Reference Manual

DP/2

parallel effects processor

ENSONIQ™

LEADING THE WORLD IN SOUND INNOVATION

READ THIS FIRST!

WARNING!

Grounding Instructions

This product must be grounded. If it should malfunction or break down, grounding provides a path of least resistance for electric current to reduce the risk of electric shock. This product is equipped with a cord having an equipment-grounding conductor and a grounding plug. The plug must be plugged into an appropriate outlet that is properly installed and grounded in accordance with all local codes and ordinances.

DANGER: Improper connection of the equipment-grounding conductor can result in the risk of electric shock. Check with a qualified electrician or service personnel if you are in doubt as to whether the product is properly grounded. Do not modify the plug provided with this product — if it will not fit the outlet, have a proper outlet installed by a qualified electrician.

CAUTION

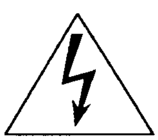
RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION : TO REDUCE THE DANGER OF ELECTRIC SHOCK

DO NOT REMOVE COVER (OR BACK)

NO USER SERVICEABLE PARTS INSIDE

REFER SERVICING TO QUALIFIED SERVICE PERSONNEL



This symbol is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electronic shock to persons.

This symbol is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

SEE IMPORTANT SAFETY INSTRUCTIONS ON BACK COVER!

ENSONIQ™

parallel effects processor DP/2

Reference Manual

Version 1.0

DP/2 Reference Manual:

Written, designed, and illustrated by: Tom Tracy, Bill Whipple, Jon Dattorro, John Senior

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USA

Printed in U.S.A.

Please record the following information:

Your Authorized ENSONIQ Dealer:________________________ Phone:______________

Your Dealer Sales Representative:____

Serial Number of Unit:________________________ Date of Purchase:______________

Your Authorized ENSONIQ Dealer is your primary source for service and support. The above information will be helpful in communicating with your Authorized ENSONIQ Dealer, and provide necessary information should you need to contact ENSONIQ Customer Service. If you have any questions concerning the use of this unit, please contact your Authorized ENSONIQ Dealer first. For additional technical support, or to find the name of the nearest Authorized ENSONIQ Repair Station, call ENSONIQ Customer Service at (610) 647-3930 Monday through Friday 9:30 AM to 12:15 PM and 1:15 PM to 6:30 PM Eastern Time. Between 1:15 PM and 5:00 PM we experience our heaviest call load. During these times, there may be delays in answering your call.

Although every effort has been made to ensure the accuracy of the text and illustrations in this manual, no guarantee is made or implied in this regard.

IMPORTANT:

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

* reorient the receiving antenna
* relocate the instrument with respect to the receiver
* move the instrument away from the receiver
* plug the instrument into a different outlet so that the instrument and receiver are on different branch circuits

"If necessary, the user should consult the dealer or an experienced radio/television technician for additional suggestions. The user may find the following booklet prepared by the Federal Communications Commission helpful: 'How to Identify and Resolve Radio-TV Interference Problems.' This booklet is available from the U.S. Government Printing Office, Washington, D.C. 20402. Stock No. 004-000-00345-4."

CAUTION! Danger of explosion if battery is incorrectly replaced. Replace only with the same or equivalent type recommended by the manufacturer. Discard used batteries according to manufacturer's instructions.

In order to fulfill warranty requirements, the DP/2 should be serviced only by an Authorized ENSONIQ Repair Station. The ENSONIQ serial number label must appear on the outside of the unit, or the ENSONIQ warranty is void.

- ENSONIQ, DP/2, DP/4, DP/4+, TS-10, TS-12, and ASR-10 are trademarks of ENSONIQ Corp.

Part # 9310 0189 01 - C **Model # MM-88**

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Welcome!

Congratulations, and thank you for purchasing the ENSONIQ DP/2 Parallel Effects Processor. The DP/2 creates 24-bit digital effects using two independent processors, and features two independent inputs and outputs with full internal mixing capabilities and discrete stereo processing. The DP/2 is equally at home in a professional recording studio, home studio, guitar rig, MIDI setup, or PA system.

The Effects

The ENSONIQ DP/2 Parallel Effects Processor has over 60 high-fidelity fully programmable digital effect algorithms. Reverb, chorusing, flanging, delay, distortion, pitch shifting and an assortment of other effects are provided with dynamic control over most of the settings. There are 600 effect presets: 300 permanent in ROM (Read Only Memory) and 300 additional presets in RAM (Random Access Memory) for you to edit or store your own creations.

Parallel Processing

The DP/2's two-in, two-out design permits stereo processing of two parallel channels (multi-processing). There is only one user interface, but up to two different input signals can each go to a separate internal signal processor. Independently configurabe inputs and outputs also allow for special types of effects, like keyed expansion and ducking.

The DP/2 can be used as one big effects box, or two separate effects boxes. The routing between the two processing units is programmable, allowing for a serial or parallel combination of effects. The DP/2 also offers paths to feedback the signal, and side-chain capability. The variable architecture and rich assortment of algorithms provides for unusual effect structures not found in fixed routing systems.

The DP/2 is equipped with an advanced digital signal processing system based on the ENSONIQ Signal Processor (ESP) chip. The ESP chip is designed specifically for digital audio signal processing, and in the DP/2, two ESP chips work in conjunction with 16-bit analog-to-digital and digital-to-analog converters to provide a studio-quality output signal.

The digital effects processing capability has been designed to complement any input source (balanced/unbalanced; +4dBu to-10dBV), and all of the algorithms (except the Guitar Tuner) can have specific parameters modulated by various MIDI and non-MIDI controllers such as a keyboard's pitch wheel, or the local CV Pedal.

Clean Up and Maintenance

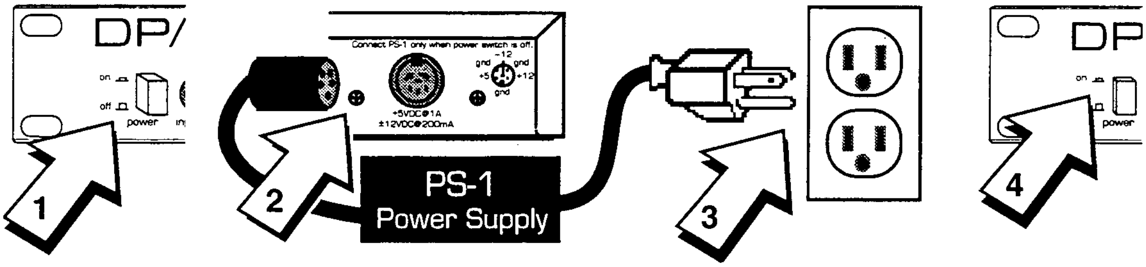
Clean the exterior of your DP/2 with a soft, lint-free, dry (or slightly damp) cloth. You can use a slightly dampened cloth (with a mild neutral detergent) to remove stubborn dirt, but make sure that the DP/2 is thoroughly dry before turning on the power. Never use alcohol, benzene, volatile cleaners, solvents, abrasives, polish or rubbing compounds.

The Manuals

The DP/2 comes with two important publications: the DP/2 User's Guide—designed to help you understand the way the DP/2 operates (in a tutorial fashion)—and the more technical DP/2 Reference Manual, where you can find specific information as you need it. You are reading the DP/2 Reference Manual.

Thank you again for choosing ENSONIQ. Enjoy the music!

Turning on the Power

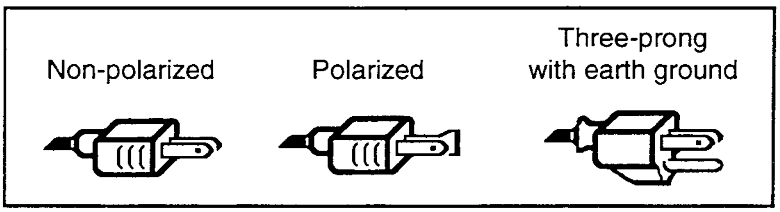


1. Before connecting the power supply, make sure the DP/2 is turned off (button out).
2. Insert the connector end of the external PS-1 Power supply into the multi-pin connector on the rear panel of the DP/2.
3. Plug the other end of the PS-1 Power Supply into a grounded AC outlet.
4. Turn the DP/2 power on and make sure the display lights up.
   If not, check your connections and power source.

When you turn the power on, the display will show "ENSONIQ * DP/2," and then go to Select mode (Select LED on). If you travel, remember the DP/2 will only operate at the voltage level listed on the PS-1 Power Supply.

Polarization and Grounding

Like many modern electrical devices, your ENSONIQ product has a three-prong power cord with earth ground to ensure safe operation. Some products have power cords with only two prongs and no earth ground. To ensure safe operation, modern products with two-prong power cords have polarized plugs which can only be inserted into an outlet the proper way.



Some products, such as older guitar amplifiers, do not have polarized plugs and can be connected to an outlet incorrectly. This may result in dangerously high voltages on the audio connections, which could cause you physical harm or damage any properly grounded equipment to which they are connected, such as your ENSONIQ product.

To avoid shock hazards or equipment damage, we recommend the following precautions:

- If you own equipment with two-pronged power cords, check to see if they are polarized or non-polarized. You might consider having an authorized repair station change any non-polarized plugs on your equipment to polarized plugs to avoid future problems.
- Exercise caution when using extension cords or plug adapters. Proper polarization should always be maintained from the outlet to the plug. The use of polarized extension cords and adapters is the easiest way to maintain proper polarity.
- Whenever possible, connect all products with grounded power cords to the same outlet ground. This will ensure a common ground level to prevent equipment damage and minimize hum in the audio output.

AC outlet testers are available from many electronic supply and hardware stores. These can be used to check for proper polarity of outlets and cords.

AC Line Conditioning

As with any computer device, the DP/2 is sensitive to sharp peaks and drops in the AC line voltage. Lightning strikes, power drops, or sudden and erratic surges in the AC line voltage can scramble the internal memory, and in some cases, damage the unit's hardware. Here are a few suggestions to help guard against such occurrences:

- A surge/spike suppressor. The cheaper of the options, a surge/spike suppressor absorbs surges and protects your gear from all but the most severe over-voltage conditions. You can get multi-outlet power strips with built-in surge/spike suppressors for little more than the cost of unprotected power strips, so using one is a good investment for all your electronic equipment.
- A line conditioner. This is the best, but by far the more expensive way to protect your gear. In addition to protecting against surges and spikes, a line conditioner guards the equipment against excessively high or low line voltages. If you use the DP/2 in lots of different locations with varying or unknown AC line conditions, you might consider investing in a line conditioner.

Guidelines for using the DP/2

Temperature Guidelines

The DP/2 contains a substantial amount of computerized and electronic circuitry that can be susceptible to damage when exposed to extreme temperature changes. When the DP/2 is brought inside after sitting in a cold climate (i.e. the back seat of your car), condensation builds up on the internal circuitry in much the same way a pair of glasses fogs up when you come inside on a cold day. If the unit is powered up as this condensation occurs, components can short out or be damaged. Excessively high temperatures also pose a threat to the unit, stressing both the internal circuits as well as the case. With this in mind, it is highly advisable to follow these precautions when storing, mounting and setting up your DP/2:

- Avoid leaving the DP/2 in temperatures of less than 50 degrees Fahrenheit or more than 100 degrees Fahrenheit.
- When bringing the DP/2 indoors after travel, allow the unit at least 20 minutes to reach room temperature before powering up. In the case of excessive outdoor temperatures (below 50 degrees Fahrenheit or above 100 degrees Fahrenheit), allow an hour or more before power up.
- Avoid leaving the DP/2 inside a vehicle exposed to direct sunlight.

Amplifying Your DP/2 Through a Home Stereo System

If you are thinking about amplifying your DP/2 through your home stereo, please be careful. A home stereo is great for playing CDs, albums or tapes — the dynamic range of these media is limited, and your speakers aren't usually subjected to extreme volume changes and frequency transients. While the dynamic range of CDs is significantly greater than LPs or tapes, the output of a CD player is still conservative compared to output of a pro-level effects processor. Running your DP/2 (or any pro-level product) through a home stereo at high volume levels can damage your stereo system and/or speakers. If your only means of amplification is your home stereo, set the **Output Knob** to around the 12 o'clock position, and try to keep your stereo volume level on the conservative side.

Powering Up Your DP/2 In a MIDI Configuration

Just as you would power up the individual components before turning on the amplifier in your home stereo system, you should first turn on the MIDI data transmitting source (processors, keyboards, modules, etc.) before you power up the receiving MIDI source. For instance, if you're using the DP/2 to receive MIDI information from a keyboard/sequencer, you would turn the keyboard on before the DP/2. This will prevent any unwanted MIDI information from being "spit" out of the transmitting source (keyboard/sequencer) during power up, which could confuse the MIDI receivers, thereby disabling them. If this should occur, turn off the receiving module, and then turn it back on.

Reinitializing the DP/2

If your DP/2 is behaving in peculiar ways (the display is showing alphanumeric characters that shouldn't be there or unexplainable Unexpected Event messages are displayed), and turning the DP/2 power off and then on again won't cure the problem, try reinitializing the DP/2.

> WARNING! THIS PROCESS WILL ERASE ALL RAM PRESETS! The 300 user presets in the internal RAM memory (locations 00-49 in Bank 1 and 00.-49. in Bank 2) are automatically loaded with the factory defaults after reinitialization. Good backup habits should be an important part of your routine. Save any important information by using the MIDI System Exclusive Dump feature of the DP/2 (see *Section 5 — Storage*), or manually write down the relevant parameters using a photocopy of the Preset Parameter Worksheet found at the end of this manual. If you fail to do so, you may accidentally lose the presets you've created.

To reinitialize the DP/2

1. While holding down the SYSTEM/MIDI button,
2. Press the B button.
3. Press the ▷ button once. The display shows:

01	Hit <WRITE> To Reinitialize!!!!

Press the CANCEL button to quit *without* reinitializing the system, or

4. Press the WRITE button to reinitialize the DP/2. Remember that by doing this you will replace *all* of the RAM Preset data in both Bank 1 and Bank 2 in the DP/2, and *all* System/MIDI parameters will be reset to their default values!

If reinitializing the DP/2 does not correct the problem, then contact an Authorized ENSONIQ Repair Station.

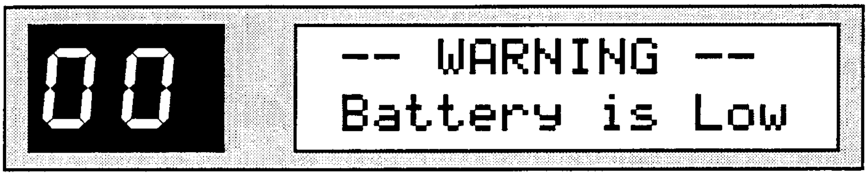
Note: If the DP/2 is sitting in an infinite loop of system errors (the display is continually cycling through errors), press the SYSTEM/MIDI button to escape this state.

Note: In the unlikely event of a system malfunction where the buttons cease to perform their normal functions, you can save your entire set-up (all Preset Banks and System parameters) with a System Exclusive dump by pressing the WRITE button. This will help you restore all of the user-defined parameters. For more information about System Exclusive dumps, see *Section 5 — Storage.*

Battery Replacement Guidelines

The reason that the DP/2 "remembers" configs, presets and system parameters, even when the power is off, is that all of its internal RAM is "battery-backed-up." The battery that keeps the DP/2 memory intact is located inside the DP/2, and when it becomes discharged, the battery must be replaced by an Authorized ENSONIQ Repair Station.

The battery that came in your DP/2 is good for up to five years. You will know when it needs replacing, because the DP/2 will tell you so. One day you will switch the power on, and instead of its usual wake-up message, the display will read:



This message will only appear for a short time, and then you can commence with normal operation. Make sure that all of your RAM presets and system parameters are saved, by using the MIDI System Exclusive Dump feature (see *Section 5 — Storage*), or manually write down the relevant parameters using a photocopy of the Preset Parameter Worksheet found at the end of this manual. Then take the DP/2 to an Authorized ENSONIQ Repair Station as soon as possible to have the battery replaced.

For more information about saving DP/2 data, see *Section 5 — Storage.*

Available Options for Your DP/2

These optional accessories are available from your Authorized ENSONIQ Dealer:

- **CVP-1 Pedal** — A *Control Voltage Foot Pedal* which can be assigned as a modulator to parameters within the DP/2. The CVP-1 Pedal makes a great "wah wah" pedal.
- **SW-10 Dual Damper Foot Switch** — Because the DP/2 has two stereo Foot Switch jacks, you can use two of these dual pedal, piano-type foot switches, for ultimate control! The foot switches can be programmed independently to act as bypass effect switches, two separately programmable modulation sources, or to step up and down through presets.

  For a full discussion of these foot switches and how to use them, see *Section 1 — Controls & Basic Functions*.

Warning!
The use of single (mono) foot switches is not recommended, and can affect the operation and performance of the DP/2.

If you are considering a foot switch for the DP/2, we strongly recommend purchasing the SW-10 Dual Foot Switch.

Need More Help?

Whether you're an aspiring programmer looking for additional information about basic effect processing techniques and MIDI theory, or a professional sound engineer working with advanced applications, you may want more detailed information that is beyond the scope of this manual. The following books can help enhance your understanding of effect processing, MIDI, and related topics. These, in addition to the numerous monthly magazines, provide a wealth of information. While we don't endorse any one of these publications, we offer this partial list as a resource for you to draw on.

The Mix Bookshelf
For prices and more information call: 1-800-233-9604

MIDI
HOW MIDI WORKS, Dan Walker
MIDI FOR MUSICIANS, Craig Anderton
MIDI SYSTEMS & CONTROL, Francis Rumsey
MIDI, THE INS, OUTS AND THRUS, Jeff Rona
THE MIDI BOOK, Steve De Furia, Joe Scacciaferro
THE MIDI HOME STUDIO, Howard Massey
THE MIDI MANUAL, David Huber
THE MIDI RESOURCE BOOK, Steve De Furia, Joe Scacciaferro
THE NEXT MIDI BOOK, Rychner & Walker
USING MIDI, Helen Casabona, David Frederick

RECORDING
BUILDING A RECORDING STUDIO, Jeff Cooper
DIGITAL DELAYS (And How to Use Them), Douglas Fraser
IMPROVING YOUR SIGNAL PROCESSING SKILLS, (cassette & manual) Bill Gibson
MASTER HANDBOOK OF ACOUSTICS, F. Alton Everest
SOUND RECORDING HANDBOOK, John Woram
SOUND REINFORCEMENT HANDBOOK, Davis & Jones

SYNTHESIS
A SYNTHESIST'S GUIDE TO ACOUSTIC INSTRUMENTS, Howard Massey
MUSIC & TECHNOLOGY, H.P. Newquist
SECRETS OF ANALOG AND DIGITAL SYNTHESIS, Steve De Furia

VIDEOS
SHAPING YOUR SOUND, (video series) Tom Lubin

Alfred Publishing Company
For prices and more information call 1-818-891-5999

MIDI
ADVANCED MIDI APPLICATIONS, GPI
BASIC MIDI APPLICATIONS, GPI
WHAT IS MIDI?, GPI

Hal Leonard Publishing
For prices and more information call 1-414-774-3630

MIND OVER MIDI, GPI

Monthly Magazines

The following magazines offer many specific articles and columns that can provide a plethora of useful information.

THE TRANSONIQ HACKER
For prices and more information about this independent news magazine for ENSONIQ Users, call 1-503-227-6848

KEYBOARD
For subscription rates and more information call 1-800-289-9919

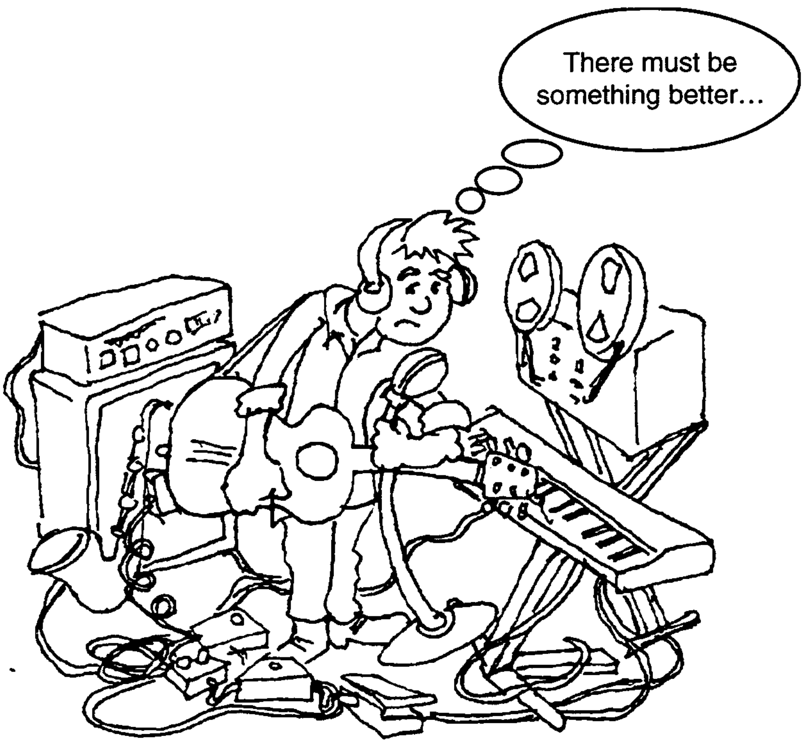
ELECTRONIC MUSICIAN
For subscription rates and more information call 1-800-888-5139

HOME & STUDIO RECORDING
For subscription rates and more information call 1-818-407-0744

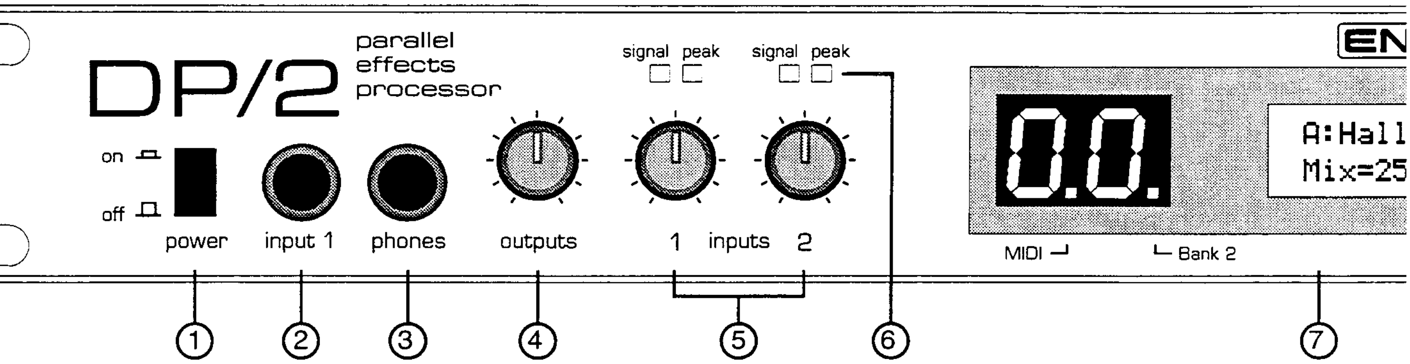
MIX
For subscription rates and more information call 1-800-888-5139

EQ
For subscription rates and more information call 1-212-213-3444

Section 1 — Controls & Basic Functions



This section provides an introduction to the DP/2's many controls and rear panel connections; a conceptual overview of the system; a guide to selecting DP/2 presets; and a discussion of editing various types of parameters. We suggest you read this section carefully — it will help you get the most out of your DP/2.



Front Panel Controls

1. Power

The power switch turns the DP/2 on and off. This switch must be in the off (button out) position when connecting the PS-1 Power Supply to the rear panel.

2. Input 1

This 1/4" mono input jack is for connecting a guitar or any high or low impedance instrument. This jack is routed to the same input circuitry as the **Input 1** jack located on the rear panel, and is electrically equivalent. When an instrument is plugged into the front panel jack, **Input 1** on the rear panel is disabled.

3. Phones

Plug headphones into this 1/4" stereo jack to listen to the DP/2 in stereo. The signal going to this jack is from the sum of the rear outputs, even if they are not connected. The rear outputs are mapped to the stereo headphone as follows: 1 is mostly to the left; 2 is mostly to the right. Headphone volume is controlled by the **Output Knob**. Plugging headphones into this jack does not turn off the audio in the outputs.

☞ **Warning:** The headphone output circuit is designed to minimize the volume differences between low and high impedance headphones. Because some headphones are more efficient than others, set the **Output Knob** accordingly — high output volume levels could damage your hearing.

4. Outputs Knob

The **Outputs** knob controls the stereo output level of each channel. If separate signals are being processed in the ENSONIQ DP/2, this knob will control their "overall" volume. The maximum output level is +17.3 dBu.

5. Input Knobs

These two input knobs control the gain applied to the input signals at Input 1 and 2 respectively. The input circuitry is designed to work with signals ranging from -18.5 dBV to +19.5 dBu. Use these knobs to set each input to the optimal level for the signal you are feeding into it.

6. Signal/Peak LEDs

The LEDs above the Input knobs indicate the level of the input signal being fed into the Analog-to-Digital Converters (ADCs) at Input 1 and 2 respectively.

- The Signal LED (green) will light when a low level signal (-30dB) is present at the input. Extremely low level input signals may not trigger this LED.
- The Peak LED (red) will light when the incoming signal reaches -6dB below the ADC clipping point.

For optimal level, adjust each **Input Knob** so that the Peak LED flashes only occasionally. Note that the Peak LEDs indicate the levels of the input signals only and will not reflect clipping in the digital processing stages.

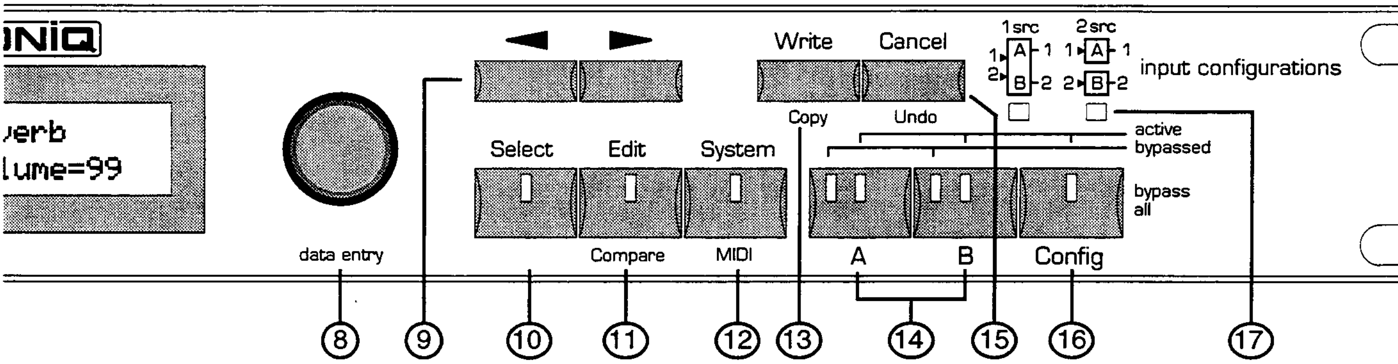
7. LED Numeric and LCD Display

In Select mode, the red, two-digit LED display shows the preset number. In Edit and System/MIDI modes, this display shows the currently active parameter number. This will also show a "--" when the preset number is invalid (i.e. when current settings are not saved).

The 32-character alphanumeric LCD display shows you information about parameters, presets, and may also ask you for additional input.

The MIDI Message Indicator (the left red decimal point in the LED Display) will light momentarily to indicate MIDI reception in all system modes.

The Bank 2 Indicator (the right red decimal point in the LED Display) will light when the preset number displayed is from Bank 2.



8. Data Entry Knob

In Select mode, turning the **Data Entry Knob** will select presets. In all other modes, the knob will change the value of the currently active parameter. In most cases, turning clockwise will increase and counterclockwise will decrease the values.

9. Left and Right Arrow Buttons

The ◁ and ▷ buttons are used to change parameters except in Select mode, where they scroll to the next preset. Also when naming presets, they are used to change the cursor position within the name.

10. Select Button

This is used to select presets which can load effects into the units and set up signal routing parameters, depending on the type of preset selected.

11. Edit Button

This is used to edit preset parameters, edit preset names and save presets. Once you've edited parameters in an algorithm, the Edit LED will flash, and pressing this button will toggle between the original (saved) preset (sollidly lit)and your newly edited version (flashing).

12. System/MIDI Button

This is used to view and modify system (or global) and MIDI parameters.

13. Write/Copy Button

The (WRITE/COPY) button is used to save or copy presets to the DP/2's internal RAM memory.

14. Unit Buttons

The two Unit buttons (**A** and **B**) correspond to the two separate signal processors in the DP/2. Use these buttons to activate a Unit for selecting presets or editing parameters. The yellow LED in each button will light when that Unit is active. When a Unit button is pressed a second time, it will be bypassed (the red LED will be lit). Pressing again will reactivate that Unit.

15. Cancel/Undo Button

The **Cancel/Undo** button is used to cancel command functions, return to the selected preset, or to undo your last unit or system parameter edit.

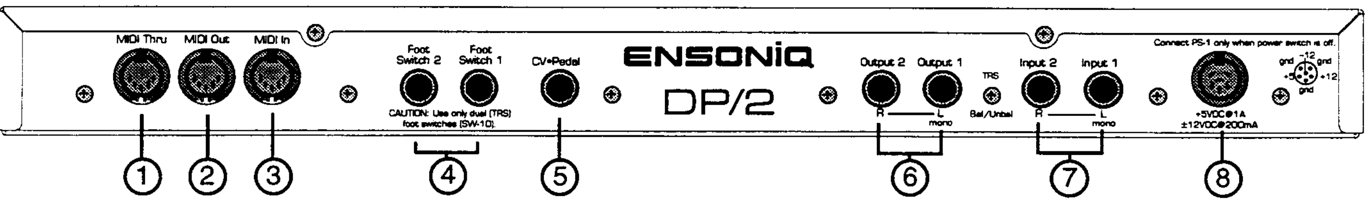
16. Config Button

This button allows you to select config presets and edit config parameters. When the Config is selected, the yellow LED above the button will light. By pressing this button a second time, you can bypass both Units (both red Unit LEDs lit). Pressing this button a third time will reactivate both Units (no red Unit LEDs lit).

17. Input Configuration LEDs

One of the LEDs beneath the diagram will be lit, to show the currently selected input configuration.

Rear Panel Connections



1. MIDI Thru

"Passes on" all MIDI (Musical Instrument Digital Interface) information received by the DP/2 to other devices. Information generated by the DP/2 itself does not go to this jack — the Thru jack merely echoes what comes in at the MIDI In jack.

2. MIDI Out

Sends out MIDI information to other instruments and computers when the System/MIDI parameter "48 Send MIDI PrgChg + Controllers" is set to "ON."

3. MIDI In

This jack receives MIDI information from other MIDI instruments or computers.

4. Foot Switch 1 and 2 Jacks

These two independent foot switch jacks are designed for dual (stereo) foot switches, and can be assigned to a number of different functions, allowing a total of four independent foot switch controllers (when two optional SW-10 Dual Foot Switches are connected).

> **Warning!**
> The use of single (mono) foot switches is not recommended, and can affect the operation and performance of the DP/2.

See "A note About the Foot Switches" later in this section.

5. CV•Pedal

This jack is for connecting an ENSONIQ Model CVP-1 Control Voltage Foot Pedal, which is assignable as a modulator to control parameters within the DP/2.

Pedal/CV Specs: 3-conductor (tip = control voltage input, ring = 424. Ohm resistor to +4.25 volts, sleeve = ground). 110. KOhm input impedance, DC coupled. Input voltage range = 0 to 4. volts DC. For use with an external control voltage, use a 2-conductor cable with the voltage on the tip and the sleeve grounded.

6. Output Jacks

The two ground compensated output jacks can be configured in either a single mono output or a mixed stereo signal.

See "A note About the Input and Output Jacks" later in this section.

7. Input Jacks

These two balanced input jacks are independent inputs and can be used in a 1 source or 2 source configuration.

See "A note About the Input and Output Jacks" later in this section.

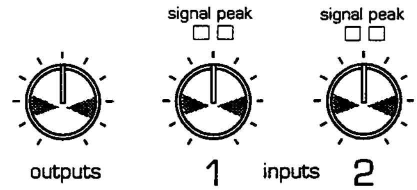
8. Line

The ENSONIQ PS-1 Power Supply is connected here. The DP/2 power switch must be off when this is connected. Use only the PS-1 Power Supply from ENSONIQ.

DP/2 RULES

Setting Levels

The input and output levels affect the volume of audio signal going into and coming out of the DP/2, and are controlled by the three knobs to the left of the display on the front panel. The two knobs on the right control the input levels for Inputs 1 and 2, the knob on the left controls the output levels for both Outputs 1 and 2.



To set the input level(s):

1. With your connections made, send a signal into the DP/2 and slowly turn the corresponding **Input Knob**(s) clockwise. The green Signal LED(s) will begin flashing as soon as a signal is detected.
2. Turn the **Input Knob**(s) clockwise until the red Peak LED above the knob begins to flash. This LED flashes when the peak level is reached, indicating that clipping is about to begin.
3. Turn the **Input Knob**(s) back down (counterclockwise) just enough so that the red LED no longer flashes. You have now attained the optimum input signal level.
4. Repeat this process for the second input (if connected).

To set the output level:

1. With your connections made and the input level(s) properly set, send a signal into the DP/2 and slowly turn the **Output Knob** clockwise. You should begin to hear signal coming through the DP/2 into your amplifier, mixer, etc.
2. Continue turning the **Output Knob** clockwise as far as you can until you hear distortion in the receiver. To optimize signal-to-noise ratio, it is best to set the output levels of the DP/2 as high as possible without distortion, turning down the receiving channel if necessary.
3. Turn the **Output Knob** down (counterclockwise) just enough until there is no distortion.

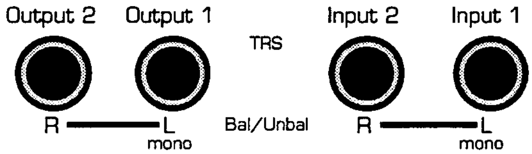
The DP/2 circuitry is designed so that if the input knob(s) are set to 11:00 and output knob is set to 2:00, and you have an input signal of +4 dBu, a +4 dBu signal will go out of the DP/2. With the input knob(s) at 2:00 and the output knob at 10:00, and an input signal of -10dBV, the output of the DP/2 will also be -10 dBV. With these settings, any incoming signals slightly above +4 dBu or -10dBV respectively, will result in clipping.

