Examples:* [PROGRAM] button, [< BLOCK] button, [VALUE/ENTER] knob/button, [DIGITAL IN] connector, etc.).

**J**

*When something important appears in the manual, an icon (like the one on the left) will appear in the left margin. This symbol indicates that this information is vital when operating the QuadraVerb 2.*

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## CHAPTER 1

# SETTING UP

## Unpacking and Inspection

Your QuadraVerb 2 was packed carefully at the factory, and the shipping carton was designed to protect the unit during shipping. Please retain this container in the highly unlikely event that you need to return the QuadraVerb 2 for servicing.

The shipping carton should contain the following items:

- This instruction manual
- Alesis QuadraVerb 2 with the same serial number as shown on shipping carton
- AC Power Supply Adapter
- Alesis warranty card

**J** *It is important to register your purchase; if you have not already filled out your warranty card and mailed it back to Alesis, please take the time to do so now.*

## AC Power Hookup

The QuadraVerb 2 comes with a power adapter suitable for the voltage of the country it is shipped to (either 110 or 220V, 50 or 60 Hz).

With the QuadraVerb 2 off, plug the small end of the power adapter cord into QuadraVerb 2's [POWER] socket and the male (plug) end into a source of AC power. It's good practice to not turn the QuadraVerb 2 on until all other cables are hooked up.

**J** *Alesis cannot be responsible for problems caused by using the QuadraVerb 2 or any associated equipment with improper AC wiring.*

## Line Conditioners and Protectors

Although the QuadraVerb 2 is designed to tolerate typical voltage variations, in today's world the voltage coming from the AC line may contain spikes or transients that can possibly stress your gear and, over time, cause a failure. There are three main ways to protect against this, listed in ascending order of cost and complexity:

- Line spike/surge protectors. Relatively inexpensive, these are designed to protect against strong surges and spikes, acting somewhat like fuses in that they need to be replaced if they've been hit by an extremely strong spike.
- Line filters. These generally combine spike/surge protection with filters that remove some line noise (dimmer hash, transients from other appliances, etc.).
- Uninterruptible power supply (UPS). This is the most sophisticated option. A UPS provides power even if the AC power line fails completely. Intended for computer applications, a UPS allows you to complete an orderly shutdown of a computer system in the event of a power outage, and the isolation it provides from the power line minimizes all forms of interference—spikes, noise, etc.

## Audio Connections

The connections between the QuadraVerb 2 and your studio are your music's lifeline, so use only high quality cables. These should be low-capacitance shielded cables with a stranded (not solid) internal conductor and a low-resistance shield. Although



quality cables cost more, they do make a difference. Route cables to the QuadraVerb 2 correctly by observing the following precautions:

- Do not bundle audio cables with AC power cords.
- Avoid running audio cables near sources of electromagnetic interference such as transformers, monitors, computers, etc.
- Do not place cables where they can be stepped on. Stepping on a cable may not cause immediate damage, but it can compress the insulation between the center conductor and shield (degrading performance) or reduce the cable's reliability.
- Avoid twisting the cable or having it make sharp, right angle turns.
- Never unplug a cable by pulling on the wire itself. Always unplug by firmly grasping the body of the plug and pulling directly outward.
- Although Alesis does not endorse any specific product, chemicals such as Tweek and Cramolin, when applied to electrical connectors, are claimed to improve the electrical contact between connectors.

## Typical Applications

The analog audio inputs and outputs are typically used in one of three ways:

- from one or two effect/aux send outputs of a mixer, and out to the effect return inputs of the mixer; or,
- from a line-level instrument (like a guitar or keyboard with either a mono or stereo output), and out to an amplifier or mixer input; or,
- from the stereo buss outputs of a mixer to a mix-down tape machine or amplifier.

When used with a mono source, the QuadraVerb is placed between the source and the mixer/amplifier. Although the source may be mono, both the [LEFT] and [RIGHT] outputs can be connected to the inputs of a mixer/amplifier if stereo processing effects are desired. Alternatively, you could use the INSERTS on your mixer to “patch in” only the left or right channel of the QuadraVerb 2. If using the effect sends of a mixer, you have the advantage of sending any of the mixer's input channels to the QuadraVerb 2's input(s), and have control over the level of each channel being sent.

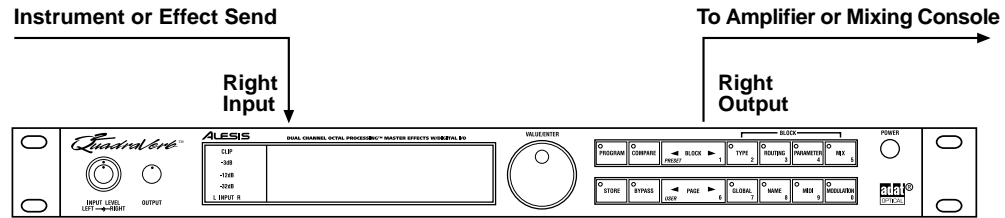
There are other combinations of input and outputs possible when you begin using the Alesis optical digital input and output. See the “Alesis Optical” section later in this chapter. For more information on interfacing with other digital audio equipment, see chapter 7.

## Interfacing Directly with Instruments

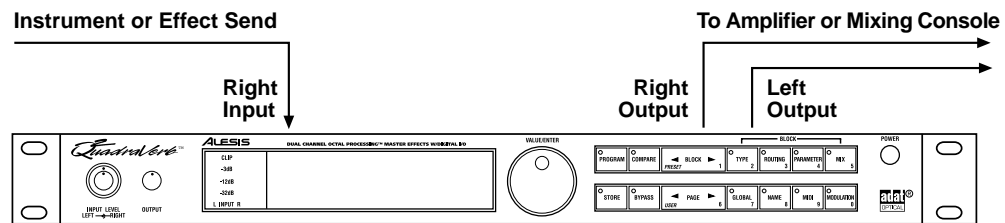
**J** *When connecting audio cables and/or turning power on and off, make sure that all devices in your system are turned off and the volume controls are turned down.*

The QuadraVerb 2 has two 1/4” balanced inputs and two 1/4” balanced outputs. These provide three different (analog) audio hookup options:

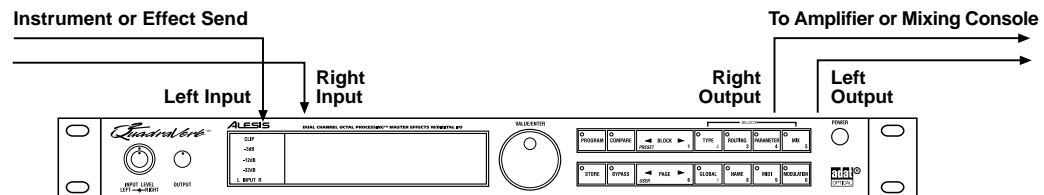
- **Mono.** Connect a cable to the [R] INPUT of the QuadraVerb 2 from a mono source, and another cable from the [R] OUTPUT of the QuadraVerb 2 to an amplification system or mixer input.



- **Mono In, Stereo Out.** While still using a mono input, you could connect two cables to the [L] and [R] OUTPUTS of the QuadraVerb 2 to a stereo amplification system or two mixer inputs.



- **Dual Mono or Stereo Source.** The QuadraVerb 2 may be used with two different instruments simultaneously, or with a stereo instrument. The hookup is the same; the difference is in the routing used within a program. A program may process the two inputs discretely, using blocks dedicated to a single channel (for example, a delay for a guitar and a gated reverb for a bass), or process them in stereo (for example, with the left and right outputs of a keyboard routed through two reverb blocks). Connect two cables to the [L] and [R] INPUTS of the QuadraVerb 2 from two mono sources or from the stereo output of the instrument, then connect two other cables from the [L] and [R] OUTPUTS of the QuadraVerb 2 to a stereo amplification system or two mixer inputs.



*Note:* In most cases when plugging an instrument directly into the QuadraVerb 2, you'll use Programs which route the "dry" signal at the input(s) directly to the output(s), where it will be mixed together with the effected signal to achieve the proper wet/dry mix at the QuadraVerb 2's outputs. If the program doesn't include this routing, you will only hear the effected signal by itself. Therefore, it may be necessary to edit such programs to add these "dry" routes when using the QuadraVerb 2 directly with an instrument. (The Factory Preset programs usually include these routes.)

## Interfacing to a Mixing Console

The QuadraVerb 2 handles mono or stereo sends at all system levels. The input circuitry of the QuadraVerb 2 can easily handle +4 dBu levels (+19 dBu peaks), while having enough input and output gain to interface with the lower -10 dBV signal levels of many recording systems.

The QuadraVerb 2 may be connected to a mixing console in several ways. Usually, it is connected to the auxiliary send and return controls of the mixer. Another method of interfacing is to connect the unit directly to the insert send and return patch points of the channel that is to be effected. Still another way of interfacing the QuadraVerb 2 to a mixer or recording console would be in-line across the output of your mixing console. This last setup would be used only if you needed to effect the entire mix.

### **Using the Aux Sends**

Generally, mixing consoles provide two types of auxiliary sends: pre-fader sends for creating a cue (headphone) mix, and individual, post-fader effect sends. Typically, if a mixer has more than two sends per channel (4, 6 or 8, perhaps), the first two sends are reserved for the cue sends, while the remaining sends are used to feed effects. If you are using a mixer with more than two sends, connect the QuadraVerb 2 using post-fader sends.

Using a mixer's aux sends poses a distinct advantage: each channel has its own level control feeding the aux output (and eventually the QuadraVerb 2 input). This allows you to make a mix of any channels you want to go to the effects by using the individual channels' aux send levels on the mixer. Most consoles also have aux master controls, which set the overall level of each aux output.

Coming back from the QuadraVerb 2's outputs into the mixer, you have two options:

- connecting to dedicated return inputs, or
- connecting to channel inputs.

The former is good if your mixer provides dedicated inputs (called returns) for effect devices like the QuadraVerb 2. If your mixer does not have these, or you have already used them all, consider connecting the QuadraVerb 2 to channel inputs or unused tape returns.

### **Setting the Effect/Dry Balance**

No matter where you connect the output of the QuadraVerb 2 into the mixer, you are in control of the balance between the mixer's channel inputs (the unaffected signal being routed to the aux sends and the Mix) and the effect returns coming from the QuadraVerb 2. The effect returns generally should only contain effected signal, and not have any unaffected signal mixed with it (since these two signals are blended together at the mixer).

If the Program you are using has the LR IN connected to the LR OUT, you may be getting some dry, unaffected signal at the return. Generally, this is not desirable, since the "dry" signal is already being heard through the original channel's fader. Therefore, in a mixer application you will want to cut the QuadraVerb 2's Program's path which connects the inputs to the outputs. This can be done in three ways:

- Go to the Mix parameters to bring down the direct level
- Go to the Routing function of each program and remove the patch cords connecting the inputs to the outputs
- Turn on the Global Direct Signal Mute function. This is the easiest method.

**J** *Most Preset Programs route the LR IN signal to the LR OUT. When connecting to a mixer's aux sends and returns, the Global Direct Signal Mute should be set to ON.*

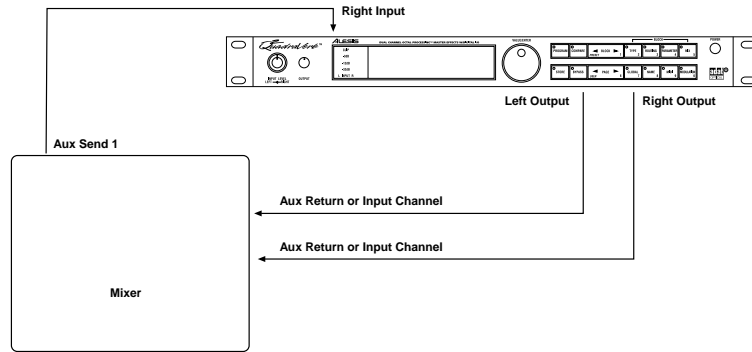
To remove all direct routings of inputs to outputs on all Programs simultaneously:

- ⌘ Press [GLOBAL].  
*The [GLOBAL] LED will be lit.*
- ⌘ Press [< PAGE] once.  
*This selects Global Page 9. The display will read:*

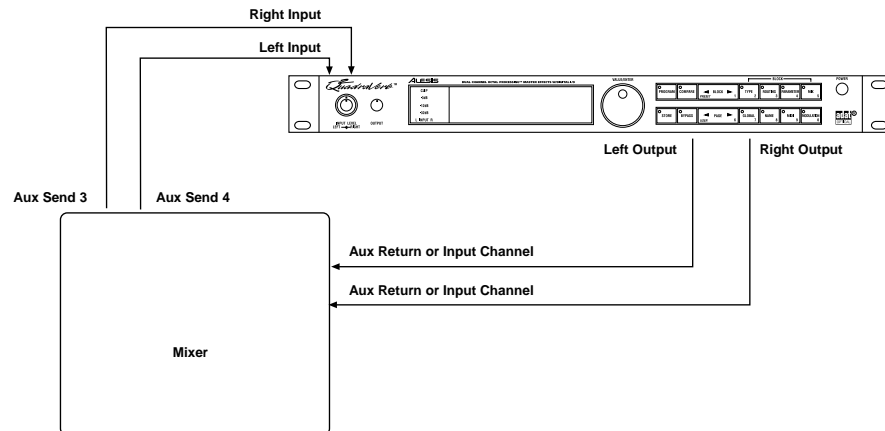
GLOBAL DIRECT SIGNAL MUTE:  
OFF

- ↩ Turn the [VALUE/ENTER] knob to the right until the display reads "ON."  
*The next Program recalled which has the inputs routed to the outputs will not display the patch cords for these connections nor will you hear any direct unaffected signal at the outputs.*

**Mono In - Stereo Out.** If you only want to feed the QuadraVerb 2 a mono input, but wish to connect both of its outputs back to the mixer, you will need three 1/4" audio cables. Connect a cable from an effect send to the [R] INPUT of the QuadraVerb 2, another cable from the [L] OUTPUT of the QuadraVerb 2 to an effect return or other mixer input, and another cable from the [R] OUTPUT of the QuadraVerb 2 to an adjacent mixer input.



**Stereo In - Stereo Out.** This connection is similar to the one described above. However, by utilizing two sends from the mixer, we add one more cord and can now send a stereo signal to the QuadraVerb 2's inputs. Example, if you connected effect sends 3 and 4 to the [L] and [R] INPUTS, and had a stereo instrument (such as a keyboard) connected to two channel inputs of the mixer (either one panned hard left and hard right), you would send the left channel to send 3 and the right channel to send 4. Alternatively, you could have two discrete effect sends between the Left and Right channel, and process each separately within the QuadraVerb 2. For example, the Left channel (from send 3) could be a chorus, and the Right (from send 4) could be a reverb. This is similar to Dual Mono, described earlier.

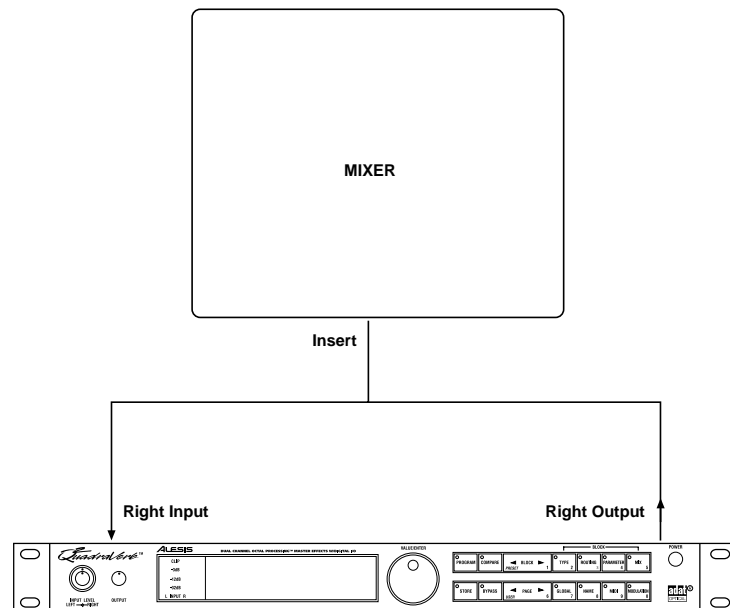


### Using Inserts

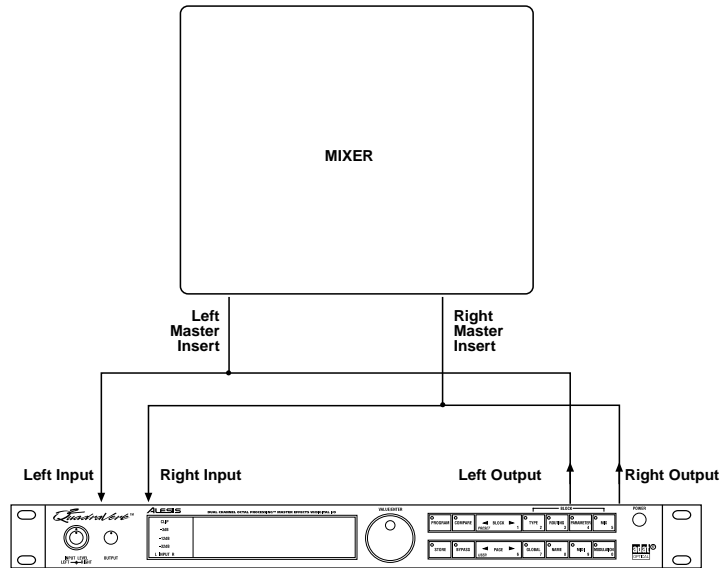
By using individual channel inserts, you can dedicate the QuadraVerb 2 to a specific channel (or pair of channels) on the mixer. The Insert connections on the back of the mixer provide a way of "inserting" external processing equipment into the signal path. The insert occurs after the input amplifier, and before the main fader; essentially it is the same as connecting the source (instrument or microphone) into the QuadraVerb 2 before the mixer's channel input. However, some mixing console's inserts come after the EQ section, and may therefore be different from the original signal.

Usually, insert connections require a special, stereo-splitting Y-cord to be connected (one stereo plug provides both send and return while two mono plugs connect separately to an input and output). These are known as TRS connectors (tip-ring-sleeve). The tip of the stereo plug typically carries the send or output of the insert jack, while the ring carries back the return. The sleeve represents a common ground for both signals. Check the manual of your mixer because some are wired differently (for example, having two separate jacks for send and receive).

**Mono.** This involves connecting a 1/4" TRS (tip-ring-sleeve) to the Insert jack of a single channel on a mixing console. The other end of the cable (which splits into two, 1/4" mono connectors) are connected to the [R] INPUT and [R] OUTPUT, respectively. If you do not hear any audio after making these connections, swap the input and output cables at the QuadraVerb 2, as these may be wired backwards. If the cable is color-coded, usually the red jack represents the send (which connects to the QuadraVerb 2's INPUT) and black is the return (which connects to the OUTPUT).

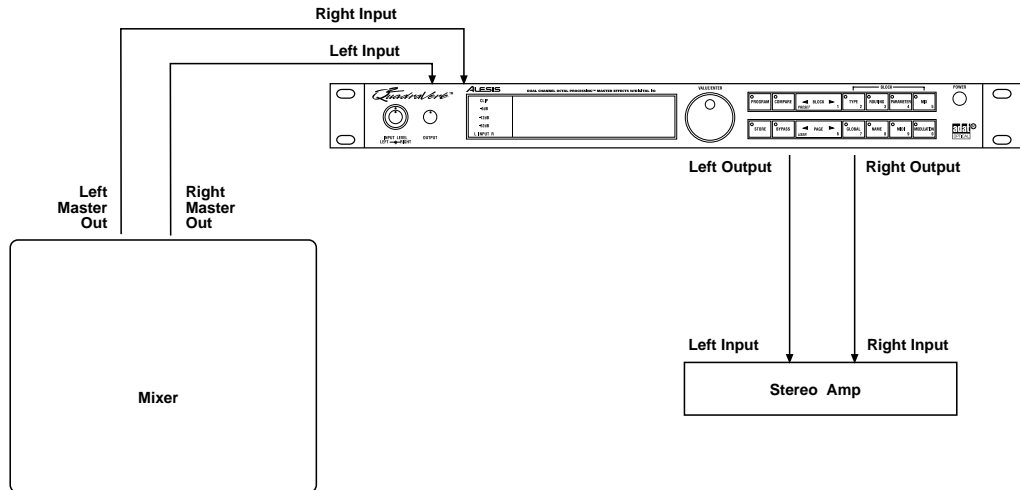


**Stereo.** In the case where a stereo instrument, such as a keyboard or sampler, is connected to two separate channels of a mixing console, you will need two 1/4" TRS cables, one for each channel. The connection is made in a similar fashion as described above.

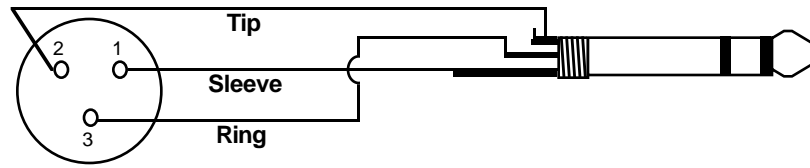


### Using Main Outputs

When you want to effect everything on the mixer, you can connect the QuadraVerb 2 between the mixer's outputs and the amplifier's or tape machine's inputs. This is done by using two 1/4" cables to connect the Left and Right Main Outputs of the mixing console to the [L] and [R] INPUTs of the QuadraVerb 2. The [L] and [R] OUTPUTs of the QuadraVerb 2 are then connected to a stereo amplifier, or two input channels of another mixing console (for sub-mixing applications).



If your mixer's outputs use XLR balanced connectors, use the illustration below as a guide for wiring cables to connect to the QuadraVerb 2's inputs.



## Avoiding Ground Loops

In today's studio, where it seems every piece of equipment has its own computer chip inside, there are many opportunities for ground loop problems to occur. These show up as hums, buzzes or sometimes radio reception, and can occur if a piece of equipment "sees" two or more different paths to ground. While there are methods to virtually eliminate ground loops and stray radio frequency interference, most of the professional methods are expensive and involve installing a separate power source just for the sound system. Here are some easy helpful hints that a professional studio installer might use to keep those stray hums and buzzes to a minimum.

- ⌘ **KEEP ALL ELECTRONICS OF THE SOUND SYSTEM ON THE SAME AC ELECTRICAL CIRCUIT.** Most stray hums and buzzes happen as a result of different parts of the sound system being plugged into outlets of different AC circuits. If any noise generating devices such as air conditioners, refrigerators, neon lights, etc., are already plugged into one of these circuits, you then have a perfect condition for stray buzzes. Since most electronic devices of a sound system don't require a lot of current (except power amplifiers), it's usually safe to run a multi-outlet box (or two) from a *SINGLE* wall outlet, and plug in all of the components of your system there.
- ⌘ **KEEP AUDIO WIRING AS FAR AWAY FROM AC WIRING AS POSSIBLE.** Many hums come from audio cabling being too near AC wiring. If a hum occurs, try moving the audio wiring around to see if the hum ceases or diminishes. If it's not possible to separate the audio and AC wiring in some instances, make sure that the audio wires don't run parallel to any AC wire (they should only cross at right angles, if possible).
- ⌘ **TO ELIMINATE HUM IF THE ABOVE HAS FAILED:**
  - A) Disconnect the power from all outboard devices and tape machines except for the mixer and control room monitor power amp.
  - B) Plug in each tape machine and outboard effects device one at a time. If possible, flip the polarity of the plug of each device (turn it around in the socket) until the quietest position is found.
  - C) Make sure that all of the audio cables are in good working order. Cables with a detached ground wire will cause a very loud hum!!
  - D) Keep all cables as short as possible, especially in unbalanced circuits.

If the basic experiments don't uncover the source of the problem, consult your dealer or technician trained in proper studio grounding techniques. In some cases, a "star grounding" scheme must be used, with the mixer at the center of the star providing the shield ground on telescoping shields, which do NOT connect to the chassis ground of other equipment in the system.

## MIDI

MIDI is an internationally-accepted protocol that allows music-related data to be conveyed from one device to another. The MIDI connections on the QuadraVerb 2 provide four different functions:



- To recall programs using MIDI program change messages
- To control various parameters inside the QuadraVerb 2 in realtime via MIDI controllers (*example*: A keyboard's mod wheel, or pedals, etc.)
- To send and receive SysEx (System Exclusive) dumps of individual programs or the entire bank of programs for storage and retrieval purposes
- To pass-on MIDI information thru the QuadraVerb 2 to another MIDI device.

To connect the QuadraVerb 2's MIDI ports to another MIDI device:

- ⌋ Connect a MIDI cable from the QuadraVerb 2's MIDI [THRU/OUT] connector to the MIDI IN connector of the other MIDI device.
- ⌋ Connect another MIDI cable from the QuadraVerb 2's MIDI [IN] connector to the other MIDI device's MIDI OUT connector.

For more information about MIDI, refer to chapter 6.

## Alesis Optical

The Alesis Optical interface provides two EIAJ fiber optic connectors for [DIG IN] and [DIG OUT]. These connectors use a proprietary Alesis multichannel format first introduced with the ADAT Multitrack Recorder. The QuadraVerb 2 can send and/or receive digital audio directly to/from an ADAT (or other devices which use the same optical interface). Digital connections provide better fidelity than the analog inputs and outputs.