+4 dBu and -10 dBV Input/Output Settings

Desired Gain	Input Setting	Output Setting
For +4 dBu	set the input knob(s) to 11:00	set the output knob to 2:00
For -10 dBV	set the input knob(s) to 2:00	set the output knob to 10:00

A Note About the Input and Output Jacks

Use standard balanced (TRS stereo cables) or unbalanced (TS mono cables) for these connections. If there is a problem with hum or buzz, see the following section on ground loops.



As the labels on the Input and Output jacks indicate, the DP/2 employs *automatic switching* on each stereo pair of inputs and outputs. That is:

- Normally, Inputs 1 and 2, are treated as stereo inputs. However, if nothing is plugged into 2, Input 1 will work as a mono input and will also provide signal to Input 2.

  Note: In some cases, you may not want to have the mono signal plugged into Input 1 sent to Input 2. To send a discrete mono signal to Input 1, connect a "dummy" cable into the Input 2 jack (a dummy cable is just a standard balanced/unbalanced cable that is not connected to any external device).

- Similarly, Outputs 1 and 2 are normally stereo outputs. However, if nothing is plugged into Output 2, the stereo signal will be summed to mono and sent to Output 1.

Using Headphones with the DP/2

Headphones can be used with the DP/2 when connected to the front panel 1/4" stereo **Phones** jack to listen to the DP/2 in stereo. The signal going to this jack is from the sum of the outputs, even if they are not connected. The rear outputs are mapped to the headphone jack as follows: 1 is mostly to the left; 2 is mostly to the right. The outputs are not routed hard left and right to the headphone jack, to provide a "mixed stereo" signal:

Left Ear Hears
Output 1 - mostly
Output 2 - a little

Right Ear Hears
Output 2 - mostly
Output 1 - a little

Headphones

Headphone volume is controlled by the **Output Knob**. Plugging headphones into the **Phones** jack does not turn off the audio in the outputs.

☞ **Warning:** The headphone output circuit is designed to minimize the volume differences between low and high impedance headphones. Because some headphones are more efficient than others, make sure you set the **Output Knob** accordingly — high output volume levels could damage your hearing.

Ground Loops

Sometimes currents flowing through the ground line generate a signal seen by another part of the circuit sharing the same ground. In other words, if there are two identical signal paths within a circuit, they can form a loop which can result in hum and/or noise. If you are using equipment that has 3-prong "grounded" AC power cords, you may suffer from a ground loop resulting from the interconnection of this equipment. The following diagram shows how cascading or "chaining" the output of one 3-prong grounded system into the input of another 3-prong grounded system with a standard, unbalanced 2 conductor cord (like a 1/4" guitar cable) can result in a ground loop.

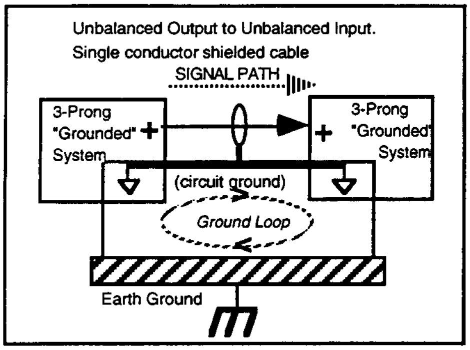


FIG. 1

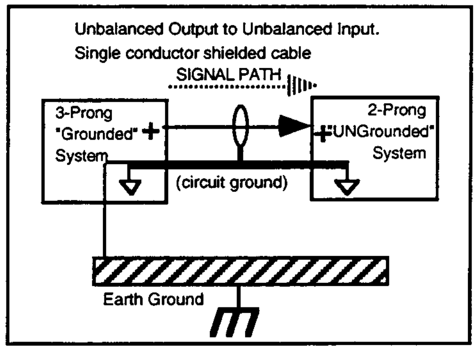


FIG. 2

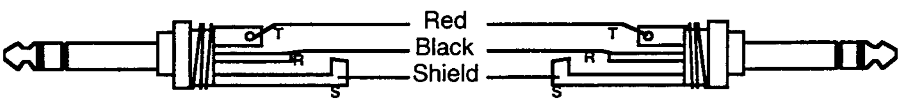
Fig. 1 shows a system interconnection where a ground loop can exist. Fig. 2 shows a system interconnection where a ground loop does NOT exist.

The DP/2 has "ground compensated" outputs, which offer the advantages of balanced outputs (minimized hum and interference), plus the advantage of a transformer isolated output (eliminates ground loop problems). The output connector "grounds" are not hooked directly to the DP/2 ground, thus eliminating the ground loop. This ground compensating scheme works on both balanced and unbalanced equipment with standard cables.

Ground loops are possible only on the inputs, and only in the following situations:

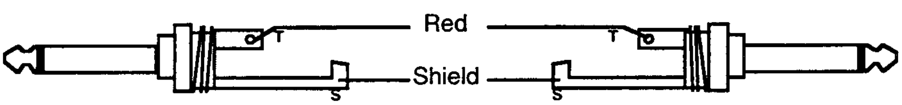
1. When a standard balanced cable is used from the preceeding piece of equipment (i.e., a standard stereo cable).

Standard Balanced Cable



2. When a standard unbalanced cable is used from the preceeding piece of equipment.

Standard Unbalanced Cable

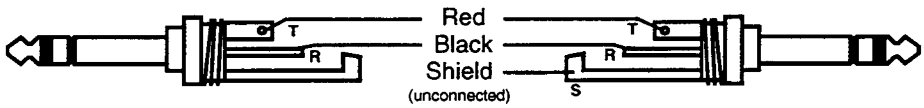


This does not mean there will always be an input ground loop problem, just the possibility.

If it exists, input ground loops can be eliminated in the following ways:

1. In balanced applications, disconnect the shield from the connector that is plugged into the output of the source device.

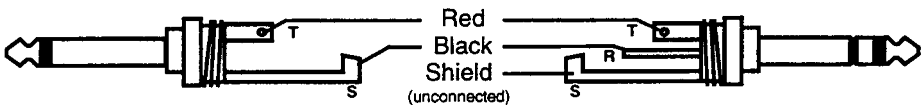
Custom Balanced Cable (to eliminate input ground loop)



OTHER 1/4" output (balanced) ENSONIQ 1/4" input (balanced)

2. In unbalanced applications, use a special cable with the shield disconnected from the connector that is plugged into the source device. Attach the source device's ground to the ring of the DP/2 input connector. The two tips connect normally.

Custom Unbalanced Cable (to eliminate input ground loop)



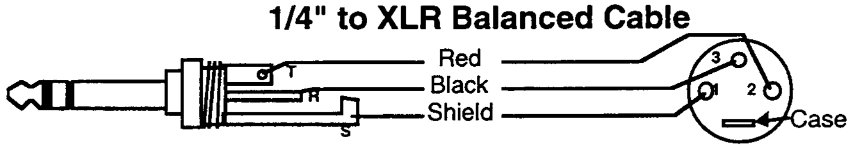
OTHER 1/4" output (unbalanced) ENSONIQ 1/4" input (balanced)

3. An audio isolation transformer will fix both balanced and unbalanced input ground loop problems, as long as the two grounds do not connect. Many of these devices have a switch on the unit that can either connect or disconnect the grounds (a ground lift switch).

Using XLR Ins and Outs with the DP/2

The DP/2 ground compensating outputs make things very easy. Use of a standard 1/4" to XLR cable will work fine with no ground loops.

1/4" to XLR Balanced Cable



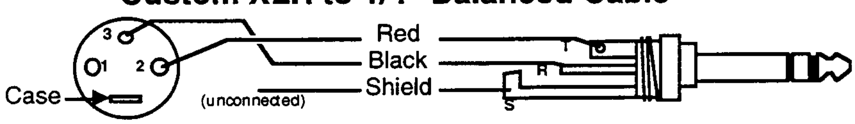
ENSONIQ 1/4" output
(ground compensated)

3-Pin XLR male input
(balanced)

As with the 1/4" to 1/4" input connections, the XLR to 1/4" cables can create some problems. Ideally, the connection of the case and pin 1 of the XLR output jack would be standard. Unfortunately, they are not. If you have an input ground loop problem with an XLR to 1/4" cable, the solutions are as follows:

- Disconnect the cable shield from pin 1 and the case connection as shown below:

Custom XLR to 1/4" Balanced Cable



3-Pin XLR female output (balanced) ENSONIQ 1/4" input (balanced)

- Use an audio isolation transformer.

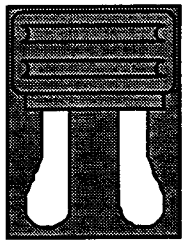
If all audio equipment adopted a compensated ground input/output scheme, ground loops would be a thing of the past.

A Note about the Foot Switches

The recommended foot switch for use with the DP/2 is the ENSONIQ SW-10 Dual Foot Switch. The SW-10 is a dual (piano-type) foot switch with two separate pedals. When the SW-10 is connected, the pedals can each be programmed independently to act as effect bypass switches, to provide two separately programmable modulation sources or to select presets.

The SW-10 is a stereo Foot Switch

and has a Stereo Plug

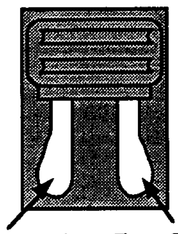


SW-10

> **Warning!**
> The use of single (mono) foot switches is not recommended, and can affect the operation and performance of the DP/2.

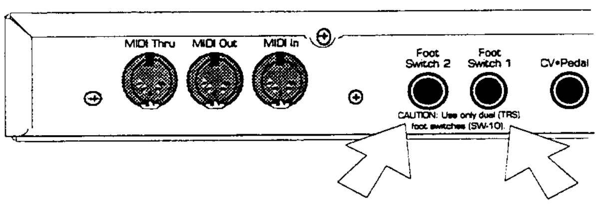
If you are considering using a foot switch, we strongly recommend the ENSONIQ SW-10 Dual Foot Switch. Why not get two?

Note: If you are using a foot switch manufactured by another company, there is a possibility that the wires inside the foot switch may be reversed. This could make the DP/2 recognize Foot Switch 1-R as left, and Foot Switch 1-L as right.



Foot Switch 1-L Foot Switch 1-R

About Mono Foot Switches



The DP/2 is designed with two stereo foot switch jacks. When any mono foot switch is plugged in, it functions like the right side of a stereo foot switch, and acts as a permanent shut-off switch for the (non-existent) left side of the foot switch. Many of the quick steps for getting around on the DP/2 require two simultaneous button presses, and will not work properly because the DP/2 reads the left foot switch connection as constantly engaged (as if a button is permanently pressed in).

If you have two mono foot switches connected, the DP/2 will assume that *two* button presses (the left sides for each foot switch) are continually engaged, and *the DP/2 will not function at all* (it will appear to be broken).

If a mono foot switch is connected to the **Foot Switch 1** jack, and the DP/2 power is switched on, you will briefly see "Button #14" in the display. If a mono foot switch is connected to the **Foot Switch 2** jack, and the DP/2 power is switched on, you will briefly see "Button #15" in the display.

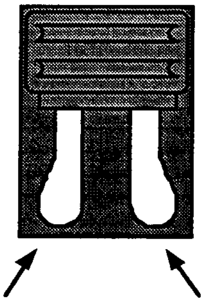
If you must use a mono foot switch, please consider performing one of the two modifications explained in "HOT MODS," found later in this section.

Using Foot Switches

An Application For Using a Stereo Foot Switch to Bypass Effects

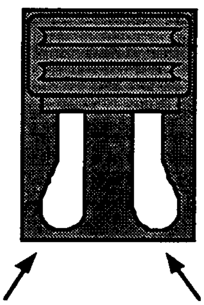
The DP/2 allows you to use the ENSONIQ SW-10 Dual Foot Switch to turn the DP/2's two effects processors on and off. To set up your foot pedal:

1. Connect the foot pedal to the **Foot Switch 1** jack on the DP/2's back panel.



2. Press the (SYSTEM/MIDI) button on the DP/2's front panel.
3. Press the ◁ or ▷ button until the large red number reads "06" and the top line of the display shows "Unit A Bypass=."
4. Turn the **Data Entry Knob** to dial in "Ftsw 1-L Toggle."
5. Press ▷ until the red number shows "13" and the display reads "Unit B Bypass=." Dial in "Ftsw 1-R Toggle."

Each foot pedal is now assigned to its own processor:



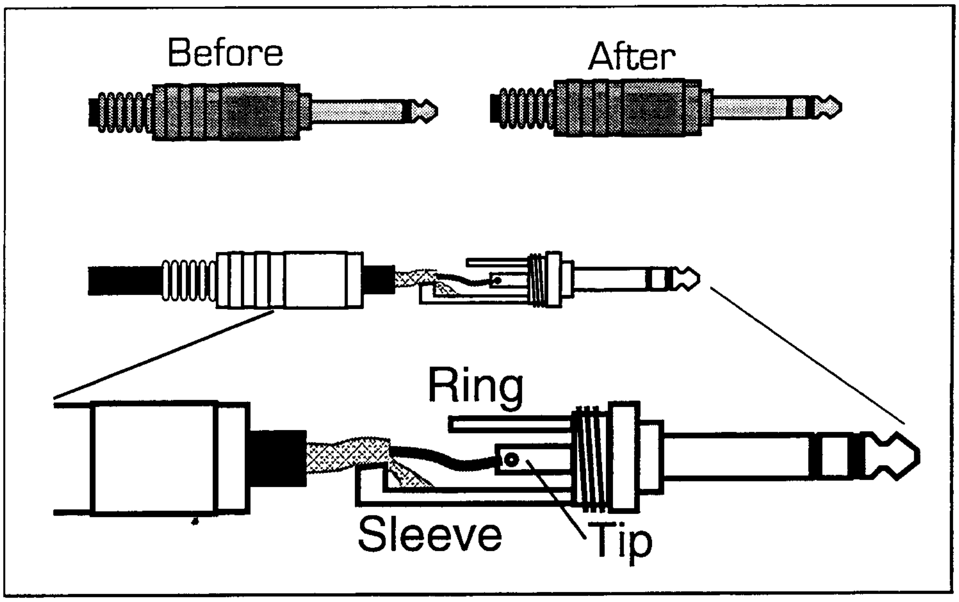
Note: Whether it's the effect being added to a sound (bypass/unbypass) — or the sound itself (kill/unkill) —is determined by each Unit's "(b)ypass and (k)ill" setting. See the description of the Bypass/Kill parameter in *Section 3—Config Parameters* for more information.

HOT MODS!

Although they are not recommended, mono foot switches such as the ENSONIQ SW-2 or SW-6 Foot Switches can be used successfully if you are willing to make either of the following modifications. If you are not comfortable performing the following modifications, we recommend asking a qualified technician for assistance:

Replace the Mono Foot Switch Plug with a Stereo Plug

The advantage of this modification is that you will eliminate the "shorted" left foot switch signal (see "About Mono Foot Switches" earlier).



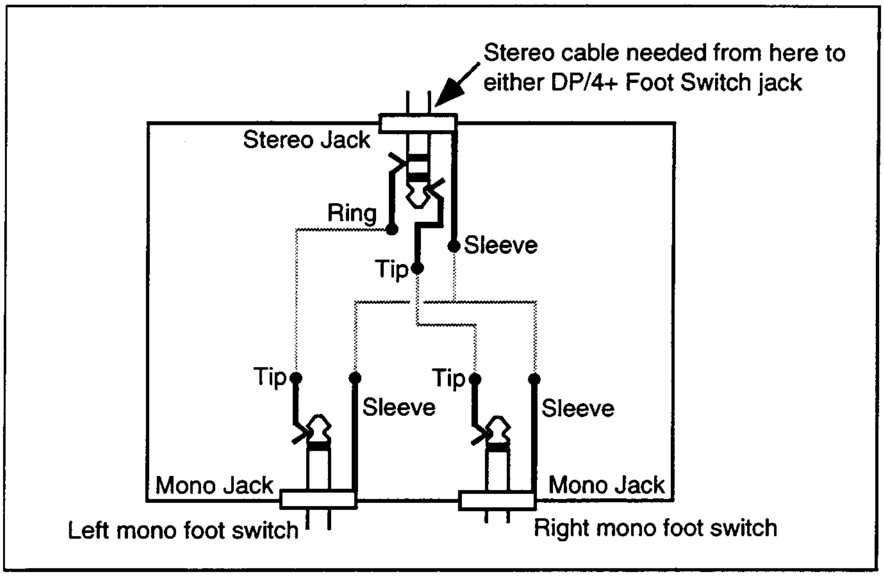
Tools/supplies required:
- soldering iron
- solder
- wire cutters
- 1/4" stereo plug

1. Unscrew the mono plug cover, and slide it out of the way (down the cable).
2. Either with wire cutters or a soldering iron, remove the wires from the mono plug.
3. Replace the mono plug cover with the stereo plug cover on the mono foot switch cable.
4. Solder the "hot" wire (the insulated wire in the center of the cable) to the tip connector, and the ground (shield) wire to the sleeve on the stereo plug as shown in the diagram.
5. Screw the stereo plug cover onto the stereo plug to complete the modification.

HOT MODS!

Build a Splitter Box to Merge Two Mono Foot Switches into One Stereo Jack

The advantage of this modification is that it will allow you to make two mono foot switches function as one stereo foot switch.



Tools/supplies required:

- soldering iron
- solder
- wire cutters
- drill and drill bits
- one plastic housing assembly (must be large enough to mount three jacks)
- one stereo jack
- two mono jacks
- shielded wire
- 1/4" stereo-to-stereo cable

1. Drill three holes in the housing assembly and mount the stereo and mono jacks.
2. Solder a wire from the tip of the left mono jack to the ring of the stereo jack.
3. Solder a wire from the tip of the right mono jack to the tip of the stereo jack.
4. Solder a wire(s) connecting the sleeves of all three jacks.
5. Connect the mono foot switch(es) to the mono jacks.
6. Connect the stereo-to-stereo cable between the stereo jack and either DP/2 Foot Switch jack.
7. You might want to mark the housing assembly to easily identify the jacks.

Tip: By doubling the above instructions, you could build a splitter box to merge four mono foot switches into two stereo jacks, for maximum DP/2 control!

DP/2 Modes

The DP/2 will always be in one of three different modes: Select, Edit, or System/MIDI. You enter one of these modes by pressing its button on the front panel; the current mode is indicated by which LED is lit.

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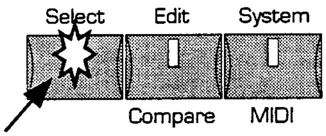
Select Mode



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Press the (SELECT) button to enter Select mode. Its LED will light. In this mode, you select Presets. These can be 1 Unit, 2 Unit, or Config Presets, depending on the current configuration.

In Select mode, presets can also be copied to new locations by pressing the (WRITE) button, and is explained in detail in *Section 5 — Storage.*

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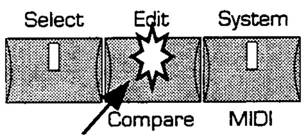
Edit Mode



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Press the (EDIT) button to enter Edit mode. Its LED will light. In this mode, you can edit (change the settings of) presets, the algorithm (effect) in each of the two Units ((A) or (B)) and its related parameters, and the config parameters (how the signals are routed). Edit mode is the easiest place to change the algorithm (by selecting a 1 Unit preset) in a single unit.

In Edit mode, presets that have been edited can be saved by pressing the (WRITE) button, and is explained in detail in *Section 5 — Storage.*

For specific information about the Algorithms and their related parameters, see *Section 2 — Algorithms.*

For more information about the Config parameters, see *Section 3 — Config Parameters.*

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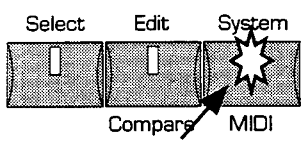
System/MIDI Mode



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Press the (SYSTEM/MIDI) button to enter System/MIDI mode. Its LED will light. In this mode, you can edit MIDI parameters, and parameters which are system-wide, or "global." The System/MIDI parameters do not change when you select different presets and configs.

In System/MIDI mode, pressing the (WRITE) button will access the MIDI System Exclusive (SysEx) data transfer function for storage of DP/2 presets and system parameters.

For more info about the System/MIDI parameters, see *Section 4 — System/MIDI.* For information about using SysEx to store DP/2 data, see *Section 5 — Storage.*

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</table>

Button Names

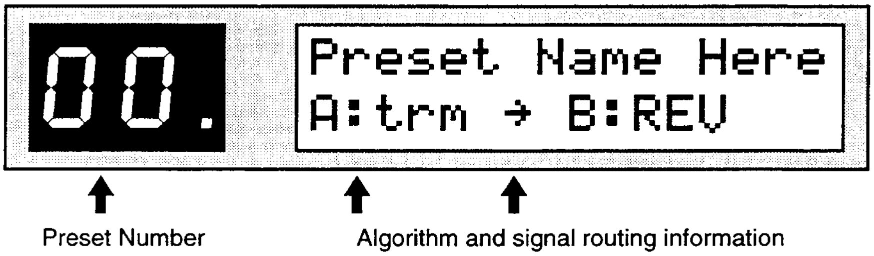
Throughout this manual, when we refer to an actual button, it will appear as a "button" in the text. For example, if the text read something like "press the Edit button," it would appear as "press the (EDIT) button." This will help you to quickly skim through familiar sections of the manual as you verify actual button presses.

About Select Mode

In Select mode, you select Presets. These can be 1 Unit, 2 Unit, or Config Presets, depending on the current config and on which unit buttons (A or B) have been pressed after pressing the SELECT button.

Input Configuration type:	What type of presets you can select:
1 source configuration	one 2-Unit Preset
2 source configuration	two 1-Unit Presets

In Select mode, the display shows the selected preset's number, name, unit algorithm(s), the currently selected unit and the signal routing. The **Data Entry Knob** and the ◁ and ▷ buttons select new presets.



- The red LED display (on the left) shows the preset's location within the DP/2 memory. The display will show numbers 00 to 99 (for Bank 1), and 00. to 99. (for Bank 2). If the lower red decimal point in the LED display is lit, the preset is located in Bank 2. If the decimal point is not lit, the preset is located in Bank 1. If any parameters have been edited (changed) within the selected preset, this display will show "- -."
- The top line of the LCD display (on the right) shows the presets name.
- The bottom line shows which algorithm (effect) is in each unit as well as signal routing information, depending on the current configuration.

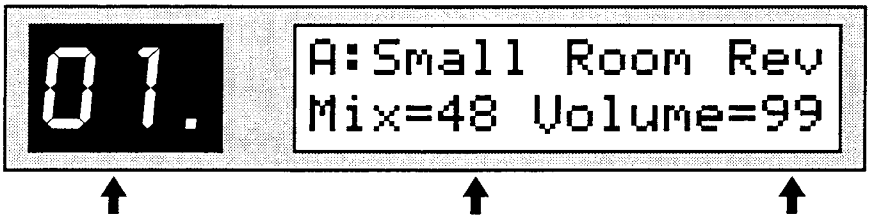
In a 2 Unit preset, you will notice that one of the abbreviations in your display is capitalized. This shows that the capitalized algorithm is located in the currently selected unit (in the diagram above, it's Unit B). If none of the algorithm abbreviations are capitalized, it means that the Config is activated. Try pressing the Unit buttons (A or B) to see the abbreviations change between lower case and upper case. When you press the CONFIG button, there are no capitalized algorithm abbreviations.

When a unit is selected by pressing its Unit button (A or B):
- its algorithm abbreviation is capitalized
- that unit will be selected for editing (if you press the EDIT button)
- pressing its button again will bypass the unit (its red LED will light)

About Edit Mode

In Edit mode, you can edit (change the settings of) presets, the algorithm (effect) in each of the Units (A or B) and its related parameters, and the config parameters (how the signals are routed). Edit mode is the easiest place to change the algorithm (by selecting a 1 Unit preset) in a single unit.

After pressing EDIT, pressing A, B, or CONFIG determines what you will be editing. The display shows:



Parameter Number — Parameters which can be selected and adjusted

- The red LED display (on the left) shows the *number* of the currently selected parameter. This will change as you press the ◁ and ▷ buttons (called scrolling) to select different parameters.
- When the algorithm name is selected, the red LED display will flash to differentiate it from the other parameter numbers. The number displayed will show the number of the 1 Unit Preset that will next be selected when the knob is turned. The display will show numbers 00 to 99 (for Bank 1), and 00. to 99. (for Bank 2). If the lower red decimal point in the LED display is lit, the preset is located in Bank 2. If the decimal point is not lit, the preset is located in Bank 1.
- The LCD display (on the right) shows one or more parameters, which can be selected and adjusted. The currently selected parameter will always be *flashing*.
- The **Data Entry Knob** is used to change the value of the selected parameter.

Edit Buffer

As soon as you change a parameter's value, you will notice that the Edit LED begins flashing. This means that you are now listening to a modified version of the algorithm in the *Edit Buffer*. The Edit Buffer is a section of RAM where edits are temporarily stored.

By pressing the COMPARE button (the EDIT button), you can toggle between the original setting (LED solidly lit) and the newly edited version (LED flashing) of that algorithm. What is visible on the display is the parameter settings that you hear.

About System/MIDI Mode

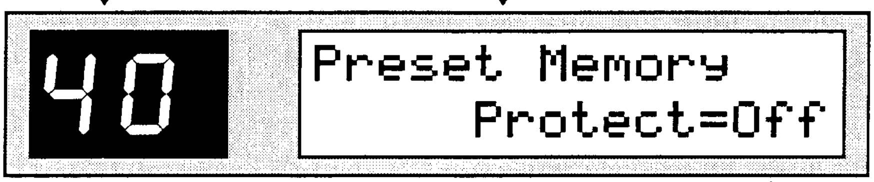
In System/MIDI mode, you can edit unit-specific MIDI parameters, and parameters which are system-wide, or "global." The System/MIDI parameters (such as MIDI channels, Controllers, and program change maps) do not change when you select different presets and configs.

For specific information about the System/MIDI parameters, see *Section 4 — System/MIDI.*

To enter System/MIDI mode, press the (SYSTEM/MIDI) button. The display shows:

Parameter Number

Parameter Name



Parameter range which can be selected and adjusted

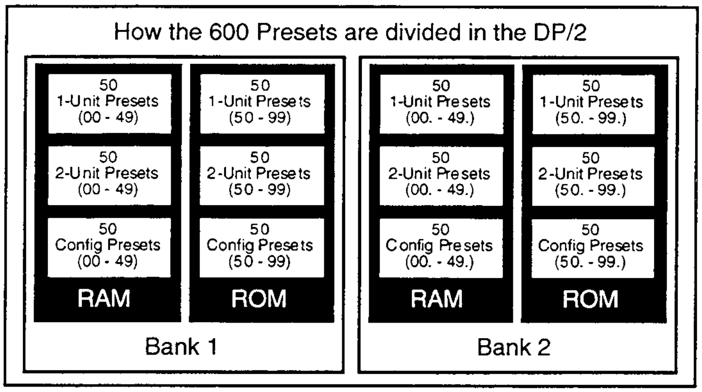
- The red LED display (on the left) shows the *number* of the currently selected parameter. There are 52 parameters in this mode.
- The LCD display (on the right) shows one or more parameters, which can be selected and adjusted. The currently selected parameter will always be *flashing.*
- The (◁) and (▷) buttons scroll through the different parameters. Use them to select the parameter you want to change.
- The **Data Entry Knob** changes the value of the selected parameter.

  Tip: Though you can scroll continuously through all 52 System/MIDI parameters using the (◁) and (▷) buttons, this might get tedious; you can use the following shortcuts to get close to the parameter(s) you desire:

- After pressing (SYSTEM/MIDI), press (A), (B), or (CONFIG) to go to unit-specific MIDI parameters relating to them.
- Press (SYSTEM/MIDI) repeatedly to go directly to several convenient locations within the remaining parameter list. Then scroll to the parameter you want.

About Presets

The DP/2 contains a total of 600 presets, divided into two banks of 300 presets each. Preset Banks 1 and 2 each contain 100 presets of each type (1 Unit, 2 Unit and Config presets), 50 in RAM and 50 in ROM.



Selecting Presets From Bank 1 and 2

Presets are manually accessed by turning the **Data Entry Knob** in both Select mode and Edit mode. Presets are also accessible via MIDI Bank Select and Program Change messages:

Select Mode:
Press (SELECT), followed by the Unit (A), Unit (B), or (CONFIG) button. Turn the **Data Entry Knob**. The red LED display shows the location of the preset that you will be selecting. If there is no dot in the display (i.e., it shows numbers 00 to 99), the preset is in Bank 1. Turn the **Data Entry Knob** until the red LED display goes past 99 (Bank 1). The red LED display will wrap back to 00, and the right LED decimal point will light to indicate that the next 100 presets (00. to 99.) are in Bank 2.

Edit Mode:
Press (EDIT), followed by the Unit (A) or Unit (B) button. Press the (◁) button repeatedly until the Algorithm Select parameter is selected. The red LED display will flash a preset number. If there is no dot in the display, it shows numbers 00 to 99 (Bank 1). Turn the **Data Entry Knob** until the number goes past 99. The red LED display will wrap back to 00, and the right LED decimal point will light to indicate that the next 100 presets (00. to 99.) are in Bank 2. After a preset is chosen (the new algorithm is being loaded into the ESP chip), the red LED number and the right LED decimal point will flash.

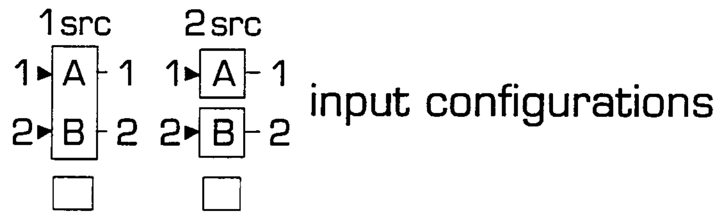
MIDI In:
When the System/MIDI Unit A, B, or Config Program Change Maps (parameters 03, 10, and 17) are turned Off, the DP/2 will respond to MIDI Bank Select and Program Changes as follows:

MIDI Bank Select and Program Change Map
Bank Select LSB 000, Program Changes 000-099 select Bank 1 presets (00-99)
Bank Select LSB 001, Program Changes 000-099 select Bank 2 presets (00.-99.)
Bank Select LSB 002 to 127 will be ignored, and will have no effect on Program Change reception.
Bank Select MSB values 000 to 127 will be ignored, and will have no effect on Program Change reception.

When the System/MIDI Unit A, B, or Config Program Change Maps (parameters 03, 10, and 17) are turned On, the DP/2 will ignore MIDI Bank Select messages, and will respond to MIDI Program Changes according to the settings in the Program Change Map.

Input Configurations

The diagram in the upper right corner of the DP/2 shows the input configurations:



1 Source Input Configuration

In a 1 Source Config, the LED beneath the 1 source input configuration diagram will light. Use **Input 1** (front or back panel) for a mono signal (such as a guitar), or **Inputs 1** and **2** if your source is a stereo signal (such as a keyboard). The choice of stereo or mono for an input is a 1 Source Config parameter, and will be covered later in this section. Remember, any mono signal (high or low impedance) can be plugged into the jack on the front panel. The **Input 1** jack (front panel) will always override the **Input 1** jack on the rear panel.

2 Source Input Configuration

In a 2 Source Config, the LED beneath the 2 source input configuration diagram will light. For your first source, use **Input 1** (front or back panel) for a mono signal. For your second source, use **Input 2** for a mono signal. Even though the input signals to the units must be mono, the effect processing can generate two *stereo* output signals.

Selecting Config Presets

Of the three Preset types (1 Unit, 2 Unit, and Config), the most powerful is the *Config Preset*. The Config preset lets you save, and later recall, the current state of the DP/2, including all algorithm, signal routing and mixing information.

Selecting a Config preset will

- Reconfigure the DP/2 inputs and outputs;
- Change the signal routing between units; and
- Load a new algorithm into each of the Units.

To select a Config preset

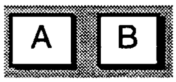
1. Press (SELECT).
2. If the Config LED is not already on, press (CONFIG).
3. Move the **Data Entry Knob**, or press the (◁) and (▷) buttons. The Select LED flashes, indicating that you are previewing presets. The display shows the available Config presets.
4. When the display is showing the preset you want to load, press (SELECT) again. This selects the preset, and the Select LED stops flashing.

Note: The first three ROM Config locations in Bank 1 (presets #50 to 52) and the first two ROM Config locations in Bank 2 (presets #50. and 51.) can be used as "starting places" for creating your own configurations, and cover common signal routing set-ups. Note that presets numbers in Bank 2 will appear with a dot in the display.

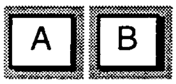
How the Config Type Affects Selecting Presets

The two different config types effectively turn the DP/2 into 1 or 2 independent effects processors:

Configuration type:



A 1 source config turns the DP/2 into one multi-effects processor, with two effects units applied to the input signal. These are called 2-Unit Presets.



A 2 source config turns the DP/2 into two independent effects processors; each of the Units processes a different input signal. These are both called 1-Unit Presets.

Preset Example:

```
Airplane Hangar
A:REV → B:rev
```

In Select mode, when viewing a 2-Unit Preset, the lower line shows the 3-letter abbreviations for the algorithms in both units, as well as a symbol showing how the units are routed.

```
Vocal Plate 1
A:Large Plate
```

In Select mode, when viewing a 1-Unit Preset, the lower line of the display shows the full name of the algorithm in the preset, and indicates which unit the preset is or will be loaded into (A or B).

If you think of the outer grey boxes shown above as different "config types," you'll notice that a config type can be either a 2-Unit preset, or a pair of 1-Unit presets. This determines what presets you can select at any given time.

There are two ways that the DP/2 shows you which type of preset you are selecting:

1. In Select mode, when you press one of the Unit buttons (A or B), either one or both of the yellow Unit LEDs will light. The number of lighted LEDs corresponds to the type of preset you will be selecting if you move the **Data Entry Knob**. The yellow Unit LEDs also tell you which unit(s) will be loaded with a new algorithm if you select a new preset.
2. The display gives you constant feedback. For both preset types, the upper line of the display shows the preset name. The lower line shows how many units are in the preset you are selecting.

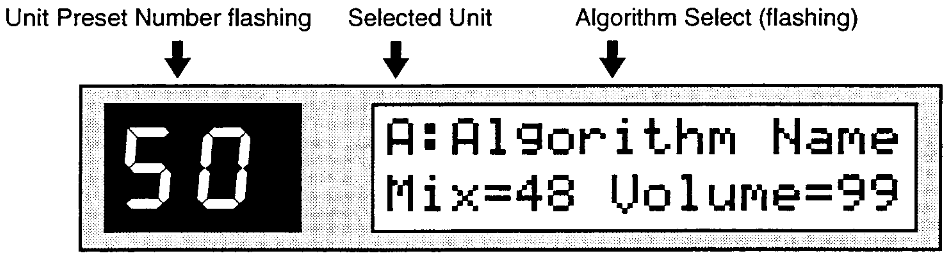
Note: The Input Configuration LEDs show what type of preset is *currently* selected, and does not show what type of preset you *will be* selecting.

Replacing the Algorithm in a Single Unit

You can use Edit mode to easily replace the algorithm in one of the units without changing the current config or affecting what is in the other unit.

To replace the algorithm in a unit:

1. Press (EDIT).
2. Press Unit (A) or (B) to select that unit for editing. The unit's yellow LED should be lit. The display shows:



   The red LED display should flash, indicating the number of the unit preset last selected. The algorithm name should be flashing in the upper line of the LCD display. If neither is flashing, press the (◁) button until this is the case.

3. Move the **Data Entry Knob** to select among the algorithms in memory. The display will change, showing the algorithm name on the top line, and the name of the 1 Unit preset that uses the algorithm on the bottom line. When you stop moving the knob, the algorithm that is showing on the display will be loaded into the Unit, and the display will change back to the one shown above.

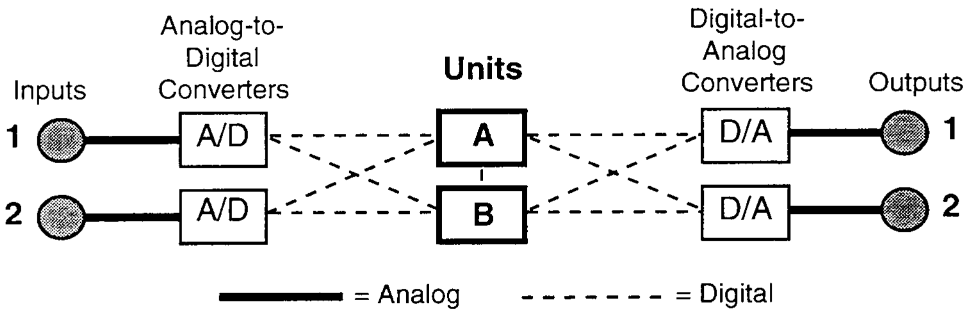
   Note: When you select algorithms in Edit mode, you are actually picking from the list of 200 1-Unit presets. As you are moving the knob, the red LED display shows the preset numbers of the 1 Unit presets you are loading into the unit. One second after you've stopped moving the knob, the algorithm is loaded into the unit and the red LED display will flash the preset number selected.

4. To change the algorithm in the other unit, simply press its unit button and repeat the above steps. Note that selecting a unit which is already active causes it to become bypassed. The red bypass LED for that unit will light. Further presses toggle out of and into bypass.

   Note: If you accidentally edit the algorithm in a unit, you will lose all of the parameter settings for the old algorithm *unless* you press (UNDO) to recall the original algorithm and its parameters. You must press (UNDO) *before* scrolling to another parameter or leaving Edit mode.

About Signal Routing

The two audio inputs are analog signals which are fed to two analog-to-digital converters. The two units are digital audio signal processors which have digital inputs and outputs. Routing between the units is digital. The output of a unit is converted back to analog audio for the output jack.

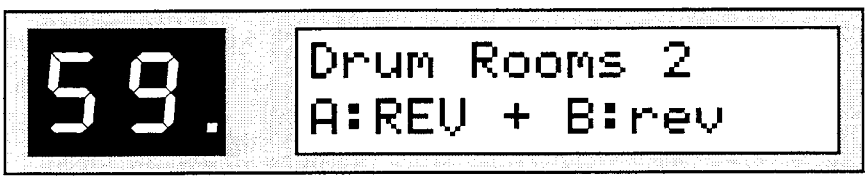


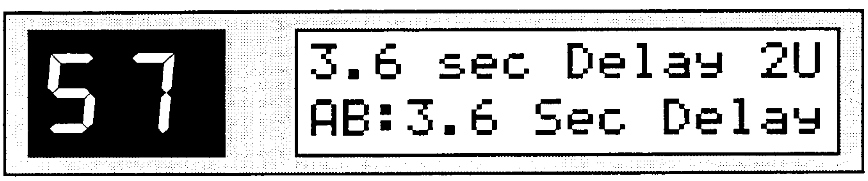
(Not all of the possible signal routings are shown.)

All of the above elements are under complete software control.

Signal Routing Between Units

Depending of the current Config, the units can be connected to each other in one of five different ways. The routing symbols (shown between each algorithm in Select mode) are:





→	Indicates a serial connection from Unit A into Unit B (top example).
+	Indicates a parallel connection between Unit A and Unit B (middle example).
⇄	Indicates a feedback 1 connection from Unit B back into Unit A.
⇆	Indicates a feedback 2 connection from Unit B back into Unit A.
AB:	Indicates that the two units are "ganged together" using an algorithm that requires two units of processing power (bottom example). The routing between units cannot be modified as long as such a "ganged" 2 Unit algorithm is active.
(blank space)	Indicates that there is no connection between units, and that they are routed separately (the blank space will not appear in a 1 Source Config, because both units are always connected together). This can only be seen in a Config Preset.

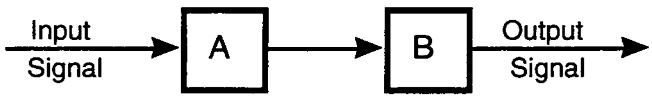
Understanding Serial, Parallel and Feedback Signal Routing

When we speak of connecting units together, we are usually referring to one of three types of signal routing, serial, parallel, or feedback. It is *very* important to understand the difference between these concepts.

Serial Routing

Serial routing means the input signal is routed through Unit A *before* being sent to the input of Unit B.

This is a *serial* signal routing between two units:

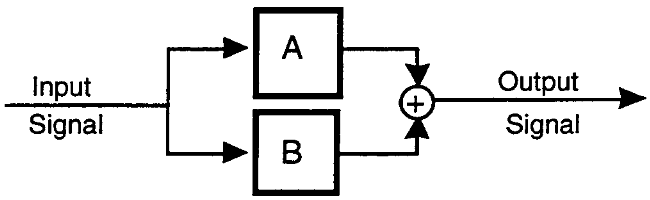


If, for example, Unit A is a chorus, and Unit B a reverb, you have the signal first going through the chorus, then into the reverb. As a result, you would hear the chorused sound with reverb applied to it.

Parallel Routing

Parallel routing means the same input signal is routed separately to the inputs of *both* units, and then their outputs are mixed together.

This is a *parallel* signal routing between two units:

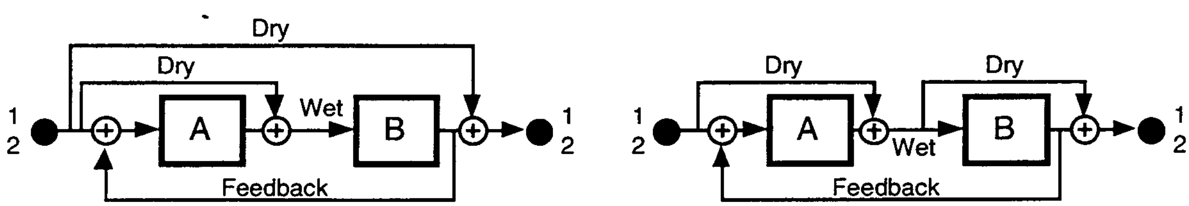


In this example, if Unit A is a chorus, and Unit B a reverb, you would hear the chorused sound *and* a sound with reverb, but the chorused sound would *not* have reverb on it, and the sound coming out of the reverb would not have chorusing.

Feedback Routing

Feedback routing (shown by a ≑ symbol) is similar to serial routing, with the addition of a feedback signal. The difference between Feedback 1 and Feedback 2 is how the dry signal is mixed into the wet signal (as shown below). Note that the feedback signal is all wet, and that it is tapped before the dry signal.

These are the two *feedback* signal routings available in the DP/2:



Feedback 1 **Feedback 2**

In these examples, if Unit A is a chorus, and Unit B a reverb, you have the signal first going through the chorus, then into the reverb. There is then an additional tap that sends the processed signal back into the beginning of Unit A (the chorus).

Bypassing Units

At some point while you're selecting various 2 Unit Presets, you may want to listen to how an individual unit's effect is processing the incoming audio signal. In this case, you would need to *bypass* the other unit.

To bypass a single unit's algorithm:

1. Press the unit button ((A) or (B)) that you want to bypass.
2. Press the same unit button again. The red LED in the unit button will be lit, and the unit will be bypassed.
3. Further presses of the unit button will toggle out of and into bypass.

To bypass both of the unit algorithms:

1. Press the (CONFIG) button.
2. Press the (CONFIG) button again. Both of the units red LEDs will be lit, and both units will be bypassed.
3. Further presses of the (CONFIG) button will toggle both units out of and into bypass.

Refer to the description of the Bypass/Kill parameter in *Section 3 — Config Parameters* for more information.

Note: Units can also be bypassed, un-bypassed, and killed remotely with MIDI Program Changes by enabling the unit's MIDI Program Change Map. See *Section 4 — System/MIDI* for more information.

Quick Tips and Shortcuts

Here are a few quick tips to find your way around the DP/2.

Tip: To get to the Algorithm Select parameter in Edit mode without having to scroll through lots of parameters, while holding down either the (◁) or (▷) button, press (CANCEL).

Tip: When there are several parameters on the screen at one time, and you want to get to the next screen without having to cursor through each parameter, while holding down either the (◁) or (▷) button, press the other arrow button.

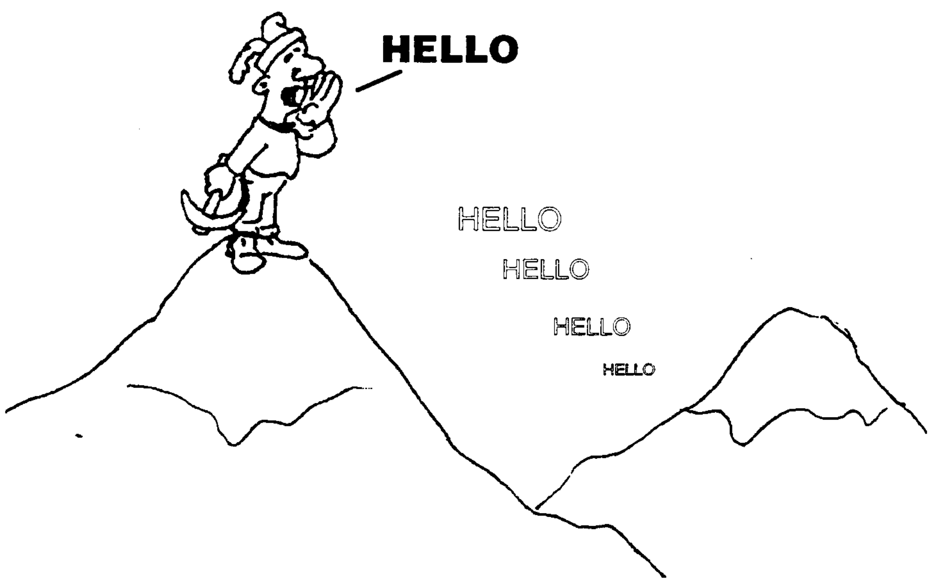
Tip: While holding down either the (◁) or (▷) button, turn the **Data Entry Knob.** This lets you move much more quickly through a long list of parameters that by repeatedly pressing the arrow buttons.

Tip: To return the last parameter you edited to its original value, press the (CANCEL) button. *This will only work as long as you still have that parameter selected.* If you have scrolled to another parameter, this undo function will no longer work.

Tip: If you accidentally change the algorithm in a unit, you will lose all of the parameter settings for the old algorithm *unless* you press (CANCEL) to recall the original algorithm and its parameters. You must press (CANCEL) *before* scrolling to another parameter or leaving Edit mode.

Tip: If you double-click on the (CANCEL) button, a signed parameter (one that has a range of -99 to +99) will put the level to +00 (the center position).

Section 2 — Algorithms



This section will teach you how the effects (called algorithms) work in the DP/2, and defines all of the parameters relating to the effect algorithms.

List of Algorithms

The following algorithms (listed in alphabetical order) are available in the DP/2:

3.6 sec DDL 2U	Keyed Expander
8 Voice Chorus	Large Plate
ADSR Env Gen	Large Room Rev
Chorus-Reverb	MultiTap Delay
CmprDistFlngRev	No Effect
De-esser	Non Lin Reverb1
DigitalTubeAmp	NonLin Reverb2
Dist-Cho-Revrb	NonLin Reverb3
Dist-Roto-Revb	Parametric EQ
Dual Delay	Phaser - DDL
Ducker / Gate	Phaser-Reverb
DynamicTubeAmp	Pitch Shift 2U
EQ-Chorus-DDL	Pitch Shift-DDL
EQ-Compressor	Pitch Shifter
EQ-DDL-withLFO	Plate-Chorus
EQ-Flanger-DDL	ReverseReverb1
EQ-Gate	ReverseReverb2
EQ-Panner-DDL	Rotating Spkr
EQ-Tremolo-DDL	Rumble Filter
EQ-Vibrato-DDL	Sine/Noise Gen
Expander	Small Plate
FastPitchShift	Small Room Rev
Flanger	Speaker Cabinet
Flanger-Reverb	Tempo Delay
Fuzz Box	Tunable Spkr 1
Gated Reverb	Tunable Spkr 2
Guitar Amp 1	VandrPolFilter
Guitar Amp 2	VCF-Distort 1
Guitar Amp 3	VCF-Distort 2
Guitar Amp 4	Vocal Remover
GuitarTuner 2U	Vocoder 2U
Hall Reverb	Wah-Dist-Revrb
InversExpander	

Understanding DP/2 Algorithms

An algorithm is a control program for the ESP chips. The ESP chips are digital signal processors that, when programmed, provide the basic sonic building blocks in the DP/2). The word "effect" could be used instead of algorithm, but some algorithms can produce several sonic effects simultaneously (e.g., the EQ-Flanger-DDL algorithm offers three different effects — an equalizer, a flanger, and a digital delay). All of the algorithms are stereo (except the ducker and the keyed expander) and have a set of parameters that are highly programmable, allowing for customization for particular applications.

Algorithms are stored within presets. Most algorithms can be found in the 1 Unit ROM preset locations. Some algorithms require more than 1 unit's worth of DSP processing power, or require specialized routing, and as a result, are stored in the 2-Unit or Config ROM presets.

Each algorithm in the DP/2 contains a complete set of parameters which determine how the algorithm will sound. The algorithm is present even if the signal is not routed through the effect (e.g. when the units are bypassed). Whenever you copy or save a preset, its algorithm parameter settings are also saved.

Programming Algorithms

The parameter settings in the DP/2 algorithms are highly programmable. There are several common parameters for every algorithm, as well as many algorithm-specific parameters. The first parameter selects which algorithm will be used. When this parameter is changed, a new algorithm is selected, which causes several important things to occur:

- a new effect preset is loaded into the ESP chips, causing the audio signal to make a smooth transition from the previous algorithm to the new algorithm,
- the parameter screens are redefined for the particular algorithm selected, and
- the parameter values are reset to their saved settings for the new algorithm.

When are New Algorithms Loaded into the ESP Chips?

When you select a new preset, the algorithm(s) and its parameters will be loaded into the ESP(s), and you will hear that algorithm.

Whenever a new algorithm is loaded into the ESP(s), the DP/2 crossfades the audio input to a dry path around each of the ESP(s), allowing for smooth effect transitions.

Note: When you bypass the units, the algorithms will *not* be changed.

Editing Algorithm Parameters

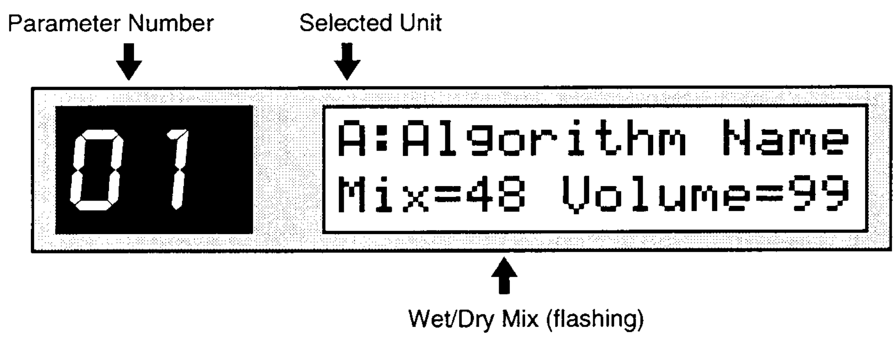
To Edit the Algorithm Parameters

1. Press the (EDIT) button, then the appropriate unit button ((A) or (B)).
2. Algorithm parameters can then be edited by pressing the (◁) and (▷) buttons to select the parameter, and turning the **Data Entry Knob** to change the parameter's value. The parameters that pertain to each algorithm are described later in this section.

If you like the algorithm that is in a unit but want to adjust the wet/dry mix, the level of that unit, reverb decay time, or any other parameter within the effect, this is done in Edit mode. In the following example, we will change the algorithm wet/dry mix and volume levels.

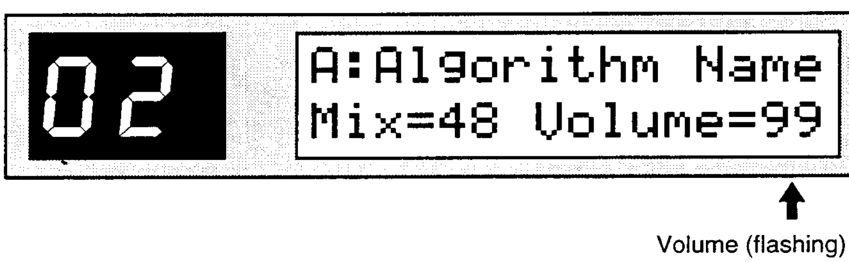
To modify the parameter levels of the algorithm in a unit:

1. Press (EDIT) (unless you are already in Edit mode).
2. Press Unit (A) or (B) to select that unit for editing. The active unit's yellow LED should be lit.
3. Press the (◁) or (▷) buttons until the display shows parameter 01, as below:



The LED display (on the left) should indicate parameter 01, which is the wet/dry mix, and the Mix value should be flashing (if not, press the (◁) or (▷) button until this is the case).

4. Move the **Data Entry Knob** to adjust the mix. A value of 00 is fully "dry" (no effect) and a value of 99 is fully "wet" (only the effect).
5. Press the (▷) button once more to move to parameter 02, Volume. The display shows:



6. Move the **Data Entry Knob** to adjust the output volume of the Unit. This parameter is used to control the levels of the various units relative to each other in a preset.

   Note: Parameters #01 Wet/Dry Mix, and #02 Volume, will be the same for all algorithms. The remaining parameters (those reached by continuing to scroll past parameter #02) will vary, depending on the algorithm.

7. To perform further edits to the algorithm, use the (◁) and (▷) buttons to select parameters and the **Data Entry Knob** to change the value of the active (flashing) parameter. You will find a wide assortment of parameters available for each algorithm, allowing you to customize the DP/2's effects to your exact needs.

Editing Algorithm Parameters

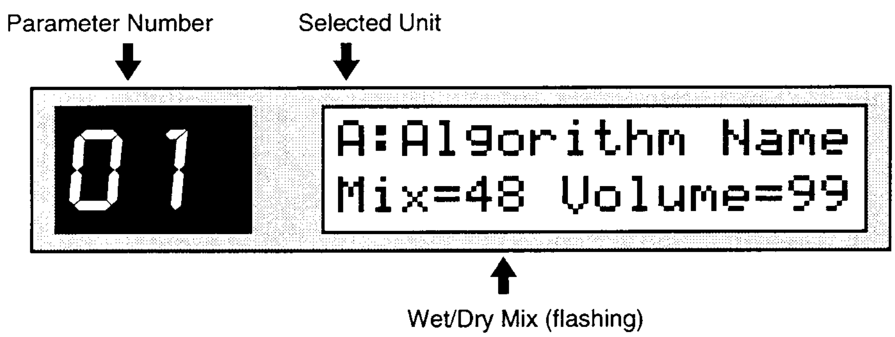
To Edit the Algorithm Parameters

1. Press the (EDIT) button, then the appropriate unit button ((A) or (B)).
2. Algorithm parameters can then be edited by pressing the (◁) and (▷) buttons to select the parameter, and turning the **Data Entry Knob** to change the parameter's value. The parameters that pertain to each algorithm are described later in this section.

If you like the algorithm that is in a unit but want to adjust the wet/dry mix, the level of that unit, reverb decay time, or any other parameter within the effect, this is done in Edit mode. In the following example, we will change the algorithm wet/dry mix and volume levels.

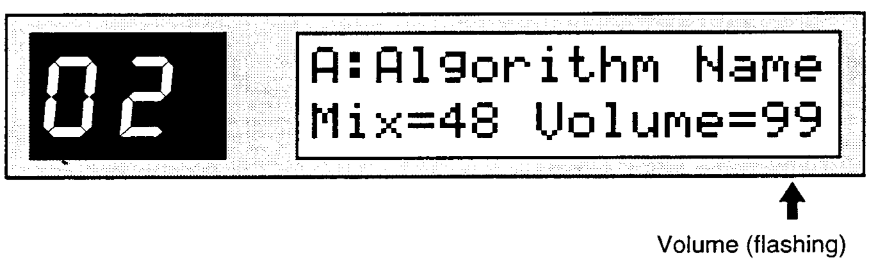
To modify the parameter levels of the algorithm in a unit:

1. Press (EDIT) (unless you are already in Edit mode).
2. Press Unit (A) or (B) to select that unit for editing. The active unit's yellow LED should be lit.
3. Press the (◁) or (▷) buttons until the display shows parameter 01, as below:



The LED display (on the left) should indicate parameter 01, which is the wet/dry mix, and the Mix value should be flashing (if not, press the (◁) or (▷) button until this is the case).

4. Move the **Data Entry Knob** to adjust the mix. A value of 00 is fully "dry" (no effect) and a value of 99 is fully "wet" (only the effect).
5. Press the (▷) button once more to move to parameter 02, Volume. The display shows:



6. Move the **Data Entry Knob** to adjust the output volume of the Unit. This parameter is used to control the levels of the various units relative to each other in a preset.

   Note: Parameters #01 Wet/Dry Mix, and #02 Volume, will be the same for all algorithms. The remaining parameters (those reached by continuing to scroll past parameter #02) will vary, depending on the algorithm.

7. To perform further edits to the algorithm, use the (◁) and (▷) buttons to select parameters and the **Data Entry Knob** to change the value of the active (flashing) parameter. You will find a wide assortment of parameters available for each algorithm, allowing you to customize the DP/2's effects to your exact needs.

Mix and Volume Parameters

All of the algorithms in the DP/2 share common Mix and Volume parameters:

01 — Mix   Range: 00 to 99

The Mix parameter (always parameter 01) controls the mix between the original (dry) signal and the fully effected (wet) signal. Setting this parameter to 00 will allow only the unprocessed signal to be heard, while a setting of 99 will eliminate the dry signal completely, with only the wet portion remaining. Some algorithms sound best with a blend of wet and dry, whereas some are optimized at a setting of 99 (all wet).

Note: When the DP/2 is used in an application where it's desirable to have only the effected (wet) signal appear in the DP/2's audio outputs, set System/MIDI parameter #46 to "Set All 1U Pset Mixes To Wet=Yes." It will automatically set all 1 Unit Preset mix levels to 99 (all wet) when they are selected—or installed in Edit mode—without altering the actual values saved in the presets. See the description of this parameter in *Section 4 — System/MIDI.*

02 — Volume   Range: 00 to 99

The Volume parameter (always parameter 02) adjusts the output volume of the signal. Setting this parameter to 00 will eliminate the signal. If Unit B is in series with Unit A, setting the Unit A volumer to zero will result in no sound for the DP/2.

Algorithm Modulators

All the algorithms allow real-time control of selectable parameters and share common modulation control parameters (except the Guitar Tuner utility algorithm). The exact numeric location of these parameters varies depending on the selected algorithm, but it is always the last eight parameters in the algorithms:

Mod1 Source
Mod2 Source   Range: Off/Controller 1 - 8

These parameters select the modulation sources used to modulate the parameter Destinations. Each algorithm has a choice of two different mod sources. Any one of the eight DP/2 System Controller sources assigned in System/MIDI mode can be selected (for more information, refer to the descriptions of the System global parameters in *Section 4 — System/MIDI*).

Mod1 Destination Parameter
Mod2 Destination Parameter   Range: Off, 01 to 34 (depending on the algorithm)

This parameter selects which algorithm parameters will be modulated by the modulation sources. The choice of parameters varies depending on the selected algorithm. Any parameter within an algorithm can be selected (except the algorithm select). Each algorithm has a choice of two different mod destinations.

Mod1 Param Range Min
Mod1 Param Range Max
Mod2 Param Range Min
Mod2 Param Range Max   Range: 00 to 99

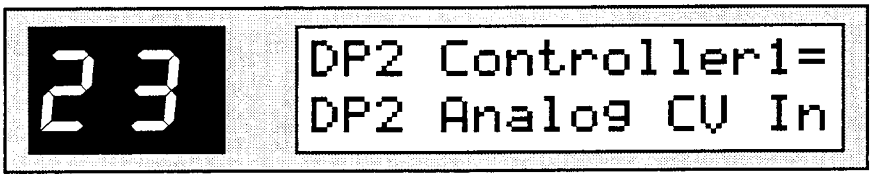
These four parameters set the minimum and maximum amount (based on a percentage of the selected parameter's range) that the Mod Destination will be modulated by the Mod Source. Inverting the amounts will also invert the modulation effect.

The eight different Mod Source controllers (two for each unit) are assigned in System/MIDI mode and are explained in more detail in *Section 4 — System/MIDI.*

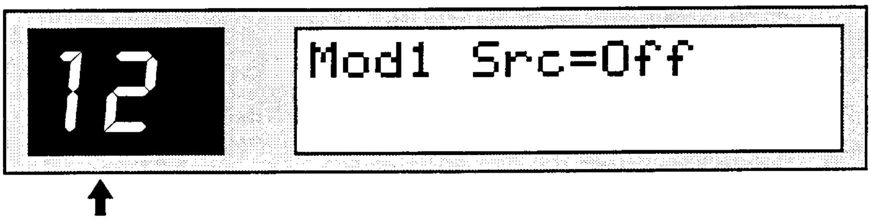
Modulating Effects Parameters with the CV Pedal

Almost any parameter of any algorithm can be modulated using one of the eight programmable controllers which you select in System/MIDI mode. This application will show you how to set up the optional CV Pedal to alter some aspect of an effect algorithm in real time. Let's suppose you want to use the CV Pedal to control the wet/dry mix of the effect in one of the units:

1. Press (SYSTEM/MIDI) until the display shows parameter #23:

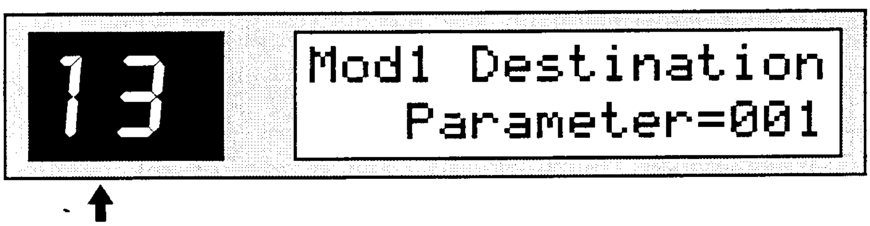


2. Set the "DP2 Controller1" parameter to "DP2 Analog CV In" as shown above (you will use this and the next seven parameters to define the eight controllers that the DP/2 will recognize as modulators).
3. Press (EDIT), and select the unit (A or B) whose mix you want to modulate using the CV Pedal.
4. Press the (▷) button until the "Mod 1 Src" parameter is showing on the display:



Parameter Number will vary, depending on the algorithm

5. Use the **Data Entry Knob** to set the parameter to "Mod 1 Src=Cntrl 1." The lower line of the display shows the name of the controller that has been selected as "Cntrl 1" in System/MIDI mode. It should read "DP2 Analog CV In."
6. Press (▷) one time. The display shows:



Parameter Number will vary, depending on the algorithm

7. Here you select which parameter in the algorithm will be modulated by the selected controller. Since Wet/Dry Mix is parameter 01 in *every* algorithm, set it to "Mod1 Destination Parameter=001."
8. Press the (▷) button, and set the Mod 1 Param Range Min=00%.
9. Press the (▷) button, and set the Mod 1 Param Range Max=99%.
10. The CV Pedal will now control the mix of the algorithm. If you scroll left to select parameter 01, you can actually see it change as you move the pedal. Remember to write (save) this preset if you want to keep it.

Note: Variations on this procedure can be used to modulate any parameter in real time with any of the eight selectable controllers.

Crossfading Effects

In a 2 Unit or Config preset, you can crossfade between two different algorithms using some controller, such as the optional CV Pedal, MIDI mod wheel, etc. Crossfading means that one algorithm is increased as the other is decreased by a controller. In the example below, we will crossfade between a Hall Reverb (Unit A) and a Dual Delay (Unit B) in a 1 Source Config.

1. Press (SYSTEM/MIDI) until the display shows parameter #23, "DP2 Controller1," and set it to "DP2 Analog CV In" (or to whichever controller you want to use).
2. Press (EDIT), then (CONFIG). Use the ◁ button to scroll to parameter #00, and set it to 1 Source Config using the **Data Entry Knob** (if it isn't already set as a 1 Source Config).
3. Press ▷ until the display shows parameter #02, AB Unit Routing, and set it to "[A+B] Parallel" using the **Data Entry Knob.**
3. Press (A) (you should still be in Edit mode), use the ◁ button to scroll to the algorithm name, and select preset #52 (Bank 1) Hall Reverb using the **Data Entry Knob.**
4. Using the ◁ and ▷ buttons, scroll right to parameter #23, and set the Mod 1 parameters to the following:

Param #:	Parameter:	Set to:
23	Mod 1 Src=	Cntrl-1
24	Mod 1 Destination	Parameter=002
25	Mod 1 Param Range Min	00%
26	Mod 1 Param Range Max	99%

This sets DP/2 Controller 1 to modulate Unit A's volume in linear fashion — low values will reduce the volume; high values will increase it.

5. Press (B), use the ◁ button to scroll to the algorithm name, and select preset #87 Dual Delay using the **Data Entry Knob.**
6. Using the ◁ and ▷ buttons, scroll right to parameter #13, and set the Mod 1 parameters to the following:

Param #:	Parameter:	Set to:
13	Mod 1 Src=	Cntrl-1
14	Mod 1 Destination	Parameter=002
15	Mod 1 Param Range Min	99%
16	Mod 1 Param Range Max	00%

This sets DP/2 Controller 1 to modulate Unit B's volume in reverse fashion — low values will increase the volume; high values will reduce it.

Now send a signal (play an instrument) into the DP/2 and move the controller you selected (be it a CV Pedal or an external MIDI Controller) and you should hear the two effects algorithms crossfade as you move the controller.

- If the two Units are routed in parallel: crossfading will only work if the Mod 1 Destination is set to parameter #02 (Volume).
- If the two Units are routed in series: crossfading will only work if the Mod 1 Destination is set to parameter #01 (Mix).

Note: If you are using a MIDI controller for any effects modulation, make sure that "Control Chan" (System/MIDI parameter #21) is set to the same MIDI channel that you are sending the controller on, and that MIDI is Enabled for the DP/2 as a whole (parameter #22), and for each of the units (parameters 01, 08, and 15). Otherwise, the MIDI controller will not be recognized by the DP/2.

3.6 Sec DDL 2U

This two unit algorithm provides a high fidelity delay longer than three seconds. **3.6 sec DDL 2U** also allows you to record a signal and play it back as a loop. By using this "Instant Replay" feature, you can play/sing along with a looped passage, creating some interesting effects.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. Delays can sound like an echo when mixed with the original signal.

03 — 3.6 Sec Delay Time   Range: 0 to 3668 ms

This parameter sets the amount of delay time. Some interesting effects can be implemented by using a real time modulation controller for this parameter.

04 — 3.6 Sec Delay Regen   Range: 00 to 99

Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delays. A setting of 71 (with a low Delay Regen Damping setting) would offer infinite repeats. Settings higher than this would offer a runaway feedback effect.

05 — 3.6 Sec Delay Pan   Range: -99 to +99

This parameter determines the location of the delay in the stereo spectrum. A value of -99 is panned far left, whereas +99 is far right.

06 — 3.6 Sec Delay Regen Damping   Range: 00 to 99

Controls the cut off of a low pass filter on the feedback signal, which reduces the amount of high frequencies to the feedback signal. The higher the number, the more the signal is damped.

07 — 3.6 Sec Delay Mode   Range: Continuous, Loop/Muted, Loop/Record, Loop/Replay

This parameter determines whether the delay will be continuous, or the "instant replay" feature is selected. When set to Continuous, any signal entering the unit will delay. When set to Loop, you can create an "instant replay" loop using any mod source defined in parameter 08.

Loop/Muted	Any signal entering the unit will delay, but because this setting is muted, you do not hear it. This is the mode you would start with when recording a loop.
Loop/Record	In this mode, you have up to 3.6 seconds to record a signal.
Loop/Replay	Once you've recorded a signal, it will loop on the audio that was input during the recording state. You can then play over top of the loop (live playing doesn't delay).

See the next page for more information about using this parameter.

08 — DelaySet   Range: Off, Controllers 1 - 8

Determines which modulation source will be used to control the loop feature. This parameter is not operative when parameter 07 is set to Continuous. When the controller is greater than 64, it goes into record; when the controller is less than 64, it goes into replay.

The playback will go into a muted state when a recording is made that is less than 300 ms.