The proprietary Alesis Optical format carries up to 8 audio channels on a single fiber optic cable. Since the QuadraVerb 2 has two channels (left and right), you may choose two of the incoming 8 channels for the QuadraVerb 2 to process. The QuadraVerb 2's Left and Right Outputs can then be routed in the digital format using any two channels (they don't have to be the same as the input channels). Fiber optic cables of various lengths are available from your Alesis dealer. The shorter the cable, the better. The OC cable is 5 meters long (16'4") and is the maximum length recommended.

## Footswitches

On the rear panel you will find two footswitch jacks labeled [ADVANCE] and [BYPASS]. Any momentary single-pole/single-throw footswitch, normally open or normally closed, will work for the two footswitch functions. These should be plugged in prior to power-up so that the QuadraVerb 2 can configure itself for the type of footswitch being used.

### Advance

The [ADVANCE] jack lets you scroll through the Programs in memory by advancing to the next higher numbered Program each time the connected footswitch is pressed. The QuadraVerb 2 will "wrap-around" whenever it reaches the end of available Programs and the Advance footswitch is pressed again. You can set a range of Programs to be used, thereby cutting off other Programs from being recalled in this manner. For example, if you set the range to be User 10 through User 24, only Programs within this range will be recalled using the Advance footswitch. If Program 24 is selected and the footswitch is pressed again, Program 10 is recalled.

To adjust the Advance Footswitch's set of Programs:

- ⌘ Press [GLOBAL].  
*The [GLOBAL] LED will light.*
- ⌘ Press [PAGE >] once.  
*This selects page 2, and the display will read:*

FOOTSWITCH: 00 Pset TO 00 Pset

- ↪ Turn the [VALUE/ENTER] knob to adjust the Program number to begin the range (Preset 00 — 99, User 00 — 99).
- ÷ Press [PAGE >] once and use the [VALUE/ENTER] knob to adjust the Program number to end the range (Preset 00 — 99, User 00 — 99).

## Bypass

The Bypass footswitch jack lets you turn the Bypass function on and off from a connected footswitch. When pressed, the [BYPASS] LED will light, indicating that Bypass mode is enabled. When pressed again, the [BYPASS] LED will turn off. For more information about Bypass mode, see chapters 2 and 5.

## Tap Tempo

Either footswitch jack can be used to provide a tap tempo source for setting delay time, provided the selected Program uses one of the two available tap tempo delay types. This requires that you have defined an Effect Block as one of the two Tap Tempo delay types, and that the desired footswitch jack has been selected for controlling tap tempo. To select a footswitch jack for use with a Tap Tempo Delay:

- ⌘ Press [GLOBAL].  
*The [GLOBAL] LED will light.*
- ⌘ Press [< PAGE] twice to select Global Page 8.  
*The display will read:*

TAP TEMPO FOOTSWITCH: NONE

- ↪ Turn the [VALUE/ENTER] knob to select either the ADVANCE or BYPASS footswitch jack, depending on which one you wish to use to control tap tempo.

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CHAPTER 2

# YOUR FIRST SESSION WITH THE QUADRAVERB 2

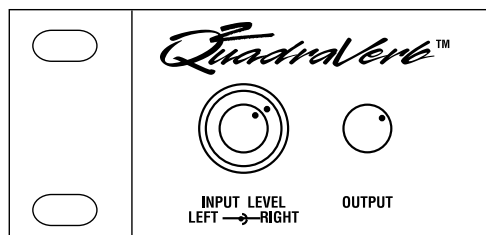
## Powering Up

After making your connections, turn on the system's power using this procedure:

- ⌚ Before turning on the QuadraVerb 2's power, check the following items:
  - Have all connections been made correctly?
  - Are the volume controls of the amplifier or mixer turned down?
- j Turn on the [POWER] switch on the front panel of the QuadraVerb 2.  
*Upon power-up, the QuadraVerb 2 will display the last selected Program, and the [PROGRAM] button's LED will be lit. If this Program has been edited, the display will indicate this by showing the word "EDITED", and by flashing the Program Number and Name in the upper display.*
- ➔ Turn on the power of the amplifier/mixer, and adjust the volume.

## Setting Levels

Proper setting of the [INPUT LEVEL] and [OUTPUT] knobs is crucial in order to achieve the maximum signal-to-noise ratio (the concentric knobs allow the Left and Right Input levels to be adjusted separately). As a good starting point, set both input and output level controls at about 2 o'clock or 65% of full. This will decrease the possibility of overload distortion and keep the amount of background noise to a minimum.



- J *For quietest operation, you should adjust the level of the source being sent to the QuadraVerb 2 so that the green [-3dB] LEDs in the QuadraVerb 2's peak meter flash, but not so loud that the red [CLIP] LEDs turn on. A nominal input sound make the -12dB LED turn on*

## The Value/Enter Knob

Located just to the right of the custom LCD display, the [VALUE/ENTER] knob is used to select Programs and adjust parameter values that appear in the display. However, it is not just a knob, it is also a button. Depending on what parameter you are editing, the [VALUE/ENTER] knob will work in either one of two ways:

**Immediate.** The desired value is selected by turning the [VALUE/ENTER] knob, and immediately takes effect. This is the case when adjusting most parameters.

**Deferred.** The desired value is selected by turning the [VALUE/ENTER] knob, but the new value will only take effect after the [VALUE/ENTER] button has been pressed. The newly selected value will flash in the display until it is selected in this manner. If you change the parameter back to its original setting, the value in the display will not flash. Also, if you go to another Page, or select another Function (by pressing any button), the parameter will be left unchanged. If you went back to look at the previous parameter, it will be set back to its original setting. This mode is used for parameters that cause architectural changes such as changing a Block's function, effect type, and routing signals.

The [VALUE/ENTER] button can also be used to step through Pages in the currently selected mode. Except when a value is flashing on and off in the display, the [VALUE/ENTER] button ordinarily doubles for the [PAGE >] button. If you change the value of a parameter that uses "deferred mode," you must press the [VALUE/ENTER] button to enter the new value (the display will stop flashing), and then you can press it again to move to the next Page (or to the next parameter, if more than one parameter appears in the display). This is a feature for power users who want to be able to move around the various pages quickly and make changes as fast as possible.

**J** *Unique Exception: When editing the Delay Time parameter of a Delay Block set to either Tap Tempo Mono Delay or Tap Tempo Ping Pong, the [VALUE/ENTER] button is used to "tap" in a tempo. See chapter 6 for more information.*

## Adjusting the Display Contrast

Occasionally, the characters in the LCD display may be difficult to read, depending on the viewing angle and existing lighting conditions. In such a situation, adjust the contrast of the LCD display using the following procedure.

- ¿ Press [GLOBAL].  
*The display will go to the Global Mode Page 1.*

ADJUST DISPLAY CONTRAST: 5

- i Adjust the contrast by turning the [VALUE/ENTER] knob.  
*The display's contrast and its value in the display will change.*

## Auditioning Internal Programs

You can audition the Programs in the QuadraVerb 2 by using the [VALUE/ENTER] knob or the front panel buttons, whenever the QuadraVerb 2 is in Program mode (the [PROGRAM] button's LED will be lit).

To select a Program using the [VALUE/ENTER] knob:

- ⌘ Press [PROGRAM].  
*The [PROGRAM] button's LED will light.*
- ⌘ Turn the [VALUE/ENTER] knob.

*Note:* The [VALUE/ENTER] knob has two modes when used for selecting Programs: Direct and Deferred. Direct mode immediately recalls the displayed Program as you turn the [VALUE/ENTER] knob. Deferred mode lets you scroll through the Programs in the display by turning the [VALUE/ENTER] knob, but you must press the [VALUE/ENTER] button to actually recall a Program. For more information on choosing between Direct and Deferred mode, see chapter 5.

To select a Program using the front panel buttons:

- ⌘ Hold the [PROGRAM] button.
- ⌘ Use the [1] through [0] buttons to directly select Programs 00 through 99.  
*These are the right-most ten buttons on the front panel, which double for [BLOCK >], [TYPE], [ROUTING], etc.*

The top line of the display will change to indicate the currently selected Program number (from 00-99) and its name.

PROGRAM: 00 "More Divisions"

The left side of the display always indicates the currently selected Program number (00 – 99). Directly beneath the two-digit Program number, the word "PRESET" will appear when the Preset bank is selected.



## Switching Between Preset and User Banks

There are two banks in the QuadraVerb 2: Preset and User. They both contain 100 Programs. However, the Preset bank cannot be permanently changed. You can edit the Preset Programs, but you can store them only in the User bank. To switch between the Preset and User bank, follow these steps:

- ⌘ Press and hold the [PROGRAM] button.
- ⌘ Use the [< BLOCK] and [< PAGE] buttons to select either PRESET or USER.  
*When the Preset bank is selected, the display will show the word "PRESET" beneath the PROGRAM Number in the lower left corner. If the User bank is selected, the word "PRESET" will not be visible.*

## Example Programs

The following are descriptions of the three example Programs in the Preset bank.

### 96: “VerbOfMyDreams”

This Program is fairly simple in design, using only 3 Blocks to provide EQ, Delay and Reverb. It is designed for a mono audio source to be connected to the [R] INPUT. In the display, you can see the R IN routed to the first Block, which is defined as a 3-Band Parametric EQ. The M output of the EQ then is routed to the next Block, which is a Mono Delay. The M output of the Delay is then routed to the next Block, which is a Room 2 Reverb. The L output of the Reverb is routed to the L OUT while the R output of the Reverb is routed to the R OUT. Also, the R IN is routed to both the L and R OUT, to combine the original dry signal with the wet signal coming from Block #3.

The EQ Block is used to tailor the sound before further processing occurs. You should make adjustments here based on the signal you feed through it; if the higher frequencies of the input signal do not sound as good through the Reverb as the mids and lows, try attenuating them in the EQ. The Delay is used to “thicken” the sound by adding a few very fast echoes. The Reverb adds ambiance, as if you were in a large chamber, and completes the total effect.

Using the Mix function, you can adjust the output level of the Reverb Block and the Direct Signal to create the balance you are looking for.

### 97: “Guitar Rack”

This Program includes two Pitch Blocks feeding a delay Block, into another Pitch Block, then into a Reverb Block, and finally yet another Pitch Block. This is a prime example of the flexibility the QuadraVerb 2 offers to programmers. The Program creates a thick, swirling, ambient effect that greatly enhances an electric guitar’s sound.

The R IN is routed to Blocks 1 and 2, as well as the L and R OUT. The first two Pitch Blocks are defined as Pitch Detune type, but are set to different detune amounts to create a thicker sound. The Delay Block is a Mono Delay which provides a very quick “slap back” by using a small delay time and no feedback. The third Pitch Block provides a Stereo Chorus which swirls the detuned, delay signal and feeds directly to the outputs. The Delay Block’s output is also fed into a Hall 1 Reverb which provides ambiance to the un-chorused signal. The Reverb’s stereo signals are routed to the QuadraVerb 2’s outputs and mixed with the Stereo Chorus’s signals. Meanwhile, the Reverb’s mono output feeds a Stereo Flanger whose stereo signals are also combined with the Stereo Chorus, Reverb and the original input signal at the outputs. Again, use the Mix parameters to create just the balance you want.

### 98: “Stereo Plates”

This Program is very simple. It routes the L and R IN through separate stereo type Reverb Blocks (Plate 1 type, to be precise). Both stereo signals coming out of the two Reverb Blocks feed the LR OUT. This creates a very clean stereo reverb effect that’s great for vocals. Play around with the Predelay and Decay parameters to adjust the attack and length of the reverb.

## Adjusting Effects Levels

Although we may not want to get started editing Programs just yet (that's left for chapter 4), it is usually necessary to have immediate control over the output levels of each Effect Block, as well as the amount of direct level going from the inputs to the outputs. These are found within Mix mode.

To adjust a Program's effect levels:

- z Press the [MIX] button.  
*The [MIX] button's LED will light.*
- j Use the [< BLOCK >] buttons to select any of the active Blocks in the display.  
*If the selected Block is routed to the LR Outputs, the display will read:*

LEVEL TO L/R:RVB=100%

*The letters RVB in the display example above indicate the selected Block is of the Reverberation type. Other types are: EQ for Equalization, PCH for Pitch and DLY for Delay.*

- ↵ Turn the [VALUE/ENTER] knob to adjust the Block's Level to the LR Outputs, from 0–100%.  
*If the selected Block is not routed to the LR Outputs, this setting will be "NONE" and you will not be able to adjust the value.*
- ÷ Press [PAGE >] to advance to Mix Page 2 and adjust the Direct Level from Input.  
*The display will read:*

DIRECT LEVEL FROM INPUT:  
100%

- f Turn the [VALUE/ENTER] knob to adjust how much of the dry (unaffected) signal you wish to hear at the QuadraVerb 2's outputs, from 0–100%.  
*If the LR Inputs are not routed to the LR Outputs, this setting will be "NONE" and you will not be able to adjust the value since there is no direct signal.*

**J** *If the Global Direct Signal Mute function is turned on, the Direct Level From Input parameter will have no audible effect, although its setting is remembered when you store the Program.*

- a Press [PAGE >] to advance to Mix Page 3 and adjust the Master Effects Level.

MASTER EFFECTS LEVEL: 100%

- D Use the [VALUE/ENTER] knob to adjust the Master Effects Level to the LR Outputs, from 0–100%.  
*This parameter comes between the combined Effect Blocks' outputs and the QuadraVerb 2's outputs, and controls the output levels of all active Blocks that are routed to the LR Outputs simultaneously. However, it does not change the individual output levels of Effect Blocks that are routed internally (i.e. to other Blocks).*



- « Use the [< BLOCK >] buttons to select other Blocks, and adjust their levels, as described above.

**J** *Mix parameters affect the Program and are only temporary unless the Program is stored into memory before a another Program is recalled. See later in this chapter for more about comparing and storing edited Programs.*

## Comparing an Edited Program to its Original Settings

The left side of the display always indicates the currently selected Program. Once a Program has been edited, the word “EDITED” appears in the lower left part of the display, just below the Program Number indicator. If the [PROGRAM] button is pressed (the [PROGRAM] button’s LED will light), the display will also indicate the Program has been altered from its stored version by flashing the Program number and name in the upper section of the display.



By pressing [COMPARE], you can temporarily access the original version of the Program you are editing; that is, the last Program saved to the currently selected location number. This allows you to hear the differences created by changing parameters in the Program.

While you are in Compare mode, the [COMPARE] button’s LED will be lit. Also, you cannot adjust any of the Type, Routing, Parameter, Modulation settings. However, you can move around the various pages and view these original settings and therefore see the actual differences in settings you have changed.

Pressing [COMPARE] a second time exits Compare mode; the [COMPARE] button’s LED will turn off and the edited version of the Program will be accessed. You can go in and out of Compare mode as often as you like, as long as the display indicates “EDITED” (i.e., if the Program hasn’t been edited yet, there is nothing to compare it to).

## Restoring an Edited Program to its Original Settings

If you decide to abort the changes you have made to a Program, this can be done in two easy steps.

To restore an edited Program to its previous, unedited version:

- ⌘ Press [PROGRAM] to select Program mode.  
*The [PROGRAM] button’s LED will light, and the display will flash the selected Program number and name.*
- ⌘ Press the [VALUE/ENTER] button.  
*This recalls the last stored version of the currently selected Program number, and the word “EDITED” disappears from the display. Consequently, any changes you had made to the Program before recalling are lost. unless you stored the edited Program into memory.*

## Storing Edited Programs

Once you are satisfied with the changes you have made to a Program, or are creating a new Program from scratch, you will need to store your edited Program back into memory. The QuadraVerb 2 will store the currently selected Program in memory (which is retained when the unit is turned off). If you edit a Program, the changes you made will still be there the next time you switch on the unit, even if you hadn't stored the edited Program into memory yet. However, if you select another Program from memory **before** storing the edited Program, your changes will be lost. Although the QuadraVerb 2 has two banks (Preset and User), you can only store Programs in the User bank.

**J** *The Preset bank cannot be permanently changed. If you edit a Program selected from the Preset bank, you will be able to make changes, but when you attempt to Store the edited Program, it will be stored into the User bank (in the selected number location).*

To store an edited Program into memory:

- i Press the [STORE] button.  
*The [STORE] button's LED will light. The display will read:*

STORE AT: XX "nnnnnnnnnnnnnnn"

*...whereby XX is a Program number from 0 to 99 and nnnn is the name of the program that will be overwritten. Both the Program number and name will be flashing.*

- i Turn the [VALUE/ENTER] knob to select the Program number location in the User bank to store the edited Program into.
- ↵ Press the [VALUE/ENTER] button.  
*The display will momentarily read:*

PROGRAM STORED

*After storing, the User Program location you chose will automatically be selected and shown in the display (example: if you edited Program 13 and stored it into location 25, Program 25 will be selected).*

## Bypassing Effects

At any time you can bypass all effects at once, thereby allowing the direct signal to pass through the QuadraVerb 2 unchanged. This can be done in two ways:

- by pressing the [BYPASS] button on the front panel; or,
- by connecting a footswitch to the [BYPASS] jack and pressing the footswitch.

Each time either the [BYPASS] button is pressed, or the footswitch connected to the [BYPASS] jack is pressed, Bypass mode is toggled on and off again. When Bypass mode is turned on, the [BYPASS] button's LED will be lit.

**J** *When Bypass is enabled, all Effect Blocks are momentarily disabled and will not have audio routed from them to the outputs. However, if the Program does not route the LR Inputs directly to the LR Outputs, you will not hear anything.*

In order for the Bypass mode to function correctly, make sure the LR Inputs are routed to the LR Outputs and that the Global Direct Signal Mute function is turned off (see below). If the inputs are not routed directly to the outputs, the [BYPASS] button acts more like a “mute” since nothing will be heard when it is enabled.

## Global Direct Signal Muting

The purpose of this feature is to satisfy the conditions of a recording studio environment. As described in chapter 1, when connecting the QuadraVerb 2 to a mixing console's aux sends and returns, it is generally desirable to remove the direct signal feed from the outputs of the QuadraVerb 2. This is because the signal coming back from the QuadraVerb 2 should only contain wet (unaffected) signal. The dry signal is then combined with the returning wet signal at the mixing console.

Since most QuadraVerb 2 Programs route the LR IN signal to the LR OUT, you will need to mute this connection when connecting to a mixer's aux sends and returns. This can be done globally for all Programs.

To remove all direct routings of inputs to outputs on all Programs simultaneously:

- 1 Press [GLOBAL].  
*The [GLOBAL] LED will light.*
- 2 Press [< PAGE] once.  
*This selects Global Page 9. The display will read:*

GLOBAL DIRECT SIGNAL MUTE: <u>OFF</u>
--

- 3 Turn the [VALUE/ENTER] knob to the right until the display reads “ON.”  
*The next Program recalled which has the inputs routed to the outputs will not display the routes (“patch cords”) for these connections nor will you hear any direct unaffected signal at the outputs.*

## CHAPTER 3

# OVERVIEW

## The Architecture of the QuadraVerb 2

The QuadraVerb 2 provides eight Effect Blocks per Program, each of which can serve as either EQ, Pitch, Delay or Reverb. The display shows these Blocks from left to right, between the LR IN (left and right inputs) and the LR OUT (outputs). The display also illustrates the routings between the Blocks. In addition to the Block functions, there is a set of Modulations which may be used in a Program. These allow MIDI messages (such as note numbers, velocity, after-touch, pitch-bend or controllers) to serve as controls over parameters in the QuadraVerb 2. This is discussed in further detail in chapter 6. For now, let's discuss the essence of Blocks and how they interact with each other.

### What is a Block?

A Block is essentially a discrete effects processor that can be used alone or in conjunction with other Blocks. Each Block has a Mono input and up to three outputs (depending on its type): left, right and mix. A Block requires that a signal be routed to it before it can effect the signal. The Block must also have its output connected either to the L OUT or R OUT (or both), or to the input of another Block (or to a series of Blocks) which is in turn connected to either the L OUT or R OUT (or both) before the Block's effect can be heard.

The four effect functions available in a Block are: Equalization, Pitch, Delay and Reverberation. Each one of these has several effect types. *Example:* Once a Block is assigned to use a Pitch function, you can choose a Chorus, Pitch Shifter, Flanger, etc. A full list of all the available effects and their parameters is shown in Chapter 5.

### Selecting and Editing Blocks

Editing a Program is done by first selecting one of the eight Blocks, and then selecting a Block function (Type, Routing, Parameter, or Mix). Use the [`< BLOCK >`] buttons to move the pointer in the display to the left or right. The pointer appears as a down-pointing triangle just above the Blocks, thereby selecting one of the eight Blocks or the L OUT or R OUT (the L OUT and R OUT are special Blocks which aren't really edited, except when adjusting the *mix*). The pointer's position indicates which Block is being edited.

There are four Block functions, which are accessed by using the four Block function buttons: [TYPE], [ROUTING], [PARAMETER], and [MIX]. Pressing any of these buttons takes you to its related parameters, and simultaneously turns on the button's LED to indicate the selected Block function.

**TYPE.** The Type function is where you go first to define a Block. A Block is defined as either EQ, Pitch, Delay, Reverberation or Off. This is done in Type Page 1. Once a Block is defined, its effect type can be selected. This is done in Type Page 2. If you need to move a defined Block to another position, you would go to Type Page 3.

**ROUTING.** Once a Block is defined, you must select its input source and route its outputs either to another Block's input or to the L OUT or R OUT. This is done in Routing Page 1. Once an input source is selected, you can adjust its level feeding to the Block's input using Routing Page 2. If you decide to delete a routing that has already been added, you will use Routing Page 3.

**PARAMETER.** This is where you adjust the selected Block's parameters. The number of pages available and the parameters found in them will be determined by the Block's type. Some effect types have only one or two parameters on a single page; while other effect types (particularly reverbs) use all nine pages, with as many as four parameters on a single page.

**MIX.** The Mix function provides three parameters on three separate pages. When a defined Block is selected, Mix Page 1 lets you adjust its level going to the LR Outputs. Page 2 lets you adjust the Direct Signal Level, and Page 3 lets you adjust the Master Effects Level. If, however, an undefined Block is selected, or the LR OUT is selected, there will only be two pages available in the Mix function (Direct Signal Level, and Master Effects Level).

Once a Block function is selected, use the [< PAGE >] buttons to scroll through the various pages within a Block function. The number of available pages will vary from function to function. The left side of the display will indicate the total number of pages within the selected Block function. The page currently being displayed will be underlined.

When you find a parameter you wish to edit, turn the [VALUE/ENTER] knob until the desired value is displayed. If the new value flashes in the display, it means you must press the [VALUE/ENTER] button to select the new value.

## Routing "Patch Cords" Between Blocks

The concept of routing involves selecting inputs for each active Block (1 through 8) and L OUT and R OUT, and then adjusting the input levels of those routings you are using. The [ROUTING] button will let you create "patch cords" that connect the input jacks to the Blocks, or the Blocks to other Blocks, and finally to the output jacks. You may select the Left Input, Right Input (analog or digital inputs), or the output of a Block: L (left), R (right) or M (mix). You can even route a Block's M (mix) output back to its own input, which can be used to create some rather stunning effects, but the level setting for doing this is critical to avoid unwanted feedback.

Keep in mind that each Block can have many inputs, coming from various sources. All routings are considered "input patch cords"; i.e., you can only make a route from the destination Block. *Example:* To set a route from Block 2 to Block 5, you'd select Block 5 (not Block 2) using the [< BLOCK >] buttons and then select Block 2's output as an input source.

*If the Global Direct Signal Mute function is turned on, any Programs which have the LR IN routed to the LR OUT will not display these "patch cords" except when editing the Program's routings. This is to avoid confusion when operating in this mode. For more information on Global Direct Signal Mute, see chapter 2.*

## J

## Setting the Routing Levels

Each time you add a “patch cord” to a Program, the QuadraVerb 2’s display automatically prompts you to set its level. This is done by adjusting the attenuation level between -48dB and +0dB, or OFF. When set at +0dB, the signal is allowed to pass through at full volume, with no attenuation. When set to OFF, the signal is fully attenuated and will not be heard at the “patch cord’s” destination.

The default level for a newly added “patch cord” is -6dB. This provides you with ample headroom for most applications. The ideal level, however, should be as high as possible without causing distortion. There are basically three places in the QuadraVerb 2 that can distort: at the input A/D converter, within the blocks, and at the output. The routing levels within a well-designed program are set so that none of these three will distort before the other ones do. If the input is clipping, it doesn’t matter if you’ve attenuated levels feeding from the input to a block--it will still distort.

Input clipping is controlled by the front panel INPUT LEVEL controls. Block clipping can be caused by routing levels to the block that are too high, or by processing within the block itself (for example, a boost in EQ, resonance in a chorus, or feedback in a delay). Output clipping can be controlled using the Master Effects Level parameter in the [MIX] function. The OUTPUT LEVEL control is analog, and cannot lower the distortion caused when the output of multiple blocks are too loud for the output D/A converter to handle.

Keep in mind when setting routing levels that each time you double the number of sources, the output rises 6 dB. If there are four blocks feeding another block, those routes may all need to be set to -12 dB to avoid clipping. On the other hand, if a program consists of a single delay or reverb block, input and output routings may be set to 0 dB without distortion.