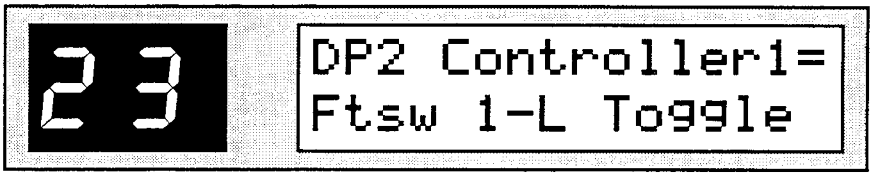
09 — Mod1 Source
10 — Mod1 Destination
11 — Mod1 Param Range Min
12 — Mod1 Param Range Max
13 — Mod2 Source
14 — Mod2 Destination
15 — Mod2 Param Range Min
16 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

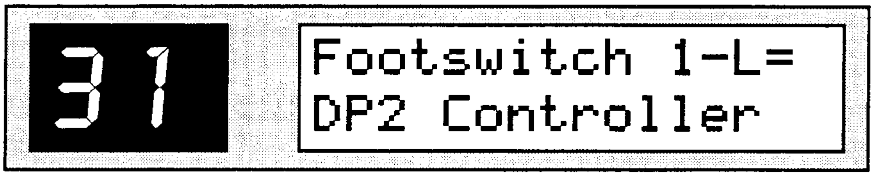
Using the Instant Replay Feature

Let's create an example using the foot switch to toggle in and out of the instant replay feature:

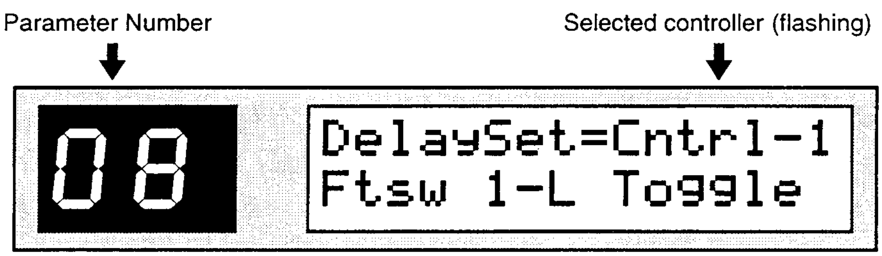
1. Press (SYSTEM/MIDI) until the display shows Parameter #23:



2. Using the **Data Entry Knob**, set the DP2 Controller1 parameter to "Ftsw 1-L Toggle."
3. Press (SYSTEM/MIDI) again. The display shows Parameter #31:



4. Using the **Data Entry Knob**, set the Footswitch 1-L to "DP2 Controller." This allows a foot switch connected to the **Foot Switch 1** jack to be used as the modulation source.
5. Press (EDIT), and use the (◁) and (▷) buttons to select parameter 08. Turn the **Data Entry Knob** and select "Ftsw 1-L Toggle." The display looks like this:



This allows you to toggle between a *recording* state and a *playback* state. In playback, it will loop on the audio which was input during the recording state (up to 3.6 seconds).

6. Press (◁) to go back to parameter 07. Turn the **Data Entry Knob** to "Mode=Loop/Muted."
7. Press the foot switch and the display shows "Mode=Loop/Record." You now have up to 3.6 seconds to record a passage.
8. Press the foot switch again. The display says "Mode=Loop/Replay." You should be hearing the passage you just played continually looping. By double-clicking the foot switch (pressing rapidly two times) you can return the display to "Mode=Loop/Muted."

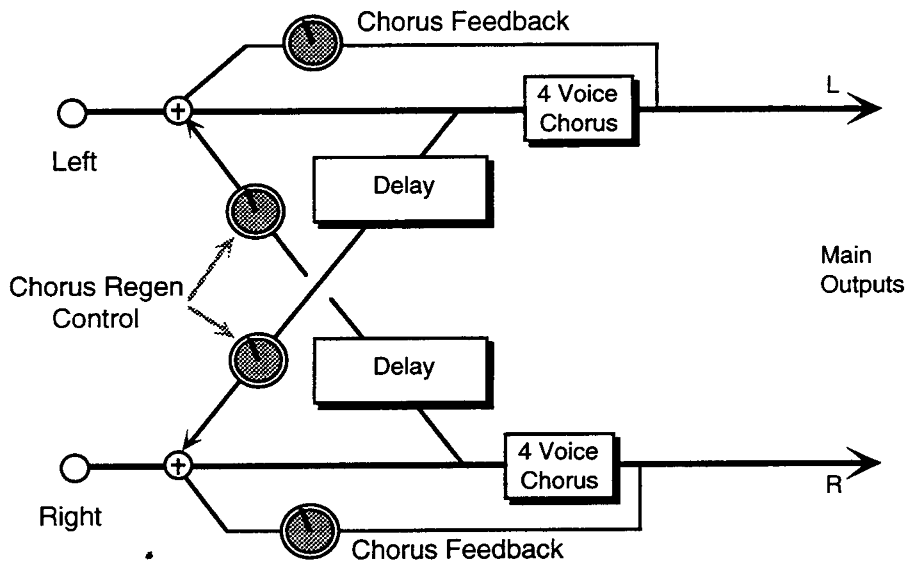
The regen parameter should be set to 71 to get infinite repeat during playback. Use less regen to have the repeats eventually fade away. Use more regen if you like runaway feedback and distortion, or if you use the damping parameter. The amount of regen is also dependent on the pan of the delay.

> **Note:** If you leave it in a recording state for longer than 3.6 seconds, then when you finally do go back to playback, it will retain the previous "legal" setting (the last 3.6 seconds).

8 Voice Chorus

8 Voice Chorus offers a symphonic chorused sound having eight different voices and using eight separately randomized LFOs. This algorithm also offers a programmable stereo delay in a cross-coupled configuration between the left and right chorused outputs (see diagram). This algorithm is good for creating an ensemble of instruments from single sources (there is no internal filtering applied to any of the chorused voices).

8 Voice Chorus Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. For this algorithm we recommend a Mix of around 50 for a starting point.

03 — 8V Chorus LFO Rate   Range: 00 to 99

Controls the rate of pitch modulation applied to the delays.

04 — 8V Chorus LFO Width   Range: 00 to 99

This parameter controls the intensity of pitch modulation applied to the delays.

05 — 8V Chorus Stereo Spread   Range: 00 to 99

This parameter offers control of the synthesized stereo field. The highest value is true stereo, intermediate values have the left and right signals mixed on both sides, and the lowest value yields only the left input channel from the right and left outputs. This parameter, though not a stereo pan, provides some interesting stereo effects when controlled by a modulation source.

06 — 8V Chorus Regen   Range: 00 to 99

Determines the amount of signal that will be fed from the output of the chorus back into the input of the chorus. A value of 00 will eliminate the regeneration effect.

07 — 8V Chorus Left Regen Time   Range: 0 to 800 ms

Controls the amount of time that the non-chorused signal will delay for the left channel.

08 — 8V Chorus Right Regen Time   Range: 0 to 800 ms

Controls the amount of time that the non-chorused signal will delay for the right channel.

09 — 8V Chorus Delay Regen   Range: 00 to 99

Determines the amount of signal that will be fed from the delay output back into the chorus input, increasing the number of repeats in the delay for high values (see diagram). A value of 00 will eliminate the delay effect.

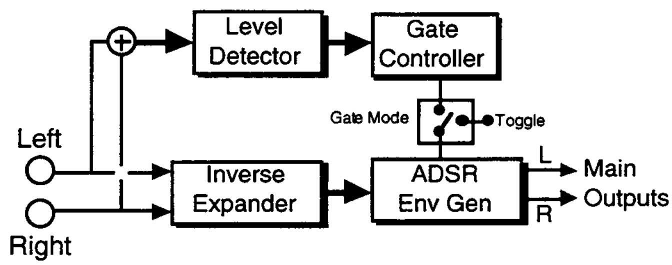
10 — Mod1 Source
11 — Mod1 Destination
12 — Mod1 Param Range Min
13 — Mod1 Param Range Max
14 — Mod2 Source
15 — Mod2 Destination
16 — Mod2 Param Range Min
17 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators covered earlier in this section.

ADSR Env Gen

ADSR Env Gen is used to "reshape" the volume characteristics of the input signal, creating new, refreshing tonal variations. Guitarists will find this algorithm useful in creating the classic "volume swell" effect, or it can be used to create dynamic staccato effects.

ADSR Env Gen Signal Routing



The level detector (an envelope follower) determines at what level the envelope will trigger, and is used to make a gate out of the input signal. The inverse expander within the algorithm is used to level the dynamics from the input signal prior to shaping via the ADSR envelope generator. There is also a dry signal (not shown) that goes directly from the input to the output and is controlled with the Mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

03 — Level Detector Off Below   Range: -96 to +00 dB

This parameter sets the lower threshold level at which the gate closes.

04 — Level Detector On Above   Range: -96 to +00 dB

Sets the upper threshold at which the gate opens. This higher second threshold prevents gate chatter.

05 — Level Detector Attack   Range: 50µs to 100ms

Sets the attack of the level detector/envelope follower once the incoming signal has been detected (i.e. determines how closely the attack is followed). Generally the attack should be short.

06 — Level Detector Release   Range: 1ms to 10.0s

Sets the amount of time after the incoming signal has ceased for the level detector to shut down. Generally these times are longer than the attack times.

07 — Expnd Ratio   Range: 1:1 to 40:1, infinity

Sets the amount of expansion. Expansion occurs below the threshold. If this is set to 3:1 for example, it will expand the change in signals below the threshold by three times in an attempt to make the signal amplitude approach the threshold level.

08 — Threshold   Range: -96 to +00 dB

Sets the inverse expander threshold level. Signals beneath this level will be expanded, while signals that are above will be unaffected. As the input signal dies away below the threshold, the expander will increase the gain of the signal.

09 — Exp Attack
Range: 50µs to 100ms

Determines the time after the initial signal amplitude has been detected for the expansion to occur.

10 — Release
Range: 1ms to 10.0s

Determines how long it takes for the expansion to be fully deactivated after the input signal rises above the threshold level. This is generally longer than the attack time.

11 — Gate Mode
Range: Auto or Manual

Detemines how the level detector and gate controller will track the signal. Auto mode follows the input signal, Manual mode allows you to key in a gate using the Toggle parameter.

12 — Toggle
Range: Off or On

Used to turn on and off the gate function when the Gate Mode is set to Manual.

13 — A
Range: 50µs to 10 sec

Sets the attack time for the ADSR Envelope shape.

14 — D
Range: 50µs to 10 sec

Sets the decay time for the ADSR Envelope shape.

15 — S
Range: 00 to 99

Sets the sustain level for the ADSR Envelope shape.

16 — R
Range: 50µs to 10 sec

Sets the release time for the ADSR Envelope shape.

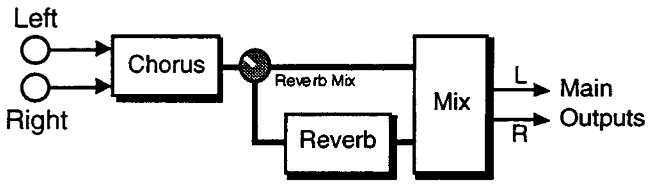
17 — Mod1 Source
18 — Mod1 Destination
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Chorus-Reverb

Chorus-Reverb combines a chorus with a large plate reverb.

Chorus-Reverb Signal Routing



The signal enters a stereo chorus, which is heard directly at the output. There is a Chorus Mix parameter (within the chorus) that determines a wet/dry mix. This signal is then routed out of the chorus into the large plate reverb. There is also a dry signal (not shown) that goes directly from the input to the output and is controlled with the Mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

03 — Reverb Mix   Range: 00 to 99

Controls the mix between the chorused signal and the reverb. Setting this parameter to 00 will allow only the chorused signal to be heard, while a setting of 99 will send all of the chorused signal to the reverb.

04 — Chorus LFO Rate   Range: 00 to 99

Controls the rate of pitch modulation to the chorus.

05 — Chorus LFO Width   Range: 00 to 99

This controls the amount of pitch deviation from the center point. A larger number pulls the pitch farther and farther away from center creating a wider modulation (larger, more obvious pitch change). Keep in mind that the width of pitch modulation is affected by the rate; as the rate is increased, the apparent pitch modulation is also increased.

06 — Chorus Center   Range: 00 to 99

Controls the nominal delay time of the chorus about which the delay modulation occurs. Adjusting this parameter will change the tonal character of the effect.

07 — Chorus Feedback   Range: -99 to +99

Controls the amount of regeneration applied to the chorused taps. The sign of the value determines the polarity of the regen. The polarity affects the tonal quality of the regeneration.

08 — Chorus Mix   Range: 00 to 99

Controls the Dry/Wet mix within the chorus itself. For starters, we recommend settings of 50.

09 — Large Plate Decay   Range: 0.40 to 140.0 sec

Controls the amount of time it takes for the reverberation to decay.

10 — Plate Predelay Time   Range: 0 to 250 ms

Controls the amount of time it takes for the original signal to be presented to the reverb. Higher values denote a longer delay.

11 — Large Plate HF Damping   Range: 00 to 99

Shapes the tone of the reverb decay. Higher settings cause the high frequency components to decay more rapidly.

12 — Large Plate HF Bandwidth   Range: 00 to 99

This parameter acts as a low pass filter on the input of the plate reverb, controlling the amount of high frequencies present. The higher the setting, the more high frequencies are allowed to pass through, offering a brighter ringing sound. Some interesting effects can be created by using a mod controller over a large range.

13 — Plate Diffsn1   Range: 00 to 99

This parameter smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

14 — Diffusion2   Range: 00 to 99

This parameter, similar to and in series with Plate Diffsn1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.

15 — Plate Decay Definition   Range: 00 to 99

Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to increase faster then the reverb decay, which can result in audible ringing.

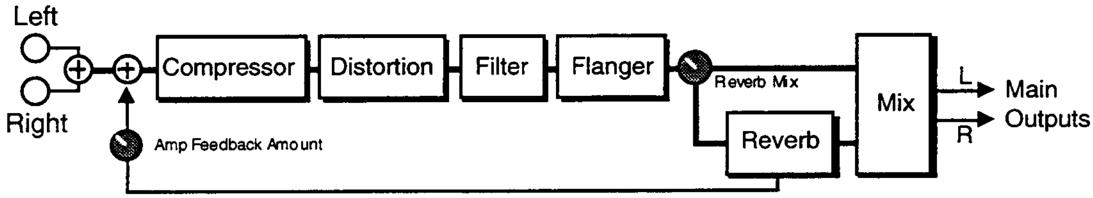
16 — Mod1 Source
17 — Mod1 Destination
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

CmprDstFlngRev

A screaming guitar effect that features not only compression, distortion, and reverb, but a flanger and high pass/low pass EQ as well.

CmprDstFlngRev Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

03 — Reverb Mix   Range: 00 to 99

Controls the mix between the original signal and the reverb. Setting this parameter to 00 will allow the signal to be heard with no reverb, while a setting of 99 will eliminate the original signal completely, with only the reverb portion remaining.

04 — Compressor Threshold   Range: -96 to +00 dB

Controls the threshold level for the compressor. As the input signal dies away below the threshold, the compressor will increase the gain of the signal, causing feedback to increase as well. To turn off, set the threshold to -96dB.

05 — Comp Attack   Range: 50µs to 100ms

Determines the attack rate after the initial signal has been detected and before the compression takes affect.

06 — Comp Release   Range: 1ms to 10.0s

Determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally set longer than the attack time (parameter 08).

07 — Distortion Level In
08 — Distortion Level Out   Ranges: 00 to 99

These two parameters control the levels going into and coming out of the distortion effect. The Distortion Level In adjusts the *intensity* of distortion. The Distortion Level Out parameter affects the output level.

09 — HighPass Fc   Range: 4 to 8000 Hz

Filters out low frequencies after the distortion signal path. The higher the value, the fewer low frequencies pass through, the thinner the sound.

10 — LowPass Fc   Range: 100 to 16 K

Filters out high frequencies after the distortion signal path. The lower the value, the fewer high frequencies pass through, the darker the sound.

11 — Amp Feedback Amount Range: -99 to +99

Controls the amount of signal applied from the output of the reverb back into the input of the compressor. The sign of the value determines the polarity of the feedback.

12 — Flanger LFO Rate Range: 00 to 99

Controls the LFO rate of the flanger.

13 — Flanger LFO Width Range: 00 to 99

Controls the width of the LFO.

14 — Flanger Center Range: 00 to 99

This parameter controls the sweep center of the flanger effect.

15 — Flanger Feedback Range: -99 to +99

Controls the amount of feedback applied from the output to the input of the flanger. The sign of the value determines the polarity of the feedback.

16 — Flanger Mix Range: 00 to 99

Controls the mix between the dry signal and the flanger. Setting this parameter to 00 will allow only the unflanged signal to be heard, while a setting of 99 will eliminate it completely, with only the flanger portion remaining.

17 — Reverb Decay Range: 0.20 to 100.0 sec

Controls the amount of time it takes for the reverberation to decay to a very low level (-60dB) after the input signal stops.

18 — Reverb HF Damping Range: 00 to 99

Controls the high frequency decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will filter out increasing amounts of high frequency energy.

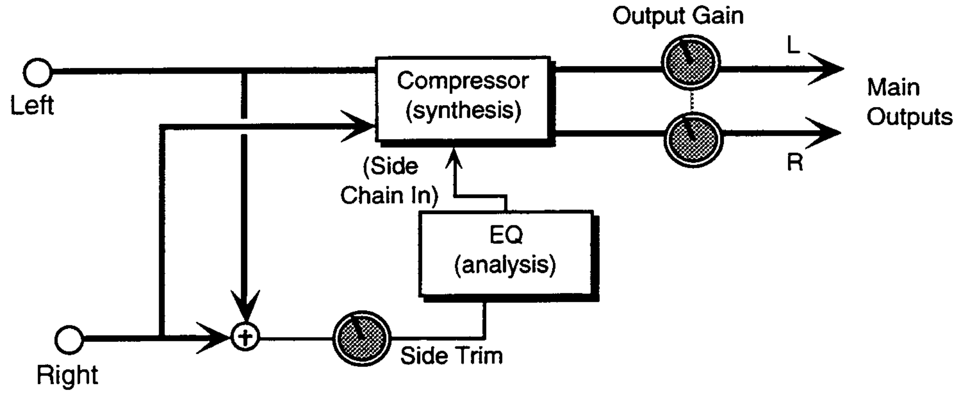
19 — Mod1 Source
20 — Mod1 Destination
21 — Mod1 Param Range Min
22 — Mod1 Param Range Max
23 — Mod2 Source
24 — Mod2 Destination
25 — Mod2 Param Range Min
26 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

De-esser

De-esser is a stereo algorithm that compresses sibilant frequencies (like the "ess" sound) as they become louder. This was designed for vocalists, but it can also be used to control the boomy sound of a guitar or the ringing sound of drums by adjusting the side-chain equalization appropriately. There is no EQ in the audio path; equalization is provided on the side chain only.

De-esser Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. For this algorithm we recommend a setting of 99.

03 — Output Gain   Range: -48 to +48 dB

Sets the amount of make up gain after the compression effect has reduced the gain. We recommend a starting application of +00 dB.

04 — Comp Ratio   Range: 1:1 to 40:1, infinity

Sets the amount of compression. The range is based on decibels (dB) above the threshold. If this is set to 4:1 for example, it will allow 1 dB increase in output level for every 4 dB increase in input level. When this is set to infinity, it acts as a limiter.

05 — Threshold   Range: -96 to +00 dB

Sets the threshold level — signals that exceed this level will be compressed, while signals that are below will be unaffected.

06 — Gain Change   Range: N/A

This read-only parameter displays a gain reduction meter.

07 — Comp Attack   Range: 50µs to 100ms

Determines the attack rate after the initial signal has been detected and before the compression takes affect.

08 — Comp Release   Range: 1ms to 10.0s

Determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally set longer than the attack time (parameter 07).

09 — Noise Gate Off Below   Range: -96 to +00 dB

This parameter sets the lower threshold level at which the noise gate shuts off the audio.

10 — Noise Gate On Above Range: -96 to +00 dB

Sets the upper threshold level at which the noise gate passes audio. This higher second threshold prevents false gate chatter.

11 — Sidechain EQ HighPass Fc Range: 4 to 8000 Hz

Controls a high pass filter frequency for the side chain EQ. This is useful for de-essing.

12 — Bass Fc Range: 0 to 1000 Hz

Sets the cutoff frequency of the lower frequency band shelving EQ.

13 — Bass Gain (loShv) Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low shelving EQ.

14 — Mid1 Fc Range: 100 to 9999 Hz

Sets the center of the mid-frequency EQ. Higher values have a brighter sound.

15 — Mid1 Gain Range: -48 to +24 dB

Sets the amount of boost or cut applied to this frequency.

16 — Mid1 Q Range: 01 to 18

This is a bandwidth control that determines the range of affected frequencies. This is equal to the cutoff frequency divided by the bandwidth. By raising the Q, you can produce a narrower bandwidth.

17 — Mid2 Fc
18 — Mid2 Gain
19 — Mid2 Q

These three parameters are identical to the previous three parameters, and can be used to control different bandwidths within the mid range.

20 — Treble Fc Range: 01KHz to 16KHz

Sets the cutoff frequency of the high shelving EQ.

21 — Treble Gain (HiShv) Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high shelving filter.

22 — Sidechain EQ Input Trim Range: -48 to +00 dB

Adjusts the input level to the side chain EQ.

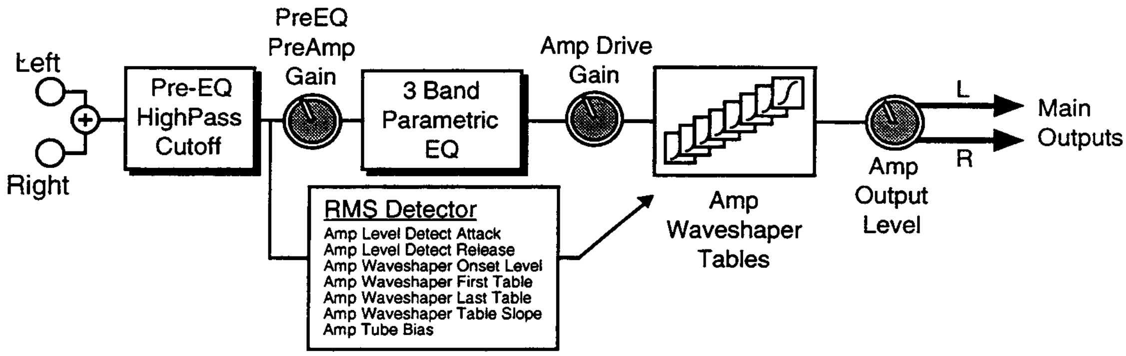
23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

DigitalTubeAmp

DigitalTubeAmp is a state-of-the-art digital guitar amp simulation. Whereas the other DP/2 Guitar Amp algorithms are based on one waveshaping table, DigitalTubeAmp offers eight different wave tables that you can dynamically change by increasing/decreasing the signal level (playing your guitar harder/softer). We highly recommend following this algorithm in series with TunableSpeaker 2.

DigitalTubeAmp Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Pre-EQHighPass Cutoff Range: 4 to 1000 Hz

Filters out the low frequencies before the preamp. The higher the value, the fewer low frequencies pass through.

04 — PreEQ PreAmp Gain Range: -42 to +48 dB

Adjusts the initial gain of the incoming signal. This parameter can be thought of as the primary distortion stage (clipping). We recommend a setting of 00 dB, since these emulations were optimized for distortion there. Lower preamp gains will result in less distortion, while higher preamp gains will yield clipping distortion. For low preamp gain, it may be desirable to use low tube bias values.

05 — Pre-EQ1 Fc Range: 5 to 9999 Hz

This parameter determines the center frequency of the parametric filter before the preamp. Higher values have a brighter sound.

06 — Pre-EQ1 Gain Range: -48 to +24 dB

Adjusts the amount of boost or cut applied to the parametric filter in front of the preamp.

07 — Pre-EQ1 Q Range: 01 to 18

Determines the width of the resonant peak at the parametric filter center frequency. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the width of the peak.

08 — Pre-EQ2 Fc
09 — Pre-EQ2 Gain
10 — Pre-EQ2 Q
11 — Pre-EQ3 Fc
12 — Pre-EQ3 Gain
13 — Pre-EQ3 Q

These parameters are identical to the previous ones, but control a second and third parametric filter before the amp.

14 — Amp Drive Gain — Range: -48 to +48dB

Adjusts the amount of boost or cut applied to the signal after the EQ. This parameter also produces distortion (clipping), but it is much less than the Pre-EQ PreAmp Gain Parameter (number 04). This parameter, in combination with Pre-EQ PreAmp Gain, produces harmonics, which are sums and differences of input frequencies, which is called "intermodulation distortion."

15 — Amp Level Detect Attack — Range: 50µs to 100ms

Sets the attack time of the RMS measurement of the input signal. RMS level determines which table is used. Generally the attack should be short.

16 — Amp Level Detect Release — Range: 1ms to 10.0s

Sets the release time of the RMS measurement after it determines which table is used. Generally these times are longer than the attack times.

17 — Amp Waveshaper Onset Level — Range: -64 to +00

This is used to set the level at which the first table kicks in.

18 — Amp Waveshaper First Table — Range: 00 to 07

Determines which table will begin when the input signal reaches the level set with the Amp Level Detect Attack parameter (number 15).

19 — Amp Waveshaper Last Table — Range: 01 to 07

Sets the highest table that you will reach. This is used to define the overall sound. A broader table range offers a more dynamic sound.

20 — Amp Waveshaper Table Slope — Range: 001 to 127

Determines how fast you will switch from one table to the next.

21 — Amp Tube Bias — Range: 00 to 99

For preamp gains approximately 00 dB, this parameter controls the emphasis of even to odd harmonics which determines the tone of the amp; mid values emphasize even harmonics and offer a warmer ("glowing tube") sound, while the highest values may sound like tubes going bad. Tube bias and preamp gain are independent parameters. For low preamp gain, it may be desirable to use low tube bias values, because this more closely imitates the operation of a real amplifier.

22 — Amp Output Level — Range: 00 to 99

This parameter controls the output level of the main amp.

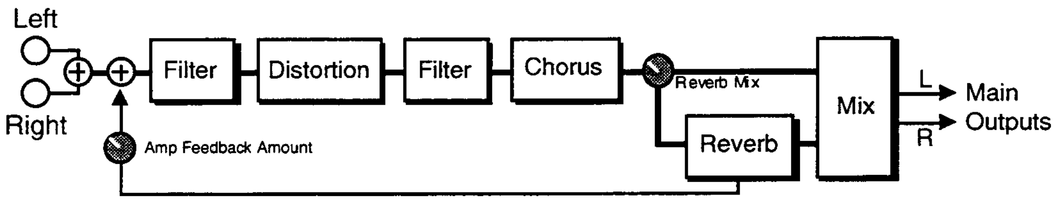
23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

Dist-Cho-Revrb

A bright guitar-effects chain-amp simulator that features distortion, chorus, and a plate reverb.

Dist-Cho-Revrb Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

03 — Reverb Mix Range: 00 to 99

Controls the mix between the processed signal and the reverb. When set to 00, the processed signal will be heard with no reverb, while a setting of 99 will send all of the processed signal to the reverb.

04 — Pre-Distortion VCF Fc Range: 01 to 99

Determines the filter cut off-frequency before the distortion. Higher values have a brighter sound. This parameter can be modulated, using a CV Pedal for a wah-wah pedal effect. To disable the distortion filter, set this parameter to 99, and set the Q parameter to 01.

05 — Pre-Distortion VCF Q Range: 01 to 30

Determines the level and width of the resonant peak at the filter cutoff point. While the Pre-Distortion VCF Fc parameter determines where (at what-frequency) this peak will occur, this parameter controls the *sharpness* of the peak. This setting is important for the auto-wah effect.

06 — Distortion Level In
07 — Distortion Level Out Ranges: 00 to 99

These two parameters control the levels going into and coming out of the distortion effect. The Distortion Level In adjusts the *intensity* of distortion. The Distortion Level Out parameter affects the output level.

08 — Distortion Mix Range: 00 to 99

Determines the Dry/Wet, or in this case, dirty/clean mix of the signal. A value of 00 yields a clean signal; 99 yields an all distortion signal.

09 — Amp Feedback Amount Range: -99 to +99

Controls the amount of signal applied from the output of the reverb back into the input of the compressor. The sign of the value determines the polarity of the feedback.

10 — Post-Distortion VCF Fc
11 — Post-Distortion VCF Q

These parameters are identical to the Pre-Distortion VCF Fc and Q parameters, and are used to control the second VCF that exists after the distortion.

12 — Chorus LFO Rate Range: 00 to 99

Simultaneously controls the rate of the four independent LFOs which modulate a short delay, creating the chorus effect. The delay modulation produces vibrato and tremolo.

13 — Width Range: 00 to 99

Controls the depth of vibrato and tremolo.

14 — Chorus Center Range: 00 to 99

Controls the nominal delay time of the chorus about which the delay modulation occurs. Adjusting this parameter will change the tonal character of the effect.

15 — Chorus Mix Range: 00 to 99

Controls the Dry/Wet mix within the chorus itself. For starters, we recommend settings of 50.

16 — Large Plate Decay Range: 0.40 to 140.0 sec.

Controls the amount of time it takes for the reverberation to decay. High values of decay sound good with this algorithm.

17 — Plate Predelay Time Range: 0 to 250 ms

Controls the amount of time it takes for the original signal to be presented to the reverb. Higher values denote a longer delay.

18 — Large Plate HF Damping Range: 00 to 99

Functions as a tone control and boosts or cuts the rate at which low frequencies will decay.

19 — Large Plate HF Bandwidth Range: 00 to 99

This parameter acts as a low pass filter on the output of the plate reverbs, controlling the amount of high frequencies present. The higher the setting, the more high frequencies are allowed to pass through, offering a brighter ringing sound. Some interesting effects can be created by using a mod controller over a large range.

20 — Plate Diffsn1 Range: 00 to 99

This parameter smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

21 — Diffusion2 Range: 00 to 99

This parameter, similar to and in series with Diffusion1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.

22 — Plate Decay Definition Range: 00 to 99

Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.

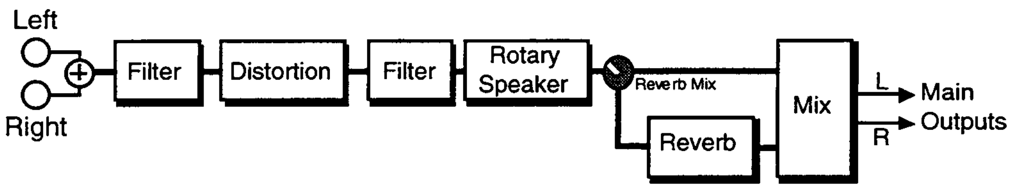
23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Dist-Roto-Revb

This effect chain combines a warm distortion with a rotary speaker and a reverb.

Dist-Roto-Revb Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

03 — Reverb Mix   Range: 00 to 99

Controls the mix between the original signal and the reverb. Setting this parameter to 00 will allow only the unprocessed dry signal to be heard, while a setting of 99 will eliminate it completely, with only the reverb portion remaining.

04 — Pre-Distortion LowPass Fc   Range: 100 Hz to 16 KHz

Attenuates the high frequency content of the signal driving the distortion at a rate of 6dB per octave starting at the corner frequency set by this parameter.

The high-frequency bandwidth acts as a low pass filter on the signal going into the distortion, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass. This functions like a tone control on a guitar.

05 — Distortion Level In
06 — Distortion Level Out   Ranges: 00 to 99

These two parameters control the levels going into and coming out of the distortion effect. The Distortion Level In adjusts the *intensity* of distortion. The Distortion Level Out parameter affects the output level.

07 — Post-Distortion VCF Fc   Range: 01 to 99

Determines the filter cut off-frequency after the distortion. Higher values have a brighter sound. This parameter can be used to emulate a speaker cabinet. To disable the distortion filter, set this parameter to 99, and set the Q parameter to 01.

08 — Post-Distortion VCF Q   Range: 01 to 30

Determines the level and width of the resonant peak at the filter cutoff point. While the Post-Dist VCF Fc parameter determines where (at what-frequency) this peak will occur, this parameter controls the *sharpness* of the peak.

09 — Distortion Mix   Range: 00 to 99

Controls the mix between the original signal and the distortion. Setting this parameter to 00 will allow only the unprocessed signal to be heard, while a setting of 99 will eliminate it completely, with only the distortion portion remaining.

10 — Rotor Speed Range: Slow or Fast

Determines how the rotating speaker will switch between slow and fast speeds. The behavior of this switch accurately reflects an actual rotary speaker, taking time to speed up or slow down, based on the value of the Inertia parameter (13). By assigning a modulation controller to this parameter, you can change between the slow and fast speeds in real time.

11 — Slow Range: 00 to 99

Determines the rate of the rotary speaker when in the "Slow" setting. This parameter determines the manual level for the rotary speaker rate when Rotor Speed=Slow, or when the selected modulator is at zero output level. Again, the higher the value, the faster the rate.

12 — Fast Range: 000 to 130

Determines the rate of the rotary speaker when in the "Fast" setting. The higher the value, the faster the rate.

13 — Rotating Speaker Inertia Range: 1 ms to 10.0 s

Determines how long it will take for the rotor effect to speed up and slow down after switching from fast to slow or vice versa. Adjust this parameter to simulate the effect of the rotary speaker gradually picking up speed.

14 — Tremolo Depth Slow
15 — Tremolo Depth Fast Ranges: 00 to 99

Control the slow and fast settings for how much the volume will fade away as the speaker rotates away from the listener. Broader ranges will create a deeper rotating speaker effect.

16 — Vibrato Depth Slow
17 — Vibrato Depth Fast Ranges: 00 to 99

These two parameers control the slow and fast settings for the amount of detuning applied to the rotating speaker. This can be used to create a "doppler" effect.

18 — Rotating Speaker Mix Range: 00 to 99

Adjusts the volume of the rotary speaker signal. A level of 00 will offer no audible speaker movement.

19 — Rotating Speaker Stereo Spread Range: 00 to 99

This parameter offers a synthesized stereo field. The highest and lowest values are true stereo, intermediate values have the left and right signals mixed on both sides. This parameter, though not a stereo pan, provides some interesting stereo effects when controlled by a modulation source.

20 — Reverb Decay Range: 0.20 to 100.0 sec

Controls the amount of time it takes for the reverberation to decay to a low level (-60 dB) after the input signal stops.

21 — Reverb HF Damping Range: 00 to 99

Controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will filter out increasing amounts of high-frequency energy.