It is also important to note that each “patch cord” in a Program uses 1% of the total DSP power available. However, if the signal is attenuated (by setting the level below +0dB), 2% of DSP is used. The following section discusses how the QuadraVerb reacts when DSP is nearly fully taxed. One way to free up more DSP power when you need it is to set some or all of your “patch cord” levels to +0dB.

### The LR IN

These appear on the far left side of the Block section in the display. They are not selectable with the [< BLOCK >] buttons. This is because we are always routing from the LR IN and, in keeping with our convention, all routing is done from the destination Block, not from the source.

### Reaching the Outputs - LR OUT

The L OUT and R OUT are separately selectable (like Blocks) for routing purposes. Whichever Block is determined to be the final effect in the chain (whether your Program has one or multiple Blocks), it needs to be routed to the LR OUT in order for you to hear it . This can be done in mono (if just one channel is needed) by routing one or more Block outputs to either the L OUT or R OUT, or in stereo by routing to both the L and R OUT.

## Limit Handling

Within the limits of the available sample memory and DSP, the Blocks can be linked in any order, in almost any combination of the four basic types. For instance, you might have two EQ Blocks followed by four pitch Effect Blocks; or one Block of each (EQ,

Pitch, Delay) type on the left channel and one of each type on the right channel, followed by a stereo reverb (a total of 7 Blocks).

Because the QuadraVerb 2 is so flexible, occasionally you will have a combination of effects and/or parameters that may overtax either the available memory or DSP power on board the unit. When you attempt to program beyond the QuadraVerb 2's capabilities, no harm will be done but one of four error messages will appear in the display indicating you have exceeded the unit's limits. This means you have to create your Program in a slightly different manner to obtain the final result you want. If you are into programming "computationally intensive" combinations of effects, you may have to learn to compromise in certain situations.

There are four categories to consider with regards to programming limitations: DSP power, Effect Memory, LFOs and Microprocessor Assistance.

**DSP.** The DSP (Digital Signal Processor) chip inside the QuadraVerb 2 is what processes all signals in the digital domain and creates all of the effects (sometimes with the assistance of the microprocessor). The DSP chip has a limit to how many instructions (per sample) it can handle. Some effects take up a large number of instructions, while others take only a few. Generally, reverbs are big DSP hogs, while most other effects use relatively small numbers of instructions. When you route "patch cords" between Blocks, each connection requires 1% of DSP power (2% if the level is not 0.0dB). If you run out of DSP power, the error message "DSP IS FULL" briefly appears. If this happens, consider substituting an effect that uses fewer instructions for one of the more DSP demanding effects you may be using.

**Effect Memory.** The on-board effect memory is primarily used for delay effects, but sometimes used by reverb effects (for pre-delay). In total you have 5.4559 seconds (or 5,455.9 ms), although the maximum delay time any single delay effect may have is 5 seconds. Once this memory has been allocated, the error message "EFFECT MEMORY IS FULL" will briefly appear when you try to apply more delay time to a delay effect than there is memory available. *Note:* If you have several delay Effect Blocks in a Program, they will share the total amount of effect memory that is available. If the total of all delay times equals that of available effect memory, you will not be able to increase any of the delay times until you decrease the delay time of one or more of the delay Blocks.

**LFOs.** The term LFO comes from the days of early synthesizers, where tremolo and vibrato effects were created using a Low Frequency Oscillator to modulate another oscillator's pitch, or the filter or amplifier volume. In effect devices, LFOs are used to create effects such as flanging, chorus and Leslie speaker emulation. The QuadraVerb 2 provides 4 LFOs per Program. Most pitch effects use 1 LFO, but some use 2. Also, some of the larger reverb effects use an LFO for a bigger sound. *Example:* If you try to add an effect using an LFO and you have already used all four LFOs in other Effect Blocks, the error message "OUT OF LFOs" briefly appears.

**Microprocessor Assistance.** Like most digital electronic music and audio products, the QuadraVerb 2 uses a microprocessor to run all basic functions like recalling, editing, comparing, storing, and renaming Programs, etc. However, some of the effect types use the microprocessor to assist the DSP chip in creating the effect, and the microprocessor is capable of assisting in this way for only one effect per Program. Three Pitch effects use the microprocessor: Phasor, Stereo Leslie and Ring Modulator. Only one of these three may be used per Program. *Example:* If you try to create a Ring Modulator effect type when either a Phasor or Stereo Leslie effect is already in use in the Program, the error message "BLOCK COMBINATION NOT ALLOWED" briefly appears.

In the back of this manual, you will find an index of effect types, listing the percentage of DSP, Effect memory and number of LFOs required by each. Use this as a reference when one or more categories of limit handling have been exceeded and you wish to learn what options exist in reorganizing the Program.



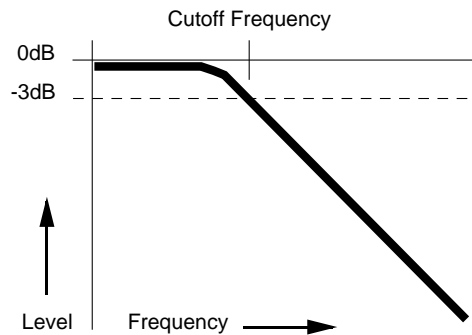
## Equalization

Equalization, or EQ, provides the ability to control the timbre, or coloration, of an audio signal by changing the frequency response of a QuadraVerb 2 audio path. Filters and EQ can be used to compensate for frequency deficiencies in microphones and other sound equipment. The EQ family of functions in the QuadraVerb 2 contains filters, equalizers, and several other highly useful studio effects.

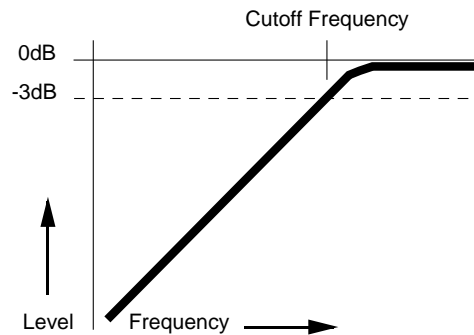
### Filters

#### Lowpass, Bandpass, and Highpass

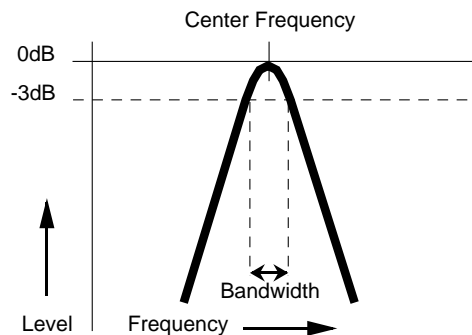
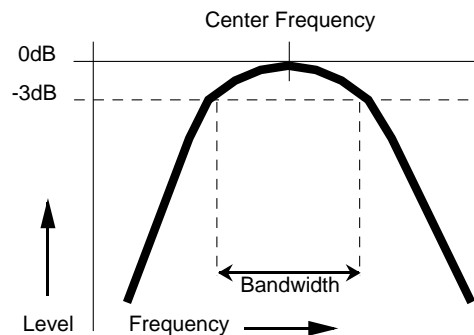
A filter allows the passing of frequencies within a certain range (called a filter's passband), and removes frequencies outside that range. The QuadraVerb 2 features a Lowpass Filter, which removes frequencies above its cutoff frequency (i.e., it allows low frequencies to pass through), and a Highpass Filter, which removes frequencies below its cutoff frequency. At the cutoff frequency, each filter attenuates the signal level by -3dB, and attenuates by 6dB per octave outside the passband. The Bandpass Filter passes frequencies near its center frequency, and removes all others. The Bandpass Filter has a bandwidth parameter to control how wide (in octaves) the passband is. Bandwidth is inversely related to the conventional "Q" value; that is, the higher the Q value, the lower the bandwidth. Bandwidth is used in the QuadraVerb 2 because it is a more familiar term to many musicians, and is more intuitive.



Lowpass Filter



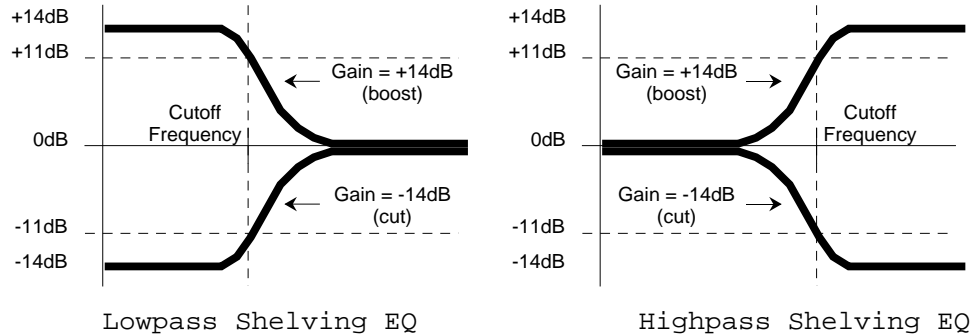
Highpass Filter

Bandpass Filter  
(low bandwidth)Bandpass Filter  
(high bandwidth)

### Shelving EQs

## Lowpass Shelf and Highpass Shelf

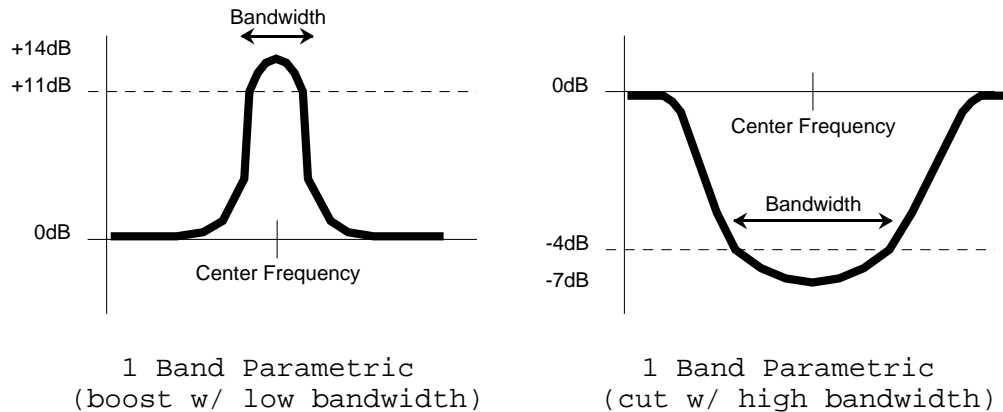
A shelving EQ can boost or cut frequencies in its pass band, while allowing the frequencies outside its pass band to go through it. The shelving EQs of the QuadraVerb 2 provide gains from -14dB to +14dB in 0.1dB increments. Shelving filters are well suited for applications in which high or low frequencies need to be boosted or cut, but the remainder of the audio signal is to pass through unchanged. Applications for shelving EQs include removing high end hiss, and “tightening up” the sound of a bass instrument. The Highpass Shelving EQ is useful to compensate for a lackluster source that needs its high end brightened. The Lowpass Shelving EQ can be set to cut low frequencies to eliminate PA rumble.



## Single Band EQs

### 1 Band Low Parametric and 1 Band High Parametric

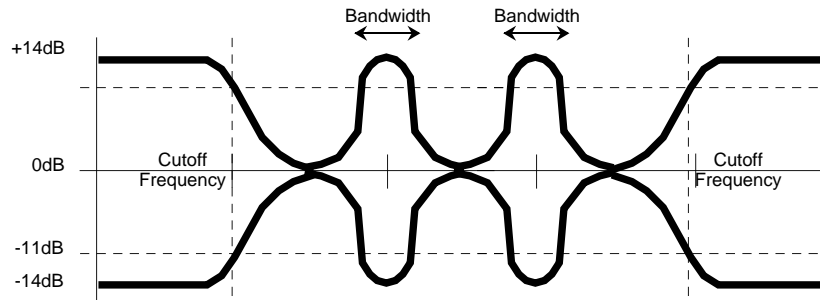
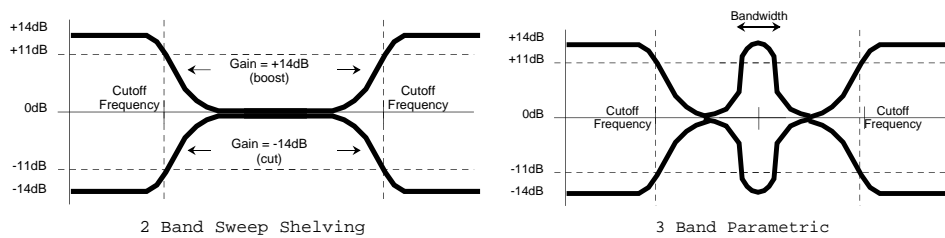
A single band parametric EQ allows frequencies within its passband to be amplified (boosted) or attenuated (cut) while allowing the remainder of the audio information to pass through unchanged. The QuadraVerb 2 provides a low frequency version (20Hz to 2000Hz), and a high frequency version (1500Hz to 10kHz). Single band parametric filters work well to highlight specific frequencies in a mix; for instance, bringing a tambourine to front and center, or boosting a repetitive bass line to achieve a realistic “inner-city throb”.



## Multiband EQs

### 2 Band Sweep Shelf EQ, 3 Band Parametric EQ, and 4 Band Parametric EQ

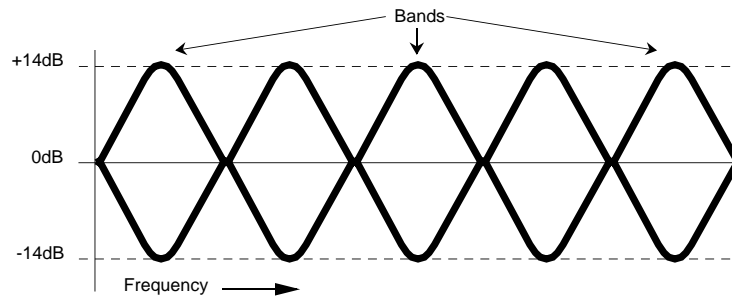
Multiband EQs can be used to flexibly and accurately tailor the tonality of a signal. The 3 band and 4 band parametric EQs in the QuadraVerb 2 can be used to boost and cut multiple bands, with the added flexibility of controlling the bandwidth of one or two mid bands. The multiband EQs provide plenty of equalization in a single Block effect. For instance, the 2 band sweep shelving EQ can be used as a loudness control by boosting both low and high frequencies (Lowpass cutoff frequency set to 150Hz, Highpass cutoff frequency set to 4kHz). The three band parametric can be applied to a mix to simultaneously eliminate hiss, boost bass, and highlight a cowbell. Although it is possible to overlap the different bands, you should be careful not to be redundant.



4 Band Parametric

### 5 Band Graphic EQ

A graphic EQ is a set of bandpass filters of fixed frequencies and of equal bandwidths, spaced evenly across the audible frequency range. The 5 Band Graphic EQ in the QuadraVerb 2 has bandwidths that are two octaves wide at 63Hz, 250Hz, 1kHz, 4kHz and 16kHz. Graphic EQs are used for general tonal shaping.



5 Band Graphic EQ

### Resonator

A resonator is a special filter whose response is so sharp that it actually imparts a pitch to any signal going through it. By tuning the resonator to your instrument and balancing their relative amplitudes you can accentuate different harmonics in a guitar tone, for instance. This can be very useful to dramatically change the character of an instrument's sound.

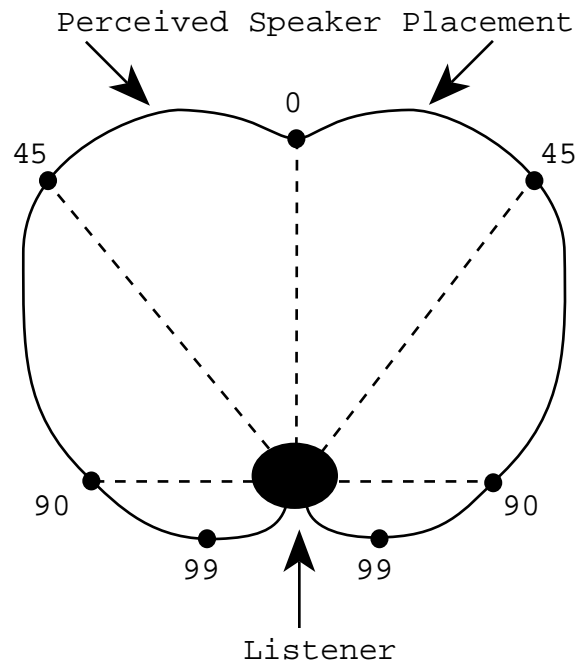
## Mono/Stereo Tremolo

The Mono Tremolo modulates the amplitude of its input signal according to three parameters: speed, which controls how often the sound changes amplitude; depth, controlling how strong the tremolos will be; and shape, which controls whether the amplitude transitions will be hard (drastic) or soft (gradual). Mono tremolo can recreate the sound of older guitar amps.

Stereo Tremolo alternates the loudness of the signal in opposite channels according to the same three parameters as the Mono Tremolo effect. On slow speeds with a soft shape, stereo tremolo provides continuous automated panning. At low to moderate depths and moderate speeds, the vibrating sound of traditional electric pianos can easily be reproduced.

## Stereo Simulator

This effect is useful for “widening” a flat sound to produce a wider stereo field. It takes a mono sound source and creates the illusion that sound is emanating in stereo from a pair of speakers placed somewhere around the listener. By doing so, the sound appears spatially distributed, and takes on a lively new ambiance. When placed at 0 degrees, there is virtually no effect. At 90 degrees, there is an illusion that the speakers are on either side of the listener. At 99 degrees, the sound appears to come from slightly behind the listener's head.

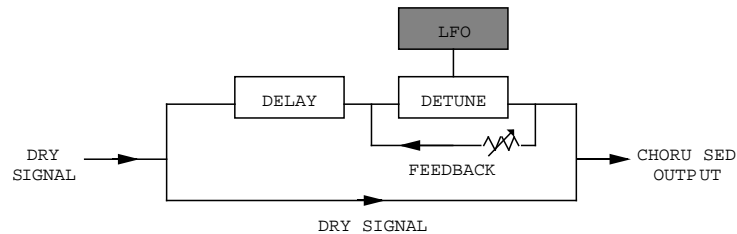


## Pitch Effects

The Pitch effects alter the pitch of a signal in various ways to produce “layered” timbres that are more complex than the original signal. Although some of these effects can sound similar to one another depending on the parameter settings, each is achieved differently and can be quite dramatic under the right circumstances. Pitch effects are achieved by splitting the signal into at least two parts, effecting the pitch of one of the parts, then mixing them back together. This eventual mixing is essential since the overall sound of the effect is achieved by the actual difference between the normal, unaffected signal and the effected signal. The various types of Pitch are:

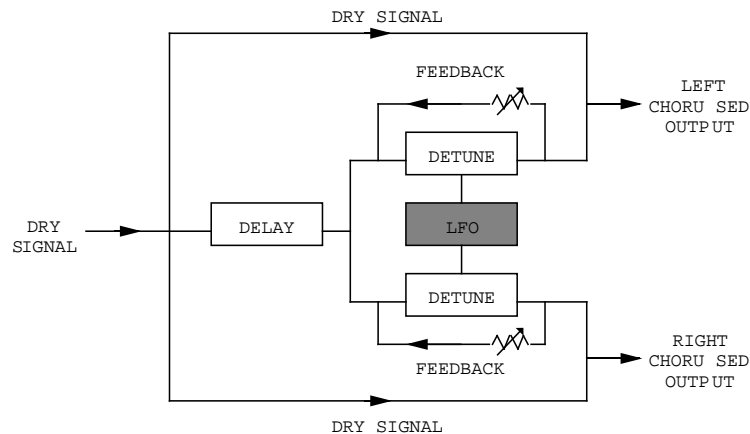
### Mono Chorus

The Chorus effect is achieved by taking part of the signal, slightly delaying it, and then slightly detuning it as well. The detuning is further effected by being modulated by an LFO which causes the detuning to vary. Many parameters are available in this scheme. The Predelay can be varied, the LFO depth can be varied, the LFO speed can be varied, and a portion of the detuned signal can be fed back to the input to increase the effect. Finally, the waveform shape of the LFO can be changed from a sine wave, to a more abrupt squarewave to make the pitch detuning more pronounced.



### Stereo Chorus

In the case of a Stereo Chorus, the signal is split into three parts with a dry signal and a separate Detuning section for both left and right channels. When the left channel is detuned sharp, the right is detuned flat, and vice versa. Once again, this causes the effect to become more pronounced and dramatic. All the parameters of the Mono Chorus are also found here, although each of the two detuned signals has its own individual Predelay amount.



### Quad Chorus

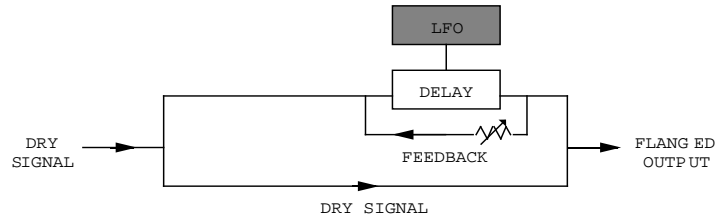
Quad Chorus modulates four delayed signals, each with its phase offset by 90°. Each of the four signals has a separate Predelay parameter, allowing you to change the “rhythm” of the phasing.

## Mono Flanging

Flanging is similar to chorusing, but modulates the delayed signal over a much shorter delay range (typically 0-12ms). This produces a “jet airplane”-like sound. Mono and stereo flanging is available, and the flange modulation sweep can be triggered in order to sync up with the rhythm of your playing.

First used in the 1960s, “Flanging” was achieved by the use of two tape recorders that would record and play back the same program in synchronization. By slowing down one tape machine, and then letting it catch up with the other, different phase cancellations would occur at different frequencies. Since the slowing down of the tape machines was done by hand pressure against the flanges of the tape supply reels, the term “Flanging” came into being.

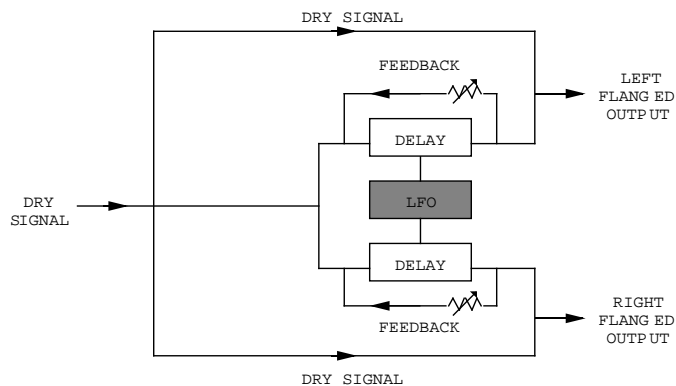
The effect of Flanging is achieved by splitting and slightly delaying one part of the signal, then varying the time delay with an LFO. The delayed signal is then mixed with the original sound to produce the “swishing” or “tunneling” sound.



Many parameters are available, from varying the speed and depth of the LFO to feeding back part of the signal to make the effect stronger.

## Stereo Flanging

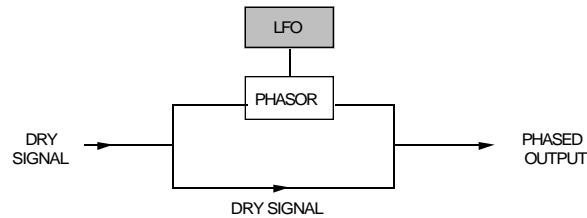
In the case of the Stereo Flange, the signal is split into three parts with a dry signal and a separate Delay section for both left and right channels with one channel flanging up while the other channel flanges down. Once again, this causes the effect to become more pronounced and dramatic.



## Phasor

Another popular effect is the Phasor or Phase Shifter. Although similar to flanging, this effect is produced differently. Again, part of the signal is split from the original

signal. The Phasor shifts the phase of different frequencies in different amounts, resulting in a comb filter effect when combined with the dry signal.



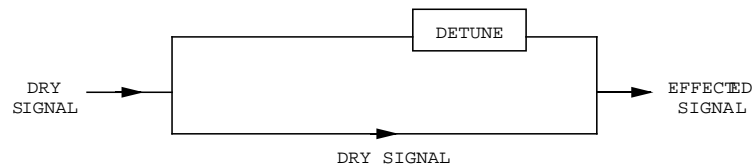
## Mono/Stereo Leslie Cabinet

With the Leslie type selected, the pitch change Block becomes a rotating speaker simulator. This effect was extremely popular during the 1960's and was achieved by mechanically rotating the speakers to produce complex timbral changes. The Leslie speaker system is most often used with Hammond-type organs, but is occasionally used for guitar amplification as well.

The stereo version gives you an additional parameter called Stereo Separation. This gives you control over how "wide" the stereo signal should appear.

## Pitch Shifter

This effect type shifts the pitch of the input signal by a fixed amount. This creates a thickening effect similar to the difference between a 12-string and 6-string guitar.



## Pitch Detune

This effects detunes the pitch of the input signal by + or - 99 cents. A cent is 1/100 of a semitone. A pitch shift of +99 cents applied to a C natural will make it sound as a C sharp. The input signal is not split, therefore if you want to create dissonance, you must route the input signal to the outputs and blend them together for the desired effect.

## Ring Modulator

This generates an outer-spacey, metallic, distorted version of the input signal by modulating it against an internally-generated digital oscillator. This shifts the frequency spectrum upwards or downwards depending on the setting of the Spectrum Shift parameter. For example, if Spectrum Shift is set to +100, the input and all its harmonics will be shifted up nonharmonically by 100 Hz.

# Delay

Delay provides a discrete repetition of a signal. By adding feedback within the effect, the delayed signal can repeat many times, with each successive decay softer than its predecessor. Each Delay type allows you to adjust delay time in milli-seconds, but also displays the equivalent tempo in BPM (beats per minute), with the exception of Multi Tap Delay. The QuadraVerb 2 offers four types of delay:

## Mono Delay and Stereo Delay

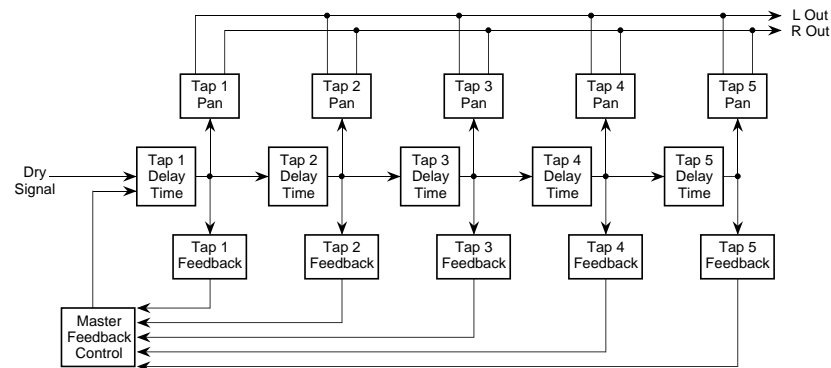
Mono Delay provides a single delay of signal up to 5000.0ms (this is subject to the limit of delay time being used by other Blocks). Feedback is also available to increase the complexity of the signal. The stereo version provides two separate delays which can be individually adjusted for delay time and Feedback.

## Ping Pong Delay

So called because the output bounces from left to right in stereo with the speed determined by the delay time.

## Multi Tap Delay

This is sort of like having five delays at once. Each of the 5 “taps” have individual volume, delay, panning and feedback controls. By adjusting the delay of each tap, you can create sophisticated rhythms. A Master Feedback control is available, making it easy to adjust the total number of “echoes” for all taps at once.



## Tap Tempo Mono Delay and Ping Pong

These are similar to the mono delay and ping pong delay types, but can have their delay time parameters set using a technique called “tap tempo” (where you literally tap in the desired tempo to establish delay time), using either the [VALUE/ENTER] button or a footswitch. You can select which footswitch jack (either [ADVANCE] or [BYPASS]) you want to use in Global Page 8.

An additional parameter, called Beat Count, is used to determine what note value your tempo tapping represents. For example, if you set the Beat Count to 1/4, then you can tap in quarter-notes to establish delay time (and tempo). If instead you set the Beat Count to 1/8T, you can play in eighth-note triplets to establish the delay time.



## Reverberation

Reverb is made up of a large number of distinct echoes, called reflections. In a natural acoustic space, each reflection's amplitude and brightness decays over time. This decaying action is influenced by the room size, the location of the sound source in the room, the hardness of the walls, and other factors. The QuadraVerb 2 offers many types of reverberation:

### Mono Room

This uses a very small amount of DSP and is useful for combining with other effects, or for dual-mono reverb applications.

### Room 1

This algorithm gives you the sound of a medium sized studio room. It doesn't use a lot of DSP, so it's useful for placing at the end of a guitar multi effect program or for using 2 reverbs at once for dual processing. Try it on percussion and brass, too.

### Hall 1

This is a simulation of a large concert hall. Halls tend to be large rooms with lots of reflective surfaces, so sounds can swim around, changing timbre over time. It doesn't need a lot of DSP, so you can use it on a guitar multi effect or a dual reverb send program.

### Plate 1

This is a simulation of a classic echo plate, a 4' by 8' suspended sheet of metal with transducers at either end used to produce reverb. Popular in the 1970's, it is still prized for its transparent sound, particularly on vocals and guitars.

### Chamber 1

Reverb chambers are small rooms made of the hardest, smoothest surfaces available (often marble). A loudspeaker is placed at one end, and 1 or 2 microphones at the other end. The decay time is adjusted by the distance from the microphones to the speaker and by sound diffusers placed in the room. This algorithm works well for keyboards and acoustic instruments.

### Room 2

This algorithm uses more DSP for a richer sound and smoother decay. It compares quite favorably to high end studio reverbs for its rich sound. The attack is also more reflective. It sounds good on drums, organs, keyboards and guitars.

### Hall 2

More processing is dedicated for a more realistic, airy concert hall. This is a classic reverb which sounds good on just about anything; try it on vocals, drums, acoustic, electric, or orchestral instruments.

### Plate 2

A richer version of Plate 1 which uses more processing power for more density and a smoother decay.

## Chamber 2

This is similar to Chamber 1, but uses more DSP for a thicker sound. The high end in particular has a glassy decay which sounds excellent on synthesizers and pianos, but try it on vocals, too. This will sound more realistic when using short decay times.

## Large Plate

This algorithm uses the most processing available for a truly realistic reverb plate simulation. It works well for a lush lead vocal, piano, or guitar, especially when looking for a classic rock and roll sound.

## Large Room

This algorithm uses the most processing for the most realistic possible sound. It has a nice smooth decay for drums and a fat attack to warm up guitars and percussion. Try gating it for that classic British progressive rock drum sound.

## Spring

Classic guitar amplifiers used a suspended spring with transducers at either end to simulate reverb. It produces a ringy, resonant sound, which can be good for vintage guitar applications. With this algorithm, you get the character of a spring without the “boing”.

## Nonlinear

An effect pioneered in the early 1980's was gated reverb. Either a live room or a digital reverb was sent through a noise gate, which was then triggered by the input signal. This created a large reverberant space which would cut off suddenly. You can get this effect in any of the other reverb algorithms, but “Nonlinear” simulates the sound of classic “gated” digital reverb programs. It is generally used for drums and percussion.

## Reverse

Another popular trick in the 80's was to record the reverb with the tape flipped over, so it would play backwards in the mix. It is a useful effect for drums and other percussive sounds — it adds space without washing out the instrument.

## Reverb Parameters

Most of the reverb effects in the QuadraVerb 2 operate under the same set of control parameters, which are listed and described in this section. However, reverbs which use more DSP provide parameters which take advantage of their extra processing power; parameters which are not found in the other, smaller reverb algorithms. For example, Reverberation Swirl is a parameter found in the Room 2 and Hall 2 types, but not in Plate 1. And the Nonlinear reverb type does not provide gating options, since it is already a “gated reverb” algorithm. Here are the reverb parameters:

### Decay

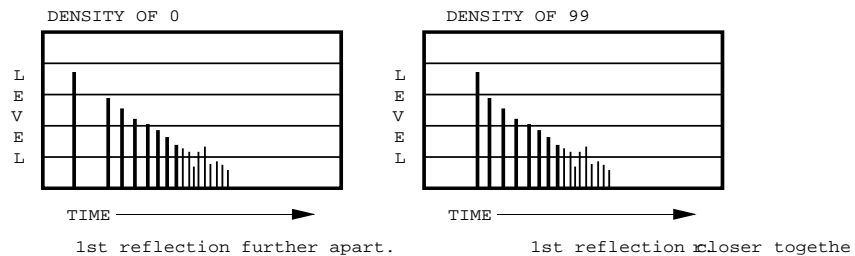
The Reverb Decay determines how long the Reverb will sound before it dies away. When using the Reverse Reverb type, Reverb Decay controls the Reverse Time.

### Damping – Hi & Lo

These two parameters allow the Decay Time to be set separately for both the low and high frequencies of the Reverb. This means that you have control over the tonal shape of the Reverb itself, being able to make the high frequencies die faster if the effect is too bright, and being able to make the lows die faster if the effect is too boomy. This allows you to simulate different surfaces of a room or hall, with softer surfaces absorbing more high frequencies and smaller rooms having faster low frequency decay.

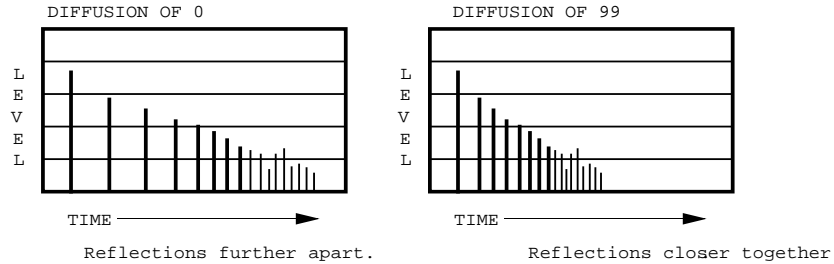
### Reverb Density

Usually, in natural acoustic spaces, some time elapses between the first reflection and all other reflections that make up the reverb sound. If you listen to only the reverb with Density set to 1, you will hear the source sound repeat (the first reflection), a short bit of silence, then the onset of the rest of the reverb. Increasing the Density shortens the time between the first reflection and the remaining reverb reflections. At the maximum setting of 99, the reverb will seem to “explode” since the first reflection will no longer be perceived as a separate echo. This parameter is very useful with percussive sounds, since an isolated first reflection may sound like an additional percussive “hit”.



### Diffusion

Diffusion determines the “thickness” of the reverb sound by altering the time that occurs between reverberant reflections. With lower diffusion settings, you may be able to actually hear the individual echoes that make up the overall reverb sound. With higher diffusion settings, the echoes increase in number and blend together. Greater diffusion works better with percussive sounds, whereas less amounts of diffusion work well with vocals and other sustained sounds.



## Input High Frequency Roll Off

This is a lowpass filter that can be set between 200Hz and 20kHz, and attenuates all frequencies above this value by 6dB per octave.

## Predelay

All the reverb effects also have pre-delay parameters. Predelay slightly delays the reverb itself up to 250 ms, so that the dry signal more easily stands out from the reverb. A bit of predelay can make certain instruments (such as snare drums) sound bigger.

## Predelay Mix

This allows you to balance the amount of predelay to direct signal as a percentage of each. This gives you the ability to hear a bit of the reverb before the loudest part of the reverb (the predelayed reverb) sounds, and makes for a bigger, smoother sounding reverb.

## Reflection Level and Spread

This is only used with the more complex reverb algorithms, such as Room 2, Hall 2, Plate 2, Chamber 2 and Large Plate. Level controls the amplitude of the early reflections. Spread controls the reflection's density, or how close they are together.

## Reverberation Swirl

This adds a type of stereo chorus-like the effect to the output of the reverb. Only used in Room 2, Hall 2 and Large Plate, this parameter is most useful for smoothing the decay of the reverb when set at a low value. When set to a high value, it creates a more dramatic detuning effect.

## Reverberation Attack

Only used in Plate 2 and Chamber 2, this controls the amount of extra signal added to the beginning of the reverb's effect, which in turn controls the intensity of the reverb's attack.

## Gating

Of all the different reverb types, only two types do not provide gating parameters (“Reverse” and “Nonlinear”). Gating is the process which abruptly cuts off the reverb’s decay for a more “choppy” sound. This effect is very popular on drums because it makes them sound HUGE. It is achieved by dropping the level of the signal very rapidly after the initial attack making a short, sharp sound.

There are four gating parameters available in most of the reverb types. These include: Gate Trigger Source, Hold Time, Gate Release Time and Gate Level. The Trigger Source can be either the Left or Right inputs, or both. Or this can be set to “OFF” when you do not wish to use the gating effect. The Hold Time determines how long the gate will be held open before it begins to turn off; this can be set from 10 to 500 ms. The rate at which it closes is determined by the Gate Release Time, which can be set from 0 to 500 ms. The Gate Level controls the level of the reverb signal after the gate closes. In other words, if the Gate Level is set to 99, then no reverb will sound after the gate turns it off. If the Gate Level is set to 50, then some reverb signal will still be present even after the gate turns off the main reverb signal.

CHAPTER 4

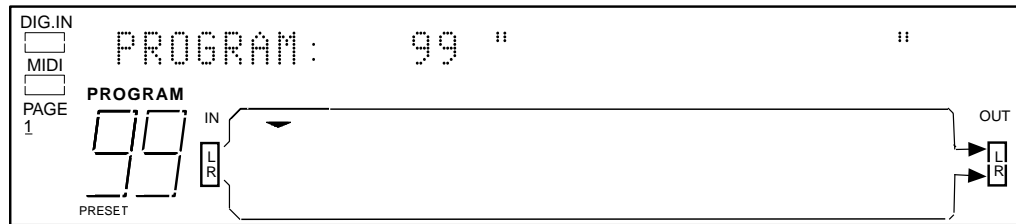
# MAKING YOUR OWN PROGRAMS

## Getting Started

Because the QuadraVerb 2's custom LCD display clearly shows the audio routings between each of the 8 possible Effect Blocks, creating and modifying your own patches is extremely easy and flexible.

There are 200 possible Program locations available, divided into two banks – Preset and User – of 100 Programs each. These Programs are numbered 0 through 99. The QuadraVerb 2 is shipped with 100 factory presets in the Preset bank; these can be modified but not permanently changed. All edited programs must be stored in the User bank .

Let's start by creating a simple single-Block program starting from scratch, then go on from there by adding other blocks. We'll use Preset #99, which has been left un-programmed for the purpose of this tutorial. Refer to Chapter 2 for information on selecting programs.



This program has the LR INs routed directly to the LR OUTs, respectively. Consequently, there is no audible change to the signal we are feeding to it. Our signal is clean (dry) now, so let's add an effect.

**J** *If the Global Direct Signal Mute function is turned on, any Programs which have the LR IN routed to the LR OUT will not display these "patch cords" except when editing the Program's routings. This is to avoid confusion when operating in this mode. For more information on Global Direct Signal Mute, see chapter 2.*

There are generally five Block Functions available for creating and editing Programs, which are: Type, Routing, Parameter, Mix and Modulation. Modulation deals with controlling various parameters in real-time via a MIDI controller. This information is covered in chapter 6. For now, let's stick to the basics, and go through the Functions one at a time.

## Programming A Single Block

### Type

**J** A Block can provide Equalization, Pitch, Delay, Reverberation or be turned OFF. A program can contain up to eight Blocks, which can be in almost any combination or order within the confines of available DSP memory.

The Block's Type determines not only what parameters are available in the Parameter function, but also what routings are available in the Routing function. Each Block can be modified according to the available parameters (based on the Block's "type"), and routed independently of the other Blocks.

- ⌘ Press the [TYPE] button.  
*The [TYPE] button's LED will light.*
- j Press the [< BLOCK] button repeatedly until the pointer is above the leftmost Block in the display (this is Block #1's position).  
*The display will show page 1 of Type mode (there are 3 pages in this mode):*

BLOCK 1 FUNCTION: OFF

- ↵ Turn the [VALUE/ENTER] knob to select EQUALIZATION as the Block Function.  
*The function will flash until you press the [VALUE/ENTER] button.*
- ÷ Press the [VALUE/ENTER] button.  
*The EQ module will now be displayed as Block 1, and the display automatically advances to Page 2.*

EQ TYPE: LOWPASS FILTER

- f Turn the [VALUE/ENTER] knob to scroll through the various EQ effect types.  
*By rotating the dial we can also see various Shelving, Parametric, and Graphic types of EQs. As we dial through the EQ types they will flash in the display. Let's select a 3 Band Parametric EQ.*

EQ TYPE: 3 BAND PARAMETRIC  
EQ

- a Press the [VALUE/ENTER] button.  
*The EQ type will now be selected for Block 1, and the display will move back to Page 1 of the Type function.*

## Routing

Routing is the Function used to connect audio between the LR IN and the Effect Blocks and the LR OUT. We must route audio to both the inputs and outputs of our EQ Block in order to hear its effect. Routing is always done by selecting the destination (where you want the audio to go to) and assigning it a source (where you want the audio to come from). After making a connection, you can then adjust the level of the source's signal.

- ¿ Use the [< BLOCK >] buttons to select the EQ Block we just created.  
*If you just finished the previous tutorial, the EQ Block should already be selected.*

The IN LR is not selectable as a Block.

**J**

- i Press [ROUTING].  
*The [ROUTING] button's LED will light and the display will read:*

ADD ROUTE FROM: INPUT LEFT

*The cursor appears under the setting (INPUT LEFT) which is flashing.*

- Turn the [VALUE/ENTER] knob to scroll through the various options.  
*The choices are INPUT LEFT, INPUT RIGHT or EQ 1 MIX. As you scroll through, notice the flashing "patch cord" which appears in the lower display connecting IN L to EQ IN or IN R to EQ IN or M out to EQ IN, depending on which option you are viewing in the upper display.*

All three options are routed to the EQ Block's input, so their nomenclature is determined by their source, i.e. FROM INPUT LEFT, FROM INPUT RIGHT, etc.

- ÷ With the "INPUT RIGHT" option selected, press the [VALUE/ENTER] button.  
*This connects the signal from the Right input of the QuadraVerb 2 (IN R) to the input of the EQ module.*

Upon pressing the [VALUE/ENTER] button in step 3, the display automatically advances to Page 2, which is where we may set the level of this connection (the volume from the right input to the EQ in). The display now reads:

LEVEL FROM: INPUT RIGHT =  
-6.0dB

*The patch cord will still flash until a level has been dialed in.*

**J**

- f Turn the [VALUE/ENTER] knob to select the input level for this connection.

## Level Setting

The input level is an important parameter, the setting of which can have a drastic impact on the resulting effect. It's default setting of -6.0dB represents the "halfway" point of its range. 0 dB is the loudest using no attenuation, while OFF mutes the route entirely.

You should set the level to accommodate the headroom the effect will need. For example, even if the front panel input levels are below clipping, if you boost an EQ band, it can distort the internal processor of the Q2. The -6.0 dB setting will accommodate most blocks, but if there is clipping internally (when the red LED's flash



on input), lower the input level. You may also need to lower the routing level if a block is receiving signal from more than one source.

On the other hand, the best signal-to-noise ratio is achieved when signals are as loud as possible without clipping. In the case of a one-block program, the routing from the block to the L and R outputs should be at 0.0 dB. Multiple-block programs will need lower settings to avoid clipping at the output blocks.

<sup>a</sup> Press the [VALUE/ENTER] button to set the level from -48.0dB to 0dB.

D Press the [BLOCK >] button to move the pointer to the OUT R Block.

*The OUT L and OUT R are selectable as separate destination Blocks.*

The display will flash:

J

ADD ROUTE FROM: EQ 1  
MIX

Note that the flashing patch cord now connects the EQ1 M to the OUT R and if we turn the [VALUE/ENTER] knob, we can see the following possible input sources for the right output:

ADD ROUTE FROM: EQ 1 IN MIX

ADD ROUTE FROM: INPUT LEFT

Note that the route from Input Right can't be added, because it's already there as part of the default patch #99.

As we scroll through the various input sources, the display will flash each one's patch cord while selected.

Let's choose EQ 1 MIX (this is the Mix output of our EQ, which is Block 1). This is not to be confused with EQ 1 IN MIX, which is the signal being received at the EQ Block's input.

« With "EQ 1 MIX" selected, press the [VALUE/ENTER] button.

Like before, the display automatically advances to page 2, and now asks us to set the level from the EQ 1 MIX output going to the Right Output.

ENTER LEVEL FROM EQ 1 MIX:  
-6.0dB

» Turn the [VALUE/ENTER] to select a level from -48.0dB to 0dB (100%).

... Press [VALUE/ENTER] button to set the level.

We have now connected the output of the EQ to our Right output.

Having connected both an input and output to our EQ Block, we can now hear the effect and change the parameters.

## Parameter

Now we will choose the desired parameters for our EQ.

- ⌘ Press the [PARAMETER] button  
*The [PARAMETER] button's LED will light.*

**J** *If the display reads "RIGHT OUTPUT SELECTED" you have not yet moved back to BLOCK 1. Press the [< BLOCK] button to move the pointer in the display to Block 1.*

The display will now show you the first of three pages:

LOW PASS: 1000Hz GAIN:  
+0.0dB

These are the only two parameters on this page: Low Pass Frequency and Gain. The cursor appears beneath the Frequency setting (in the display above it is set to 1000Hz; your display may read something different).

- j Turn the [VALUE/ENTER] knob to select a frequency between 200 and 10000Hz.
- Press the [PAGE >] button to move the cursor to the Gain parameter.
- ÷ Turn the [VALUE/ENTER] knob to set the Low Pass gain from -14 to +14 dB.  
*Set the Gain below 0dB to "cut" or above 0dB to "boost" the selected Frequency; a setting of 0dB has no effect.*
- f Press the [PAGE >] button to advance to Parameter Page 2 of this EQ Block.  
*The display will look like this:*

MID BAND: 1000Hz +0.0dB  
1.00OCT

This page has three parameters: Mid Band Frequency, Gain and Q (or bandwidth).

- a Turn the [VALUE/ENTER] knob to select a frequency from 200 to 10000Hz.
- D Press the [PAGE >] button to move the cursor to the Gain parameter.
- « Turn the [VALUE/ENTER] knob to set the Mid Band gain from -14 to +14 dB.  
*Set the Gain below 0dB to "cut" or above 0dB to "boost" the selected Frequency; a setting of 0dB has no effect.*
- » Press the [PAGE >] button to move the cursor to the Q parameter.
- ... Turn the [VALUE/ENTER] knob to set the Mid Band EQ's bandwidth from .20 to 2.50 octave.  
*Set the Q to a small number to effect the specific frequency you selected, or to a larger number to affect a wider range of frequencies centered around the selected frequency.*

Continue editing the remaining EQ parameters in a similar manner.

## Mix

Once we have setup our EQ Block the way we want, we should then go to the Mix function to adjust the Block's output level. This, however, is only necessary if all we want to do is EQ our input signal. That would be fine, except we have 7 more Blocks we can use to stack effects almost any way we want. In the next section we will see how to route this EQ Block into other Blocks. Most of the time, EQ Blocks are used to tailor the input signal before going into another Block or set of Blocks. However, they can also be useful at the end of the effects chain just before the LR OUT (especially when you use the parametric and graphic EQs.

However, since this is just a tutorial, let's look at just how we go about mixing the EQ'd signal with the original input signal.

- ⌘ Press the [MIX] button  
*The [MIX] button's LED will light.*

**J** *If the display reads "DIRECT LEVEL FROM INPUT" you have not yet moved back to BLOCK 1. Press the [< BLOCK] button to move the pointer in the display to Block 1.*

The display will now show you the first of three pages:

LEVEL TO L/R: EQ=100%

- ⌘ Turn the [VALUE/ENTER] knob to adjust how much of the EQ's output is heard at the LR OUT.
- Press the [PAGE >] button to advance to Mix Page 2.  
*The display will read:*

DIRECT LEVEL FROM INPUT:  
100%

- ÷ Turn the [VALUE/ENTER] knob to adjust how much of the direct input signal is being routed to the LR OUT. (For an EQ, the level would normally be 0%)  
*By adjusting this parameter, you can create an ideal balance between the original unaffected signal at the inputs, and the effected signal coming from the EQ Block.*

**J** *If the Global Direct Signal Mute function is turned on, the Direct Level From Input parameter will appear to have no effect. However, any changes you make will be remembered if you store the edited Program. For more information on Global Direct Signal Muting, see chapter 2.*

## Programming Multiple Blocks

When you start adding more Blocks, the same principles apply as described in the previous section. However, the routing options begin to increase, since, as you add more Blocks, the number of input sources increases. Let's take our simple 1 Block Program and add two more Blocks. We'll route the output of the EQ into a Mono Delay, then through a stereo reverb, such as Hall 1, before going to the main outputs. First, we'll add the new Blocks. Then, we'll reroute the signal so that they are chained together in series.

### Defining New Blocks

Adding the two additional Blocks is done in the same way as described earlier, except this time we'll choose Delay and Reverb as the functions.

- ⌘ Press [TYPE].  
*The [TYPE] button's LED will light.*
- ⌘ Use the [< BLOCK >] buttons to move the pointer in the display to Block 2.  
*The display will look like this:*

BLOCK 2 FUNCTION: OFF

- ↵ Turn the [VALUE/ENTER] knob until the display reads "DELAY", and press the [VALUE/ENTER] button.  
*The display advances to Type Page 2.*

DELAY TYPE: MONO DELAY

Since this is the type of effect we want, let's move on and add the Reverb.

- ÷ Press [BLOCK >] so that the pointer is above Block 3.
- f* Turn the [VALUE/ENTER] knob until the display reads "REVERBERATION", and press the [VALUE/ENTER] button.  
*The display advances to Type Page 2.*

REVERB TYPE: PLATE 1

- a* Turn the [VALUE/ENTER] knob until the display reads "HALL 1", and press the [VALUE/ENTER] button.  
*The display will flash until the [VALUE/ENTER] button is pressed.*

REVERB TYPE: HALL 1

### Deleting Unnecessary Routings

Now that we've added our new Blocks, we want to reroute the signals so that the EQ output no longer directly feeds the LR OUT, but instead goes into the Delay Block, and then into the Reverb Block. Then we'll route the Reverb Block to the LR OUT so we can hear our effects at work. The first step, however, is to delete the unnecessary routings before adding the new ones.