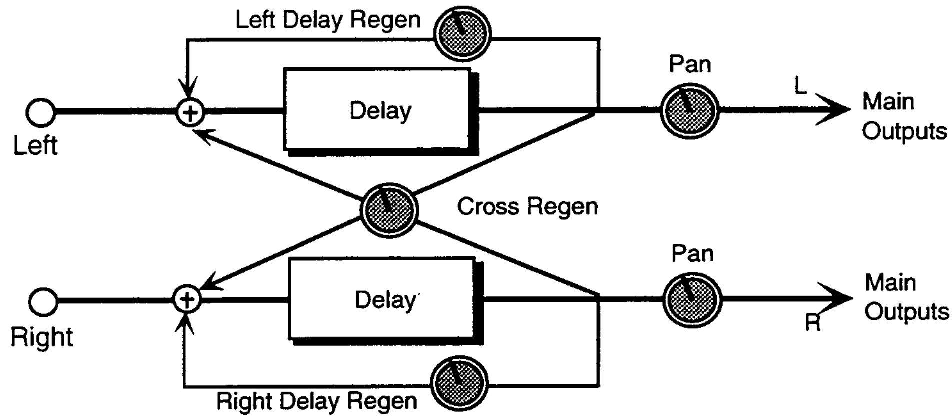
22 — Mod1 Source
23 — Mod1 Destination
24 — Mod1 Param Range Min
25 — Mod1 Param Range Max
26 — Mod2 Source
27 — Mod2 Destination
28 — Mod2 Param Range Min
29 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Dual Delay

Dual Delay features a professional quality high fidelity stereo digital delay. This algorithm splits the available memory into two equal delay lines to retain a true stereo image within the delay.

Dual Delay Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. The Dual Delay sounds best with a Mix of wet and dry.

03 — Left Input Delay Time   Range: 0 to 840 ms

Determines the amount of delay time between the original signal and the left input delay.

04 — Left Input Delay Time (fine)   Range: 0.00 to 0.99 ms

Acts as a fine tune (in milliseconds) for the delay time between the original signal and the left input delay.

05 — Left Input Delay Regen   Range: 00 to 99

This parameter determines the amount of signal from the left delay time that will be fed from the output back into the input, increasing the number of repeats in the delay.

06 — Left Input Delay Pan   Range: -99 to +99

This parameter determines the location of the left input delay in the stereo spectrum. A value of -99 is panned far left, whereas +99 is far right.

07 — Right Input Delay Time
Range: 0 to 840 ms

Determines the amount of time between the original signal and the right input delay.

08 — Right Input Delay Time (fine)
Range: 0.00 to 0.99 ms

Acts as a fine tune (in milliseconds) for the delay time between the original signal and the right input delay.

09 — Right Input Delay Regen
Range: 00 to 99

Determines the amount of signal from the right delay time that will be fed from the output back into the input, increasing the number of repeats in the delay.

10 — Right Input Delay Pan
Range: -99 to +99

This parameter determines the location of the right input delay in the stereo spectrum. A value of -99 is panned far left, whereas +99 is far right.

11 — Dual Delay Cross Regen
Range: -99 to +99

This parameter allows you to feedback the delayed signals to their opposite sides (if both delay pans are set to opposite values); the left voice crosses to the right voice, and the right voice crosses to the left voice. A setting of +99 or -99 will cause infinite delay.

12 — Dual Delay Regen Damping
Range: 00 to 99

Controls the cut off of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The higher the number, the more the signals are damped.

13 — Mod1 Source
14 — Mod1 Destination
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

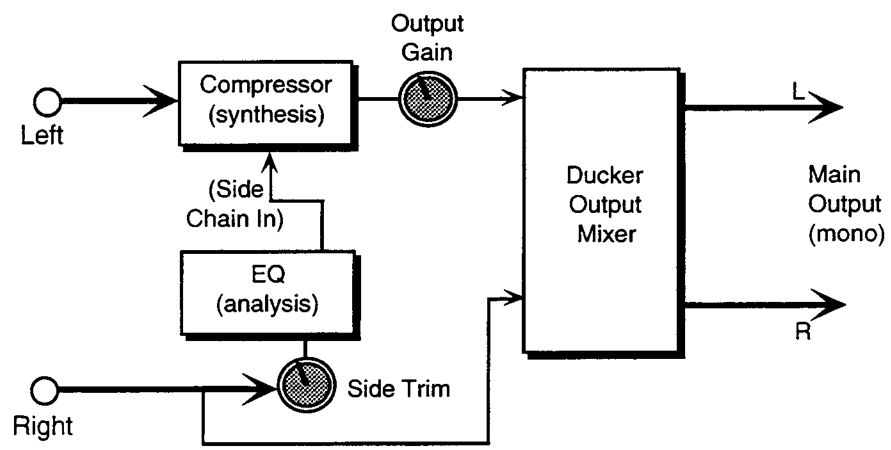
Ducker / Gate

Ducker / Gate is a compressor that automatically lowers the level of a signal (such as music) when another signal (like an announcer voice-over) comes in. When the voice-over leaves, the level of the original signal is restored. This algorithm is useful for voice-overs, Rap, and DJ work. In order for this algorithm to work properly, the music (audio source to be ducked) must be plugged into Input 1 (left), and the voice-over is plugged into Input 2 (right). In this set up, Input 2 is considered the side chain to a traditional compressor. This algorithm employs an internal mixer which mixes left and right inputs to a mono output.

The gate function is achieved for high compression ratios. In this application, a transient signal source, such as a snare drum, can gate on and off some other music signal in Input 1, to achieve an externally controllable staccato effect.

☞ **Important:** This special algorithm is only made available in the DP/2 as a ROM Config Preset (location #89 in Bank 1), because it requires special input signal routing.

Ducker / Gate Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, earlier in this section. The mixer works as in all the other algorithms, and is different from the ducker output mixer, shown above.

03 — Output Gain   Range: -48 to +48 dB

Sets the amount of cut (negative values) or boost (positive values) applied to the ducker on the output volume. We recommend a starting application of +00 dB.

04 — Ducker Output Mix   Range: 00 to 99

Mixes the output of the music signal (Input 1) with the output of the ducker (voice-over) signal (Input 2) to a mono output. This is the internal mixer which is shown above.

05 — Comp Ratio   Range: 1:1 to 40:1, infinity

Sets the amount of compression. The range is based on decibels (dB) above the threshold. If this is set to 4:1 for example, it will allow 1 dB increase in output level for every 4 dB increase in input level. When this is set to infinity, it acts as a limiter.

06 — Threshold   Range: -96 to +00 dB

Sets the threshold level. Signals that exceed this level will be compressed, while signals that are below will be unaffected.

07 — Gain Change   Range: N/A

This read-only parameter displays a gain reduction meter.

08 — Comp Attack
Range: 50µs to 100ms

Determines the attack rate after the initial signal has been detected and before the compression takes affect.

09 — Comp Release
Range: 1ms to 10.0s

Determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally set longer than the attack time (parameter 08).

10 — Noise Gate Off Below
Range: -96 to +00 dB

Sets the lower threshold level at which the noise gate shuts off the audio.

11 — Noise Gate On Above
Range: -96 to +00 dB

Sets the upper threshold level at which the noise gate passes audio. This higher second threshold prevents false "turn ons."

12 — Bass Fc
Range: 0 to 1000Hz

Sets the cutoff frequency of the lower frequency band shelving filter.

13 — Bass Gain (loShv)
Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low shelving filter.

14 — Mid1 Fc
Range: 100 to 9999 Hz

Sets the center of the mid-frequency parametric. Higher values have a brighter sound.

15 — Mid1 Gain
Range: -48 to +24 dB

Sets the amount of boost or cut applied to this frequency parametric.

16 — Mid1 Q
Range: 01 to 18

This bandwidth control determines the width of the resonant peak at the center of the frequency band. Higher values produce a narrower bandwidth.

17 — Mid2 Fc
18 — Mid2 Gain
19 — Mid2 Q

These three parameters are identical to the previous three parameters, and can be used to control different bandwidths within the mid range.

20 — Treble Fc
Range: 01KHz to 16KHz

Sets the cutoff frequency of the upper frequency band high shelving EQ.

21 — Treble Gain (HiShv)
Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high shelving filter.

22 — Side Chain EQ Input Trim
Range: -48 to +00 dB

Adjusts the input level to the side chain EQ, which performs analysis on the input signal so as to selectively compress it.

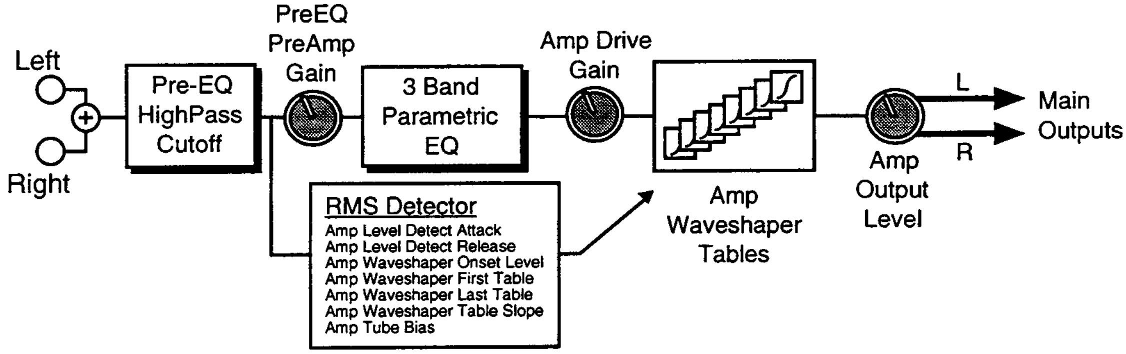
23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

DynamicTubeAmp

DynamicTubeAmpis identical to DigitalTubeAmp, but the order of the eight amp waveshape tables is reversed. We recommend following this algorithm in series with Tunable Speaker2.

DynamicTubeAmp Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Pre-EQHighPass Cutoff Range: 4 to 1000 Hz

Filters out the low frequencies before the preamp. The higher the value, the less low frequencies pass through.

04 — PreEQ PreAmp Gain Range: -42 to +48 dB

Adjusts the amount of boost or cut applied to the incoming signal. This parameter can be thought of as the primary distortion stage (clipping). We recommend a setting of 00 dB, since these emulations were optimized for distortion there. Lower preamp gains will result in less distortion, while higher preamp gains will yield clipping distortion. For low preamp gain, it may be desirable to use low tube bias values.

05 — Pre-EQ1 Fc Range: 5 to 9999 Hz

This parameter determines the center frequency of the parametric filter before the preamp. Higher values have a brighter sound.

06 — Pre-EQ1 Gain Range: -48 to +24 dB

Adjusts the amount of boost or cut applied to the parametric filter in front of the preamp.

07 — Pre-EQ1 Q Range: 01 to 18

Determines the width of the resonant peak at the parametric filter center frequency. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

08 — Pre-EQ2 Fc
09 — Pre-EQ2 Gain
10 — Pre-EQ2 Q
11 — Pre-EQ3 Fc
12 — Pre-EQ3 Gain
13 — Pre-EQ3 Q

These parameters are identical to the previous ones, but control a second and third parametric filter before the preamp.

14 — Amp Drive Gain Range: -48 to +48dB

Adjusts the amount of boost or cut applied to the signal after the EQ. This parameter also produces distortion (clipping), but it is much less than the Pre-EQ PreAmp Gain Parameter (number 04). This parameter, in combination with Pre-EQ PreAmp Gain, produces sums and differences of harmonics, which is called "intermodulation distortion."

15 — Amp Level Detect Attack Range: 50µs to 100ms

Sets the attack time of the RMS measurement of the input signal. RMS level determines which table is used. Generally the attack should be short.

16 — Amp Level Detect Release Range: 1ms to 10.0s

Sets the release time of the RMS measurement after it determines which table is used. Generally these times are longer than the attack times.

17 — Amp Waveshaper Onset Level Range: -64 to +00

This is used to set the level at which the first table kicks in.

18 — Amp Waveshaper First Table Range: 00 to 07

Determines which table will begin when the input signal reaches the level set with the Amp Level Detect Attack parameter (number 15).

19 — Amp Waveshaper Last Table Range: 01 to 07

Sets the highest table that you will reach. This is used to define the overall sound. A broader table range offers a more dynamic sound.

20 — Amp Waveshaper Table Slope Range: 001 to 127

Determines how fast you will switch from one table to the next.

21 — Amp Tube Bias Range: 00 to 99

For preamp gains approximately 00 dB, this parameter controls the emphasis of even to odd harmonics which determines the tone of the amp; mid values emphasize even harmonics and offer a warmer ("glowing tube") sound, while the highest values may sound like tubes going bad. Tube bias and preamp gain are independent parameters. For low preamp gain, it may be desirable to use low tube bias values, because this more closely imitates the operation of a real amplifier.

22 — Amp Output Level Range: 00 to 99

This parameter controls the output level of the main amp.

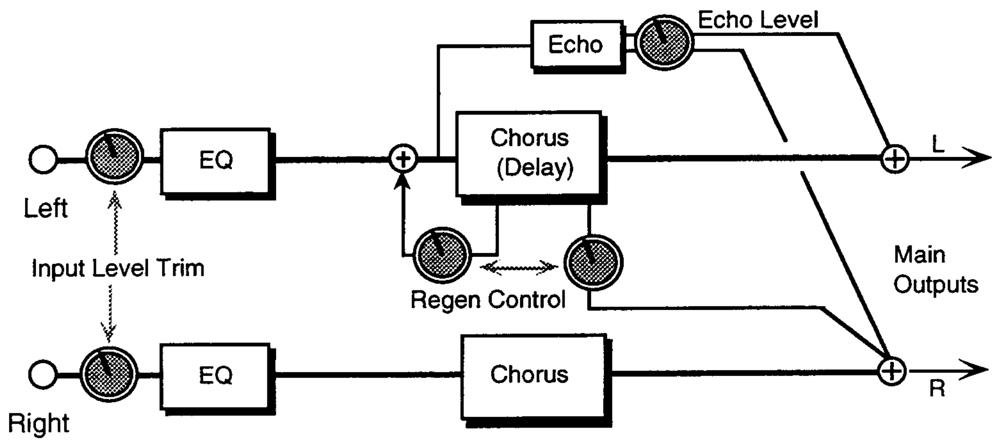
23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

EQ-Chorus-DDL

EQ-Chorus-DDL combines an EQ with a chorus and a digital delay. This is the industry standard chorus effect, and provides two very long delays in addition to the modulated chorus delays. This algorithm sounds great with a guitar, but try it with any source!

EQ-Chorus-DDL Signal Routing



The signal enters a programmable EQ, which is preceded by an input level trim (parameter 17). The signal is then routed to the chorus which is heard directly at the output. There is also a delayed unchorused signal (sharing the same delay lines) that is routed back into the chorus. There is also a second signal from the delay line that is routed to the right side. There are two discrete echo times available before the chorus delay line. These are unchorused echoes. The signal from the echoes are routed directly to the outputs. There is also a dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found in the beginning of this section.

03 — Chorus LFO Rate Range: 00 to 99

Controls the rate of pitch modulation to the chorus.

04 — Chorus LFO Width Range: 00 to 99

Controls the depth of pitch modulation. Keep in mind that the width of pitch modulation is affected by the rate; as the rate is increased, the apparent pitch modulation is also increased.

05 — Chorus Center Range: 00 to 99

Controls the nominal delay time of the chorus about which the delay modulation occurs. Adjusting this parameter will change the tonal character of the effect. This delay time is not related to the regen delays or the echo delays.

06 — Left/Right LFO
Range: Out-of-Phase or In-Phase

When this parameter is In-Phase, the left and right chorus delays will modulate together. When set to Out-of-Phase, the chorus delay on the left channel will increase, as the chorus delay on the right will decrease, and vice-versa.

07 — Chorus Left Delay Time
Range: 0 to 1500 ms

Controls the time delay for the left channel regen delay, which is independent of the chorus effect.

08 — Chorus Right Delay Time
Range: 0 to 1500 ms

Controls the time delay for the right channel regen delay, which is independent of the chorus effect.

09 — Chorus Delay Regen
Range: -99 to +99

Controls the amount of regeneration applied to the delay time taps. The sign of the value determines the polarity of the regen. The polarity affects the tonal quality of the regeneration.

10 — Chorus Left Echo Time
Range: 0 to 1500 ms

Controls the left chorus echo time. Higher settings yield a deeper echo. There are two discrete echoes, one to the left and one to the right.

11 — Chorus Right Echo Time
Range: 0 to 1500 ms

Controls the right chorus echo time. Higher settings yield a deeper echo.

12 — Chorus Echo Level
Range: 00 to 99

Controls the volume of the discrete echo for both the left and right sides. Higher values offer louder echo, while a value of 00 will eliminate the echo. For sustained sounds, mid echo levels yield a "poor man's reverb."

13 — Bass Fc
Range: 0 to 1000 Hz

Sets the cutoff frequency of the lower frequency band shelving filter.

14 — Bass EQ Gain
Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low shelving filter.

15 — Treble Fc
Range: 01KHz to 16KHz

Sets the cutoff frequency of the high shelving filter.

16 — Treble EQ Gain
Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high shelving filter.

17 — EQ Input Level Trim
Range: -24 to +00 dB

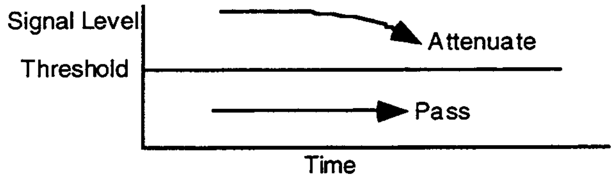
Adjusts the input volume of the EQs to eliminate the possibility of clipping boosted signals.

18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

EQ-Compressor

EQ-Compressor combines an EQ with a full feature stereo compressor. For high compressor ratios, this algorithm functions as a limiter. This algorithm operates by compressing (attenuating) signals above the threshold and passing the signals below the threshold. For higher ratios and lower thresholds, this algorithm can be used to create sustain. EQ exists in both signal and side chain paths, in contrast to the Expander which has filtering in only the side chain path.



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. We recommend a setting of 99.

03 — Compressor Gain   Range: -48 to +48 dB

This parameter boosts the compressed signal level.

04 — Compressor Ratio   Range: 1:1 to 40:1, infinity

Sets the amount of compression. The range is based on decibels (dB) above the threshold. If this is set to 4:1 for example, it will allow 1 dB increase in output level for every 4 dB increase in input level. When this is set to infinity, it acts as a limiter.

05 — Compressor Threshold   Range: -96 to +00 dB

Sets the threshold level. Signals that exceed this level will be compressed, while signals that are below will be unaffected. To turn off the compressor, set the level to +00 dB.

06 — Gain Change   Range: N/A

This read-only parameter displays a gain reduction meter.

07 — Comp Attack   Range: 50µs to 100ms

Determines the attack rate after the initial signal has been detected and before the compression takes affect.

08 — Comp Release   Range: 1ms to 10.0s

Determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally chosen longer than the attack time (parameter 06).

09 — Comp Noise Gate Off Below   Range: -96 to +00 dB

Sets the lower threshold level at which the noise gate shuts off the audio.

10 — Comp Noise Gate On Above   Range: -96 to +00 dB

Sets the upper threshold level at which the noise gate passes audio. This higher second threshold prevents false "turn ons."

11 — Gate Release Time   Range: 1ms to 10.0s

Determines how long it takes for the gate to be fully released after the input signal drops below the threshold level. Lower settings yield a quick gate.

12 — Bass Fc   Range: 0 to 1000 Hz

Sets the cutoff frequency of the lower frequency band shelving filter.

13 — Bass EQ Gain   Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low shelving filter.

14 — Treble Fc   Range: 01KHz to 16KHz

Sets the cutoff frequency of the upper frequency band high shelving filter.

15 — Treble EQ Gain   Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high shelving filter.

16 — EQ Input Level Trim   Range: -24 to +00 dB

Adjusts the input volume of the EQs, to eliminate the possibility of clipping boosted signals.

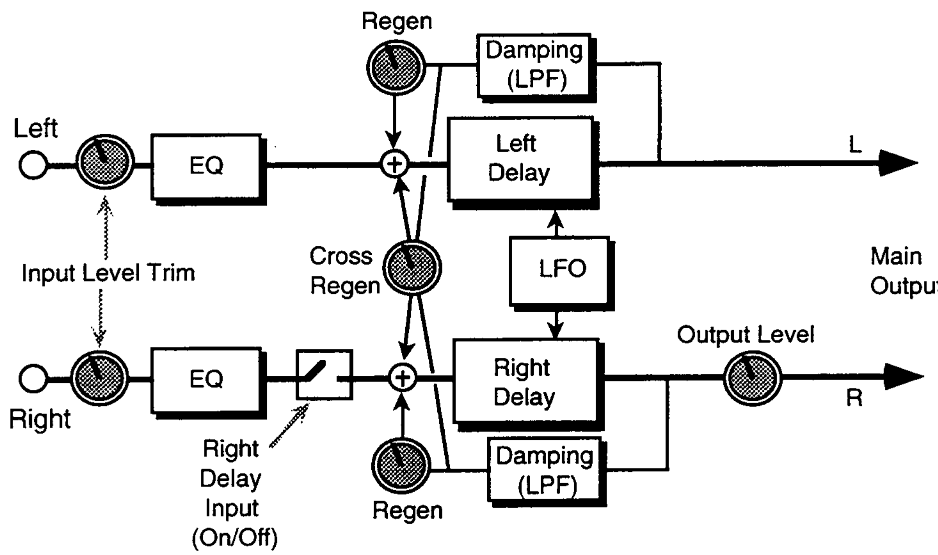
17 — Mod1 Source
18 — Mod1 Destination
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

EQ-DDL-withLFO

EQ-DDL-withLFO features a parametric EQ and a stereo digital delay (similar to Dual Delay) that provides LFO modulation of a wide range of delays. This algorithm sounds great with an electric piano, but try it with any source!

EQ-DDL-withLFO Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — DDL+LFO Left Delay Time   Range: 0 to 845 ms

Determines the amount of time between the input signal and the left delay output.

04 — DDL+LFO Right Delay Time   Range: 0 to 845 ms

Determines the amount of time between the input signal and the right delay output. Set this differently from parameter 03 to achieve syncopated repeats.

05 — DDL+LFO LFO Rate   Range: 00 to 99

Controls the rate of pitch modulation which is the LFO. To achieve a chorusing effect, this rate must be very slow.

06 — DDL+LFO LFO Width   Range: 00 to 99

Controls the excursion of pitch modulation. Since the rate is usually very slow, then the width is usually large.

07 — Left/Right LFO   Range: Out-of-Phase or In-Phase

When this parameter is In-Phase, the left and right choruses will modulate their detunes together. When set to Out-of-Phase, the detune on the left channel will go up while the detune on the right will go down.

08 — DDL+LFO Delay Regen Range: -99 to +99

Controls the amount of regeneration applied to the delay time taps. The sign of the value determines the polarity of the regen.

09 — DDL+LFO Delay Cross Regen Range: -99 to +99

Allows you to feedback the delayed signals to their opposite sides; the left voice crosses to the right voice, and the right voice crosses to the left voice. A setting of +99 or -99 will cause infinite repeats. Be careful, if the delay regen is set too high, it may cause this parameter to "blow up." Also, too high of a setting in parameters 08 and 09 will cause a DC offset, which will make this algorithm shut down.

10 — DDL+LFO Regen Damping Range: 00 to 99

Adjusts the cut off of a low pass filter on the feedback signal, which controls the amount of damping to the feedback signals. The higher the number, the more the signals are damped.

11 — DDL+LFO Right Delay Input Range: Off or On

Disables the input into the right side delay line. The right delay line will still get input from the Cross Regen. This allows a ping-pong delay effect.

12 — DDL+LFO Right Output Level Range: 00 to 99

This parameter controls the right output signal level.

13 — Bass Fc Range: 0 to 1000 Hz

Sets the cutoff frequency of the lower frequency band shelving filter.

14 — Bass EQ Gain Range: -48 to +24 dB

This parameter sets the amount of boost or cut applied to the low shelving filter.

15 — Treble Fc Range: 01KHz to 16KHz

This parameter sets the cutoff frequency of the high shelving filter.

16 — Treble EQ Gain Range: -48 to +24 dB

This parameter sets the amount of boost or cut applied to the high shelving filter.

17 — EQ Input Level Trim Range: -24 to +00 dB

Allows you to adjust the input volume of the EQs to avoid clipping signals.

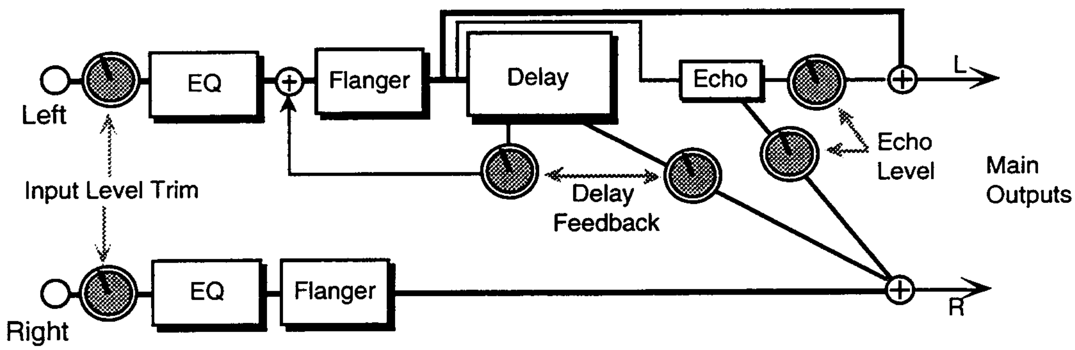
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

EQ-Flanger-DDL

EQ-Flanger-DDL combines an EQ with a flanger and a digital delay. Use flanging to get that "jet aircraft woosh" effect.

EQ - Flanger - DDL Signal Routing



The signal enters an input level trim (parameter 20) followed by a programmable EQ, and then is routed to the flanger. The flanger is routed directly to the output. The left channel signal passes through the delay and is routed back into the flanger. Another signal from the delay is routed to the output on the right side. One feedback parameter (12) controls both delay levels. There are two discrete echoes that are sent to the left and right outputs respectively. The signal from both echoes has one level control (parameter 15). There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. We recommend a Mix setting of 99.

03 — Flanger LFO Rate   Range: 00 to 99

Controls the rate of modulation of the flanger effect.

04 — Flanger LFO Width   Range: 00 to 99

Controls the range of the high to low frequency sweep in the flanger effect.

05 — Flanger Center   Range: 00 to 99

Controls the sweep center of the flanger effect. The larger the flanger center, the wider will be the available width.

06 — Flanger Feedback   Range: -99 to +99

Controls the amount of feedback applied from the output to the flanger input. The sign of the value determines the polarity of the feedback.

07 — Flanger Notch Depth   Range: -99 to +99

Controls the depth of the notches created by the flanging effect. A setting of +00 will disable the flanging effect, and also provide a doppler effect for wide, moderately slow LFO rates.

08 — Left/Right LFO   Range: Out-of-Phase or In-Phase

Determines whether the flanger on the left and right channels is modulating in or out-of phase.

09 — Flanger Sample & Hold Rate   Range: Off, 001 to 100

Controls the sample rate of a sample and hold network. This is applied to the LFO within the flanger. When in hold, the effect will be to create momentarily fixed notches within the frequency spectrum (if the notch depth is not 00). A setting of 001 will have the largest space between samples. Higher values will increase the number of holds per second, making the flanging flow more smoothly. The sample and hold function can be turned off.

10 — Flanger Left Delay Time   Range: 0 to 1500 ms

Controls the time delay for the left channel regen delay. This is the "ping."

11 — Flanger Right Delay Time   Range: 0 to 1500 ms

Controls the time delay for the right channel regen delay. This is the "pong."

12 — Flanger Delay Feedback   Range: -99 to +99

Controls the level of the delay time taps. The sign of the value determines the polarity of the feedback.

13 — Flanger Left Echo Time   Range: 0 to 1500 ms

Controls the flanger echo time for the left side. Higher values yield a deeper echo.

14 — Flanger Right Echo Time   Range: 0 to 1500 ms

This parameter controls the flanger echo time for the right side.

15 — Flanger Echo Level   Range: 00 to 99

Controls the volume of the discrete echoes. A setting of 00 would eliminate any audible echo.

16 — Bass Fc   Range: 0 to 1000 Hz

Sets the cutoff frequency of the lower frequency band shelving filter.

17 — EQ Gain   Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low shelving filter.

18 — Treble Fc   Range: 01KHz to 16KHz

Selects the cutoff of the upper frequency band high shelving filter.

19 — EQ Gain   Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high shelving filter.

20 — EQ Input Level Trim   Range: -24 to +00 dB

Adjusts the input volume of the EQs to eliminate the possibility of clipping boosted signals.

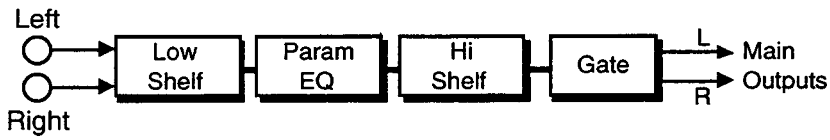
21 — Mod1 Source
22 — Mod1 Destination
23 — Mod1 Param Range Min
24 — Mod1 Param Range Max
25 — Mod2 Source
26 — Mod2 Destination
27 — Mod2 Param Range Min
28 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

EQ-Gate

EQ-Gate combines a parameteric EQ with a gate.

EQ-Gate Signal Routing



The signal enters a programmable EQ, which is preceded by a low shelving filter. The signal is then routed to a high shelving filter, then to a gate which is heard directly at the output. There is also a dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

03 — Bass Fc — Range: 0 to 1000 Hz

Sets the center of the low frequency EQ.

04 — Bass Gain (loShv) — Range: -48 to +24 dB

Sets the amount of boost or cut applied to this low frequency parametric.

05 — Mid1 Fc — Range: 5 to 9999 Hz

Sets the center of the mid frequency parametric.

06 — Mid1 Gain — Range: -48 to +24 dB

Sets the amount of boost or cut applied to this mid frequency parametric.

07 — Mid1 Q — Range: 01 to 18

This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.

08 — Treble Fc — Range: 01 to 16 KHz

Sets the center frequency of the high frequency parametric.

09 — Treble Gain (HiShv) — Range: -48 to +24 dB

Sets the amount of boost or cut applied to this high frequency parametric.

10 — EQ Input Level Attenuation — Range: -24 to +00 dB

Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.

11 — Noise Gate Off Below — Range: -96 to +00 dB

Sets the lower threshold level at which the noise gate shuts off the audio.

12 — Noise Gate On Above   Range: -96 to +00 dB

Sets the upper threshold level at which the noise gate passes the audio. The higher second threshold prevents false "turn ons."

13 — Gain Change   Range: N/A

This read-only parameter displays the amount of gain reduction in real time.

14 — Gate Release Time   Range: 1ms to 10.0s

This parameter sets the amount of time after the signal has elapsed for the noise gate to shut down. For a longer sustain, set this parameter higher.

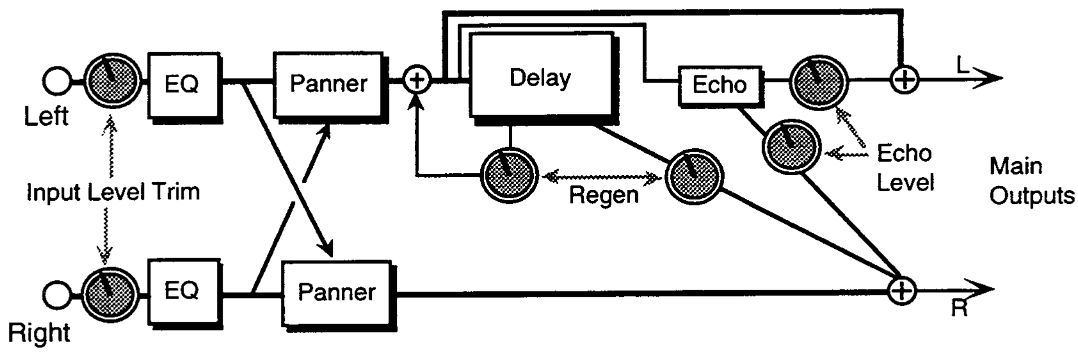
15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

EQ-Panner-DDL

EQ-Panner-DDL combines an EQ with a panning effect and a digital delay. If this algorithm doesn't sound like it's panning, check parameter 05 to see if it's in-phase or out-of-phase; a mono signal will only work "in-phase."

EQ-Panner-DDL Signal Routing



The signal enters an input level trim (parameter 17) followed by a programmable EQ, and is then routed to the panner. The panner is routed directly to the output. The left channel signal passes through the digital delay and is routed back into the delay. There is another signal from the delay that is routed to the output on the right side. One regen parameter (09) between the delay sends controls both delay levels. There are two discrete echoes that are sent to the left and right outputs respectively. The signal from both echoes has one level control. This configuration of delays and echoes provides the "ping-pong"effect. There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Panner Rate   Range: 00 to 99

This parameter sets the rate of panning from left to right. Higher values create a faster movement. High values of this parameter used in conjunction with the Sample & Hold rate yield interesting staccato effects.

04 — Panner Width   Range: 00 to 99

Sets the width of the excursion from left to right. Because this algorithm features a multi-shaped LFO, optimal settings are around 50. Higher values create a wider separation of the LFO curve.

05 — Left/Right LFO   Range: Out-of-Phase or In-Phase

This parameter selects an in-phase (like windshield wipers) or an out-of-phase (opposing wipers) LFO. In-Phase pans both left and right to left, then right. Out-Of-Phase pans left to left and right to right, and then left to right and right to left; at the halfway point, both channels are in the center, and a stereo signal becomes mono. Switch between the two settings until it sounds right for your routing config.

06 — Panner Sample & Hold Rate Range: Off, 001 to 100

Controls the sample rate of a sample and hold network applied to the LFO within the panner. When in hold, the stereo image will be momentarily fixed (if the width is not 00). A setting of 001 will have the largest space between holds. Higher values will increase the number of holds per second, making the panning flow more smoothly. This parameter can also be turned off.

07 — Panner Left Delay Time Range: 0 to 1500 ms

Controls the time delay for the left channel regen delay, independent of the pan effect.

08 — Panner Right Delay Time Range: 0 to 1500 ms

Controls the time delay for the right channel regen delay, independent of the pan effect.

09 — Panner Delay Regen Range: -99 to +99

Controls the amount of regen applied to the delay time taps. The sign of the value determines the polarity of the regen. A value of +00 will eliminate any audible delay.

10 — Panner Left Echo Time Range: 0 to 1500 ms

This parameter controls the echo time for the left side. Higher settings yield a slower echo. There are two discrete echoes, one to the left and one to the right.

11 — Panner Right Echo Time Range: 0 to 1500 ms

This parameter controls the echo time for the right side.