- ⌘ Press [ROUTING].  
*The [ROUTING] button's LED will light.*
- ⌘ Press the [BLOCK >] button until the pointer appears above R OUT.
- ⌘ Press [< PAGE] to advance to Routing Page 3.  
*The display will look like this:*

DELETE ROUTE FROM: INPUT  
RIGHT

- ⌘ Turn the [VALUE/ENTER] knob until the display reads "EQ 1 MIX" and press the [VALUE/ENTER] button.  
*The display will flash until the [VALUE/ENTER] button is pressed. At the same time, the "patch cord" connecting the EQ out to the R OUT will disappear.*

## Patching In The New Blocks

Now its time to reroute the signal so the effects form a chain.

- ⌘ Press [ROUTING].  
*The [ROUTING] button's LED will light.*
- ⌘ Press the [< BLOCK] button until the pointer appears above Block 2.
- ⌘ Turn the [VALUE/ENTER] knob until the display looks like this:

ADD ROUTE FROM: EQ 1 MIX

- ⌘ Press [VALUE/ENTER] to select the route.  
*The display will flash until the [VALUE/ENTER] button is pressed. At the same time, the "patch cord" connecting the EQ to the Delay will appear. Also, the display will automatically advance to Routing Page 2, where you can adjust the level of this signal going into the Delay in Block 2. This defaults to -6.0dB, which is the halfway point between maximum and minimum. This gives you enough overhead so that, if later you need more gain, you can raise this signal to a higher level.*
- f Press [VALUE/ENTER] to set the input level. The display will automatically return to the "Add Route" page
- a Press the [BLOCK >] button until the pointer appears above Block 3.
- D Turn the [VALUE/ENTER] knob until the display looks like this:

ADD ROUTE FROM: DLY 2 MIX

- « Press [VALUE/ENTER] to select the route.  
*The display will flash until the [VALUE/ENTER] button is pressed. At the same time, the "patch cord" connecting the Delay to the Reverb will appear. Set the level on page 2 as before.*

- » Press the [BLOCK >] button until the pointer appears above L OUT, and turn the [VALUE/ENTER] knob until the display looks like this:

ADD ROUTE FROM: RVB 3 LEFT

- ... Press [VALUE/ENTER] to select the route.  
*The display will flash until the [VALUE/ENTER] button is pressed. At the same time, the “patch cord” connecting the Reverb to the L OUT will appear.*

- ① Press [BLOCK >] and then press [VALUE/ENTER] to route the right output of the Reverb in Block 3 to the R OUT.  
*The display should look like this before pressing [VALUE/ENTER]:*

ADD ROUTE FROM: RVB 3 RIGHT

## Adjusting Parameters

At this point, the tutorial ends, since there are so many directions you can go in. Each type of effect has its own set of parameters. Using the steps in the first section of this chapter, you can get an idea of how to edit the other effect's parameters, even though they each perform a different function. Refer to chapter 5 for a list of controls, including a list of the parameters used in each of the available effects.

We highly recommend experimentation in this area. Because the QuadraVerb 2 is so flexible, you'll be able to create effect combinations never before possible. *Example:* You can route a Block's output to its own input to create interesting “feedback” effects. However, the level of this “patch cord” is very important to avoid unwanted clipping. This is why when such a “feedback” connection is made, the level is automatically set to 100% attenuation (-48dB) to avoid sudden, unexpected loud levels. Only through experimentation, and examination of the Preset Programs, can you start to create truly innovative effect Programs.

## Moving and Swapping Blocks

A third page is included in Type mode which allows you to move a defined Block to another position. If you defined Block #1 as a Delay, and later decide to first have the input signal routed through an EQ, you can move the Delay Block to make room for it. First select the Block to be moved, press [PAGE >] to advance the cursor to the next field, and select the Block # you want to move it to. Then to execute the move, press the [VALUE/ENTER] button. The Block is moved and the routings repatched to recreate the same signal path. Now you can redefine Block #1 as an EQ. If you move a Block to a position already occupied by another Block, the two Blocks will swap positions, and their routings will automatically be swapped as well to maintain the identical signal path.

Since blocks can be routed in any order you like, Block Move is primarily for visual feedback. The input could be routed to Block 3 first, for example, then to Blocks 1 and 2 and to the outputs. If you prefer, you can rearrange the blocks so they are in the proper order on the screen.

To move a defined Block to another position:

- ¿ Press [TYPE].  
*The [TYPE] button's LED will light.*

- j Press [PAGE >] twice to advance to Type Page 3.  
*The display will look like this:*

```
MOVE BLOCK: 1 INTO BLOCK 2
```

*The right half of the display will flash.*

- ↪ Use the [VALUE/ENTER] knob to select which Block you want to move (1–8).
- ÷ Press [PAGE >] to advance the cursor to the next field.

```
MOVE BLOCK: 1 INTO BLOCK 2
```

- f Turn the [VALUE/ENTER] knob to select where you want to move it to (1–8).
- a Press [VALUE/ENTER] to execute the move.  
*If the selected destination is already occupied by a defined Block, the Blocks will slide into the now-vacant positions.*

## Changing Effect Types

If you already have defined a Block and assigned its routings, you can still change its Function and Effect Type. Usually, this doesn't disturb anything else. However, if the Block was defined as a mono effect and you change it to a stereo effect, the QuadraVerb 2 will make the necessary changes in routings, and vice-versa.

Here's an example: Let's say you have a Mono Delay Block that has its mono output (M) routed to the R OUT. Now you go into Type mode and change it to a Stereo Delay. As soon as you press the [VALUE/ENTER] button and make the change, the display momentarily reads:

```
CONVERTING MONO ROUTES TO  
STEREO
```

Now, the Block has three outputs: L (left), R (right), and M (mono). In this case, the QuadraVerb 2 automatically assigned the R output of the Delay to the R OUT. Since the M output of the Delay Block had originally gone to the R OUT, the QuadraVerb 2 assumed you wanted to continue having the Block go to this output. So, logically, it routed the Block's new R output to the R OUT.

Let's say we did the opposite; i.e., change a stereo effect into mono. The exact opposite would happen. Once we changed the stereo effect into a mono effect type, the display would momentarily read:

```
CONVERTING STEREO ROUTES  
TO MONO
```

In this case, if only one of the effects outputs had been routed to one of the QuadraVerb 2's OUTS, it would be rerouted to come from the new mono Block's M output. If both the Left and Right outputs of the stereo Block were connected to the LR OUT, then the mono effect's M out would now be connected to both the L OUT and R OUT.





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## CHAPTER 5

## DESCRIPTION OF CONTROLS

## Front Panel

## Input Level

**Left and Right:** These concentric knobs (one within the other) control the incoming level of our analog signal, and the relative balance between the two input channels. This “gain-staging” allows us to maintain a superior signal-to-noise ratio and eliminate any overloading of the QuadraVerb 2’s inputs. This feeds directly to the LED ladder.

## Output Level

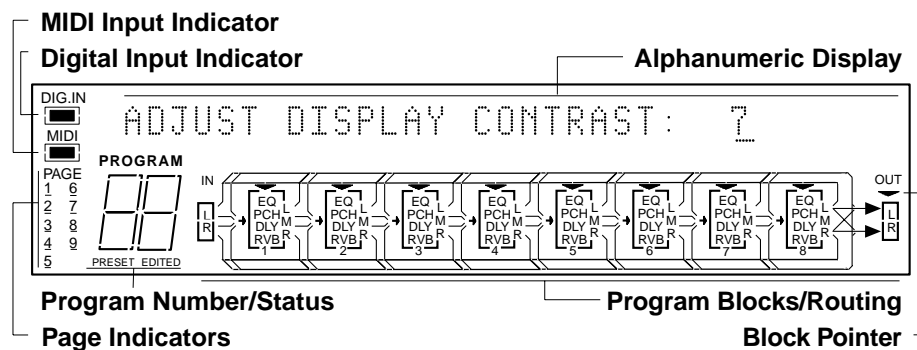
This controls how much signal we are returning to our mixing console, or whatever else it’s connected to. Matching levels throughout the recording/mixing chain will allow us to get the cleanest possible recordings .

## LED Meter

This is a visual indicator of level. It is a four step LED (light emitting diode) ladder with steps at -32dB, -12dB, -3dB (green LEDs), and the red LED indicating CLIP. The dB markings refer to how much headroom is remaining before overload. As with all meters, a certain amount of experimentation is necessary before you feel comfortable, as different types of signals with sharper attack times will affect the LEDs differently. The meters have several sources, so they can detect overloads caused by internal gain or feedback, as well as by the [INPUT LEVEL] controls. Internal clipping is shown when the red LEDs flash; if the red LEDs flash while the -3 LEDs do not light, the levels within the QuadraVerb 2’s program should be lowered.

## Display

The QuadraVerb 2 is equipped with a custom LCD display, which provides (among other things) a complete view of the signal routing used by the current Program.



*Note: This illustration shows all display flags and routes (“patch cords”) turned on at once. Actual displays show only some of these at any time.*

**DIG. IN.** Illuminates if the INPUT AUDIO SOURCE is switched to OPTICAL DATA in the Global functions.

**MIDI.** Illuminates if MIDI signals are present at the MIDI IN.

**Alphanumeric Display.** This 32 character display indicates types of effects, parameters of effect, mode, etc. It also allows us to select and enter types and amounts of an effect. Includes a 14 character entry section for naming the Programs.

**PAGE 1-9.** Indicates the total number of pages available in the currently selected mode, with an underline to indicate the current page. Some functions have more pages than others. A REVERB effect may have seven pages of parameters, the MIDI button will have six pages, for example. Some pages have more than one parameter field. Use the [< PAGE >] buttons to advance the cursor through each parameter field in a page, and to access the other pages in a selected function.

**PROGRAM (00-99).** Indicates the PROGRAM number currently selected.

**PRESET.** Indicates if the PROGRAM selected is a PRESET and if any modifications have been made to the PRESET.

**EDITED.** This indicates that the currently selected Program has been edited (modified from its original version as it is stored in memory). You must Store a Program after editing it, if you want to make those changes permanent. Otherwise, if you select another Program before saving the edited one, your edits will be lost.

**LR IN.** This represents the inputs of the QuadraVerb 2 which need to be routed to the OUT LR before it can pass along its signal. Along the way the signal may be routed through a variety of effects. These inputs are fed directly by the [L] and [R] INPUT connectors on the rear of the unit.

**Blocks 1-8.** Eight selectable, discrete effects processors that can be used together or separately. Choices are Off, Equalization (EQ), Pitch (PCH), Delay (DLY) and Reverberation (RVB).

**LR OUT.** These are the outputs of the QuadraVerb 2. They feed the [L] and [R] OUTPUT connectors on the rear of the unit.

**Patch Cords.** A fully programmable series of connections, representing patch cords, that connect the LR IN to a variety of effects (Blocks), Block outputs to other Blocks, and then to the LR OUT.

## Value/Enter

The [VALUE/ENTER] knob is used to adjust values in the display. However, it is also a button. Depending on what you are editing, the [VALUE/ENTER] knob will work in either one of two modes:

**Immediate.** The desired value is selected by turning the [VALUE/ENTER] knob, and immediately takes effect. This is normally the case when selecting Programs (see next page) and for adjusting most of the numeric parameters.

**Deferred.** The desired value is selected by turning the [VALUE/ENTER] knob, but the new value will only take effect after the [VALUE/ENTER] button has been pressed. The newly selected value will flash in the display until it is selected in this manner. If you change the parameter back to its original setting, the value in the display will not flash. Also, if you go to another Page, or select another Function (by pressing any button), the parameter will be left unchanged. If you went back to look at the previous parameter, it will be set back to its original setting. This mode is used for parameters that cause architectural changes such as changing a Block's function, effect type, and routing signals.

**For Power Users.** The [VALUE/ENTER] button can also be used to step through Pages in the currently selected mode. Except when a value is flashing on and off in the display, the [VALUE/ENTER] button ordinarily doubles for the [PAGE >] button. If you change the value of a parameter that uses “deferred mode,” you must press the [VALUE/ENTER] button to enter the new value (the display will stop flashing), and then you can press it again to move to the next Page (or to the next parameter, if more than one parameter appears in the display). This is a feature for power users who want to be able to move around the various pages quickly and make changes as fast as possible.

**J** *Unique Exception: When editing the Delay Time parameter of a Delay Block set to either Tap Tempo Mono Delay or Tap Tempo Ping Pong, the [VALUE/ENTER] button is used to “tap” in a tempo . See chapter 6 for more information.*

## Program

This button is used, in conjunction with the [VALUE/ENTER] knob, to select the Program number in the display. When its LED is lit, turning the [VALUE/ENTER] knob selects a different Program, using one of two modes: Direct or Deferred. Direct mode immediately recalls the displayed Program as you turn the [VALUE/ENTER] knob. Deferred mode lets you scroll through the Programs in the display by turning the [VALUE/ENTER] knob, but you must press the [VALUE/ENTER] button to actually recall a Program. You can choose between either Direct or Deferred mode by going to Program Page 2, using the following steps:

To select either Direct or Deferred Program Selection Mode:

- 1 Press the [PROGRAM] button.  
*The [PROGRAM] button’s LED will light.*
- 2 Press [PAGE >] to advance to Program Page 2.  
*The display will look like this:*

PROGRAM SELECTION MODE:  
 DIRECT

- 3 Turn the [VALUE/ENTER] knob to select either Direct or Deferred mode.

You can also directly select Programs using the front panel buttons. By holding the [PROGRAM] button, you can press [PRESET] or [USER] (stenciled in gray) to select either the Preset or User bank (100 Programs in each). Also while holding the [PROGRAM] button and using the [1] through [0] buttons (stenciled in gray), you can directly recall any Program (from 00–99) in the current bank (Preset or User) you happen to be in.

## Store

Once you have created a new Program you can press [STORE] to store it. The [STORE] button’s LED will light. You can now dial in a Program number from the User bank, using the [VALUE/ENTER] knob, that you want to store it into (you cannot store a Program into the Preset bank, since this is stored in ROM). Press either the [STORE] button or the [VALUE/ENTER] button and your Program is stored. The display will momentarily read “PROGRAM STORED”. The [STORE] button’s LED will turn off and you will be returned to whatever mode you were in prior to storing. To cancel the operation without storing, press any other button before pressing [STORE] or [VALUE/ENTER].

**J** *Once a Program is stored, the previous Program in the selected location is replaced, and cannot be retrieved. Before storing, you may want to transmit the entire contents of the User bank into a SysEx storage device. See chapter 6 for more info.*

## Compare

Once a Program has been edited, the word "EDITED" will appear in the display below the Program number. If [COMPARE] is pressed, its LED will light and you will temporarily hear (and see) the original version of the Program. This allows you to compare the changes you've made with the original.

## Bypass

Each time the [BYPASS] button is pressed, Bypass mode is toggled on and off. When Bypass mode is engaged, its LED will light. This bypasses the effects chain allowing you to make an instant comparison between the effected signal and the clean signal. Bypass mode can also be enabled using the [BYPASS] footswitch. For more information, see the Footswitch section in chapter 6.

**J** *When Bypass is enabled, all Effect Blocks are momentarily disabled and will not have audio routed from them to the outputs. However, if the Program does not route the LR Inputs directly to the LR Outputs, you will not hear anything.*

In order for Bypass mode to function correctly, make sure the LR Inputs are routed to the LR Outputs and that the Global Direct Signal Mute function is turned off (see below). If the inputs are not routed directly to the outputs, the [BYPASS] button acts more like a "mute" since nothing will be heard when it is enabled.

## < Block >

This rocker button allows you to position the Block pointer above Blocks 1 through 8 or above L OUT or R OUT. These buttons are used in conjunction with the [TYPE], [ROUTING], [PARAMETER] and [MIX] buttons to determine which Block you wish to edit.

When you are editing Block parameters, the QuadraVerb 2 remembers which page you were in last for each Block. This makes programming easier since, when you go from one Block to another (which may be of a different effect type with different parameters/pages), you will be back in the same page you left when you return to the previous Block to do more editing.

## < Page >

This rocker button allows you to scroll back and forth through the pages indicated under PAGE (maximum of 9) in the display. Sometimes a Page has more than one parameter in it; in this situation, the [< PAGE >] buttons will step through each parameter field separately before moving on to the next page.

## Type

Use the [TYPE] button to define the type of effect in the selected Block. Once a Block has been chosen and the [TYPE] button is pressed, we can dial through with the [VALUE/ENTER] knob to choose the function of that Block from EQUALIZATION, PITCH, DELAY, REVERBERATION or OFF. Once the general function of a Block has been selected by pressing the [VALUE/ENTER] button, the display will automatically advance to the second page of the Type function where we can define the specific effect of the Block. *Example:* If it is a Reverberation Block, you can select Hall 1, Plate 1, Non Linear, etc., using the [VALUE/ENTER] knob.

Another page is included in Type mode which allows you to move a defined Block to another position. If you defined Block #1 as a Delay, and later decide to first have the input signal routed through an EQ, you can move the Delay Block to make room for it. First select the Block to be moved, press [PAGE >] to advance the cursor to the next field, and select the Block # you want to move it to. Then to execute the move, press the [VALUE/ENTER] button. The Block is moved and the routings repatched to recreate the same signal path. Now you can redefine Block #1 as an EQ. If you move a Block to a position already occupied by another Block, the two Blocks will swap positions, and their routings will automatically be swapped as well to maintain the identical signal path.

Selection	Parameter	Page #
Defined Blocks	Block Function	1
	Effect Type	2
	Move Block	3
Undefined Blocks	Block Function	1
	Move Block	2

## Parameter

Once a type of effect has been chosen, pressing the [PARAMETER] button allows us to see and choose the available parameters for that specific effect. *Example:* This is where you define specific attributes of a Block; the boost or cut of an EQ, or the decay time of a Reverb. Some effects have many more parameters and these can be found by [PAGE]ing through while in Parameter mode.

## Routing

In order for us to hear our signal through the QuadraVerb 2 it must be routed from input to output. Also in order for us to hear our effects, the signal must be routed through the Blocks that contain our chosen effects.

Pressing the [ROUTING] button selects Routing Page 1. This is where you can dial through, with the [VALUE/ENTER] knob, all the possible connections. Once you find the route you want, press [VALUE/ENTER] to select it. This automatically advances the display to Routing Page 2, where you can enter a level for that connection. Page 3 allows you to delete any unwanted patches.

Selection	Parameters	Page #
Defined Blocks, L OUT, R OUT	Add Route From:	1
	Level From:	2
	Delete Route From:	3
Undefined Blocks	None ("BLOCK IS OFF")	1

## Mix

Allows us to create a mix of the input signal, the levels of effects sent to the outputs, and the Master effects level. If an unassigned Block is selected, or the LR OUT, the Mix function will have only two pages. These let you control the Direct signal level being routed to the outputs, and adjust the master effects level of all Blocks routed to the outputs. Normally, these are adjusted as a percentage of the total level.

When a defined Block is selected, a new Page 1 is added, and the other two pages move down (Page 1 becomes page 2, Page 2 becomes Page 3). Page 1 will now let you adjust the selected Block's level going to the LR OUT.

Selection	Parameters	Page #
Defined Blocks	Level to L/R	1
	Direct Level From Input	2
	Master Effects Level	3
L OUT, R OUT, Undefined Blocks	Direct Level From Input	1
	Master Effects Level	2

If the selected Block is not routed to the LR OUT, the Level to L/R parameter value will be “NONE”. If the inputs are not routed to the outputs in the current Program, the Direct Level will be “NONE”. Likewise, if none of the defined Blocks are routed to the LR OUT, the Master Effects Level will be “NONE”.

## Global

When the [GLOBAL] button is pressed, its LED will light. The [GLOBAL] button gives us access to the following settings which affect the entire instrument:

**Page 1: Adjust Display Contrast: 1–10.** This controls the contrast of the LCD display. The value range is 1–10, and the default is 5. Use the [VALUE/ENTER] knob to adjust the contrast to a comfortable setting.

ADJUST DISPLAY CONTRAST: 5

**Page 2: Footswitch Range .** This page has two values which determine the start and end point of the range of Programs that are selectable via the [ADVANCE] footswitch connector. Both values have a range of 00—99 Preset (Pset) and 00—99 User. For more information, see the Footswitch section in chapter 6.

FOOTSWITCH: 00 Pset TO 00 Pset

**Page 3: VU Meter Peak Hold.** When set to “ON”, the VU meters’ peak LED indicators will remain lit for about a second.

VU METER PEAK HOLD: OFF

**Page 4: Input Audio Source.** Normally, this is set to “ANALOG DATA” for using the analog input connectors on the rear panel. If using the Alesis Optical DIGITAL [IN] connector, however, this should be switched to “OPTICAL DATA”. When this happens, the [DIG.IN] indicator will light. At the same time, the Sample Clock Source parameter (see below) is automatically switched to “OPTICAL”.

INPUT AUDIO SOURCE: ANALOG  
DATA

**Page 5: Sample Clock Source.** Normally, when using the analog inputs, this parameter is set to Internal, so that the QuadraVerb 2 uses its own “INTERNAL” sample clock for processing effects in the digital domain. However, if you are recording digitally onto ADAT using the QuadraVerb 2’s optical connectors, and you are using either an Alesis BRC or AI-2 synchronizer, you must feed the master 48 kHz clock signal back to the QuadraVerb 2 so that it is in sync with the recording system. This is done by connecting a BNC cable from the clock source (BRC or AI-2) to the [48kHz IN] connector on the QuadraVerb 2, and then setting the Sample Clock Source parameter to “48kHz INPUT”. If the Input Audio Source parameter is set to “OPTICAL”, this parameter will automatically be set to “OPTICAL” and you will not be able to change its setting.

SAMPLE CLOCK  
SOURCE: INTERNAL

**Page 6: Dig In/Out Channels.** When using the Alesis Optical DIGITAL [IN] connector, (Input Audio Source must be set to “OPTICAL DATA”, see above), this page allows you to select which of the 8 channels on the optical buss will be routed to the L and R IN. Both inputs can be set to receive any channel on the buss, from 1 to 8. You can even route the same channel to both inputs. Since the QuadraVerb 2 has two outputs (left and right), you get to chose which of the 8 channels on the outgoing optical buss they should be sent on. Both outputs can be routed to any channel on the buss, from 1 to 8. However, you cannot assign both outputs to the same channel on the optical buss. For the outputs to be routed at all, the Digital Output Mode must be set to “NORMAL” (see below).

DIG CH IN: L=1,R=2 OUT: L=1,R=2

**Page 7: Digital Output Mode.** This determines how the DIGITAL [OUT] connector will function. When set to “NORMAL”, the QuadraVerb 2’s LR OUT is routed to the DIGITAL [OUT] connector. When set to “THRU”, whatever is received at the DIGITAL [IN] connector is passed on to the DIGITAL [OUT] connector without any processing occurring on the digital signal within the QuadraVerb 2. Use this mode when you are connected between to Alesis Optical devices (such as ADAT) between which you wish to bounce digital audio.

DIGITAL OUTPUT MODE: NORMAL



**Page 8: Tap Tempo Footswitch.** This lets you choose which footswitch jack (either [ADVANCE] or [BYPASS]) will be used to provide a tap tempo source for setting delay time, provided the selected Program uses one of the two available tap tempo delay types. This requires that you have defined an Effect Block as one of the two Tap Tempo delay types, and that the desired footswitch jack has been selected for controlling tap tempo.

**Page 9: Global Direct Signal.** The purpose of this feature is to satisfy the conditions of a recording studio environment. When connecting the QuadraVerb 2 to a mixing console's aux sends and returns, it is generally desirable to remove the direct signal feed from the outputs of the QuadraVerb 2. Since most QuadraVerb 2 Programs route the LR IN signal to the LR OUT, you will need to remove this connection from the Program's routing when connecting to a mixer's aux sends and returns. However, this can be done globally for all Programs using this function. If it is set to "ON", the inputs will be heard at the outputs, if they are routed this way in the current Program. By setting this function to "MUTE", you can remove this connection for all Programs without having to delete the "patch cords" in each that route the inputs to the outputs.

GLOBAL DIRECT SIGNAL: ON

When set to "MUTE", recalling a Program which has the LR IN routed to the LR OUT will not display those "patch cords" so as to avoid confusion. You can, however, still add or delete these connections, as well as adjust the Direct Signal Level in Mix mode, and these changes will be remembered when you store the edited Program. However, as long as the Global Direct Signal function is set to "MUTE", you will not hear the effect of any of these edit.

### Name

Selecting this button allows us to name, edit a name or rename a Program. Names can contain up to 14 Characters and are selected by turning the [VALUE/ENTER] knob. The cursor is positioned by using the [< PAGE >] buttons. Or, you can press the [VALUE/ENTER] button to advance the cursor to the right; this helps make naming your Programs a fast operation. Once chosen, the name can be stored along with the Program using the [STORE] button (see above). Here is a chart of available characters:

	!	"	#	\$	%	&	'	(	)	*	+	,	-	.	/	0	1	2	3
4	5	6	7	8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G
H	I	J	K	L	M	N	O	P	Q	R	S	T	U	V	W	X	Y	Z	[
¥	]	^	_	`	a	b	c	d	e	f	g	h	i	j	k	l	m	n	o
p	q	r	s	t	u	v	w	x	y	z	{		}	Æ	¨				

### MIDI

When the [MIDI] button is pressed, its LED will light and the display will move to MIDI Page 1. There are a total of 6 pages in all, containing various MIDI related parameters. For more information about MIDI parameters, see chapter 6.

**Page 1: MIDI Channel.** This sets the basic MIDI channel of the QuadraVerb 2. This is the channel used to receive MIDI program change messages for recalling Programs, and for receiving controller, pitch-bend, note or velocity information for controlling various effect parameters in realtime from an external MIDI source (such as a controller or sequencer). The MIDI channel can be set between 1—16, or set to Omni (this lets the QuadraVerb 2 respond to all 16 channels).

MIDI CHANNEL: 1

**Page 2: MIDI Program Change.** This determine whether or not the QuadraVerb 2 will respond to incoming MIDI program change messages. This can be set to either “ON” (recognize), “OFF” (ignore) or “TABLE” (use Program Table). When turned on, the QuadraVerb will recognize program change messages received on the same MIDI channel it is set to receive on, and respond by calling up the equivalent Program number from the User bank. When set to “TABLE”, the Program that is recalled is based on how the Program Table is set up (see below).

MIDI PROGRAM CHANGE: OFF

**Page 3: Program Table.** The Program Table allows you to intercept incoming program change messages and have them recall specific Programs (in either the Preset bank or the User bank) which may not be the same number value as the program change message received. There are 128 different possible MIDI program change messages (000 through 127). However, the QuadraVerb 2 has 200 Programs to choose from. Therefore, the Program Table allows us to choose which of the 200 Programs will be recalled when certain program change numbers are received. The first value indicates the MIDI program change you wish to remap (000—127). The second value in the display indicates the Program you wish to be recalled (00—99 Preset (Pset) and 00—99 User). You can remap each of the 128 program change numbers, if so desired.

PROGRAM TABLE: 00 = 00 User

**Page 4: MIDI Thru.** This determines whether information received at the [MIDI IN] connector should be routed to the [MIDI OUT] connector. When “ON”, the [MIDI OUT] connector “echoes” any MIDI messages it receives on the [MIDI IN] connector.

MIDI THRU: OFF

**Page 5: System Exclusive Receive.** This determines whether the QuadraVerb 2 will respond to any received MIDI SysEx messages. SysEx (System Exclusive) is used to control the various parameters of the QuadraVerb 2. When “ON”, the QuadraVerb 2 will respond to SysEx messages. When off, SysEx messages will be ignored.

SYSTEM EXCLUSIVE RECEIVE:  
OFF

**Page 6: Send MIDI Program.** This page lets you dump out a single Program or the current Program being used/edited), or the entire User bank via MIDI. The data is sent as SysEx information. This can be sent to a MIDI storage device, or to another QuadraVerb 2. Select either “ALL” or a Program number from 00—99 (User bank only) or EDIT (the currently selected Program which is in the edit buffer). When ALL is selected, simply pressing the [VALUE/ENTER] button starts the MIDI dump. The display will read “MIDI PROGRAM DUMP IN PROGRESS...” indicating that all 100 User Programs are being sent out the [MIDI OUT] connector. If EDIT or 00—99 is selected, a second value appears which lets you select which Program location to send the selected Program to. This is useful when you are connected to another QuadraVerb 2, and wish to transfer a Program from one to the other but do not want to store it in the same location.

SEND MIDI PROGRAM: 00 TO 00

## Modulation

The QuadraVerb 2 allows you to set up to 8 modulation routings which can control almost any parameter within a Program in realtime. This is especially useful for live applications, where the performer can actually interact or automate the changes in one or more effects.

When the [MODULATION] button is pressed, its LED will light and the display will advance to Modulation Page 1. There are a total of 9 pages in Modulation mode. Pages 1 through 3 let you select a Modulator (or modulation patch) which provides the means for routing a particular MIDI (such as pitch bend) or one of the Internal Generators to modulate one of the Block parameters used in the current Program. In each of the first three page, the first field lets you select a Modulator from 1 to 8.

MOD 1 TO:DLY 1-DELAY TIME

The second field of page 1 lets you choose a Block in the current Program. The Block Type will be shown as an abbreviation (EQ, PCH, DLY, REV) along with its location number (i.e. DLY-1 means Block 1 which is a Delay type). The third field is where you choose a particular parameter to be modulated. The list of available parameters depends on the selected Block's type.

MOD 1 SOURCE: PITCH BEND

The second field of page 2 lets you select a Source to modulate the chosen Target.

The second field of page 3 lets you adjust the amount of modulation (Amplitude), either positively or negatively, from -99 to +99.

MOD 1 AMPLITUDE: +0

Pages 4 through 9 allow you to define the two Internal Generators. These can be used as an alternative to external MIDI control, allowing the audio signal at the QuadraVerb 2's inputs to control the effect parameters in realtime.

GEN 1 TYPE: INPUT ENVELOPE

Page 4 lets you choose what type each of the two Input Generators should be. There are two types to choose from: Input Envelope and Peak Follower. The first field lets you select which Input Generator you wish to edit. The remaining pages are used to setup the parameters for the Input Generators.

For more information about the Input Generators, see chapter 6.

## Power

The [POWER] switch turns the AC power to the unit ON (in) or OFF (out).

## Rear Panel

### Power

This is a 4 pin DIN plug which connects to Alesis AC09-25D Power Supply (supplied). The power supply is then connected to an AC outlet delivering a nominal 120VAC

(North American versions). The correct power supply must be used AT ALL TIMES. Any other power supply might create a fire risk and/or permanently damage your unit. This damage would NOT be covered under your warranty.

### **MIDI In**

This is a 5 pin DIN standard MIDI plug which connects to any MIDI compatible equipment such as a MIDI sequencer that will send program changes and controller information to the unit.

### **MIDI Thru/Out**

This is a 5 pin DIN standard MIDI plug which connects to any MIDI compatible equipment such as a keyboard or another effects device. Provided for sending system exclusive commands for storing programs. It also relays all messages received on the MIDI IN if MIDI THRU is enabled.

### **Bypass - Bypass Footswitch**

This is a 1/4" phone jack which connects to a footswitch, either normally-open or normally-closed. When the footswitch is pressed the signal will bypass the effects chain allowing an instant comparison between the dry and wet signal. The red LED on the [BYPASS] button on the front panel will illuminate.

### **Advance - Program Advance Footswitch**

This is a 1/4" phone jack which connects to a footswitch, either normally-open or normally-closed. When the footswitch is pressed the currently selected program number is advanced by one. When 99 is reached it will roll-over to 00. However, you can create a range of Programs to be used in conjunction with this footswitch. See Footswitch Range, earlier in this chapter.

### **48kHz in - Sample Clock Input**

This is a BNC connector which connects to an external master clock from an external device to synchronize digital output. It needs to be used when recording the digital outputs of the QuadraVerb 2 to an ADAT system that uses either an Alesis BRC or AI-2 synchronizer. Once connected, it is necessary to set the QuadraVerb 2's Sample Clock Source to "48kHz INPUT". See chapter 7 for more information.

### **Digital In - Digital Audio In**

This is an EIAJ fiber optic jack which connects to the ADAT optical interface for receiving two of any eight digital audio channels. See chapter 7 for more information.

### **Digital Out - Digital Audio Out**

This is an EIAJ fiber optic jack which connects to the ADAT optical interface for transmitting on two of any eight digital audio channels. See chapter 7 for more information.

### **Left/Right In - Analog Audio In**

These are 1/4" TRS phone jacks which connect to analog source devices such as the effects sends of mixing consoles. These are true stereo balanced inputs and are represented by the IN Block on the front panel LCD display. They may be used with nominal input levels from -20dBV (guitar level) to +4dBu.

## Left/Right Out - Analog Audio Out

These are 1/4" TRS phone jacks which connect to analog devices such as the effects returns on a mixing console. These are true stereo balanced outputs and are represented by the OUT Block on the front panel LCD display. These outputs are servo balanced, meaning that if one leg of the signal is grounded, the other leg will automatically increase 6dB to maintain the nominal output level.

## Effect Parameters

The following charts are designed to act as a “road map” to assist you in locating parameters and their value ranges. All of the following parameters may be found by first selecting the appropriate effect type, pressing the [PARAMETER] button, and using the [< PAGE >] buttons to navigate through the various pages. The number of pages available depends on the effect type you’ve selected. The total number of pages will be shown in the left side of the display, and the currently selected page number will have an underline beneath it.

### Equalization

EQ Type	Parameters	Page#
Lowpass Filter	Lowpass Fc (frequency): 20Hz-10kHz	1
Bandpass Filter	Bandpass Fc: 200Hz-10kHz, BW (bandwidth): 0.20-2.50 Octaves	1
Highpass Filter	Highpass Fc: 100Hz-20kHz	1
Lowpass Shelf	Lowpass: 20Hz-10kHz, Gain: $\pm 14$ dB	1
Highpass Shelf	Highpass: 100Hz-20kHz, Gain: $\pm 14$ dB	1
1 Band Low Parametric	Bandpass: 20Hz-2000Hz, Gain: $\pm 14$ dB, BW: 0.20-2.00 Octaves	1
1 Band High Parametric	Bandpass: 200Hz-10kHz, Gain: $\pm 14$ dB BW: 0.20-2.50 Octaves	1
2 Band Sweep Shelf	Lowpass: 20Hz-10kHz, Gain: $\pm 14$ dB	1
	Highpass: 100Hz-20kHz, Gain: $\pm 14$ dB	2
3 Band Parametric	Lowpass: 20Hz-10kHz, Gain: $\pm 14$ dB	1
	Mid Band: 200Hz-10kHz, Gain: $\pm 14$ dB, BW: 0.20-2.50 Octaves	2
	Highpass: 100Hz-20kHz, Gain: $\pm 14$ dB	3
4 Band Parametric	Lowpass: 20Hz-10kHz, Gain: $\pm 14$ dB	1
	Low Mid: 20Hz-2kHz, Gain: $\pm 14$ dB, BW: 0.20-2.00 Octaves	2
	High Mid: 1500Hz-10kHz, Gain: $\pm 14$ dB, BW: 0.20-2.50 Octaves	3
	Highpass: 100Hz-20kHz, Gain: $\pm 14$ dB	4
5 Band Graphic	63Hz/250Hz/1kHz/4kHz/16kHz: $\pm 14$ dB <i>Note: All bands have a two octave bandwidth.</i>	1
Resonator	Resonator Tuning: A0-A5 (4 Octaves), Decay 0-99	1
Mono Tremolo	Speed: 0-99, Depth: 0-99, Shape: Soft/Hard	1
Stereo Tremolo	Speed: 0-99, Depth: 0-99, Shape: Soft/Hard	1
Stereo Simulator	Stereo Spread: 0-99 Degrees	1

## Pitch

Pitch Type	Parameters	Page#
Mono Chorus	Speed: 0-99, Depth: 0-99, Feedback: 0-99	1
	Chorus PreDelay: 0-100mS	2
	Chorus Shape: Sine/Square	3
Stereo Chorus	Speed: 0-99, Depth: 0-99, Feedback: 0-99	1
	PreDelay Chorus 1: 0-100mS, 2: 0-100mS	2
	Chorus Shape: Sine/Square	3
Quad Chorus	Speed: 0-99, Depth: 0-99, Feedback: 0-99	1
	PreDelay Chorus 1: 0-100mS, 2: 0-100mS	2
	PreDelay Chorus 3: 0-100mS, 4: 0-100mS	3
Mono Flanging	Speed: 0-99, Depth: 0-99, Feedback: 0-99	1
	Flanging Shape: Sine/Triangle	2
Stereo Flanging	Speed: 0-99, Depth: 0-99, Feedback: 0-99	1
	Flanging Shape: Sine/Triangle	2
Phasor	Phasor Speed: 0-99, Depth: 0-99	1
Mono Lezlie	Lezlie Motor: On/Off, Speed: Slow/Fast	1
	High Rotor Level: -12dB to +6dB	2
Stereo Lezlie	Lezlie Motor: On/Off, Speed: Slow/Fast	1
	High Rotor Level: -12dB to +6dB	2
	Stereo Separation: 0-99	3
Pitch Shifter	Fine: 0-99, Coarse: $\pm 12$ (semi tone)	1
Pitch Detune	Detune Amount: $\pm 99$	1
Ring Modulator	Spectrum Shift: $\pm 100$	1

## Delay

Delay Type	Parameters	Page #
Mono Delay	Delay Time: 0.1-5000mS, Beat Count: 1/2, 1/4., 1/4, 1/4T, 1/8., 1/8, 1/8T, 1/16, 1/32 (. = dotted, T = triplet)	1
	Feedback: 0-99	2
Stereo Delay	Left Delay: 0.1-5000mS, Beat Count: 1/2, 1/4., 1/4, 1/4T, 1/8., 1/8, 1/8T, 1/16, 1/32	1
	Left Feedback: 0-99	2
	Right Delay: 0.1-5000mS, Beat Count: 1/2, 1/4., 1/4, 1/4T, 1/8., 1/8, 1/8T, 1/16, 1/32	3
	Right Feedback: 0-99	4
Ping Pong Delay	Delay Time: 0.1-5000mS, Beat Count: 1/2, 1/4., 1/4, 1/4T, 1/8., 1/8, 1/8T, 1/16, 1/32	1
	Feedback: 0-99	2
Multi Tap Delay	Tap 1, Del: 0.1-5000mS, Vol: Off/-48-0dB, L<->R: <99-<50>-99, FB: 0-99	1
	Tap 2, Del: 0.1-5000mS, Vol: Off/-48-0dB, L<->R: <99-<50>-99, FB: 0-99	2
	Tap 3, Del: 0.1-5000mS, Vol: Off/-48-0dB, L<->R: <99-<50>-99, FB: 0-99	3
	Tap 4, Del: 0.1-5000mS, Vol: Off/-48-0dB, L<->R: <99-<50>-99, FB: 0-99	4
	Tap 5, Del: 0.1-5000mS, Vol: Off/-48-0dB, L<->R: <99-<50>-99, FB: 0-99	5
	Master Feedback 0-99	6
Tap Tempo Mono Delay	Delay Time: 0.1-5000mS, Beat Count: 1/2, 1/4., 1/4, 1/4T, 1/8., 1/8, 1/8T, 1/16, 1/32	1
	Feedback: 0-99	2
Tap Tempo Ping Pong	Delay Time: 0.1-5000mS, Beat Count: 1/2, 1/4., 1/4, 1/4T, 1/8., 1/8, 1/8T, 1/16, 1/32	1
	Feedback: 0-99	2



## Reverberation

Reverb Type	Parameters	Page #
Mono Room	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	5
	Gate Release Time: 0-500mS	6
	Gate Level: 0-99	7
Room 1	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	5
	Gate Release Time: 0-500mS	6
	Gate Level: 0-99	7
Hall 1	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	5
	Gate Release Time: 0-500mS	6
	Gate Level: 0-99	7
Plate 1	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	5
	Gate Release Time: 0-500mS	6
	Gate Level: 0-99	7
Chamber 1	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	5
	Gate Release Time: 0-500mS	6
	Gate Level: 0-99	7
Room 2	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Reflection Level: 0-99, Spread: 0-99	5
	Reverberation Attack: 0-99	6
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	7
	Gate Release Time: 0-500mS	8
	Gate Level: 0-99	9

Reverb Type	Parameters	Page #
Hall 2	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Reflection Level: 0-99, Spread: 0-99	5
	Reverberation Swirl: 0-99	6
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	7
	Gate Release Time: 0-500S	8
	Gate Level: 0-99	9
Plate 2	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Reflection Level: 0-99, Spread: 0-99	5
	Reverberation Attack: 0-99	6
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	7
	Gate Release Time: 0-500mS	8
	Gate Level: 0-99	9
Chamber 2	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Reflection Level: 0-99, Spread: 0-99	5
	Reverberation Attack: 0-99	6
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	7
	Gate Release Time: 0-500mS	8
	Gate Level: 0-99	9
Large Plate	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Reflection Level: 0-99, Spread: 0-99	5
	Reverberation Swirl: 0-99	6
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	7
	Gate Release Time: 0-500mS	8
	Gate Level: 0-99	9
Large Room	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	3
	Predelay: 1-250mS, Mix: <99-<50>-99>	4
	Reflection Level: 0-99, Spread: 0-99	5
	Reverberation Swirl: 0-99	6
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	7
	Gate Release Time: 0-500mS	8
	Gate Level: 0-99	9

Reverb Type	Parameters	Page #
Spring	Decay: 0-99, Damping Hi: 0-99, Lo: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	2
	High Frequency Roll Off: 200Hz-20,000Hz	4
	Predelay: 1-250mS, Mix: <99-<50>-99>	5
	Gate Trig: Off/Left/Right/L&R, Hold: 10-500 mS	7
	Gate Release Time: 0-500mS	8
	Gate Level: 0-99	9
Nonlinear	Gate Hold Time: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	
	Input High Frequency Roll Off: 200Hz-20,000Hz	2
	Predelay: 1-250mS, Mix: <99-<50>-99>	3
Reverse	Reverse Time: 0-99	1
	Reverb Density: 0-99, Diffusion: 0-99	
	Input High Frequency Roll Off: 200Hz-20,000Hz	2
	Predelay: 1-250mS, Mix: <99-<50>-99>	3

## CHAPTER 6

# ADVANCED APPLICATIONS

## MIDI Functions

The QuadraVerb 2 provides many MIDI functions, including being able to respond to program changes, sending and receiving Program information via SysEx dumps, and realtime control over various effect parameters via MIDI controllers. For more information about basic MIDI connections, see chapter 1.

### Global MIDI Channel

The Global MIDI Channel is used to receive program change messages, as well as other MIDI events for use with the realtime modulation capabilities provided in the QuadraVerb 2.

To set the QuadraVerb 2's MIDI channel:

- ⌘ Press [MIDI].  
*The [MIDI] LED will light and the display will read:*

MIDI CHANNEL: 1

- ⌘ Turn the [VALUE/ENTER] knob to set the Global MIDI Channel to either 1 through 16, or OMNI (all 16 channels simultaneously).

### Receiving Program Changes

In order to recall programs or control parameters on the QuadraVerb 2 from a MIDI control device (keyboard, drum pad, guitar or bass controller, sequencer, etc.):

- ⌘ Connect the control device's MIDI OUT to the QuadraVerb 2's [MIDI IN].  
*This provides a one-way connection, since this is all we need to control the QuadraVerb via MIDI (we don't require the QuadraVerb to send MIDI information back to the other device).*

- ⌘ Press [MIDI].  
*The [MIDI] LED will light.*
- ⌘ Press [PAGE >] to access MIDI Page 2.  
*The display will read:*

MIDI PROGRAM CHANGE: OFF

- ⌘ Turn the [VALUE/ENTER] knob to turn MIDI Program Changes "ON".  
*Incoming MIDI program change messages received on the same channel the QuadraVerb 2 is set to will recall the same numbered User Program in the QuadraVerb 2.*

### Program Change Table

Alternatively, you can use a Program Change Table in the QuadraVerb 2 which re-maps incoming program change messages so that they recall QuadraVerb 2

Programs of a different number. This is especially useful since MIDI program changes only allow for 128 different programs to be recalled (0–127). Using the Program Table, you can have any incoming MIDI program change message recall any of the 200 Programs in the QuadraVerb 2. This is done by assigning the MIDI Program Change parameter (described above) to “TABLE”; then select a MIDI program change number and assign it a Program. *Example:* If you programmed the Table so that program message 101 was mapped to Preset 33, and message 102 was mapped to User 39, then sending a MIDI program change message of 101 to the QuadraVerb 2 would recall Program Preset 33, and sending a message of 102 would recall Program User 39.

To incorporate and modify the Program Change Table:

⌘ Press [MIDI].  
The [MIDI] LED will light.

⌘ Press [PAGE >] to access MIDI Page 2.  
The display will read:

MIDI PROGRAM CHANGE: ON

↔ Turn the [VALUE/ENTER] knob to the right to set the MIDI Program Change parameter to “TABLE”.

MIDI PROGRAM CHANGE: TABLE

⌘ Press [PAGE >] to access MIDI Page 3.  
You can edit the program change table, which can substitute alternate Programs (Preset 00-99 and User 00-99) for each of the 128 MIDI program change messages.

PROGRAM TABLE: 000 = 00 User

*f* Turn the [VALUE/ENTER] knob to select a program change number from 000–127 to be re-mapped.

*a* Press [PAGE >] to advance the cursor to the second field in the display.

PROGRAM TABLE: 000 = 00 User

*D* Use the [VALUE/ENTER] knob to select a QuadraVerb 2 Program for the selected MIDI program change message to be re-mapped to (Preset 00–99 or User 00–99).

## SysEx Storage

To send and receive program information via SysEx dumps using a computer, or some other SysEx storage device (such as the Alesis DataDisk), or another QuadraVerb 2:

- ⌘ Connect the other device's MIDI OUT to the QuadraVerb 2's [MIDI IN].
- ⌘ Connect the QuadraVerb 2's [MIDI OUT] to the other device's MIDI IN.  
*This provides two-way communication between the devices.*
- ⌘ Press [MIDI].  
*The [MIDI] LED will light.*
- ÷ Press [< PAGE] to select MIDI Page 6.  
*The display will read:*

SEND MIDI PROGRAM: 00 TO 00

- f Use the [VALUE/ENTER] knob to select a User Program from 00-99, or the current Program being used (EDIT), or ALL User Programs.  
*To send all User Programs, turn the [VALUE/ENTER] knob all the way to the right to select "ALL".*

Optionally, when sending a single Program to another QuadraVerb 2, you can choose to send it to a different location. When you select a Program to be sent, the second field (which represents the destination location) is linked and moves along with it. To select a different location number to send this Program to:

- a Press [PAGE >] and use the [VALUE/ENTER] knob to select a destination location (00-99 or EDIT).  
*If you chose to send "ALL" User Programs, the second field will disappear.*
- D Set the receiving MIDI device to receive or record the MIDI information about to be sent from the QuadraVerb 2.
- « Press the [VALUE/ENTER] button to transmit.  
*If sending "ALL" User Programs, during the dump the display will read:*

MIDI PROGRAM DUMP IN  
PROGRESS...

*If sending a single Program, the display will not change when the [VALUE/ENTER] button is pressed, since this effectively takes no time at all.*

- » When you send a SysEx dump back to the QuadraVerb 2, it will automatically go into receive mode (you do not have to do anything special). When this occurs, the display will momentarily read:

RECEIVING MIDI DATA...

If you don't want the Q2 to receive SysEx dumps, go to MIDI Page 5, and set System Exclusive Receive to OFF.

## MIDI Thru

In order to pass on MIDI information from a control device thru the QuadraVerb 2 to another MIDI device:

- ⌘ Connect the control device's MIDI OUT to the QuadraVerb 2's [MIDI IN].
- ⌘ Connect the QuadraVerb 2's [MIDI OUT] to the MIDI IN of the other device you wish to control.
- ↩ Press [MIDI].  
*The [MIDI] LED will light.*
- ÷ Press [PAGE >] four times to advance to MIDI Page 4.  
*The display will read:*

MIDI THRU: OFF

- f* Turn the [VALUE/ENTER] knob to the right to turn MIDI Thru "ON".

MIDI THRU: ON

## Realtime Modulation Functions

You can use various MIDI messages (such as note number, velocity, aftertouch, or pitch-bend, etc.) to control parameters in the QuadraVerb 2, such as delay time, room size, etc. This is accomplished by way of the QuadraVerb 2's MIDI Modulation Matrix. Modulation parameters are stored in the Program, and therefore can be setup differently for each of the Programs in memory.

Use the [MODULATION] button to program which MIDI messages control which parameters in each Program, and by how much. This is done by following three steps:

- ⌘ Select a Modulator (or modulation patch), of which there are 8 per Program;
- ⌘ Choose the Target; i.e. what is to be modulated;
- ↩ Choose the Source; i.e. what is doing the modulating; and,
- ÷ Set the Amplitude, or amount, of modulation.

Keep in mind that a target may be modulated by more than one source. Likewise, a single source can modulate as many as 8 different targets. It is this kind of flexibility that matrix modulation affords.

**J** *Modulation sources must be received on the same MIDI channel the QuadraVerb 2 is set to in order for the Modulators to function properly.*

## Selecting the Modulator

Once you press the [MODULATION] button, you can turn the [VALUE/ENTER] knob to select one of the eight Modulators. Think of each Modulator as a modulation patch which routes a Source to a Target using a particular amplitude.

- ⌘ Press [MODULATION].  
The [MODULATION] button's LED will light, and the display will read:

```
MOD 1 TO:DLY 1-TAP 1 DELAY
      TIME
```

The cursor appears beneath the Modulator number. This is Modulation Page 1, which allows you to choose a Target (see below).

- ⌘ Turn the [VALUE/ENTER] knob to select a Modulator (1–8).

At any time you can move between the 8 Modulators by first moving the cursor back to the far left of the display (using the [< PAGE >] buttons), and then turning the [VALUE/ENTER] knob to select a different Modulator.

## Choosing a Target

Once you have selected a Modulator, you must then select the Target to be modulated. The Target is broken down into two parts: first, you choose one of the active Blocks in the current Program, then you choose a parameter within the Block.

- ⌘ With [MODULATION] selected (its LED should be lit), press [PAGE >].  
The cursor will advance to the second field in the display:

```
MOD 1 TO:DLY 1-TAP 1 DELAY
      TIME
```

The cursor appears beneath the Modulator Target Block.

- ⌘ Turn the [VALUE/ENTER] knob to select a Block.  
The display will look different depending on each Block's defined type. In the display example above, the first Block is a Delay (DLY). The number (1) tells us this is Block #1. If you turn the [VALUE/ENTER] knob, you will see other options defining the Block's type and number (example: PCH 2, EQ 3, RVB 4, etc.).
- Press [PAGE >] to advance the cursor to the next field in the display.

```
MOD 1 TO:DLY 1-TAP 1 DELAY
      TIME
```

The cursor appears beneath the Modulator Target Parameter.