12 — Panner Echo Level Range: 00 to 99

Controls the volume of the discrete echo for both the left and right sides. Higher values offer louder echo, while a value of 00 will eliminate the echo.

13 — Bass Fc Range: 0 to 1000 Hz

Selects the cutoff frequency of the low EQ.

14 — Bass EQ Gain Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low EQ.

15 — Treble Fc Range: 01KHz to 16KHz

Selects the cutoff frequency of the high EQ.

16 — Treble EQ Gain Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high EQ.

17 — EQ Input Level Trim Range: -24 to +00 dB

Adjusts the input volume to the EQs to eliminate clipping signals.

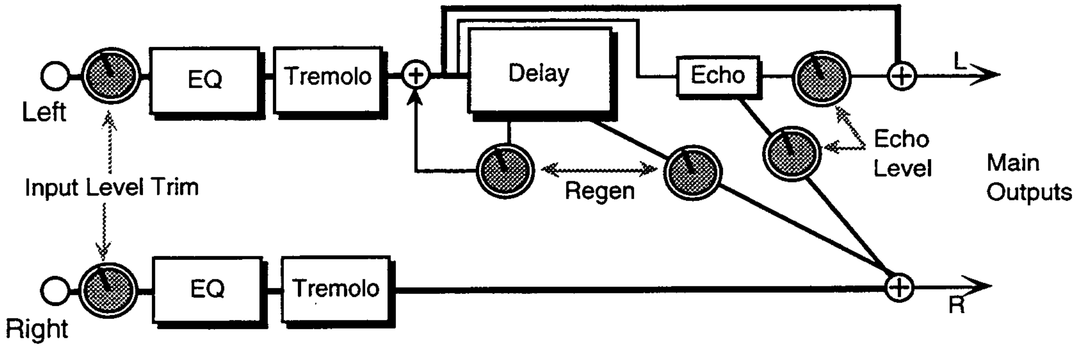
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

EQ-Tremolo-DDL

EQ-Tremolo-DDL combines an EQ and a tremolo effect, which is a pulsating change in volume, with a digital delay.

EQ-Tremolo-DDL Signal Routing



The signal enters an input level trim (parameter 17) followed by a programmable EQ, and is then routed to the tremolo. The tremolo is routed directly to the output. The left channel signal passes through the digital delay and is routed back into the delay. There is another signal from the delay that is routed to the output on the right side. One Regen parameter (09) between the delay sends controls the left and right delay level. This constitutes the "ping-pong" effect. There are two echoes that are sent to the left and right outputs respectively. The signal from the two discrete echoes has one level control. There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Tremolo Rate   Range: 000 to 200

Sets the rate of modulation. Mid values create a faster wavering sound. High values will raise the level up into the audio range, creating a ring modulation (amplitude modulation) effect. This parameter, when used in conjunction with the Sample & Hold Rate parameter, can create some interesting staccato effects.

04 — Tremolo Depth   Range: 00 to 99

This parameter sets the depth of amplitude modulation. Because this algorithm features a multi-shaped LFO, optimal settings are around 50. Higher values create a wider separation of the LFO curve.

05 — Left/Right LFO   Range: Out-of-Phase or In-Phase

Controls whether the left and right channels of the stereo tremolo will modulate in or out-of-phase.

06 — Tremolo Sample & Hold Rate Range: Off, 001 to 100

Controls the sample rate of a sample and hold network applied to the LFO within the tremolo. When in "Hold," the effect will be to fix the instantaneous amplitude (if the depth is not 00). A setting of 001 will have the largest space between holds. Lower settings create a staccato effect, whereas higher values will increase the amount of samples, making the tremolo flow more smoothly. This parameter can also be turned off.

07 — Tremolo Left Delay Time Range: 0 to 1500 ms

This parameter controls the time delay for the left channel regen delay, independent of the tremolo effect.

08 — Tremolo Right Delay Time Range: 0 to 1500 ms

This parameter controls the time delay for the right channel regen delay.

09 — Tremolo Delay Regen Range: -99 to +99

Controls the amount of regen applied to the delay time taps. The sign of the value determines the polarity of the regen. A value of +00 will eliminate the audible delay.

10 — Tremolo Left Echo Time Range: 0 to 1500 ms

Controls the tremolo echo time for the left side. Higher settings yield a slower echo.

11 — Tremolo Right Echo Time Range: 0 to 1500 ms

This parameter controls the tremolo echo time for the right side.

12 — Tremolo Echo Level Range: 00 to 99

Controls the volume of the discrete echo for both the left and right sides.

13 — Bass Fc Range: 0 to 1000 Hz

Selects the cutoff frequency of the low EQ.

14 — Bass EQ Gain Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low EQ.

15 — Treble Fc Range: 01KHz to 16KHz

Selects the cutoff frequency of the high EQ.

16 — Treble EQ Gain Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high EQ.

17 — EQ Input Level Trim Range: -24 to +00 dB

Adjusts the input volume to the EQs to prevent clipping boosted signals.

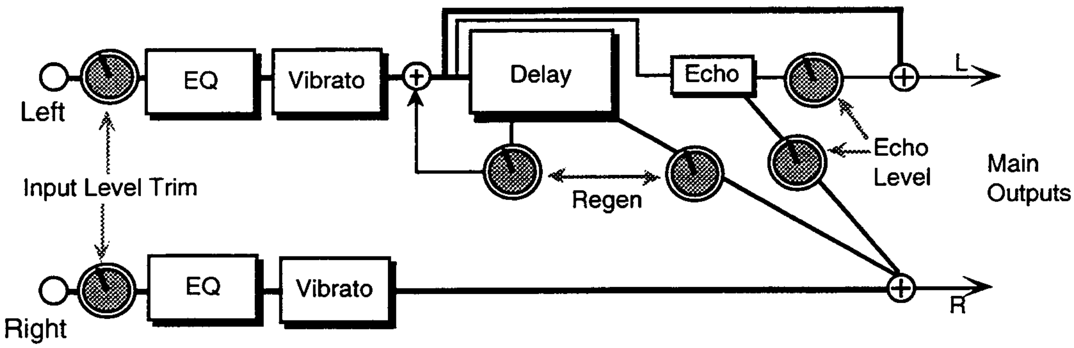
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

EQ-Vibrato-DDL

EQ-Vibrato-DDL combines an EQ and a vibrato effect (a pitch shifter modulating over a very small range), with a digital delay. Many vintage guitar amplifiers offered a vibrato control, but don't feel that this algorithm is limited to guitars; try this with other sources as well. There is a sample & hold parameter that doesn't hold the instantaneous pitch shift, but if set properly will provide a "chirping" effect when acting on the input signal.

EQ-Vibrato-DDL Signal Routing



The signal enters a programmable EQ, which is preceded by an input level trim (parameter 17). The signal is then routed to the vibrato. The vibrato is routed directly to the output. The vibrato also passes through the delay which is then regenerated back into the delay. A different delay signal is routed to the output on the right side. This constitutes a "ping-pong" delay effect. The Regen parameter between the delay sends controls the delay feedback amount. There are also two echoes that are sent to the left and right outputs respectively. The signal from the two echoes has one level control. There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. This algorithm sounds best when set to 100% wet (Mix=99).

03 — Vibrato Rate Range: 00 to 99

Sets the rate of modulation. Higher values create a faster vibrating rate.

04 — Vibrato Width Range: 00 to 99

This parameter adjusts the amount of the modulation.

05 — Left/Right LFO Range: Out-of-Phase or In-Phase

Controls the vibrato pitch direction of the left and right channels. When Out-of-Phase, the (quadrature) pitch change on the left channel will lag 90° from the right. When In-Phase, both channels will change pitch together.

06 — Vibrato Sample & Hold Rate Range: Off, 001 to 100

Controls the sample rate of a sample and hold network. This is applied to the LFO within the vibrato. When in hold (low values), it causes rhythmic chirps in the pitch of the audio signal. Higher values will increase the number of holds per second, making the vibrato flow more smoothly. The sample and hold function can also be turned Off.

07 — Vibrato Left Delay Time Range: 0 to 1500 ms

Controls the time delay on the left regenerated delay.

08 — Vibrato Right Delay Time Range: 0 to 1500 ms

Controls the time delay on the right non-regenerated delay.

09 — Vibrato Delay Regen Range: -99 to +99

Controls the amount of positive or negative feedback applied to the regenerated delay. The sign of the value determines the polarity of the feedback. A value of +00 will eliminate any feedback. This parameter controls both left and right levels.

10 — Vibrato Left Echo Time Range: 0 to 1500 ms

Controls the echo time for the left side. Higher settings yield a deeper echo. There are two discrete echoes, one to the left and one to the right.

11 — Vibrato Right Echo Time Range: 0 to 1500 ms

Controls the echo time for the right side.

12 — Vibrato Echo Level Range: 00 to 99

Controls the volume of the discrete echo for both the left and right sides. A setting of 00 will eliminate any audible echo.

13 — Bass Fc Range: 0 to 1000 Hz

Selects the cutoff frequency of the low shelving filter.

14 — Bass EQ Gain Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low shelving filter.

15 — Treble Fc Range: 01KHz to 16KHz

Selects the cutoff of the upper frequency band high shelving filter.

16 — Treble EQ Gain Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high shelving filter.

17 — EQ Input Level Trim Range: -24 to +00 dB

Adjusts the input volume before the EQs to eliminate the possibility of clipping boosted signals.

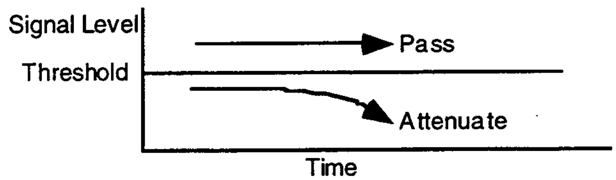
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

Expander

Expander performs downward expansion of an input signal's dynamic range. For high expansion ratios this algorithm functions as a gate. This algorithm operates by reducing the level of signals below the threshold and passing the signals above the threshold. The Threshold is a definable parameter. This algorithm can be used to eliminate noise. There is no EQ in the audio path; high and low pass filtering are provided on the side chain only. This algorithm possesses two special features:

1. The ADSR (envelope generator) in this algorithm has Attack, Sustain, and Release (the sustain is new and is called the Hold Time).
2. This algorithm contains a trigger mask function. This function is used primarily to extract a click track from drum tracks. Once triggered, this function inserts a zero signal level into the side chain detector for an amount of time determined by the user. This function becomes triggered if Trigger Mask is enabled and if side chain signal falls below the Trigger Mask Threshold.



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. We recommend a setting of 99.

03 — Exp Ratio — Range: 1:1 to 1:40, infinity

Sets the amount of expansion. The range is based on decibels (dB) below the threshold. If this is set to 1:4 for example, it will expand changes in signals below the threshold by a factor of four. When this is set to infinity, it acts as a gate. A setting of 1:1 offers no expansion.

04 — Exp Threshold — Range: -96 to +00 dB

This parameter sets the threshold level. Signals that exceed this level will be unaffected, while signals that are below will be expanded. To turn off the expander, set the level to -96 dB.

05 — Gain Change — Range: N/A

This read-only parameter displays the amount of gain reduction in real time.

06 — Exp Attack — Range: 50µs to 100ms

Determines the attack rate after the initial signal has been detected and before the expansion takes affect.

07 — Exp Release — Range: 1ms to 10.0s

Determines the release rate after the signal has been detected below the threshold level. This is generally chosen longer than the attack time (parameter 06).

08 — Expander Gate Hold Time Range: 1ms to 10.0s

This is the detection sustain time in the ADSR which constitutes attack, sustain, and release.

09 — Sidechain EQ Gain Range: -48 to +48 dB

Controls the amount of boost applied to the output signal of the high/low pass filter. This accounts for insertion loss through those filters.

10 — HighPass Fc Range: 4 to 8000 Hz

This sets the cutoff frequency of the lower frequency band high pass shelving filter.

11 — LowPass Fc Range: 100 Hz to 16 KHz

Sets the amount of boost or cut applied to the low pass filter.

12 — Trigger Mask Range: Off or On

This parameter enables the trigger mask function. Once triggered, the side chain detector will see no input signal for a duration specified by parameter 13.

13 — Time Range: 1ms to 10.0s

Sets the duration over which the side chain detector will be blacked out. This parameter is useful for isolating the first beat of a drum track.

14 — Trig Mask Lower Threshold Range: -96 to +00 dB

This sets the trigger mask threshold level. Signals that fall below this level will trigger the mask function. The trigger mask function uses the Expander Threshold (04) as upward hysteresis. Therefore, the Trigger Mask Threshold should always be set lower than the Expander Threshold.

15 — Expander Output Gain Range: -48 to +48 dB

Sets the amount of loss (negative values) or gain (positive values) applied to the expander on the output volume. We recommend a starting application of +00 dB.

16 — Mod1 Source
17 — Mod1 Destination
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

FastPitchShift

FastPitchShift has a transport delay of only 10 msec and a maximum detune ratio of one semitone. Try shifting the voices slightly in both positive and negative values (parameters 03 and 06) to create a fat sound. This algorithm can be used for pitch correction (for instance, try hooking up a mod wheel for MIDI control).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. This algorithm sounds best with a Mix of wet and dry. Try using a modulation controller for the Mix parameter to bring in or fade out the pitch shifted signal.

03 — PitchShifter Vc 1 Fine   Range: -99 to +99

This parameter allows you to fine tune the pitch of Voice 1.

04 — PitchShifter Vc 1 Level   Range: 00 to 99

Adjusts the volume of Voice 1. A setting of 00 would eliminate any audible pitch shift.

05 — PitchShifter Vc 1 Pan   Range: -99 to +99

Allows you to assign the location of Voice 1 in the stereo field. A value of -99 would be far left, and +99 would be far right.

06 — PitchShifter Vc 2 Fine   Range: -99 to +99

This parameter allows you to fine tune the pitch of Voice 2.

07 — PitchShifter Vc 2 Level   Range: 00 to 99

Adjusts the volume of Voice 2. A setting of 00 would eliminate any audible pitch shift.

08 — PitchShifter Vc 2 Pan   Range: -99 to +99

This parameter allows you to assign the location of Voice 2 in the stereo field. A value of -99 would be far left, and +99 would be far right.

09 — PitchShifter LFO Rate   Range: 00 to 99

Controls the rate of pitch modulation which creates a chorusing effect. To achieve chorusing, this rate must be very low.

10 — PitchShifter LFO Width   Range: 00 to 99

Controls the excursion (amount) of pitch modulation. Since the rate is usually very low, then the width is usually very large.

11 — Mod1 Source
12 — Mod1 Destination
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

Flanger

Flanger is a fat digital flanger. The DP/2 offers two different flanger algorithms. This flanger has deeper notches and requires less feedback than the EQ-Flanger-DDL algorithm.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. The flange depth can be changed somewhat by changing the Mix level.

03 — Flanger LFO Rate Range: 00 to 99

This parameter controls the rate of modulation of the flanger notches.

04 — Flanger LFO Width Range: 00 to 99

Controls the range of the high to low frequency sweep about the flanger center in the flanger effect.

05 — Flanger Center Range: 00 to 99

This parameter controls the sweep center of the flanger effect.

06 — Flanger Regen Range: -99 to +99

Controls the amount of feedback applied from the output to the input of the flanger. The sign of the value determines the polarity of the feedback.

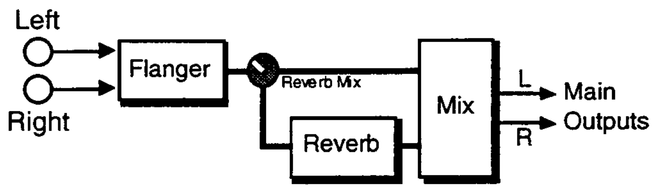
07 — Mod1 Source
08 — Mod1 Destination
09 — Mod1 Param Range Min
10 — Mod1 Param Range Max
11 — Mod2 Source
12 — Mod2 Destination
13 — Mod2 Param Range Min
14 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

Flanger-Reverb

Flanger-Reverb combines a flanger with a plate reverb.

Flanger-Reverb Signal Routing



The signal enters a stereo flanger, which is heard directly at the output. There is a signal that is routed out of the flanger into the large plate reverb. There is also a dry signal (not shown) that goes directly from the input to the output and is controlled with the Mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

03 — Reverb Mix   Range: 00 to 99

Controls the mix between the flanged signal and the reverb. Setting this parameter to 00 will allow only the flanged signal to be heard, while a setting of 99 will send all of the flanged signal to the reverb.

04 — Flanger LFO Rate   Range: 00 to 99

Controls the rate of pitch modulation to the flanger.

05 — Flanger LFO Width   Range: 00 to 99

Controls the width of pitch modulation. Keep in mind that the width of pitch modulation is affected by the rate; as the rate is increased, the apparent pitch modulation is also increased.

06 — Flanger Center   Range: 00 to 99

Controls the sweep center of the flanger effect. The larger the flanger center, the wider will be the available width.

07 — Flanger Feedback   Range: -99 to +99

Controls the amount of regeneration applied to the delay time taps. The sign of the value determines the polarity of the regen. The polarity affects the tonal quality of the regeneration.

08 — Flanger Notch Depth   Range: -99 to +99

Controls the depth of the notches created by the flanging effect. A setting of +00 will disable the flanging effect, and also provide a doppler effect for wide, moderately slow LFO rates.

09 — Left/Right LFO   Range: Out-of-Phase or In-Phase

Determines whether the flanger on the left and right channels is modulating in or out-of phase.

10 — Large Plate Decay   Range: 0.40 to 140.0 sec.

Controls the amount of time it takes for the reverberation to decay. High values of decay sound good with this algorithm.

11 — Plate Predelay Time Range: 0 to 250 ms

Controls the amount of time it takes for the original signal to be presented to the reverb. Higher values denote a longer delay.

12 — Large Plate HF Damping Range: 00 to 99

Shapes the tone of the reverb decay. Higher settings cause the high frequency components to decay more rapidly.

13 — Large Plate HF Bandwidth Range: 00 to 99

This parameter acts as a low pass filter on the output of the plate reverbs, controlling the amount of high frequencies present. The higher the setting, the more high frequencies are allowed to pass through, offering a brighter ringing sound. Some interesting effects can be created by using a mod controller over a large range.

14 — Plate Diffsn1 Range: 00 to 99

This parameter smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

15 — Diffusion2 Range: 00 to 99

This parameter, similar to and in series with Diffusion1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.

16 — Plate Decay Definition Range: 00 to 99

Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.

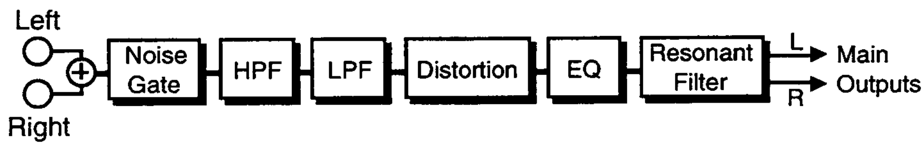
17 — Mod1 Source
18 — Mod1 Destination
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Fuzz Box

The **Fuzz Box** algorithm offers an assortment of distorted tones, covering the gamut from vintage stomp boxes to modern distortion units.

Fuzz Box Signal Routing



The input signal enters a noise gate, and is immediately followed by a high-pass, then a low-pass filter. The signal then enters the distortion, where the fuzz box can be "sculpted." The signal exits the distortion, is followed by the EQ section, and is then routed into a resonant filter (used to create a speaker cabinet or wah effect), which is heard directly at the output. The end user has the freedom to say "cool," or any other verbage to express his/her joy. There is also a dry signal (not shown) that goes directly from the input to the output and is controlled with the Mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

03 — HighPass Fc Range: 4 to 1000 Hz

Filters out low frequencies before the distortion signal path. The higher the value, the fewer low frequencies pass through, the thinner the sound.

04 — LowPass Fc Range: 100 to 16 K

Filters out high frequencies before the distortion signal path. The lower the value, the fewer high frequencies pass through, the darker the sound.

05— Distortion Level In
06 — Distortion Level Out Ranges: 00 to 99

These two parameters control the levels going into and coming out of the distortion effect. The Distortion Level In adjusts the *intensity* of distortion. Distortion Level In will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the Distortion Level Out (06) down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.

The Distortion Level Out parameter controls the gain coming out of the distortion effect. Generally, if the Distortion Level In (05) is set high, set this parameter lower to control the volume.

07 — Rectifier Mix Range: 00 to 99

Mixes the original input signal with a full-wave-rectified version. A setting of 99 yields harsh, metallic overtones.

08 — Softness Range: -99 to +99

Sets the clipping characteristic between soft or tube-like (+99) to hard or transistor-like (-99). The differences are most noticeable for moderate settings of parameter 05 (Distortion Level In).

09 — Harmonic Mod
Range: -99 to +99

Allows the level of the the input signal to modulate the harmonic content. As you play harder, more even harmonics will be generated (similar to the response of a tube amplifier).

10 — Offset
Range: -99 to +99

Clips the positive and negative peaks of the input signal asymmetrically, resulting in even harmonics. Postive settings yield more positive clipping; negative settings yield more negative clipping. Extreme settings actually chop off the negative or positive halves of the signal.

11 — Gain Bass
Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low shelving EQ.

12 — Gain Treble
Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high shelving EQ.

13 — Mid1 Fc
Range: 50 to 9999 Hz

Sets the center of the mid-frequency EQ. Higher values have a brighter sound.

14 — Mid1 Gain
Range: -48 to +24 dB

Sets the amount of boost or cut applied to this frequency.

15 — Mid1 Q
Range: 01 to 18

This is a bandwidth control that determines the range of affected frequencies. This is equal to the cutoff frequency divided by the bandwidth. By raising the Q, you can produce a narrower bandwidth.

16 — Mid2 Fc
17 — Mid2 Gain
18 — Mid2 Q

These three parameters are identical to the previous three parameters, and can be used to control different bandwidths within the mid range.

19 — Post-Distortion VCF Fc
Range: 01 to 99

Determines the filter cut off frequency after the distortion. Higher values have a brighter sound. This parameter can be used to emulate a speaker cabinet. To disable the distortion filter, set this parameter to 99, and the Post-Distortion VCF Q parameter to 01.

20 — Post-Distortion VCF Q
Range: 01 to 30

Determines the level and width of the resonant peak at the filter cutoff point. While the Post-Dist VCF Fc parameter determines where (at what-frequency) this peak will occur, this parameter controls the *sharpness* of the peak.

21 — Noise Gate Off Below
Range: -96 to +00 dB

Sets the lower threshold level at which the noise gate shuts off the audio.

22 — Noise Gate On Above
Range: -96 to +00 dB

Sets the upper threshold level at which the noise gate passes the audio. The higher second threshold prevents false "turn ons."

23 — Gain Change Range: N/A

This read-only parameter displays the amount of gain reduction in real time.

24 — Gate Release Time Range: 1ms to 10.0s

This parameter sets the amount of time after the signal has elapsed for the noise gate to shut down. For a longer sustain, set this parameter higher.

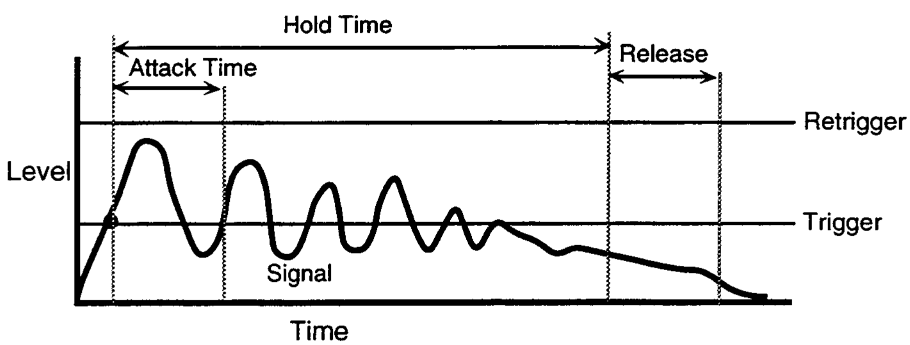
25 — Mod1 Source
26 — Mod1 Destination
27 — Mod1 Param Range Min
28 — Mod1 Param Range Max
29 — Mod2 Source
30 — Mod2 Destination
31 — Mod2 Param Range Min
32 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

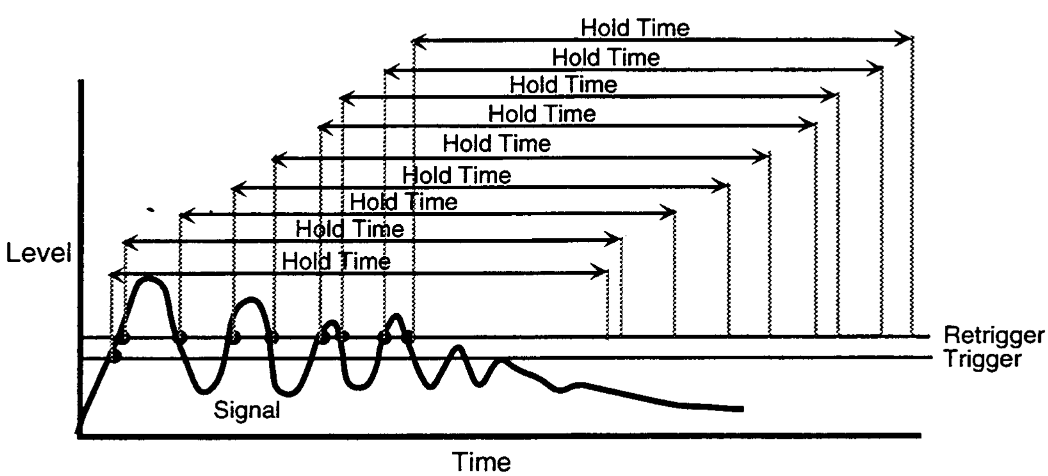
Gated Reverb

Gated Reverb provides an excellent gated reverb. When the output of a reverb is muted partway through it's decay, it creates a gated sound. To achieve this gated effect, both the Gated and Reverse reverbs must gate a number of internal parameters, not just the output amplitude envelope. It is however, the output amplitude over which the user has control. The DP/2 offers a highly controllable gated reverb, optimized for percussive instruments, but useful for any signal. The gate is first opened when the input signal passes the trigger threshold. This trigger threshold should be set as low as possible, so that none of the input signal is missed. The gated reverb is distinguished from the reverse reverb by retriggering whenever the input signal passes a retrigger (user programmable) threshold (see diagrams). The gate will stay open as long as the input signal remains above the retrigger threshold, and all the input signals will be accumulated under this gate until the total input signal level falls below the retrigger threshold. When this happens, the Hold Time will begin (as shown in the diagram below). The reason for two thresholds is to eliminate false retriggering and to ensure precise hold time durations. If you desire a separate gate on each and every note, use the Non Lin reverbs. The topology for the Gated Reverb is derived from the Plate Reverb.

Gated Reverb with a High Retrigger Threshold



Gated Reverb with a Low Retrigger Threshold



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Attack Range: 1ms to 10.0s

Sets the attack time of the gated reverb once the incoming signal has reached the trigger level. Generally the attack should be short and not set longer than the Hold Time. This parameter should not be used to achieve a reverse reverb envelope, because here the attack volume increases whereas in the Reverse reverb the attack volume accelerates.

04 — Hold Time Range: 1ms to 10.0s

Sets the amount of time that the reverb will hold after the retrigger and before the release. The Hold Time will begin again if retriggered (see diagrams).

05 — Decay Range: 0.20 to 100.0 sec.

Sets the decay rate much like in the Reverse Reverb algorithm. In general, the decay rate is set very high. Decay rate is not offered as a controllable parameter in the Reverse Reverb, but the DP/2 brings it out here for special effect when low values are used.

06 — Release Time Range: 1ms to 10.0s

Sets the amount of time after the Hold Time has elapsed for the gated reverb to shut down. Generally these times are very short.

07 — Trigger Threshold Range: -96 to +00 dB

Sets the signal level that triggers the gated reverb. When the incoming signal reaches this value, it triggers (starts) the gated reverb. Higher values would require a stronger incoming signal. Set this parameter as low as possible to work with your particular source, but not too low so as to cause false triggering.

08 — Retrigger Threshold Range: -96 to +00 dB

This parameter sets the level at which the gated reverb will retrigger. For precise Hold Time that begins at the onset of the incoming source, this parameter should be set higher than the incoming signal to prevent retriggering (as shown in the diagrams). After the incoming signal reaches the trigger threshold, the gated reverb is activated. Every time the signal reaches the retrigger threshold, the gated reverb will retrigger causing the Hold Time to restart.

If the level of this parameter is set lower than the incoming signal, the gated reverb will continue to retrigger as shown in the diagrams. With a high Decay Rate (parameter 05), this adds a cavernous quality to percussion instruments.

09 — HF Damping Range: 00 to 99

Controls the rate of attenuation of high frequencies in the decay of the reverb. Increasing the value of this parameter will gradually filter out increasing amounts of high frequency energy. We recommend a setting of 00.

10 — Diffusion 1 Range: 00 to 99

Smears the transients, so as to diffuse and smooth the sound. Lower values will cause impulsive sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding). Recommended setting is approximately 50.

11 — Diffusion 2 Range: 00 to 99

This parameter, similar to and in series with Diffusion 1, performs the same way but controls lower frequency ranges. Recommended setting is approximately 50.

12 — Decay Definition Range: 00 to 99

Controls the rate of echo density build up in the reverb decay. If set too high, the echo density will build at a rate that exceeds the decay rate. A general rule of thumb: Definition should not exceed the Decay Rate. We recommend settings between 25 and 50.

13 — Slapback Range: 0 to 500 ms

Controls the delay time of an internal dry stereo signal to create a slapback. In general, the slapback is greater or equal to the Hold Time (parameter 04) to achieve a reverse effect.

14 — Slapback Level Range: 00 to 99

Adjusts the volume of the slapback (internal dry) signal. A value of 00 would eliminate any audible slapback.

15 — Early Reflections 1
16 — Early Reflections 2
17 — Early Reflections 3
18 — Early Reflections 4 Ranges: -99 to +99

These parameters control four early reflection levels. Setting these levels to lower values will produce a wetter sound.

19 — Left/Right Balance Range: -99 to +99

Controls the left/right stereo balance of the gated reverb signal. A setting of -99 would offer hard left, whereas a setting of +99 would offer hard right. A setting of +00 would place the reverb in the center of the stereo spectrum.

20 — Mod1 Source
21 — Mod1 Destination
22 — Mod1 Param Range Min
23 — Mod1 Param Range Max
24 — Mod2 Source
25 — Mod2 Destination
26 — Mod2 Param Range Min
27 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, in the beginning of this section.

Guitar Amp 1, Guitar Amp 2

These algorithms recreate the warm sound of a tube guitar amplifier. They do this by emulating tube distortion characteristics. These algorithms are good for all stringed instruments. Guitar Amp 1 offers more distortion than Guitar Amp 2.

Guitar Amp 1 is designed for Hard Rock sounds.
Guitar Amp 2 is optimized for "bluesy" type sounds.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Amp Preamp Gain Range: -48 to +48 dB

Adjusts the amount of boost or cut applied to the incoming signal. We recommend a setting of 00 dB, since these emulations were optimized for distortion there. Lower preamp gains will result in less distortion, while higher preamp gains will yield clipping distortion. For low preamp gain, it may be desirable to use low tube bias values.

04 — Output Level Range: 00 to 99

This parameter controls the output level of the main amp before the output EQ.

05 — Amp Tube Bias Range: 00 to 99

For preamp gains approximately 00 dB, this parameter controls the emphasis of even to odd harmonics which determines the tone of the amp; mid values emphasize even harmonics and offer a warmer ("glowing tube") sound, while the highest values may sound like tubes going bad. Tube bias and preamp gain are independent parameters. For low preamp gain, it may be desirable to use low tube bias values, because this more closely imitates the operation of a real amplifier.

06 — Pre-EQ Input Level Trim Range: -24 to +00 dB

Controls the input level to the pre-amp EQ to eliminate the possibility of clipping boosted signals.

07 — Pre-EQ High Pass Cutoff Range: 4 to 1000 Hz

Filters out the low frequencies before the preamp. The higher the value, the less low frequencies pass through.

08 — Pre-EQ Fc Range: 100 to 9999 Hz

This parameter determines the center frequency of the parametric filter before the preamp. Higher values have a brighter sound.

09 — Pre-EQ Gain Range: -48 to +24 dB

Adjusts the amount of boost or cut applied to the parametric filter in front of the preamp.

10 — Pre-EQ Q Range: 01 to 18

Determines the width of the resonant peak at the parametric filter center frequency. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

11 — Noise Gate Off Below Range: -96 to +00 dB

Sets the lower threshold level at which the noise gate shuts off the audio.

12 — Noise Gate On Above Range: -96 to +00 dB

Sets the upper threshold level at which the noise gate passes the audio. The higher second threshold prevents false "turn ons."

13 — Gate Release Time Range: 1ms to 10.0s

This parameter sets the amount of time after the signal has elapsed for the noise gate to shut down. For a longer sustain, set this parameter higher.

14 — Speaker High Pass Cutoff Range: 4 to 1000 Hz

This parameter filters out the low frequencies of the main amp prior to the speaker. The higher the value, the less low frequencies pass through.

15 — OutEQ1 Fc Range: 50 to 9999 Hz

This parameter determines the filter center frequency of the parametric in the main amp stage. Higher values have a brighter sound.

16 — OutEQ1 Gain Range: -48 to +24 dB

This parameter adjusts the amount of boost or cut applied to the main amp parametric.

17 — OutEQ1 Q Range: 01 to 18

Determines the width of the resonant peak of the filter center. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

18 — OutEQ2 Fc Range: 50 to 9999 Hz

This parameter determines the filter center frequency of the second parametric in the main amp stage. Higher values have a brighter sound.

19 — OutEQ2 Gain Range: -48 to +24 dB

This parameter adjusts the amount of boost or cut applied to the second main amp parametric.

20 — OutEQ2 Q Range: 01 to 18

This parameter determines the width of the resonant peak of the second filter center.

21 — Speaker Low Pass Cutoff Range: 2.0 to 16.0 KHz

Filters out the high frequencies of the speaker. The lower the value, the less high frequencies pass through. This speaker filter is less selective than the speaker cabinet emulation algorithms.