- ÷ Turn the [VALUE/ENTER] knob to select a Parameter to be modulated in the selected Block.  
The parameters available will of course depend on the selected Block's type.

Here is a list of possible Target Parameters, listed with each effect type:



Type	Effect	Target
EQ	Lowpass Filter	Frequency
	Bandpass Filter	Bandpass Frequency, Bandpass Width
	Highpass Filter	Frequency
	Lowpass Shelf	Frequency, Gain
	Highpass Shelf	Frequency, Gain
	1 Band Low/High Parametric	Bandpass Gain Bandpass Width
	2 Band Sweep Shelf	Lowpass Frequency, Lowpass Gain Highpass Frequency, Highpass Gain
	3 Band Parametric EQ	Lowpass Frequency, Low pass Gain Bandpass Frequency, Bandpass Gain Bandpass Width Highpass Frequency, Highpass Gain
	4 Band Parametric EQ	Lowpass Frequency, Lowpass Gain Low BP Frequency, Low BP Gain Low BP Bandwidth High BP Frequency, High BP Gain High BP Bandwidth Highpass Frequency, Highpass Gain
	5 Band Graphic EQ	63 Hz Band Gain, 250 Hz Band Gain 1KHz Band Gain, 4KHz Band Gain 16KHz Band Gain
	Resonator	Tuning, Decay
Mono/Stereo Tremolo	Speed, Depth, Shape	
Stereo Simulator	Stereo Spread	
Pitch	Mono Chorus	Speed, Depth, Feedback
	Stereo Chorus	Speed, Depth, Feedback
	Quad Chorus	Speed, Depth, Feedback
	Mono Flanging	Speed, Depth, Feedback
	Stereo Flanging	Speed, Depth, Feedback
	Phasor	Speed, Depth
	Mono Lezlie	Speed, Depth, Motor On/Off Speed Slow/Fast, High Rotor Level
	Stereo Lezlie	Speed, Depth, Motor On/Off Speed Slow/Fast, High Rotor Level Stereo Separation
	Pitch Shifter	Pitch Fine, Pitch Coarse
	Pitch Detune	Pitch Detune
	Ring Modulator	Spectrum Shift
Delay	Mono Delay	Delay Time, Feedback
	Stereo Delay	Left Feedback, Left Delay Time
	Ping Pong Delay	Delay Time, Feedback
	Multi Tap Delay	Tap 1–5 Delay Time, Tap 1–5 Level Tap 1–5 Panning, Tap 1–5 Feedback Master Feedback
	Tap Tempo Mono Delay Tap Tempo Ping Pong	Delay Time Feedback

Type	Effect	Target
Reverberation	Mono Room Room 1 Hall 1 Plate 1 Chamber 1 Spring	Input Hi Rolloff Predelay Time, Predelay Mix Decay, Density, Diffusion Damping Low Damping High Gate Trigger On/Off, Gate Hold Time Gate Release Time, Gate Level
	Room 2 Hall 2 Plate 2 Chamber 2	Input Hi Rolloff Predelay Time, Predelay Mix Reflection Level, Reflection Spread Decay, Density, Diffusion Damping Low Damping High Reverberation Attack Gate Trigger On/Off, Gate Hold Time Gate Release Time, Gate Level
	Large Plate Large Room	Predelay Time, Predelay Mix Reflection Level, Reflection Spread Decay, Density, Diffusion Damping Low Damping High Reverberation Swirl Gate Trigger On/Off, Gate Hold Time Gate Release Time, Gate Level
	Nonlinear Reverse	Input Hi Rolloff Predelay Time, Predelay Mix Reverb Time (or Gate Time), Diffusion, Density

### Choosing a Source

Once you have chosen a Modulator and a Target, it is time to select a Source. This is done in Modulation Page 2. If you have been following along so far, let's continue where we left off from the last set of steps:

- i After selecting the Target Block and Parameter, press [PAGE >] to advance to Modulation Page 2.  
*The display will look like this:*

MOD 1 SOURCE: PITCH BEND

*The cursor appears beneath the Modulator, indicating that you can now turn the [VALUE/ENTER] knob to select any of the 8 Modulators.*

- j Press the [PAGE >] button again to advance the cursor to the next field.
- k Use the [VALUE/ENTER] knob to select a Modulation Source.  
*Choose from any of the following Modulation Sources:*

Pitch Bend	Aftertouch	Note Number	Note Velocity
Controllers 000–123	Local Generator 1	Local Generator 2	

### Setting the Amplitude

Once a Modulator's Target and Source have been defined, it is then time to set the Amplitude. This is the amount of modulation control the Source will have over the

Target. This number can be set to 00 (for no effect) all the way to +99 (for maximum effect) or down to -99 (for maximum negative effect; i.e., inverted control). If you have been following along so far, let's continue where we left off from the last set of steps:

- ⌘ After selecting the Target and Source, press [PAGE >] to advance to Modulation Page 3.  
The display will look like this:

MOD 1 AMPLITUDE: +0

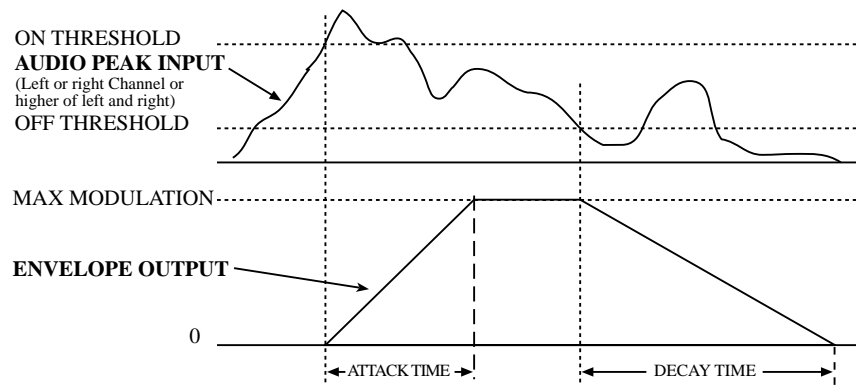
The cursor appears beneath the Modulator, indicating that you can now turn the [VALUE/ENTER] knob to select any of the 8 Modulators.

- ⌘ Press the [PAGE >] button again to advance the cursor to the next field.
- ↵ Use the [VALUE/ENTER] knob to set the Modulator's amplitude from -99 to +99.

## Local Generators

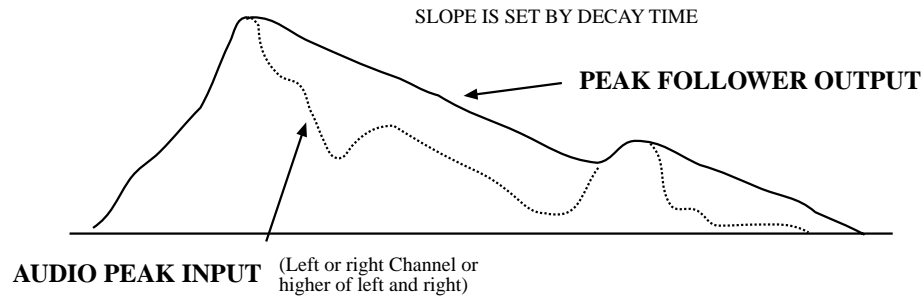
The QuadraVerb 2 is equipped with two Local Generators which can be used as Modulation Sources. Each Local Generator can be set to either one of two types: Input Envelope or Peak Follower. Both allow different ways of having the audio signal at the QuadraVerb 2's inputs modulate the parameters of a Program. Both Local Generators may be triggered by either the Left Input or Right Input, or both Left and Right Inputs. They may also be triggered by the Digital Input, if Optical Data is selected as the Input Audio Source (on Global Page 4).

**Input Envelope.** This uses an envelope, similar to the type found in most synthesizers, which is triggered once the audio signal at the QuadraVerb 2's input(s) exceeds the Attack Threshold you have chosen (-32 to 0dB). Once this happens, the envelope opens up at a rate determined by the Attack Time (which can be set from 0.00–10.00 seconds). Once the input level has dipped down below the Release Threshold (-50 to 0dB), the envelope closes at a rate determined by the Release Time (0.00–10.00 seconds).



**Peak Follower.** The peak follower will follow the audio input to its loudest point and, as the input level drops, the peak follower drops at a constant rate determined by the Decay time (0.20–10.00 seconds). Once the input level exceeds the Peak Follower's level, the Peak Follower is raised to accommodate the higher level, and then (as the input signal again falls) proceeds to fall again using the same Decay time. Since the input levels may be too low at times for the Peak Follower to have ample effect, its Gain may be set to multiply the input's level from x1 to x99.

Peak Follower 1 and 2 have identical features and work as shown below:



Modulation Page 4 allows you to select the type of each Input Generator. The remaining pages in Modulation Mode will appear differently in the display depending on which type of Input Generators you have selected. This may be confusing at times, since both generators could be set to Input Envelope type, or you could have Input Generator 1 set to an Input Envelope type while Input Generator 2 is set to a Peak Follower type. Once you are in Modulation Pages 4 through 7, you can select which of the two Input Generators you want to edit (much the same way you would select a Modulator in Modulation Pages 1 through 3).

To select the Input Generators' types and edit their parameters:

- ⌘ Press [MODULATION].  
*The [MODULATION] button's LED will light.*
- ⌘ Press [PAGE >] seven times (or until Modulation Page 4 is underlined).  
*The display should look like this:*

GEN 1 TYPE: INPUT ENVELOPE

*The cursor appears beneath the Generator, indicating that you may now turn the [VALUE/ENTER] knob and select either Input Generator 1 or 2, depending on which you want to define. In the display example above, Input Generator 1 is defined to be an Input Envelope.*

- Press the [PAGE >] button to advance the cursor to the next field.
- ÷ Turn the [VALUE/ENTER] knob to select either Input Envelope or Peak Follower to be the selected Input Generator's type.
- f Press the [PAGE >] button to advance to Modulation Page 5.  
*If the Generator's type is Input Envelope, the display will look like this:*

GEN 1 ENV TRIGGER SOURCE:  
LEFT

*If the Generator's type is Peak Follower, the display will look like this:*

GEN 1 PEAK FOLLOWER SOURCE:  
LEFT

*The cursor appears beneath the Generator, indicating that you may now turn the [VALUE/ENTER] knob and select either Input Generator 1 or 2, depending on which one you want to edit.*

*You may now select what the trigger source of the selected Input Generator will be.*

- a Press [PAGE >] to advance the cursor to the next field, and turn the [VALUE/ENTER] knob to set the selected Input Generator's trigger source to either LEFT, RIGHT or L&R (both inputs).
- D Press [PAGE >] to advance to Modulation Page 6.  
*If the selected Input Generator's type is Input Envelope, the display will look like this:*

GEN 1 ENV ATK. THRESHOLD:  
+0dB

*If the selected Input Generator's type is Peak Follower, the display will look like this:*

GEN 1 PEAK FOLLOWER GAIN: x 1

*The cursor appears beneath the Generator, indicating that you may now turn the [VALUE/ENTER] knob and select either Input Generator 1 or 2, depending on which one you want to edit.*

- « Press [PAGE >] to advance the cursor to the next field, and turn the [VALUE/ENTER] knob to adjust either the Envelope Attack Threshold (-32 to +0dB) or the Peak Follower's Gain (x1 to x99).
- » Press [PAGE >] to advance to page 7.  
*If the selected Input Generator's type is Input Envelope, the display will look like this:*

GEN 1 ENV REL. THRESHOLD:  
+0dB

*If the selected Input Generator's type is Peak Follower, the display will look like this:*

GEN 1 PEAK FOLW. DECAY:  
0.20Sec

*Again the cursor appears beneath the Generator, indicating that you may now turn the [VALUE/ENTER] knob and select either Input Generator 1 or 2, depending on which one you want to edit.*

- ... Press [PAGE >] to advance the cursor to the next field, and turn the [VALUE/ENTER] knob to adjust either the Envelope Release Threshold (-50 to +0dB) or the Peak Follower's Decay Time (from 0.20 to 10.00 seconds).

If the selected Input Generator's type is input envelope:

- ⑪ Press [PAGE>] to advance to Page 8.

The display will look like this:

GEN 1 ENV ATTACK TIME: 0.00Sec

- ⑫ PRESS [PAGE>} to advance the cursor to the next field and turn the [VALUE/ENTER] knob to adjust the input envelope's attack time (0.00 to 10.00 seconds). This controls the rate that the envelope's control will affect the selected target parameter once it has crossed the attack threshold.

- ⑬ Press [PAGE>} to advance to Page 9.  
If the selected Input Generator's type is Input Envelope, the display will look like this:

GEN 1 ENV RELEASE TIME:  
0.00Sec

- ⑭ Press [PAGE>} to advance the cursor to the next field, and turn the [VALUE/ENTER] knob to adjust the input envelope's release time (0.00 to 10.00 seconds). This controls the rate that the envelope's value will fall after the source level has crossed the release threshold.

## Footswitch Controls

Two footswitch jacks, [ADVANCE] and [BYPASS], are provided on the back panel of the QuadraVerb 2.

### Program Advance

The [ADVANCE] jack allows the QuadraVerb 2's Programs to be advanced remotely from a footswitch. The Program numbers included can be selected by setting the Footswitch Range, found on Global Page 2. The range can be as small or large as you like, including either the Preset or User banks or both. *Example:* If the Footswitch Range is from Preset 89 to User 14, the Programs will continuously cycle from Preset 89 to User 14 whenever the Advance footswitch is triggered.

To set the Footswitch Range:

- ⌘ Press [GLOBAL].  
The [GLOBAL] button's LED will light.

- ⌘ Press [PAGE >} to advance to Global Page 2.  
The display will look like this:

FOOTSWITCH: 00 Pset TO 00 Pset

- ↵ Turn the [VALUE/ENTER] knob to select the Program number to be the beginning point of the Footswitch Range (Preset 00–99 or User 00–99).
- ÷ Press [PAGE >} to advance the cursor to the next field, and use the [VALUE/ENTER] knob to select the Program number that is to be the end point of the Footswitch Range.

When the last Program of the Footswitch Range is reached and the Advance footswitch is pressed again, the first Program in the Footswitch Range will once again be selected.

## Bypassing Effects

The [BYPASS] jack allows you to bypass the effects of the QuadraVerb 2 via a footswitch. This is the same as pressing the [BYPASS] button on the front panel. Each time either the [BYPASS] button is pressed, or the footswitch connected to the [BYPASS] jack is pressed, Bypass mode is toggled on and off again. When Bypass mode is turned on, the [BYPASS] button's LED will be lit.

**J** *When Bypass is enabled, all Effect Blocks are momentarily disabled and will not have audio routed from them to the outputs. However, if the Program does not route the LR Inputs directly to the LR Outputs, you will not hear anything.*

In order for Bypass mode to function correctly, make sure the LR Inputs are routed to the LR Outputs, and the Global Direct Signal Mute function is turned off. If the inputs are not routed directly to the outputs, the [BYPASS] button acts more like a “mute” since nothing will be heard when it is enabled.

## Controlling Delay Time via Tap Tempo

When using either of the two Delay types which allow for “tap tempo” adjustments of delay time, either of the two footswitch jacks may be assigned to provide tap tempo control. This is an alternative way of setting the delay time of a “tap tempo” type delay rather than tapping the [VALUE/ENTER] knob. This can be a great utility for live applications, where the delay time can be adjusted to match your playing tempo simply by tapping your footswitch to the beat of the music.

To select either the [BYPASS] or [ADVANCE] footswitch to control tap tempo:

- ⌘ Press [GLOBAL].  
*The [GLOBAL] button's LED will light.*
- ⌘ Press [PAGE <] twice to move to Global Page 8.  
*The display will read:*

**TAP TEMPO FOOTSWITCH: NONE**

- ↪ Turn the [VALUE/ENTER] knob to select either the ADVANCE or BYPASS footswitch jack, depending on which one you wish to use to control tap tempo.

When either BYPASS or ADVANCE is selected, that footswitch will not perform its normal function until the Tap Tempo Footswitch parameter is set to the other footswitch jack or to “NONE”.

## CHAPTER 7

# ALESIS OPTICAL INTERFACE

## Overview

The Alesis optical digital interface provides the means to communicate with other digital audio devices, such as the Alesis ADAT Multitrack Recorder. The Alesis optical interface uses a proprietary 8 channel buss, from which the QuadraVerb 2 can provide processing for 2 channels. This means you can come into the QuadraVerb 2's digital input from one or two tracks of an ADAT, and go out of the QuadraVerb 2's digital out to one or two ADAT tracks.

Alternatively, you may wish to use the QuadraVerb 2's analog inputs to process an instrument and record digitally direct to ADAT. This is possible because while the QuadraVerb 2's inputs are set to either analog or digital, both its analog and digital outputs are always active. However, you must determine which 2 channels (of the digital buss' 8 channels) to assign the QuadraVerb 2's digital outputs to.

Going digital is the preferred method of recording if you use the QuadraVerb 2 with an ADAT. Since the DSP effects are created in the digital realm, why not let them remain there while being transferred to a tape medium? Also, if you connect the QuadraVerb 2's digital output to an Alesis AI-1, you can route the digital audio to other devices using either the AES/EBU or S/PDIF standards.

This chapter is designed to illustrate the precise ways of connecting the QuadraVerb 2 to various digital audio peripherals, and how to manipulate the QuadraVerb 2's digital routings. It is divided into two parts. The first half of this chapter details the connections to various digital audio sources and destinations. The second half talks about the routing possibilities that are available on the QuadraVerb 2 for processing digital audio sources and transmitting to digital audio recorders.

## Digital Clock Synchronization

Digital signals are self-clocking; they carry a word clock along with them. Usually the receiving unit automatically adjusts its sample clock to follow the rate of the incoming digital signal. When the receiving unit must synchronize to another clock--for example, an ADAT synchronized to SMPTE time code--a conflict may occur between the clocks that will result in audible "clicks" in the signal. For this reason, in more complex setups, the QuadraVerb 2 features a 48 kHz IN jack. This is to synchronize the sample clock of the QuadraVerb 2's analog-to-digital converter to the BRC or AI-2. If you're using the QuadraVerb 2 only to process signals that come via the optical input from the ADATs, this connection is not necessary because the optical clock coming in is already indirectly synchronized by the BRC. The 48 kHz connection IS necessary if the QuadraVerb 2 is the originator of the optical for the rest of the system, from its analog inputs, while the rest of the system is seeing the BRC as the master.

A complete diagram of every single possible setup with every possible input would be needlessly complex. Just remember that there can only be one master clock. If the BRC is the master, the QuadraVerb 2 must be synchronized to its clock via the optical or the 48 kHz in. If you have a single ADAT with no BRC, the QuadraVerb 2 becomes the master clock source whenever the ADAT is set to DIGITAL IN and the QuadraVerb 2's Input Audio Source is set to ANALOG.



**J** *Note: If you experience problems (clicks, etc.) with an ADAT system set to digital input, it may be that the ADAT's clock source is set to internal, even though its Digital In is pressed. (This will only happen if you've set it this way for making tape copies) Hold SET LOCATE and press DIGITAL IN to toggle the clock to digital.*

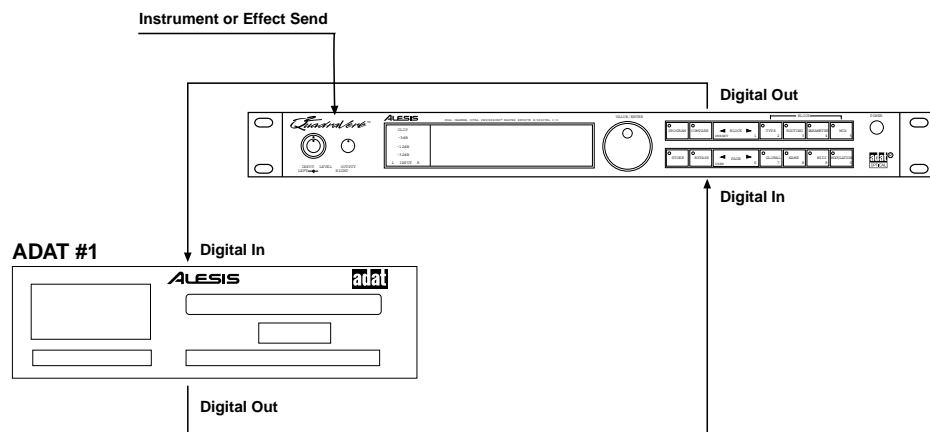
**Note that the front panel level controls have no effect on a digital signal.** The [INPUT LEVEL] controls will affect an analog input being converted to digital, and the [OUTPUT LEVEL] control affects the analog outputs only. If you need to change the levels in all-digital mode, you must do so by editing the program using page 2 of the Routing function.

**When the input is set to Digital, the meters will show only the level of the signals on Channels 1 and 2 of the ADAT Optical interface, regardless of which channels are selected by the QuadraVerb 2.** However, internal clipping within a program will still be shown by the Clip LEDs.

## Connections

### To a Single ADAT

Connection between the QuadraVerb 2 and one or more ADATs is a simple task of creating a digital audio network that loops all the devices together. When using a single ADAT, simply connect two fiber optic cables between the devices to provide two-way communication.



To connect the QuadraVerb 2 to a single ADAT:

- i Connect an EIAJ fiber optic cable (such as Alesis OC, OA, or O1) from the [DIGITAL OUT] of the QuadraVerb 2 to the DIGITAL IN of the ADAT.  
*To test the cable and QuadraVerb 2 digital output, plug one cable end into the QuadraVerb 2. The other end should emit a soft red light (it is not dangerous to look directly at this light).*
- j Connect another fiber optic cable from the QuadraVerb 2's [DIGITAL IN] to the DIGITAL OUT of the ADAT.

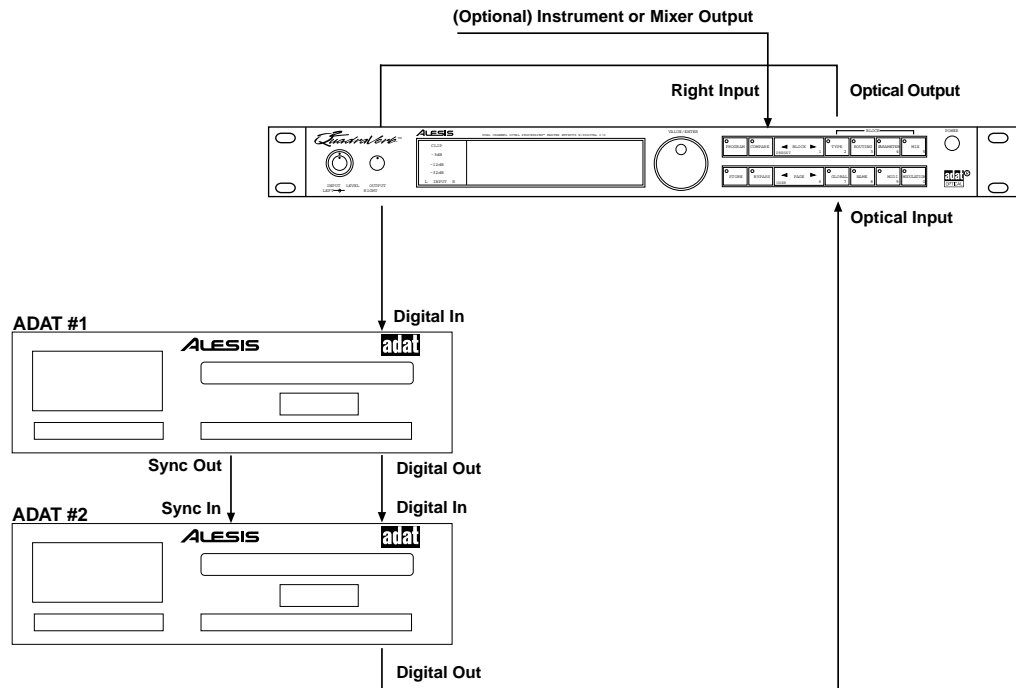
### To Two or More ADATs

If using two or more ADATs, they must be connected together so that they are in sync with each other. This is accomplished by connecting each ADAT's SYNC OUT connector to the SYNC IN connector of the next ADAT in the chain, as described in the ADAT Reference Manual.

The fiber optic cables that make up the digital buss need to be connected in the same order as the Sync cables. The QuadraVerb 2 is inserted in the fiber optic loop from the output of the second ADAT to the input of the master ADAT. The QuadraVerb 2 lets you turn its digital output into a “throughput” of the digital input. This allows you to move audio through the QuadraVerb 2, even when you don’t want to use it to process anything, without having to repatch your fiber optic cables. More information about digital audio routing can be found in the next section.

To connect the QuadraVerb 2 to two ADATs:

- ¿ Connect an EIAJ fiber optic cable (available from your dealer) from the [DIGITAL OUT] of the QuadraVerb 2 to the [DIGITAL IN] of the first ADAT.
- ¡ Connect another fiber optic cable from the first ADAT’s [DIGITAL OUT] to the second ADAT’s [DIGITAL IN].
- ↪ Connect another fiber optic cable from the second ADAT’s [DIGITAL OUT] to the QuadraVerb 2’s [DIGITAL IN].
- ÷ Connect a dual male, 9 pin D cable from the first ADAT’s [SYNC OUT] to the second ADAT’s [SYNC IN].