22 — Mod1 Source
23 — Mod1 Destination
24 — Mod1 Param Range Min
25 — Mod1 Param Range Max
26 — Mod2 Source
27 — Mod2 Destination
28 — Mod2 Param Range Min
29 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

Guitar Amp 3

Guitar Amp 3 combines an inverse expander with a bright distortion for amp lead sounds. The inverse expander may be thought of as a compressor that amplifies all signals below the threshold. This algorithm is good for heavy metal and hard rock guitar solos.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Preamp Gain   Range: -48 to +48 dB

Adjusts the amount of boost or cut applied to the EQ'd incoming signal. Lead sounds are obtained using high gain.

04 — Output Level   Range: 00 to 99

This parameter controls the output level before the output EQ.

05 — PreEQ Input Level Trim   Range: -24 to +00 dB

Controls the input level to the preamp EQ to eliminate the possibility of clipping boosted signals.

06 — Pre-EQ Fc   Range: 100 to 9999 Hz

Determines the filter center frequency of the parametric in the preamp stage. Higher values have a brighter sound.

07 — Pre-EQ Gain   Range: -48 to +24 dB

This parameter adjusts the amount of boost or cut applied to the preamp parametric.

08 — Pre-EQ Q   Range: 01 to 18

Determines the width of the resonant peak at the filter center. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

09 — ExpndRatio   Range: 1:1 to 40:1, infinity

Sets the amount of inverse expansion. Expansion occurs below the threshold. If this is set to 3:1 for example, it will expand the change in signals below the threshold by three times in an attempt to make the signal amplitude approach the threshold level.

10 — Threshold   Range: -96 to +00 dB

Sets the inverse expander threshold level. Signals beneath this level will be expanded, while signals that are above will be unaffected. As the input signal dies away below the threshold, the expander will increase the gain of the signal.

11 — Gain Change   Range: N/A

This read only parameter shows the level of the signal.

12 — Noise Gate Off Below   Range: -96 to +00 dB

Sets the lower threshold level at which the noise gate shuts off the audio.

13 — Noise Gate On Above   Range: -96 to +00 dB

Sets the upper threshold level at which the noise gate passes audio. This higher second threshold prevents false "turn ons."

14 — Gate Release Time   Range: 1ms to 10.0s

Sets the amount of time after the signal has elapsed for the noise gate to shut down. For a longer sustain, set this parameter higher.

15 — Speaker High Pass Cutoff   Range: 4 to 1000 Hz

Filters out the low frequencies of the main amp prior to the speaker. The higher the value, the less low frequencies pass through.

16 — OutEQ1 Fc   Range: 50 to 9999 Hz

Determines the filter center frequency of the parametric in the main amp stage. Higher values have a brighter sound.

17 — OutEQ1 Gain   Range: -48 to +24 dB

This parameter adjusts the amount of boost or cut applied to the main amp parametric.

18 — OutEQ1 Q   Range: 01 to 18

Determines the width of the resonant peak at the filter center. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak. This parameter is equal to the cutoff frequency divided by the bandwidth.

19 — OutEQ2 Fc   Range: 50 to 9999 Hz

Determines the filter center frequency of the second parametric in the main amp stage. Higher values have a brighter sound.

20 — OutEQ2 Gain   Range: -48 to +24 dB

Adjusts the amount of boost or cut applied to the second main amp parametric.

21 — OutEQ2 Q   Range: 01 to 18

Determines the width of the resonant peak of the second filter center.

22 — Speaker Low Pass Cutoff   Range: 2.0 to 16.0 KHz

This parameter filters out the high frequencies of the speaker. The lower the value, the less high frequencies pass through. True speaker emulations are provided as separate algorithms.

23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

Guitar Amp 4

Guitar Amp 4 is designed to recreate the warm sound and "touch" of vintage class "A" tube guitar amplifiers. This is done by carefully emulating their distortion characteristics. The waveshaping table used in creating this guitar amp simulation is a symmetrical table. A symmetrical table produces odd harmonics (no even harmonics). The Amp Tube Bias parameter is very important to this algorithm as it is used to dynamically alter the symmetry, thus producing even harmonics.

We recommend following this algorithm with any of the speaker algorithms.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Amp Preamp Gain   Range: -48 to +48 dB

Adjusts the amount of boost or cut applied to the incoming signal. We recommend a gain of 20 dB, since these emulations were optimized for distortion there. Lower preamp gains will result in less distortion, while higher preamp gains will yield more distortion. For low preamp gain, it may be desirable to use low tube bias values.

04 — Output Level   Range: 00 to 99

This parameter controls the output level of the main amp before the output EQ.

05 — Amp Level Detect Attack   Range: 50µs to 100ms

Controls the time it takes for the incoming signal to get to the Amp Tube Bias. Generally the attack should be short.

06 — Amp Level Detect Release   Range: 1ms to 10.0s

Sets the amount of time after the incoming signal has ceased for the amp level to shut down. Generally these times are longer than the attack times.

07 — Amp Tube Bias   Range: 00 to 99

For preamp gains approximately 00 dB, this dynamic parameter controls the emphasis of even to odd harmonics which determines the tone of the amp; mid values emphasize even harmonics and offer a warmer ("glowing tube") sound, while the highest values may sound like tubes going bad. Tube bias and preamp gain are independent parameters. For low preamp gain, it may be desirable to use low tube bias values, because this more closely imitates the operation of a real amplifier.

08 — Pre-EQ InputLevel Trim Range: -18 to +06 dB

Controls the input level to the pre-amp EQ to eliminate the possibility of clipping boosted signals.

09 — Pre-EQHighPass Cutoff Range: 4 to 1000 Hz

Filters out the low frequencies before the preamp. The higher the value, the less low frequencies pass through.

10 — Pre-EQ Fc   Range: 5 to 9999 Hz

This parameter determines the center frequency of the parametric filter before the preamp. Higher values have a brighter sound.

11 — Pre-EQ Gain
Range: -48 to +24 dB

Adjusts the amount of boost or cut applied to the parametric filter in front of the preamp.

12 — Pre-EQ Q
Range: 01 to 18

Determines the width of the resonant peak at the parametric filter center frequency. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

13 — Noise Gate Off Below
Range: -96 to +00 dB

Sets the lower threshold level at which the noise gate shuts off the audio. This parameter also automatically sets the noise gate to turn back on at 6 dB higher than the defined range, thus preventing hysteresis.

14 — Gate Release Time
Range: 1ms to 10.0s

This parameter sets the amount of time after the signal has elapsed for the noise gate to shut down. For a longer sustain, set this parameter higher.

15 — Speaker HighPass Cutoff
Range: 4 to 1000 Hz

This parameter filters out the low frequencies of the main amp prior to the speaker. The higher the value, the less low frequencies pass through.

16 — OutEQ1 Fc
Range: 5 to 9999 Hz

This parameter determines the filter center frequency of the parametric in the main amp stage. Higher values have a brighter sound.

17 — OutEQ1 Gain
Range: -48 to +24 dB

This parameter adjusts the amount of boost or cut applied to the main amp parametric.

18 — OutEQ1 Q
Range: 01 to 18

Determines the width of the resonant peak of the filter center. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

19 — OutEQ2 Fc
Range: 5 to 9999 Hz

This parameter determines the filter center frequency of the second parametric in the main amp stage. Higher values have a brighter sound.

20 — OutEQ2 Gain
Range: -48 to +24 dB

This parameter adjusts the amount of boost or cut applied to the second main amp parametric.

21 — OutEQ2 Q
Range: 01 to 18

This parameter determines the width of the resonant peak of the second filter center.

22 — Speaker Low Pass Cutoff
Range: 2.0 to 16.0 KHz

The parameter acts like a speaker, and filters out the high frequencies of the guitar signal. The lower the value, the less high frequencies pass through. This speaker filter is less selective than the speaker cabinet emulation algorithms.

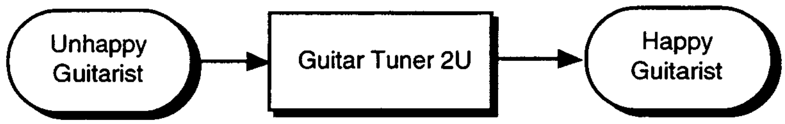
23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

GuitarTuner 2U

GuitarTuner 2U is a utility algorithm specifically optimized to tune a guitar, or a bass guitar. In Select mode, when a preset that uses this algorithm is selected, it automatically takes you to parameter 03, for instant tuning ability.

GuitarTuner 2U Signal Routing



01 — Mix
Range: 00 to 99

This parameter could be thought of as a reversed volume control. When this parameter is set to 99, the signal is muted. If it is desired that the algorithm pass signal, set the Mix to 00 (or bypass the unit).

02 — Volume
Range: 00 to 99

Adjusts the volume of the dry external signal — 00 is silent and 99 is full volume.

03 — Note
Range: A to G#

This parameter automatically detects the note being played, and interprets if the signal is sharp or flat. When the meter rests on the center line, you are tuned to the displayed note.

04 — Range
Range: Bass or Guitar

This parameter optimizes the frequency-detection range for bass (low signals) and guitar (high signals).

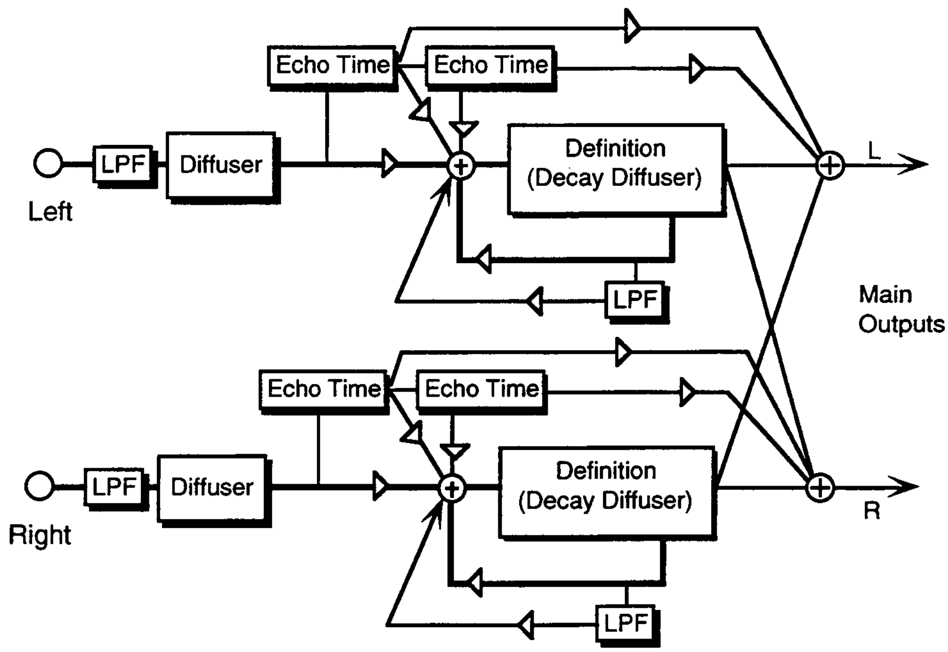
06 — Reference
Range: A438 to A445

Determines the reference pitch based on A=440 (the default setting). Different countries use different reference points, and this parameter allows the fine adjustment of that point.

Hall Reverb

Hall Reverb is a large acoustic space, and provides a high density reverb.

Hall Reverb Signal Routing



The signal enters a low pass filter, and goes directly through the diffusers which smear the signal. The signal is then routed to a larger decay diffuser, known as Definition, and is diffused over a period of time (creating a decay). There are taps from both the left and right Definition that are routed to the output to create a synthesized stereo output. A signal from the Definition goes through a low pass filter followed by a low frequency decay parameter, which controls the rate of decay of the low frequencies. There is also a parameter at this stage that controls the decay time of both the left and right signals. The left and right signals are routed back into the Definition. There are two echo times between the diffuser and the definition that can be routed directly to the output, or sent back through the definition. There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

These parameters are explained in detail under the Mix and Volume Parameters description, found in the beginning of this section. Reverbs sound best with a Mix of wet and dry.

03 — Decay
Range: 0.70 to 250.0 sec.

Controls the amount of time it takes for the reverberation to decay away to a very low level after the input signal stops. Higher values are recommended for the hall reverb.

04 — Predelay Time
Range: 0 to 450 ms

Controls the amount of time it takes for the original signal to be presented to the reverb. Higher values denote a longer delay.

05 — LF DecayTime
Range: -99 to +99

Functions as a tone control and boosts (when set to a positive value) or cuts (when set to a negative value) the rate at which low frequencies will decay.

06 — HF Damping
Range: 00 to 99

Controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.

07 — HF Bandwidth
Range: 01 to 99

The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass. This functions like a tone control on a guitar.

08 — Diffusion1
Range: 00 to 99

This parameter smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

09 — Diffusion2
Range: 00 to 99

This parameter, similar to and in series with Diffusion1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.

10 — Decay Definition
Range: 00 to 99

Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate. A general rule of thumb is this: Definition should not exceed the LF Decay Time added to the Decay Time.

11 — Detune Rate
Range: 00 to 99

Controls the LFO rate of detuning introduced into the reverberation decay. Detuning creates a slight oscillating pitch shift into the decay, giving it a more natural sound by breaking up resonant modes.

12 — Detune Depth   Range: 00 to 99

Controls the depth of the detuning, that is, how much the pitch will change. Low values yield a metallic sound. Some sounds may require very low values, while others sound more natural with higher values.

13 — Primary Send   Range: -99 to +99

Controls the level of the diffused input signal into the reverb definition.

14 — Ref 1 Time   Range: 0 to 120 milliseconds

Controls the delay time for the first pre-echo. Pre-echoes are the first sounds which have been reflected back from the walls or reflective "live" surfaces. Higher values delay the diffused signal more.

15 — Ref 1 Level   Range: 00 to 99

Controls the level of the first pre-echo. This pre-level controls the echo send to the Definition.

16 — Ref 1 Send   Range: 00 to 99

Controls the level of the first pre-echo, with the echo routed directly to the output.

17 — Ref 2 Time   Range: 0 to 120 milliseconds

Controls the delay time for the second pre-echo.

18 — Ref 2 Level   Range: 00 to 99

Controls the level of the second pre-echo. As a signal continues to bounce off the different reflective surfaces (walls), it decreases in volume. Set this parameter to a lower value than Ref 1 Level, in order to create a natural sounding echo.

19 — Ref 2 Send   Range: 00 to 99

Controls the level of the second pre-echo, with the echo routed directly to the output.

20 — Position Balance (1)
21 — Position Balance (2)
22 — Position Balance (3)   Ranges: -99 to +99

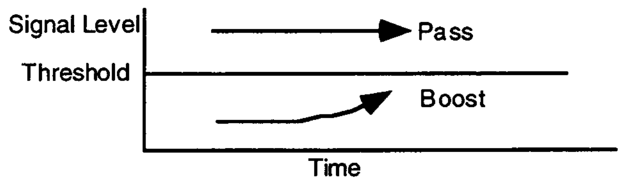
These parameters simulate the depth of the hall. Think of these parameters as three different microphones placing at various distances within the hall (parameter 20 is closest to the front, and parameter 22 is farthest from the front). When the range (volume) is higher for parameter 20, the sound appears closer to the front, whereas a higher setting for parameter 22 appears farther from the front, suggesting a deeper (wetter) hall.

23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

These modulation control parameters are identical for all of the algorithms and are explained in detail under the Algorithm Modulators description, found in the beginning of this section.

InversExpander

InversExpander creates sustain by expanding the signal so that the signal levels above threshold are passed and levels below threshold are boosted to create a more even sound. A traditional expander would have the opposite effect: that is a signal level below threshold would be attenuated. An inverse expander is much like a compressor in so far as they both can be used to create sustained sounds, and de-emphasize transient signals. EQ exists in both signal and side chain paths, in contrast to the Expander which has filtering in only the side chain path.



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. We recommend a Mix setting of 99.

03 — Expnd Ratio   Range: 1:1 to 40:1, Infinity

This parameter sets the amount of expansion. The range is based on decibels (dB) below the threshold. If this is set to 3:1 for example, it will expand the changes in signals that are below the threshold level 3 dB for each 1 dB change in the output. We recommend starting with settings near 1:1 (a setting of exactly 1:1 disables expansion).

04 — Threshold   Range: -96 to +00 dB

This sets the threshold level. Signals below this level will be boosted, while signals that are above will be unaffected. As the input signal dies away below the threshold, the expander will increase the signal gain. To turn off the inverse expander set the threshold to -96dB.

05 — Gain Change   Range: N/A

This read-only parameter displays a gain increase meter.

06 — Exp Attack   Range: 50µs to 100ms

Determines the time after the initial signal amplitude has been detected for the expansion to occur.

07 — Exp Release

Range: 1ms to 10.0s

Determines how long it takes for the expansion to be fully deactivated after the input signal rises above the threshold level. This is generally longer than the attack time.

08 — Exp Noise Gate Off Below

Range: -96 to +00 dB

This parameter sets the lower threshold level at which the noise gate shuts off the audio.

09 — Comp Noise Gate On Above

Range: -96 to +00 dB

Sets the upper threshold level at which the noise gate passes audio. This second parameter provides hysteresis.

10 — Bass Fc

Range: 0 to 1000 Hz

Sets the cutoff frequency of the lower frequency band shelving filter.

11 — Bass EQ Gain

Range: -48 to +24 dB

Sets the amount of boost or cut applied to the low shelving filter.

12 — Treble Fc

Range: 01KHz to 16KHz

Sets the cutoff frequency of the upper frequency band high shelving filter.

13 — Treble EQ Gain

Range: -48 to +24 dB

This parameter sets the amount of boost or cut applied to the high shelving filter.

14 — EQ Input Level Trim

Range: -24 to +00 dB

Adjusts the input volume of the EQs, to eliminate the possibility of clipping boosted signals.

15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

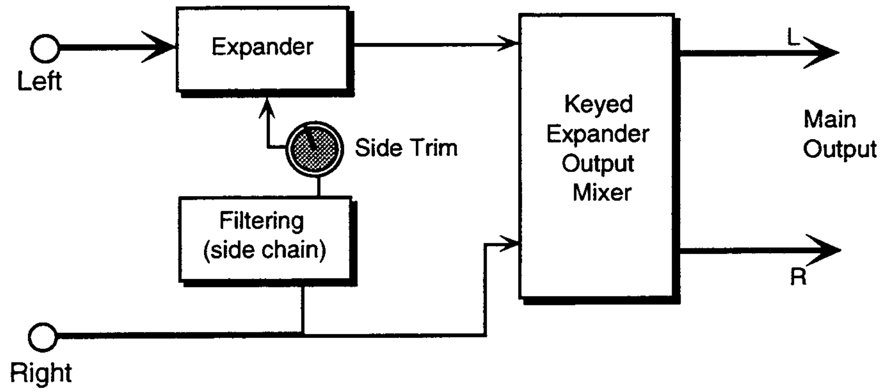
See the descriptions under the Algorithm Modulators at the beginning of this section.

Keyed Expander

Keyed Expander operation is identical to the Expander. The only difference is that the left signal (Input 1) is expanded as determined by the key. The key is the right channel signal (Input 2). This effect is often used in studios to "tighten" rhythm tracks (e.g., a rhythm guitar in Input 1 is tightened by a different signal, like a drum machine connected to Input 2).

☞ **Important:** This special algorithm is only made available in the DP/2 as a ROM Config Preset (location #96 in Bank 1), because it requires special input signal routing.

Keyed Expander Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. The Mixer works as in all the other algorithms, and is distinct from the output mixer shown in the diagram above.

03 — Exp Ratio — Range: 1:1 to 1:40, infinity

Sets the amount of expansion. The range is based on decibels (dB) below the threshold. If this is set to 1:4 for example, it will expand changes in signals below the threshold by a factor of four. When this is set to infinity, it acts as a gate. A setting of 1:1 offers no expansion.

04 — Exp Threshold — Range: -96 to +00 dB

Sets the threshold level. Signals that exceed this level will be unaffected, while signals that are below will be expanded. To turn off the expander, set the level to -96 dB.

05 — Gain Change — Range: N/A

This read-only parameter displays the amount of gain reduction in real time.

06 — Exp Attack — Range: 50µs to 100ms

Determines the attack rate after the initial signal has been detected and before the expansion takes affect.

07 — Exp Release — Range: 1ms to 10.0s

Determines the release rate after the signal has been detected below the threshold level. This is generally chosen longer than the attack time (parameter 06).

08 — Expander Gate Hold Time — Range: 1ms to 10.0s

This is the detection sustain time in the ADSR which constitutes attack, sustain, and release.

09 — Sidechain EQ Gain   Range: -48 to +48 dB

Controls the amount of boost applied to the output signal of the high/low pass filter. This accounts for insertion loss through those filters.

10 — HighPass Fc   Range: 4 to 8000 Hz

Sets the cutoff frequency of the lower frequency band high pass shelving filter.

11 — LowPass Fc   Range: 100 Hz to 16 KHz

Sets the amount of boost or cut applied to the low pass filter.

12 — Trigger Mask   Range: Off or On

Enables the trigger mask function. Once triggered, the side chain detector will see no input signal for a duration specified by parameter 13.

13 — Trigger Time   Range: 1ms to 10.0s

This parameter sets the duration over which the side chain detector will be blacked out. This parameter is useful for isolating the first bar of a drum track.

14 — Trigger Mask Threshold   Range: -96 to +00 dB

Sets the trigger mask threshold level. Signals that fall below this level will trigger the mask function. The trigger mask function uses the Expander Threshold (04) as upward hysteresis. Therefore, the Trigger Mask Threshold should always be set lower than the Expander Threshold.

15 — Expander Output Mix   Range: 00 to 99

Mixes the output of the left signal (Input 1) with the output of the right signal (Input 2). This is the output mixer which is shown in the diagram.

16 — Expander Output Gain   Range: -48 to +48 dB

Sets the amount of cut (negative values) or boost (positive values) applied to the expander on the output volume. We recommend a starting application of +00 dB.

17 — Mod1 Source
18 — Mod1 Destination
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

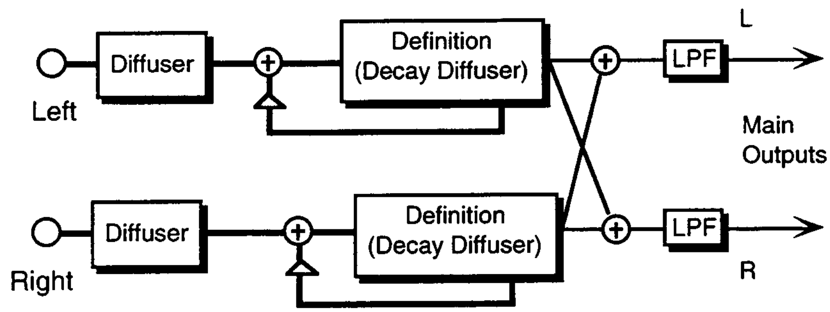
See the descriptions under the Algorithm Modulators in the beginning of this section.

Large Plate

A plate reverb takes the vibrations from a metal plate and uses them to create a metallic sounding reverb. Large plate reverbs are often used to enhance a vocalist's performance.

Large Plate simulates a larger plate reverb.

Large Plate Reverb Signal Routing



The two plate reverb algorithms share exactly the same signal routing topology. The internal values of the components (not user programmable) differentiate the large and small plate reverbs. The signal goes directly through the diffusers which smear the signal. The signal is then routed to a larger decay diffuser, known as Definition, and is diffused over a period of time (creating a decay). The signal is then routed to the output, and then goes through a low pass filter. There is a parameter that controls the decay time of both the left and right signals (shown as triangles above). This signal is then routed back into the definition. There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section.

03 — Decay   Range: 0.40 to 140.0 sec.

Controls the amount of time it takes for the reverberation to decay. High values of decay sound good on these algorithms.

04 — Predelay Time   Range: 0 to 430 ms

Controls the amount of time it takes for the input signal to be presented to the plate reverb. A value of 0 would offer no delay.

05 — HF Damping   Range: 00 to 99

Increasing the value of this parameter will gradually filter out increasing amounts of high frequency energy. Higher values yield an abrupt decay. This parameter controls the cut off of a low pass filter in series with the decay within the definition.

06 — HF Bandwidth   Range: 01 to 99

This parameter acts as a low pass filter on the output of the plate reverbs, controlling the amount of high frequencies present. The higher the setting, the more high frequencies are allowed to pass through, offering a brighter ringing sound. Some interesting effects can be created by using a mod controller over a large range.

07 — Diffusion 1 Range: 00 to 99

Smears the input signal to create a smoother sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear, making the echoes less apparent.

08 — Diffusion 2 Range: 00 to 99

This Diffuser, similar to and in series with the previous one, offers control over lower frequency ranges. Plate reverbs tend to sound metallic, and the diffusers help to smear the signal, eliminating the metallic sound.

09 — Decay Definition Range: 00 to 99

Controls the rate at which echo density increases with time. Higher values can cause the echo density to build at a rate that exceeds the decay rate. For the best results, try to select the highest value that works with your sound source.

10 — Early Ref Level 1
11 — Early Ref Level 2
12 — Early Ref Level 3
13 — Early Ref Level 4 Ranges: -99 to +99

Control four early reflection levels. Setting these levels to lower values will produce a wetter sound. These four reflection levels are close to the input of the Decay Definition.

14 — Left/Right Balance Range: -99 to +99

Controls the left/right stereo balance of the plate reverb signal. A setting of -99 would offer hard left, whereas a setting of +99 would offer hard right. A setting of +00 would place the reverb in the center of the stereo spectrum.

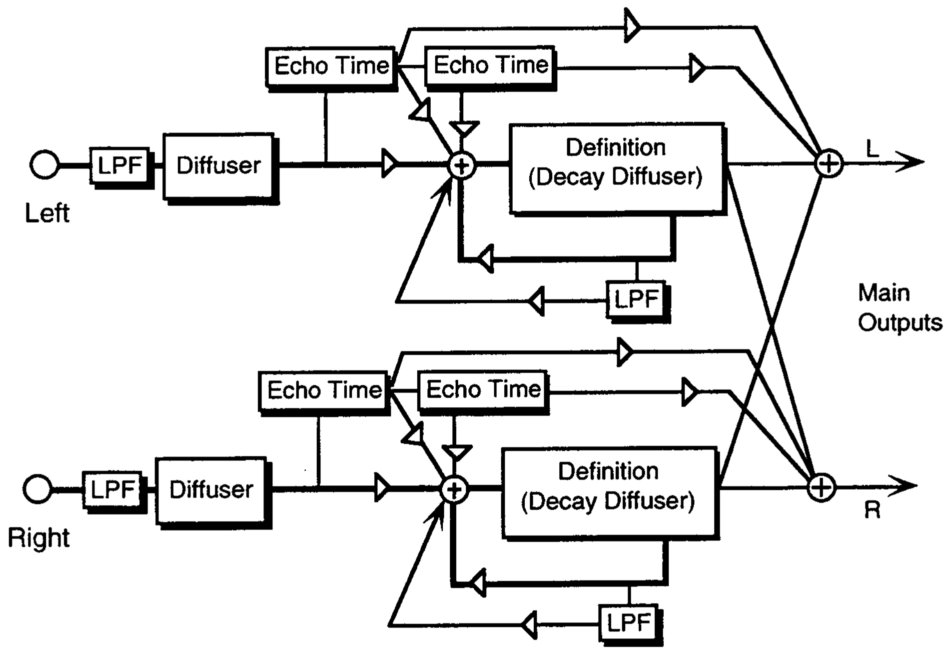
15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Large Room Rev

Large Room Rev, larger than Small Room Rev, provides ambience.

Large Room Rev Signal Routing



The signal enters a low pass filter, and goes directly through the diffusers which smear the signal. The signal is then routed to a larger decay diffuser, known as Definition, and is diffused over a period of time (creating a decay). There are taps from both the left and right Definition that are routed to the output to create a synthesized stereo output. A signal from the Definition goes through a low pass filter followed by a low frequency decay parameter, which controls the rate of decay of the low frequencies. There is also a parameter at this stage that controls the decay time of both the left and right signals. The left and right signals are routed back into the Definition. There are two echo times between the diffuser and the definition that can be routed directly to the output, or sent back through the definition. There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the Mix parameter (01).

01 — Mix
02 — Volume

These parameters are explained in detail under the Mix and Volume Parameters description, found in the beginning of this section. Reverbs sound best with a Mix of wet and dry.

03 — Decay

Range: 0.20 to 150.0 sec.

Controls the amount of time it takes for the reverberation to decay away to a very low level after the input signal stops. In room reverbs, we don't recommend higher settings, which tend to create an infinite and unnatural sustain. Since most ambient room reverbs don't naturally have a large decay, set this low for the best sound.

04 — Predelay Time

Range: 0 to 450 ms

Controls the amount of time it takes for the original signal to be presented to the reverb. Higher values denote a longer delay.

05 — LF DecayTime Range: -99 to +99

Functions as a tone control and boosts (when set to a positive value) or cuts (when set to a negative value) the rate at which low frequencies will decay.

06 — HF Damping Range: 00 to 99

Controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.

07 — HF Bandwidth Range: 01 to 99

The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass. This functions like a tone control on a guitar.

08 — Diffusion1 Range: 00 to 99

This parameter smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

09 — Diffusion2 Range: 00 to 99

This parameter, similar to and in series with Diffusion1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.

10 — Decay Definition Range: 00 to 99

Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate. A general rule of thumb is this: Definition should not exceed the LF Decay Time added to the Decay Time.

11 — Detune Rate Range: 00 to 99

Controls the LFO rate of detuning introduced into the reverberation decay. Detuning creates a slight oscillating pitch shift into the decay, giving it a more natural sound by breaking up resonant modes.

12 — Detune Depth Range: 00 to 99

Controls the depth of the detuning, that is, how much the pitch will change. Low values yield a metallic sound. Some sounds may require very low values, while others sound more natural with higher values.

13 — Primary Send Range: -99 to +99

Controls the level of the diffused input signal into the reverb definition.

14 — Ref 1 Time Range: 0 to 120 milliseconds

Controls the delay time for the first pre-echo. Pre-echoes are the first sounds which have been reflected back from the walls or reflective "live" surfaces. Higher values delay the diffused signal more.

15 — Ref 1 Level Range: 00 to 99

Controls the level of the first pre-echo. This pre-level controls the echo send to the Definition.

16 — Ref 1 Send Range: 00 to 99

Controls the level of the first pre-echo, with the echo routed directly to the output.

17 — Ref 2 Time Range: 0 to 120 milliseconds

Controls the delay time for the second pre-echo.

18 — Ref 2 Level Range: 00 to 99

Controls the level of the second pre-echo. As a signal continues to bounce off the different reflective surfaces (walls), it decreases in volume. Set this parameter to a lower value than Ref 1 Level, in order to create a natural sounding echo.

19 — Ref 2 Send Range: 00 to 99

Controls the level of the second pre-echo, with the echo routed directly to the output.

20 — Position Balance (1)
21 — Position Balance (2)
22 — Position Balance (3) Ranges: -99 to +99

These parameters simulate the depth of the room. Think of these parameters as three different microphones placing at various distances within the room (parameter 20 is closest to the front, and parameter 22 is farthest from the front). When the range (volume) is higher for parameter 20, the sound appears closer to the front, whereas a higher setting for parameter 22 appears farther from the front, suggesting a deeper (wetter) room.

23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

These modulation control parameters are identical for all of the algorithms and are explained in detail under the Algorithm Modulators description, found in the beginning of this section.

MultiTap Delay

MultiTap Delay produces four independent controllable delays. This algorithm requires only one unit, allowing the other three units to be free for other algorithms.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. This algorithm sounds best with a Mix of wet and dry.

03 — MultiTap 1 Time
07 — MultiTap 2 Time
11 — MultiTap 3 Time
15 — MultiTap 4 Time

Ranges: 0 to 1834 ms

These four parameters set the amount of delay time for the independent delays. Experiment with the different settings to find the right mix for your sound source and application. Some interesting effects can be created by using a real time modulation controller for these parameters.

04 — MultiTap 1 Level
08 — MultiTap 2 Level
12 — MultiTap 3 Level
16 — MultiTap 4 Level

Ranges: 00 to 99

These four parameters adjust the volume of the delayed signals against the original dry signal. Levels of 00 will offer no audible delay.

05 — MultiTap 1 Regen
09 — MultiTap 2 Regen
13 — MultiTap 3 Regen
17 — MultiTap 4 Regen

Ranges: 00 to 99

These parameters determine the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delays. A setting of 99 would offer an infinite delay.

06 — MultiTap 1 Pan
10 — MultiTap 2 Pan
14 — MultiTap 3 Pan
18 — MultiTap 4 Pan

Ranges: -99 to +99

These parameters determine the location of the four controllable delays in the stereo spectrum. A value of -99 is panned far left, and +99 is far right.

19 — MultiTap Regen Damping

Range: 00 to 99

Controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The higher the number, the more the signals are damped.

20 — Mod1 Source
21 — Mod1 Destination
22 — Mod1 Param Range Min
23 — Mod1 Param Range Max
24 — Mod2 Source
25 — Mod2 Destination
26 — Mod2 Param Range Min
27 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, in the beginning of this section.

No Effect (Bypass Effect)

No Effect will bypass the unit, providing no effect. Whether or not this utility algorithm passes audio (bypass) or squelches it (kill) is controlled in the Edit/Config parameters and is explained in more detail in *Section 3 — Config Parameters.*

01 — Mix

Range: 00 to 99

Controls the mix of silence with a dry audio signal. In other words, this algorithm has two signals, one that is silent, and a signal that is not. When this parameter is set to 00, you select the audible signal. When set to 99, you select the silent signal. This parameter could be thought of as a reversed volume control.

02 — Volume

Range: 00 to 99

Adjusts the volume of the dry external signal — 00 is silent and 99 is full volume.

03 — Mod1 Source
04 — Mod1 Destination
05 — Mod1 Param Range Min
06 — Mod1 Param Range Max
07 — Mod2 Source
08 — Mod2 Destination
09 — Mod2 Param Range Min
10 — Mod2 Param Range Max

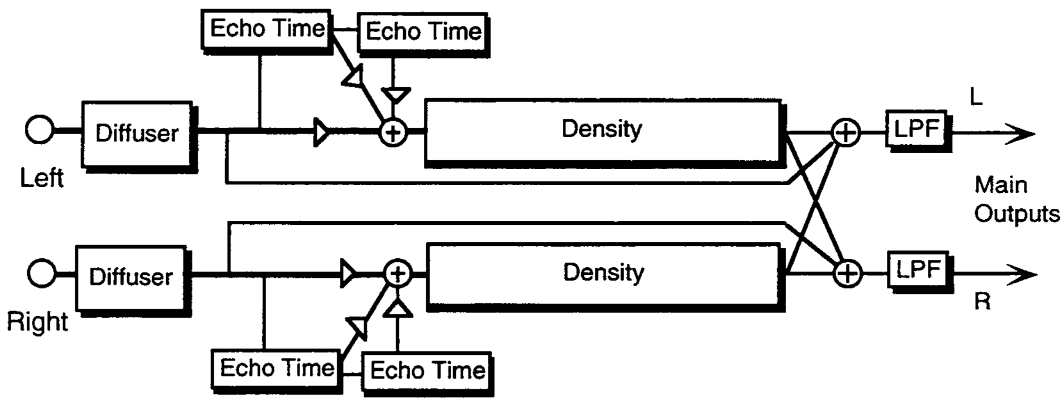
See the descriptions under the Algorithm Modulators in the beginning of this section.