To Two or More ADATs With a BRC or AI-2

If your ADAT system has an Alesis BRC Master Remote Controller, the fiber optic cables that make up the digital buss need to be connected in the same order as the Sync cables. If your ADAT system uses the BRC or the Alesis AI-2 Synchronizer, the QuadraVerb 2's digital clock must be synchronized to the clock coming from the BRC or AI-2. Connect a BNC-to-BNC cable (such as the Alesis BN cable) between the BRC's 48 kHz CLOCK OUT and the QuadraVerb 2's 48 kHz CLOCK IN.

**J** *If only using one ADAT without the BRC or AI-2, it is not necessary to connect the 48 kHz Clock.*

In Global mode ([GLOBAL] button's LED lit), page 5 lets you select the Sample Clock Source. This can either be Internal or Sample Clock. When set to Internal, the QuadraVerb 2 uses its own internal sample clock as a reference for its DSP. However, if you are recording the QuadraVerb 2's digital output to ADAT using the BRC Master Remote Controller, the QuadraVerb 2 must receive a 48 kHz clock signal from the BRC in order to maintain perfect sync with the ADAT system. This requires that you connect a BNC-to-BNC cable between the BRC's 48 kHz Clock Out to the QuadraVerb 2's 48 kHz Clock In. When you are ready to record onto ADAT from the QuadraVerb 2, be sure to set the QuadraVerb 2's Sample Clock Source function to Sample Clock.

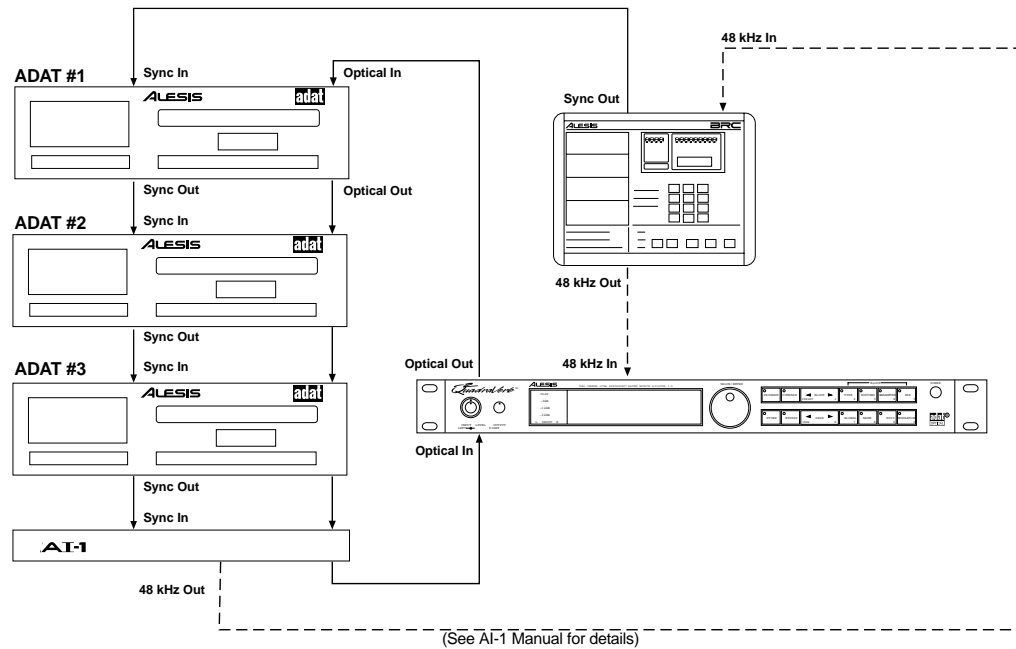
To select the Sample Clock Source:

- ⌘ Press [GLOBAL].  
*The [GLOBAL] button's LED will light.*
- j Press [PAGE >] 5 times, so that Page 5 is underlined in the display.  
*The display will read:*



**J** *If the Input Audio Source is set to Optical Data, then the Sample Clock Source will be set to Optical and cannot be changed.*

- ↵ Turn the [VALUE/ENTER] knob to select either "INTERNAL" or "48 kHz IN".



## To the AI-1

If you wish to route the digital output to a device that uses AES/EBU or S/PDIF standards, you can connect the QuadraVerb 2 to the Alesis AI-1. This can be done directly, by connecting the QuadraVerb 2's [DIGITAL OUT] to the AI-1's DIGITAL IN and the QuadraVerb 2's [DIGITAL IN] connected to the AI-1's DIGITAL OUT. Alternatively, if you are using multiple ADATs and the BRC (as described above), you can simply insert the AI-1 into the fiber optic chain, either before or after the QuadraVerb 2.

In the BRC/ADAT system illustration above, the AI-1 is inserted between the last ADAT in the chain and the QuadraVerb 2. In this scenario, you can route the QuadraVerb 2's digital output to any ADAT in the chain, or the AI-1 (for routing to another digital audio peripheral using either AES/EBU or S/PDIF standards). You can also route audio to the digital inputs of the QuadraVerb 2 coming from any ADAT in the chain or the AI-1 (for processing digital audio from some other digital audio peripheral). More information about digital audio routing can be found in the next section.

## From the QuadraSynth

The Alesis QuadraSynth and S4 sound module both have Alesis Optical digital outputs. These devices provide their four analog audio outputs on the first four channels of the 8 channel digital buss. Simply connect a single fiber optic cable between the QuadraSynth's DIGITAL OUT connector and the QuadraVerb 2's [DIGITAL IN] connector. If the ADATs are being controlled by a BRC or AI-2, the QuadraSynth's clock must be controlled by the 48 kHz OUT of the BRC or AI-2. Connect a BNC cable from the BRC or AI-2 to the QuadraSynth's 48 kHz IN jack.

## Routings

### To specific ADAT tracks

To route the QuadraVerb 2's digital outputs to specific ADAT tracks depends on which of the 8 channels of the buss you use. If you want to record in stereo onto tracks 1 and 2 of ADAT (or 9 and 10 on ADAT #2 in a chain), route the QuadraVerb 2's Left output to channel 1 and its Right output to channel 2. Likewise, if you were recording onto tracks 5 and 6, then it would be channels 5 and 6 that you would use.

To route the QuadraVerb 2's Left and Right outputs to the digital buss:

⌘ Press [GLOBAL].  
*The [GLOBAL] LED will light.*

⌘ Press [PAGE >] six times to advance to Global Page 6.  
*The display will read:*

```
DIG CH IN: L=1,R=2 OUT: L=1,R=2
```

⌘ Press the [PAGE >] button two more times to advance the cursor to the third field in the display.

```
DIG CH IN: L=1,R=2 OUT: L=1,R=2
```

÷ Use the [VALUE/ENTER] knob to select an output on the digital buss for the Left channel to be routed to.  
*You can select any of the 8 channels, except for the channel the Right output is assigned to (you cannot route both the Left and Right outputs to the same channel of the digital buss).*

f Press the [PAGE >] button to advance the cursor to the next field.

```
DIG CH IN: L=1,R=2 OUT: L=1,R=2
```

a Use the [VALUE/ENTER] knob to select an output on the digital buss for the Right channel to be routed to.  
*You can select any of the 8 channels, except for the channel the Left output is assigned to.*

D Record-enable those same track numbers (in this case, 1 and 2) on the connected ADAT.  
*You will be able to monitor the QuadraVerb 2's output on those tracks of the connected ADAT.*

## From ADAT through the QuadraVerb 2 back to ADAT

When routing ADAT tracks to the QuadraVerb 2 to be recorded back to ADAT, how you go about it depends on whether or not you are using a BRC. If you have one or more ADATs with no BRC, then you simply connect the digital buss of the ADATs and QuadraVerb 2 in a loop (described earlier in this chapter).

With the BRC, you can select up to eight tracks of one ADAT to be output on the digital buss (which connects to all ADATs and the QuadraVerb 2 as described earlier) for track bouncing. In this situation, the BRC determines (by choosing your digital source tracks) which ADAT in the chain is transmitting. Only one ADAT can transmit on the digital buss at any single moment. All other ADATs are put into a “digital thru” mode, whereby they can receive all eight channels (even record them) and pass them along to their digital outputs without adding anything.

Although the BRC does not determine which tracks are going to the QuadraVerb 2, it does determine which ADAT is transmitting. Therefore, even if you do not intend to route the same tracks to the digital buss for bouncing (it is unlikely that you would want to do this at the same time, anyway), you must use the BRC to choose at least one of the tracks on the ADAT you wish to process audio in the QuadraVerb 2. The QuadraVerb 2 then intercepts any two of the 8 channels coming from the transmitting ADAT, and then processes them and routes them to the same or different channel numbers, to be recorded back onto another ADAT in the beginning of the loop.

This means that if you connect QuadraVerb 2 to the digital buss between the last and first ADAT in a multi-unit/BRC system, you can process any two channels; BUT you can record the QuadraVerb 2's output only to the same ADAT that is transmitting or any ADAT that comes before it in the chain. This is because the transmitting ADAT does not route the digital audio it receives “thru” to its output.

*Example:* Let's say you have four ADATs and a BRC. If you were to select tracks 9 and 10 (tracks 1 and 2 on ADAT #2) as your source tracks for digital bouncing on the BRC, you thereby designate ADAT #2 as the transmitter on the digital buss. QuadraVerb 2 can now intercept any of that ADAT's 8 channels, process them, and output them back onto two channels, which connects back to the first ADAT in the chain. The QuadraVerb 2's output could therefore be recorded onto ADAT #1 and #2, but not ADAT #3. This is because ADAT #2 is not routing its digital input thru to its output, because it is routing its tape signal instead. If you wanted to record the QuadraVerb 2's digital output onto ADAT #3, the easiest solution is to swap tapes between ADAT #1 and #3.

## From the QuadraSynth through the QuadraVerb 2 back to ADAT

The QuadraSynth (and S4 sound module) provides its four audio outputs (Main Left/Right and Aux Left/Right) on the first four channels of the Alesis Optical digital buss. If you connect the QuadraSynth's digital output to the digital input of the QuadraVerb 2, you would select channels 1 and 2 if you wanted to process the two Main outputs, or channels 3 and 4 if you wanted to process the Aux channels.

If you are using the QuadraSynth along with an ADAT system and the BRC or an AI-2, you will need to supply the QuadraSynth with a 48 kHz Clock source from the BRC or AI-2 (see earlier in the chapter for more information about the 48 kHz Clock). The QuadraVerb 2 does not need to receive a 48 kHz Clock from the BRC or AI-2, since it will be synchronizing to the QuadraSynth's clock coming in on its digital inputs, which in turn will be synchronized to the 48 kHz clock coming from the BRC or AI-2. This is extremely important to make sure that all the digital devices are synchronized to the same sample clock.

CHAPTER 8

# TROUBLE-SHOOTING

## Trouble-Shooting Index

If you are experience problems while operating the QuadraVerb 2, please use the following table to locate possible causes and solutions before contacting Alesis customer service for assistance.

Symptom	Cause	Solution
The display does not light when the ON/OFF switch is turned on.	No power.	Check that the power cable is plugged in properly.
Display is blank.	Display contrast is set either too high or low.	Adjust the display contrast. Press [GLOBAL], turn [VALUE/ENTER].
No audio is heard.	The digital input is selected.	Press [GLOBAL], press [PAGE >] four times, then turn [VALUE/ENTER] knob counterclockwise.
	Bypass is on with no direct routings set.	Select another program with routings.
	Bypass is on with Global Direct Signal Mute On.	Go to GLOBAL page 9 and set it to OFF.
	Bypass is on, with Direct Level From Input set to NONE.	Go to MIX and set Direct Level to 100% .
	Program being edited has no input or output routings.	Go to ROUTING page 3 (Delete Route) and check the existence of routings for each block and the L/R OUT by rotating the [VALUE] control to see all inputs to that block.
Effect output level is low on a single Program.	Master Effects Level is below 100%.	Press MIX and raise the level.
Clicks are heard in the audio using digital I/O.	Digital clock conflict.	Provide a master 48 kHz clock or repatch the system. Set the receiving unit's clock to external.
	ADAT clock set improperly.	Hold SET LOCATE and press DIGITAL IN until the display says "DIG".
Distortion (red LEDs flash before meter is full).	Internal DSP is clipping.	Lower the LEVEL of any rate feeding a block that is overloading.
Unit does not respond to front panel controls.	Unknown software conflict, cosmic rays, or static electricity.	Disconnect MIDI input. Power down and power up. If this doesn't work, try reinitialization, then total reset (see later in this chapter).



MIDI Program Changes are off by 1.	Transmitter doesn't display true MIDI Program Change numbers.	Change the transmitter to display 0-127, or setup a Program Table in the Q2.
------------------------------------	---	--

## Error Messages

Although the digital signal processing power in the QuadraVerb 2 is powerful and features plenty of memory, the programmability of the unit makes it possible to come across various error messages when exceeding its limitations. This is the QuadraVerb 2's limit handling software kicking in. Depending on the message you get, you may have to modify your Program by altering or removing unnecessary Blocks in order to complete the task. The error message will vary depending on what limitations you are attempting to exceed.

If you are using two reverb Blocks and you try to create a third, the following message will appear:

DSP IS FULL

This means there is not enough processing power available to create the additional effect. But there may be enough available to add an EQ, two pitch effects and two delays (or some other combination).

If you have created four pitch Blocks and you try to create a fifth, the following message will appear:

OUT OF LFOs

This means that there are no more available LFOs (low frequency oscillators), which are required by all pitch effects. However, the larger reverb effects (Plate 2, Room 2, Chamber 2 and Hall 2) use 1 LFO each. This means if you had three Pitch Blocks and a reverb Block using one of the larger reverbs, you would not be able to add a fourth Pitch Block. Also, the Pitch type "Mono Lezlie" uses two LFOs, which means that when using it you will only have two LFOs available.

While the QuadraVerb 2 has a custom DSP chip that processes all the effects, some of the effects available require that the main microprocessor assist in handling some of this responsibility. On occasion, you may be using enough effects simultaneously that no more processing power is available from the main microprocessor; *example*: when you try to create more than three Multi Tap Delays in a Program. In these cases, the following message will appear:

BLOCK COMBINATION NOT  
ALLOWED

Finally, if a Delay Block has its delay time to 5 seconds (5,000.0 ms), and you try to create another Delay Block and set its delay time to 460 ms or higher, you will get a "Effect Memory is Full" message. There is actually 5,455.9 ms of delay memory available. Although you could create 8 Delay Blocks, they all share the available delay memory. Therefore, when setting the delay time of each Delay Block you would only have available the amount of delay memory not being used by the other Delay Blocks. When all memory is being used, the following message will appear:

EFFECT MEMORY IS FULL

## Checking the Software Version

The current software version may be determined by holding [GLOBAL] and pressing [PROGRAM]. The display will indicate the currently installed software version.

## Re-initializing

To re-initialize the QuadraVerb 2, hold down both [PROGRAM] and [COMPARE] while turning on the power. This will reset all Global parameters to their default settings, and will initialize the edit buffer so that the current Program is set to its default settings (no Blocks/routings). However, none of the Programs in the User bank or the Preset bank are changed when re-initializing the unit this way.

## Total Reset

A more drastic version of re-initializing the QuadraVerb 2 will erase all User Programs, as well as resetting all Global parameters. Hold down [PROGRAM], [COMPARE], and [<BLOCK] simultaneously while turning on the power.

## Maintenance/Service

### Cleaning

Disconnect the AC cord, then use a damp cloth to clean the QuadraVerb 2's metal and plastic surfaces. For heavy dirt, use a non-abrasive household cleaner such as Formula 409 or Fantastik. **DO NOT SPRAY THE CLEANER DIRECTLY ONTO THE FRONT OF THE UNIT AS IT MAY DESTROY THE LUBRICANTS USED IN THE SWITCHES AND CONTROLS!** Spray onto a cloth, then use cloth to clean the unit.

### Warranty Information

This product is warranted by Alesis to the original purchaser against defects in material and workmanship for a period of 1 year for parts and 90 days for labor from the date of purchase. Complete terms of the Limited Warranty are stated on the Warranty Card packed with the product. Please retain a copy of your dated sales receipt for proof of warranty status should repairs be necessary. The warranty is valid in the country of purchase only.

### Refer All Servicing to Alesis

We believe that the QuadraVerb 2 is one of the most reliable multi-effects processors that can be made using current technology, and should provide years of trouble-free use. However, should problems occur, **DO NOT** attempt to service the unit yourself. High voltage/high current DC voltages are present at several points within the chassis. Service on this product should be performed only by qualified technicians. **NO USER-SERVICEABLE PARTS INSIDE.**

### Obtaining Repair Service

Before contacting Alesis, check over all your connections, and make sure you've read the manual.

**Customers in the USA:** If the problem persists, call Alesis USA at 1-800-5-ALESIS and request the Product Support department. Talk the problem over with one of our technicians; if necessary, you will be given a return authorization (RA) number and instructions on how to return the unit. All units must be shipped prepaid and COD shipments will not be accepted.

For prompt service, indicate the RA number on the shipping label. If you do not have the original packing, ship the QuadraVerb 2 in a sturdy carton, with shock-absorbing materials such as styrofoam pellets (the kind without CFCs) or “bubble-pack” surrounding the unit. Shipping damage caused by inadequate packing is not covered by the Alesis warranty.

Tape a note to the top of the unit describing the problem, include your name and a phone number where Alesis can contact you if necessary, as well as instructions on where you want the product returned. Alesis will pay for standard one-way shipping back to you on any repair covered under the terms of this warranty. Next day service is available for a surcharge.

Field repairs are not normally authorized during the warranty period, and repair attempts by unqualified personnel may invalidate the warranty.

Service address for customers in the USA:

Alesis Product Support  
3630 Holdrege Avenue  
Los Angeles, CA 90016

**Customers outside the USA:** Contact your local Alesis dealer for warranty assistance. Do not return products to the factory unless you have been given specific instructions to do so. Note that the warranty is valid only in the country of origin.

# MIDI IMPLEMENTATION CHART

Function		Transmitted	Recognized	Remarks
Basic Channel	Default Changed	X * * * * *	1 — 16	
Mode	Default Messages Altered	X X * * * * *	Mode 3 X	
Note Number	True Voice	X * * * * *	0 — 127 0 — 127	
Velocity	Note On Note Off	X X	O X	
After Touch	Key's Ch's	X X	O O	
Pitch Bender		X	O	
Control Change	0 — 123	X	O	
Prog Change	True #	X * * * * *	O 0 — 127 0 — 99 Preset, 00-27 User	
System Exclusive		O	O	
System Common	Song Pos Song Sel Tune	X X X	X X X	
System Realtime	Clock Commands	X X	X X	
Aux Messages	Local On/Off All Notes Off Active Sense Reset	X X X X	X X X X	
Notes				

Mode 1: OMNI ON, POLY  
Mode 2: OMNI ON, MONO

Mode 3: OMNI OFF, POLY  
Mode 4: OMNI OFF, MONO

O : Yes  
X : No

# SPECIFICATIONS

## Electrical

Frequency Response:	±1dB from 20Hz to 20kHz
Dynamic Range:	>90 dB "A" wtg., 20 Hz-22 kHz
Distortion (THD+N):	<0.009% @ 1kHz, nominal level (-12 dBfs) <0.005% @ peak level
Crosstalk:	<90 dB below full scale
Power requirements:	9VAC, 13.5W (Alesis P4).

## Physical

Dimensions:	19" x 1.75" x 6" (single rack space)
Weight:	4.2 lbs.

## Inputs

Number of Channels:	2
Format:	1/4" TRS 3-conductor jacks, balanced/unbalanced
Nominal Level:	-20 dBV to +4 dBu, front-panel adjustable
Maximum Input Level:	+19dBu
Input Impedance (nominal):	500K unbalanced, 20K balanced

## A/D - D/A Conversions

A/D converter:	16 bit Sigma-Delta, 128 times oversampling
D/A converter:	18 bit Delta-Sigma, 64 times oversampling
Sampling Frequency:	48kHz (variable from 40.4 kHz to 50.8 kHz under external control)

## Outputs

Number of Channels:	2
Format:	1/4" TRS 3-conductor jacks, servo balanced/unbalanced
Maximum Level:	+17.5dBu
Nominal Level:	-20 dBV to +4 dBu, front-panel adjustable
Output Impedance (nominal):	500 ohms

## Digital Input/Output

Input and Output Type:	ADAT MultiChannel Optical Digital Interface (AES/EBU and S/PDIF formats available using optional AI-1 Digital Interface and Sample Rate Converter)
48 kHz In:	BNC jack for word clock connection to BRC, AI-2, or AI-1
Number of Channels:	2 (selectable to/from any of the 8 ADAT Optical channels)
Connectors:	EIAJ fiber optic jacks

## Processing and Memory

User Programs (RAM):	100
----------------------	-----

Factory Preset Programs (ROM):	100
Internal processing resolution:	24 bit accumulator
Delay memory:	5 seconds
EQ effects:	Lowpass Nonshelving, Bandpass Nonshelving, Highpass Nonshelving, Lowpass Shelving, Bandpass Shelving Low & High, Highpass Shelving, 2-Band Sweepable, 3-Band Parametric, 4-Band Parametric, 5-Band Graphic, 1-Band Resonator, Mono Tremolo, Stereo Tremolo, Pseudo Stereo Expander
Pitch effects:	Mono Chorus, Stereo Chorus, Quad Chorus, Mono Flanging, Stereo Flanging, Phasor, Mono Lezlie, Stereo Lezlie, Pitch Shift, Pitch Detune, Ring Modulator
Delays:	Mono Delay, Stereo Delay, Ping Pong Delay, Multi Tap Delay, Tap Tempo Mono Delay, Tap Tempo Ping Pong Delay
Reverbs:	Mono Room, Room 1, Hall 1, Plate 1, Chamber 1, Room 2, Hall 2, Plate 2, Chamber 2, Large Plate, Large Room, Spring, Nonlinear, Reverse

## Front Panel

Controls	Input Level (Left/Right) Output														
Keypad	<table> <tr> <td>Program</td> <td>Store</td> </tr> <tr> <td>Compare</td> <td>Bypass</td> </tr> <tr> <td>&lt; Block &gt;</td> <td>&lt; Page &gt;</td> </tr> <tr> <td>Type</td> <td>Global</td> </tr> <tr> <td>Routing</td> <td>Name</td> </tr> <tr> <td>Parameter</td> <td>MIDI</td> </tr> <tr> <td>Mix</td> <td>Modulation</td> </tr> </table>	Program	Store	Compare	Bypass	< Block >	< Page >	Type	Global	Routing	Name	Parameter	MIDI	Mix	Modulation
Program	Store														
Compare	Bypass														
< Block >	< Page >														
Type	Global														
Routing	Name														
Parameter	MIDI														
Mix	Modulation														
Indicators	Dual, 4-stage LED VU meters Custom LCD display														

## Rear Panel

Input (Left and Right)	1/4" 3-conductor
Output (Left and Right)	1/4" 3-conductor
Bypass	1/4" (accepts normally open or normally closed momentary switches, such as the Alesis PD)
Advance	1/4" (as above)
Digital (In and Out)	EIAJ fiber optic cable jacks (Toslink connectors)
MIDI (In and Out/Thru)	5 pin DIN
Power	4 pin Barrel

# EFFECTS PROCESSING INDEX

Function	Effect	% of DSP	% of Memory	LFOs
EQ	Lowpass Filter	6	1	0
	Bandpass Filter	15	1	0
	Highpass Filter	6	1	0

	Lowpass Shelf	8	1	0
	Highpass Shelf	8	1	0
	1 Band Low Parametric	15	1	0
	1 Band High Parametric	7	1	0
	2 Band Sweep Shelf	15	1	0
	3 Band Parametric	30	1	0
	4 Band Parametric	35	1	0
	5 Band Graphic	34	1	0
	Resonator	3	1	0
	Mono Tremolo	3	1	1
	Stereo Tremolo	7	1	1
	Stereo Simulator	14	1	0
Pitch	Mono Chorus	4	1	1
	Stereo Chorus	8	1	1
	Quad Chorus	20	1	1
	Mono Flanging	3	1	1
	Stereo Flanging	8	1	1
	Phasor <sup>1</sup>	17	1	0
	Mono Lezlie	10	1	2
	Stereo Lezlie <sup>1</sup>	16	1	1
	Pitch Shift	7	1	1
	Pitch Detune	7	1	1
	Ring Modulator <sup>1</sup>	15	1	0
	Delay	Mono Delay <sup>2</sup>	2	3
Stereo Delay		4	6	0
Ping Pong Delay <sup>2</sup>		3	6	0
Multi Tap Delay <sup>3</sup>		15	14	0
Reverb	Mono Room	28	11	0
	Room 1	33	11	0
	Hall 1	33	11	0
	Plate 1	33	11	0
	Chamber 1	40	11	0
	Room 2	76	11	0
	Hall 2	61	11	1
	Plate 2	64	11	1
	Chamber 2	64	11	1
	Large Plate	67	11	1
	Large Room	91	11	1
	Spring	41	11	0
	Nonlinear	38	–	0
	Reverse	38	–	0

<sup>1</sup>Only one per Program.

<sup>2</sup>Incl. Tap Tempo versions

<sup>3</sup>Max. 3 per Program