NonLin Reverb1, 2, 3

Non Linear reverbs can be used to obtain blooming reverb, gated reverb, reverse reverb and early reflections. In general, they do not produce an exponentially decaying reverb. Unlike the hall, room and plate reverbs, Non Lin 1, 2, and 3 pass the input signal through the reverb diffusers only once. For this reason the reverb diffusers are called *Density*, to distinguish them from the other reverb diffusers (called Definition). Density controls the *amount* of echo density, as opposed to the rate of increase of echo density. Other reverbs give limited control of early reflections. For more control, try using these algorithms in serial or parallel with other reverbs to emphasize the early reflections. The Non Lin reverbs purposely impose a coloration on the resulting sound.

NonLin Reverb1 is optimized for shorter duration effects (approx. 0.5 sec.).
NonLin Reverb2 offers approx. a 1.5 sec. duration.
NonLin Reverb3 is sonically similar to Non Lin 1, but there is less stereo movement, making it better suited for drum tracks.

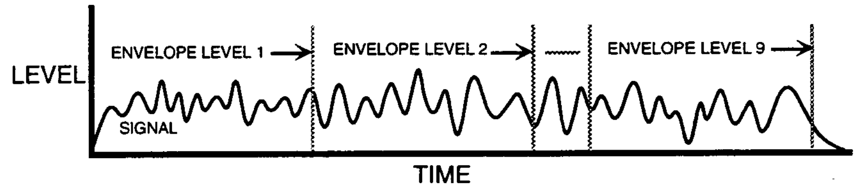
NonLin Reverb Signal Routing



The signal goes directly through a diffuser which smears the signal. The signal is then routed to a decay diffuser (Density), and is diffused over a period of time. Within the density the signal goes through a high frequency damper. The signal is then routed to the output. After the density, the signal passes through a low pass filter. There are two echo times between the diffuser and the density. There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.



03 — Envelope Level 1
04 — Envelope Level 2
05 — Envelope Level 3
09 — Envelope Level 4
07 — Envelope Level 5
08 — Envelope Level 6
09 — Envelope Level 7
10 — Envelope Level 8
11 — Envelope Level 9 Ranges: 00 to 99

These parameters control the output tap levels sequenced in time across the density from input to output. Envelope Level 1 is tapped right after the diffusers and before the echoes (see the topology). If this is undesirable, set Envelope Level 1 to 00. Envelope Levels 8 and 9 are positioned at the very end of the Density; setting these too high can cause excessive ringing. Envelope Levels 8 and 9 are also very dry. Set all nine tap levels to find the envelope for your application. We recommend the average Envelope Level not to exceed a value of 45 to prevent overdriving these three reverbs.

12 — NonLin HF Damping Range: 00 to 99

The HF Damping is located within the density. This parameter selects the amount of high frequency energy to be filtered out.

13 — NonLin HF Bandwidth Range: 01 to 99

The high frequency bandwidth parameter acts as a low pass filter on the output signal, controlling the amount of high frequencies that will be heard. The higher the setting, the more high frequencies are heard. This works the same way that a tone control would work on a guitar.

14 — NonLin Diffusion1 Range: 00 to 99

This parameter smears the input signal transients of higher frequency ranges. Higher values are recommended for smoother percussion. Very low values will give a highly repetitive echo-like sound. Diffusion1 and 2 exist within each diffuser block (see diagram).

15 — NonLin Diffusion2 Range: 00 to 99

Diffusion2 is similar to Diffusion1, but offers control of lower frequencies. In general a setting of 50 can be considered an equal mix of dry/diffused sound; this setting is a good starting point.

16 — NonLin Density 1 Range: 00 to 99

Density 1 controls the number of echoes.

17 — NonLin Density 2 Range: 00 to 99

Density 2 controls the number of echoes in a lower frequency range. In general, to get the smoothest sound, Density 2 is usually less than the value of Density 1.

18 — NonLin Primary Send Range: -99 to +99

Controls the level of the diffused input signal which is nearly instantaneous with respect to the input. This signal is injected directly into the Density at the specified level.

19 — Reflection 1 Time Non Lin 1, 3 Range: 0 to 600 ms

Non Lin 2 Range: 0 to 85 ms

Controls the amount of time it takes for the first pre-echo to be injected into the Density. Pre-echoes are the sounds which have been reflected back from the walls or other reflective surfaces.

20 — Reflection 1 Send Range: -99 to +99

This parameter controls the level of the first pre-echo.

21 — Reflection 2 Time Non Lin 1, 3 Range: 0 to 600 ms

Non Lin 2 Range: 0 to 85 ms

This controls the amount of time it takes for the second pre-echo to be injected into the Density.

22 — Reflection 2 Send Range: -99 to +99

This parameter controls the level of the second pre-echo. Experiment with both positive and negative on all echoes to change the tonal character of the results.

23 — Left/Right Balance Range: -99 to +99

Controls the left/right stereo balance of the reverb signal. A setting of -99 would offer hard left, whereas a setting of +99 would offer hard right. A setting of +00 would place the reverb in the center of the stereo spectrum.

24 — Mod1 Source
25 — Mod1 Destination
26 — Mod1 Param Range Min
27 — Mod1 Param Range Max
28 — Mod2 Source
29 — Mod2 Destination
30 — Mod2 Param Range Min
31 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

Parametric EQ

Parametric EQ offers a minimum phase four band parametric EQ.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. For this algorithm, we recommend a setting of 99.

03 — Bass Fc Range: 0 to 1000 Hz

Sets the center of the low frequency EQ.

04 — Bass Gain (loShv) Range: -48 to +24 dB

Sets the amount of boost or cut applied to this low frequency parametric.

05 — Mid1 Fc Range: 50 to 9999 Hz

Sets the center of the mid frequency parametric.

06 — Mid1 Gain Range: -48 to +24 dB

Sets the amount of boost or cut applied to this mid frequency parametric.

07 — Mid1 Q Range: 01 to 18

This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. This parameter is equal to the cutoff frequency divided by the bandwidth. By raising the value, you can produce a narrower bandwidth.

08 — Mid2 Fc
09 — Mid2 Gain
10 — Mid2 Q

These three parameters are identical to the previous three parameters, and are used to control different bandwidths within the mid range.

11 — Treble Fc Range: 01 to 16 KHz

Sets the center frequency of the high frequency parametric.

12 — Treble Gain (HiShv) Range: -48 to +24 dB

Sets the amount of boost or cut applied to this high frequency parametric.

13 — EQ Input Level Attenuation Range: -24 to +00 dB

Adjusts the input level trim to the EQs to eliminate the possibility of clipping boosted signals.

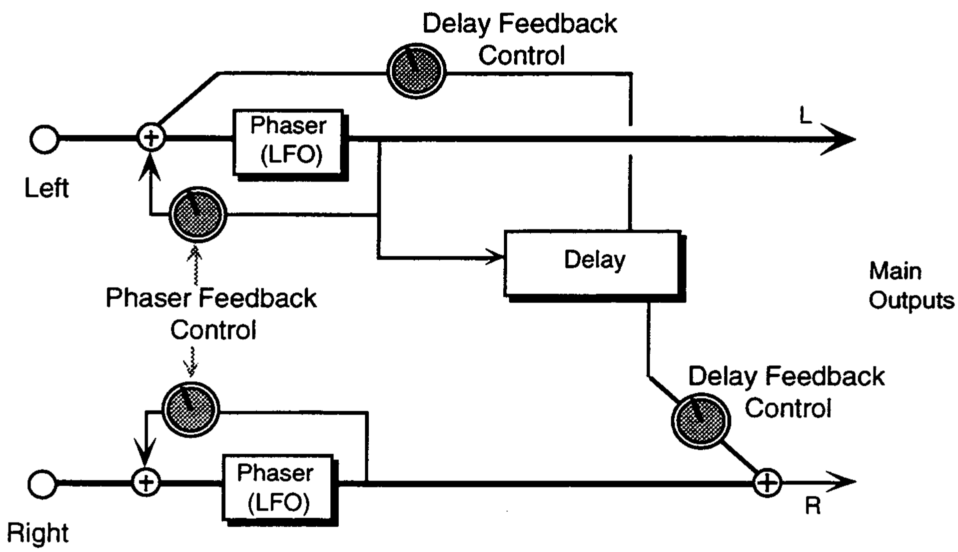
14 — Mod1 Source
15 — Mod1 Destination
16 — Mod1 Param Range Min
17 — Mod1 Param Range Max
18 — Mod2 Source
19 — Mod2 Destination
20 — Mod2 Param Range Min
21 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

Phaser - DDL

Phaser - DDL combines a phaser with a digital delay. The phaser creates non-harmonically spaced moving notches in the signal spectrum, whereas a flanger creates harmonic spacing. This phaser implements a stereo twelve pole phasing network to achieve time delay which is a function of frequency (i.e., phase delay); this is what differentiates the phaser from the flanger. The phasing effect is achieved within the phaser topology, so it does not depend upon the external mix. A delay is included at the left output of the phaser which feeds back into the phaser (see the diagram). Setting the phaser delay feedback parameter (shown at the knobs) to 00 will disable this delay function. The delay feedback also controls the delay feed forward level of another tap sent to the right channel. This delay topology achieves a 1.5 second ping-pong effect, and is very effective as a "poor man's reverb."

Phaser - DDL Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. For this algorithm we recommend a Mix setting of 99.

03 — Phaser LFO Rate   Range: 00 to 99

The LFO is within the phaser network. This parameter controls the rate of modulation of the phaser poles. The higher the value, the faster the rate. Lower values work best with sustained signals.

04 — Phaser LFO Width   Range: 00 to 99

Controls the width of the notch *excursion*. For large excursions set this parameter to 99. Doing so can give a very high "woosh" and a very low "woosh."

05 — Phaser Center   Range: -99 to +99

Controls the phaser pole center. High values raise the nominal spectral location of the "woosh" sound, while low values lower the "woosh." The range from high to low is controlled with the phaser width.

06 — Phaser Feedback   Range: -99 to +99

Controls the amount of feedback applied to the left and right channel phaser. The sign of the value determines the polarity of the feedback.

07 — Phaser Notch Depth   Range: -99 to +99

Controls the depth of the notches created by the phasing. Deep notches occur in the phased spectrum when the parameter is set to 99. When this parameter is set to +00, there exists no phasing (i.e. notches), but there is a doppler effect with higher LFO rates.

08 — Left/Right LFO   Range: Out-of-Phase or In-Phase

Determines whether the phaser on the left and right channels is modulating in or out-of phase.

09 — Phaser Sample & Hold Rate   Range: Off, 001 to 100

Controls the sample rate of a sample and hold network applied to the LFO within the phaser. When in hold, the effect will be to create momentarily fixed notches within the frequency spectrum (if the notch depth is not +00). A setting of 001 will have the largest space between samples. Higher values will increase the number of holds per second, making the phasing flow more smoothly. The sample and hold function can be turned off.

10 — Phaser Left Delay Time   Range: 0 to 1600 ms

This parameter sets the delay time for the left side. This is the "ping."

11 — Phaser Right Delay Time   Range: 0 to 1600 ms

Controls the feed forward delay time for the right side. This is the "pong."

12 — Phaser Delay Feedback   Range: -99 to +99

Controls the feedback amount for the delay effect. The sign of the value determines the polarity of the feedback. A value of +00 will eliminate the delay effect. This parameter also controls the feed forward level (see diagram).

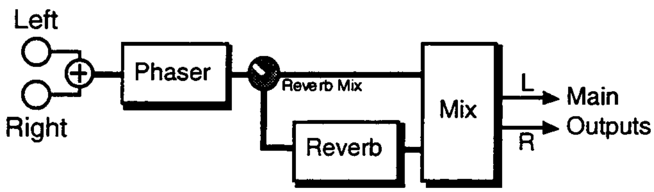
13 — Mod1 Source
14 — Mod1 Destination
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found earlier in this section.

Phaser-Reverb

Phaser-Reverb combines a phaser with a plate reverb.

Phaser-Reverb Signal Routing



The signal enters a stereo phaser, which is heard directly at the output. The signal is then routed out of the phaser into the large plate reverb. There is also a dry signal (not shown) that goes directly from the input to the output and is controlled with the Mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

03 — Reverb Mix   Range: 00 to 99

Controls the mix between the phased signal and the reverb. Setting this parameter to 00 will allow only the phased signal to be heard, while a setting of 99 will send all of the phased signal to the reverb.

04 — Phaser LFO Rate   Range: 00 to 99

The LFO is within the phaser network. This parameter controls the rate of modulation of the phaser poles. The higher the value, the faster the rate. Lower values work best with sustained signals.

05 — Phaser LFO Width   Range: 00 to 99

Controls the width of the notch *excursion*. For large excursions set this parameter to 99. Doing so can give a very high "woosh" and a very low "woosh."

06 — Phaser Center   Range: -99 to +99

Controls the phaser pole center. High values raise the nominal spectral location of the "woosh" sound, while low values lower the "woosh." The range from high to low is controlled with the phaser width.

07 — Phaser Feedback   Range: -99 to +99

Controls the amount of feedback applied to the left and right channel phaser. The sign of the value determines the polarity of the feedback.

08 — Phaser Notch Depth   Range: -99 to +99

Controls the depth of the notches created by the phasing. Deep notches occur in the phased spectrum when the parameter is set to 99. When this parameter is set to +00, there exists no phasing (i.e. notches), but there is a doppler effect with higher LFO rates.

09 — Large Plate Decay   Range: 0.40 to 140.0 sec.

Controls the amount of time it takes for the reverberation to decay. High values of decay sound good on these algorithms.

10 — Plate Predelay Time   Range: 0 to 250 ms

Controls the amount of time it takes for the original signal to be presented to the reverb. Higher values denote a longer delay.

11 — Large Plate HF Damping   Range: 00 to 99

Shapes the tone of the reverb decay. Higher settings cause the high frequency components to decay more rapidly.

12 — Large Plate HF Bandwidth   Range: 00 to 99

This parameter acts as a low pass filter on the input of the plate reverb, controlling the amount of high frequencies present. The higher the setting, the more high frequencies are allowed to pass through, offering a brighter ringing sound. Some interesting effects can be created by using a mod controller over a large range.

13 — Plate Diffsn1   Range: 00 to 99

This parameter smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

14 — Diffusion2   Range: 00 to 99

This parameter, similar to and in series with Diffusion1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.

15 — Plate Decay Definition   Range: 00 to 99

Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate.

16 — Mod1 Source
17 — Mod1 Destination
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

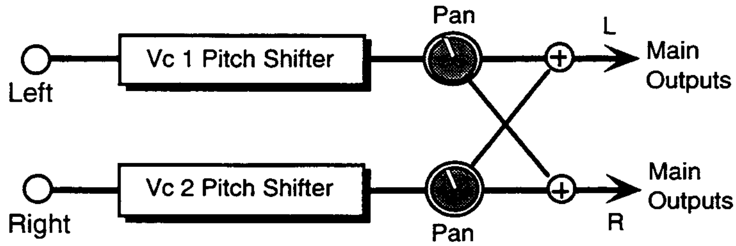
See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Pitch Shift 2U

Pitch Shifters allow you to change the pitch of a signal to any pitch desired within a range of one octave in either direction. **Pitch Shift 2U** is a splicer-type incorporating zero crossing detection.

Pitch Shift 2U has incorporated one ESP chip for zero crossing (pitch) detection for splice synchronization, having an optimal detection range of 55 to 555Hz. Splicer type pitch shifters are popular because for low pitch shift ratios, splicing is infrequent. These pitch shifters can create very interesting stereo fields — by panning each of two pitch shifted voices selectively, and because of the inherent time delay modulation of the algorithm. Pitch Shift 2U takes the left channel input as Voice 1, and the right channel input as voice 2.

Pitch Shift 2U Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. These algorithms sound best with a Mix of wet and dry. Try using a modulation controller for the Mix parameter to bring in or fade out the pitch shifted signal.

03 — PitchShifter Vc 1 Semi Range: -12 to +12

Allows you to adjust the pitch of Voice 1 up to an octave above or below the original pitch in semi-tones (half steps).

04 — PitchShifter Vc 1 Fine Range: -99 to +99

This parameter allows you to fine tune the pitch of Voice 1.

05 — PitchShifter Vc 1 Level Range: 00 to 99

Adjusts the volume of Voice 1. A setting of 00 would eliminate any audible pitch shift.

06 — PitchShifter Vc 1 Pan   Range: -99 to +99

Allows you to assign the location of output Voice 1 in the stereo field. A value of -99 would be far left, and +99 would be far right.

07 — PitchShifter Vc 2 Semi   Range: -12 to +12

Adjusts the pitch of Voice 2 up to an octave above or below the original pitch in semi-tones.

08 — PitchShifter Vc 2 Fine   Range: -99 to +99

This parameter allows you to fine tune the pitch of Voice 2.

09 — PitchShifter Vc 2 Level   Range: 00 to 99

Adjusts the volume of Voice 2. A setting of 00 would eliminate any audible pitch shift.

10 — PitchShifter Vc 2 Pan   Range: -99 to +99

Allows you to assign the location of output Voice 2 in the stereo field. A value of -99 would be far left, and +99 would be far right.

11 — PitchShifter LFO Rate   Range: 00 to 99

This parameter controls the rate of pitch modulation which creates a chorusing effect. To achieve chorusing, this rate must be very low.

12 — PitchShifter LFO Width   Range: 00 to 99

This parameter controls the excursion of pitch modulation. Since the rate is usually very low, then the width is usually very large.

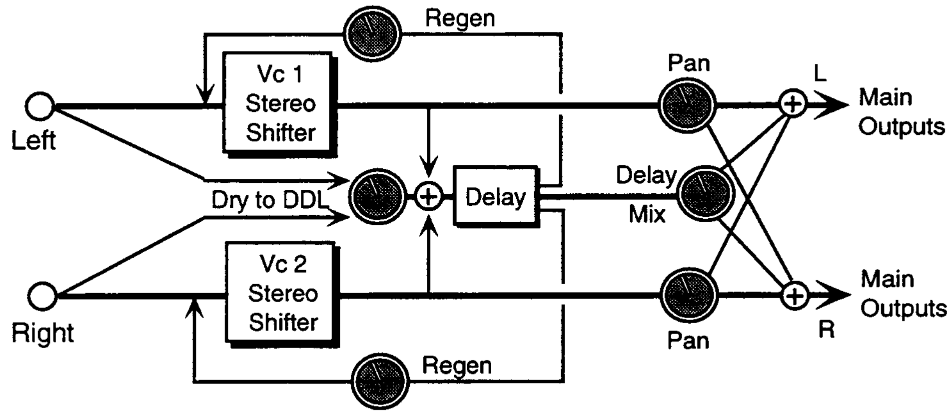
Mod1 Source
Mod1 Destination
Mod1 Param Range Min
Mod1 Param Range Max
Mod2 Source
Mod2 Destination
Mod2 Param Range Min
Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

PitchShift-DDL

PitchShift-DDL combines a pitch shifter with a digital delay. PitchShift-DDL uses a continual crossfading technique of pitch shifting. This technique maintains the stereo field exactly. Of the one unit pitch shifters, this one works best for large pitch shift ratios, in some circumstances. Another feature of this algorithm is a digital delay that feeds back into the pitch shift. This feature allows spiraling upward or downward pitch shifts.

PitchShift-DDL Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters earlier in this section. This algorithm sounds best with a Mix of wet and dry. Try using a modulation controller for the Mix parameter to fade in or fade out the pitch shifted signal.

03 — PitchShift Vc 1 Semi Range: -12 to +12

Adjusts the pitch of Voice 1 up to an octave above or below the original pitch in semi-tones.

04 — PitchShift Vc 1 Fine Range: -99 to +99

This parameter allows you to fine tune the pitch of Voice 1.

05 — PitchShift Vc 1 Level Range: 00 to 99

This parameter adjusts the volume of Voice 1.

06 — PitchShifter Vc 1 Pan Range: -99 to +99

Allows you to assign the location of Voice 1 in the stereo field. A value of -99 would be far left, and +99 would be far right.

07 — PitchShift Vc 2 Semi Range: -12 to +12

Allows you to adjust the pitch of Voice 2 up to an octave above or below the original pitch in semi-tones.

08 — PitchShift Vc 2 Fine Range: -99 to +99

Allows you to fine tune the pitch of Voice 2. Slight shifts create something like a chorused effect.

09 — PitchShift Vc 2 Level Range: 00 to 99

This parameter adjusts the volume of Voice 2.

10 — PitchShifter Vc 2 Pan Range: -99 to +99

Allows you to assign the location of Voice 2 in the stereo field. A value of -99 would be far left, and +99 would be far right.

11 — PitchShift Dry Level to DDL Range: 00 to 99

Lets you bypass the pitch shifter with an internal dry signal, and send it through the digital delay. Higher values would send more of the dry signal to the delay. The purpose of this parameter is to mix the dry signal appropriately with the pitch shifted delay signals.

12 — PitchShift Left Delay Time Range: 0 to 1500 ms

Controls the amount of time for the pitch shifted signal to delay from the left input.

13 — PitchShift Right Delay Time Range: 0 to 1500 ms

Controls the amount of time for the pitch shifted signal to delay from the right input.

14 — PitchShift Delay Mix Range: 00 to 99

Controls the mix between the delay signal and the pitch shifted signal. A setting of 00 would be all pitch shifter, and no delay. A setting of 99 will be all delay and no direct pitch shift.

15 — PitchShift Delay Regen Range: -99 to +99

Controls the amount of feedback from the output of the delay back into the input of the pitch shifter. This allows you to create special effects with ascending/descending delays.

16 — Mod1 Source
17 — Mod1 Destination
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

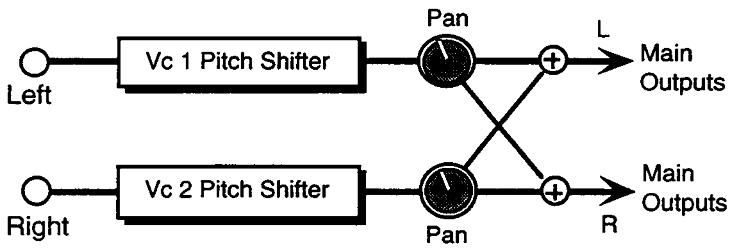
See the descriptions under the Algorithm Modulators earlier in this section.

PitchShifter

A pitch shifter allows you to change the pitch of a signal to any pitch desired within a range of one octave in either direction. **PitchShifter** offers a 1 unit splicer type pitch shifter.

Try the different pitch shifters until you find the one that works best with your sound source, and for your application. A "splicer type" of pitch shifter will drop or add small sections of the original signal to the effect. **PitchShifter** uses only one unit, but does not incorporate zero crossing detection. This pitch shifter is best used for a doubling effect. Splicer type pitch shifters are popular because for low pitch shift ratios, splicing is infrequent. These pitch shifters can create very interesting stereo fields by panning each of two pitch shifted voices selectively, and because of the inherent time delay modulation of the algorithm. This pitch shifters takes the left channel input as Voice 1, and the right channel input as voice 2.

PitchShifter Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. These algorithms sound best with a Mix of wet and dry. Try using a modulation controller for the Mix parameter to bring in or fade out the pitch shifted signal.

03 — PitchShifter Vc 1 Semi   Range: -12 to +12

Allows you to adjust the pitch of Voice 1 up to an octave above or below the original pitch in semi-tones (half steps).

04 — PitchShifter Vc 1 Fine   Range: -99 to +99

This parameter allows you to fine tune the pitch of Voice 1.

05 — PitchShifter Vc 1 Level   Range: 00 to 99

Adjusts the volume of Voice 1. A setting of 00 would eliminate any audible pitch shift.

06 — PitchShifter Vc 1 Pan Range: -99 to +99

Allows you to assign the location of output Voice 1 in the stereo field. A value of -99 would be far left, and +99 would be far right.

07 — PitchShifter Vc 2 Semi Range: -12 to +12

Adjusts the pitch of Voice 2 up to an octave above or below the original pitch in semi-tones.

08 — PitchShifter Vc 2 Fine Range: -99 to +99

This parameter allows you to fine tune the pitch of Voice 2.

09 — PitchShifter Vc 2 Level Range: 00 to 99

Adjusts the volume of Voice 2. A setting of 00 would eliminate any audible pitch shift.

10 — PitchShifter Vc 2 Pan Range: -99 to +99

Allows you to assign the location of output Voice 2 in the stereo field. A value of -99 would be far left, and +99 would be far right.

11 — Delay vs Quality Range: Long/Smoother or Short/Coarser

Allows you to choose between a long/smoother setting, or a short/coarser setting. A smooth setting would sound best with slower sustaining chords, whereas a coarse setting would enhance a rapidly played musical passage. Depending on your sound source and musical needs, set this parameter accordingly. This parameter actually controls the effect transport delay; smooth yields a long transport delay, coarse yields a short transport delay.

PitchShifter LFO Rate Range: 00 to 99

This parameter controls the rate of pitch modulation which creates a chorusing effect. To achieve chorusing, this rate must be very low.

PitchShifter LFO Width Range: 00 to 99

This parameter controls the excursion of pitch modulation. Since the rate is usually very low, then the width is usually very large.

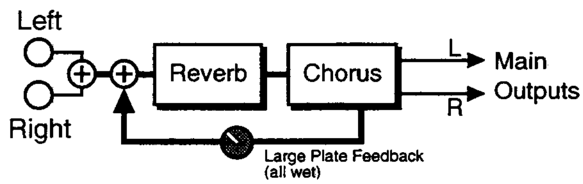
Mod1 Source
Mod1 Destination
Mod1 Param Range Min
Mod1 Param Range Max
Mod2 Source
Mod2 Destination
Mod2 Param Range Min
Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

Plate-Chorus

Plate-Chorus combines a plate reverb with a chorus.

Plate-Chorus Signal Routing



The signal enters a programmable plate reverb, and it is then routed to the chorus which is heard directly at the output. The Large Plate Feedback signal is tapped out of the chorus (all wet) and routed back into the reverb. There is also a dry signal (not shown) that goes directly from the input to the output and is controlled with the Mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section.

03 — Large Plate Decay   Range: 0.40 to 140.0 sec.

Controls the amount of time it takes for the reverberation to decay. High values of decay sound good with this algorithm.

04 — Plate Predelay Time   Range: 0 to 250 ms

Controls the amount of time it takes for the input signal to be presented to the plate reverb. A value of 0 would offer no delay.

05 — Large Plate HF Damping   Range: 00 to 99

Shapes the tone of the reverb decay. Higher settings cause the high frequency components to decay more rapidly. Increasing the value of this parameter will gradually filter out increasing amounts of high frequency energy. This parameter controls the cut off of a low pass filter in series with the decay within the definition.

06 — Large Plate HF Bandwidth   Range: 00 to 99

This parameter acts as a low pass filter on the output of the plate reverb, controlling the amount of high frequencies present. The higher the setting, the more high frequencies are allowed to pass through, offering a brighter ringing sound. Some interesting effects can be created by using a mod controller over a large range.

07 — Plate Diffsn1   Range: 00 to 99

Smears the input signal to create a smoother sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear, making the echoes less apparent.

08 — Diffusion2   Range: 00 to 99

This Diffuser, similar to and in series with the previous one, offers control over lower frequency ranges. Plate reverbs tend to sound metallic, and the diffusers help to smear the signal, eliminating the metallic sound.

09 — Plate Decay Definition Range: 00 to 99

Controls the rate at which echo density increases with time. Higher values can cause the echo density to build at a rate that exceeds the decay rate. For the best results, try to select the highest value that works with your sound source.

10 — Large Plate Feedback Range: -99 to +99

Controls the amount of feedback allowed to pass from the chorus (all wet) to the input of the plate reverb. The sign of the value determines the polarity of the regen.

11 — Chorus LFO Rate Range: 00 to 99

Controls the rate of pitch modulation to the chorus.

12 — Chorus LFO Width Range: 00 to 99

Controls the width of pitch modulation. Keep in mind that the width of pitch modulation is affected by the rate; as the rate is increased, the apparent pitch modulation is also increased.

13 — Chorus Center Range: 00 to 99

Controls the nominal delay time of the chorus about which the delay modulation occurs. Adjusting this parameter will change the tonal character of the effect.

14 — Chorus Feedback Range: -99 to +99

Controls the amount of regeneration applied to the delay time taps. The sign of the value determines the polarity of the regen. The polarity affects the tonal quality of the regeneration.

15 — Chorus Mix Range: 00 to 99

Controls the Dry/Wet mix within the chorus itself. For starters, we recommend settings of 50.

16 — Mod1 Source
17 — Mod1 Destination
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

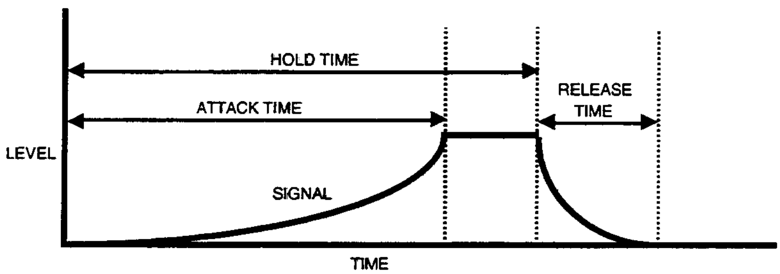
See the descriptions under the Algorithm Modulators, found in the beginning of this section.

ReverseReverb1

ReverseReverb1 produces a reverberation that gradually increases, simulating a backwards sound with a maximum duration of several seconds. When a signal enters this algorithm, the plate reverb (from which this algorithm is derived) is almost instantaneously turned on, and then the output volume is ramped up. This algorithm will only trigger one time. ReverseReverb1 is triggered by an input signal level (threshold) determined by the user. Once triggered, the reverse envelope will proceed to completion, and ignore subsequent trigger levels. If you are looking for a reverse effect that will retrigger, try using Reverse Reverb 2. The topology of the Reverse Reverb is similar to the Plate Reverb.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section.



03 — Envelope Hold Time   Range: 1ms to 10.0s

Controls the amount of time that the reverse will sound after it has been triggered. Rule of thumb: Don't set the hold time much longer than the attack time (see diagram above).

04 — Envelope Attack   Range: 1ms to 10.0s

Sets the duration over which the reverb builds. It is recommended that you set this value less than the hold time (parameter 03).

05 — Envelope Release   Range: 1ms to 10.0s

Determines the release time after the hold time has elapsed. Generally this time is very short. Lower values offer an abrupt cutoff.

06 — Trigger Threshold   Range: -96 to +00 dB

Set this parameter as low as possible to work with your particular sound source. To eliminate false triggering, it should not be set too low. When the input signal rises above this threshold, the reverse envelope will begin.

07 — HF Damping   Range: 00 to 99

This parameter sounds best when it's set to low values. It has the same function as in the Plate Reverb, which is to filter out more and more high frequency energy. For the most natural sounding reverse effect, we recommend a setting of 00.

08 — Diffusion 1   Range: 00 to 99

Diffusion 1 smears the input signal making a smoother sounding reverb. This parameter controls the high frequency ranges. For percussion sounds, high values are recommended.

09 — Diffusion 2   Range: 00 to 99

Similar to and in series with Diffusion 1, this parameter controls lower frequency ranges.

10 — Decay Definition   Range: 00 to 99

Controls the rate at which echo density is increased with time. If set too high, the echo density will build at a rate that exceeds the decay rate. This can be used for a special effect.

11 — Slapback   Range: 0 to 530ms

Controls the delay time of an internal dry signal to create a slapback. This effect helps to simulate a backwards reverb, since now the dry signal appears at the end. In general, we recommend the Mix (parameter 01) be set all wet (99) for this effect. Rule of thumb: Set this parameter at about the same value as the Envelope Hold Time (parameter 03).

12 — Slapback Level   Range: 00 to 99

Adjusts the volume of the slapback (internal dry) signal. A value of 00 would eliminate audible slapback.

13 — Mod1 Source
14 — Mod1 Destination
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, in the beginning of this section.

ReverseReverb2

ReverseReverb2 is identical to Reverse Reverb, except this algorithm will retrigger by an user definable input signal level (threshold). Once triggered, the reverse envelope will proceed to completion, unless retriggered by subsequent input signals. If you are looking for a reverse effect that will not retrigger, try using the previous Reverse Reverb algorithm.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section.

03 — Envelope Hold Time   Range: 1ms to 10.0s

Controls the amount of time that the reverse will sound after it has been triggered. Rule of thumb: Don't set the hold time much longer than the attack time.

04 — Attack   Range: 1ms to 10.0s

This sets the duration over which the reverb builds. It is recommended that you set this value less than the hold time (parameter 03).

05 — Release   Range: 1ms to 10.0s

Determines the release time after the hold time has elapsed. Generally this time is very short. Lower values offer an abrupt cutoff.

06 — Trigger Threshold   Range: -96 to +00 dB

Set this parameter as low as possible to work with your particular sound source. To eliminate false triggering, it should not be set too low. When the input signal rises above this threshold, the reverse envelope will begin.

07 — Pre-Trigger Memory   Range: 0 to 530 ms

Used to capture transients which occur before the trigger. This parameter is critical to the sound quality. You can determine how much pretrigger sound will be injected into the reverse reverb tank.

08 — HF Damping   Range: 00 to 99

This parameter sounds best when set to low values. Its function is to filter out more and more high frequency energy. For the most natural sounding reverse effect, we recommend a setting of 00.

09 — Diffusion 1   Range: 00 to 99

Smears the input signal making a smoother sounding reverb. This parameter controls the high frequency ranges. For percussion sounds, high values are recommended.

10 — Diffusion 2   Range: 00 to 99

Similar to and in series with Diffusion 1, this parameter controls lower frequency ranges.

11 — Decay Definition   Range: 00 to 99

Controls the rate at which echo density is increased with time. If set too high, the echo density will build at a rate that exceeds the decay rate. This can be used for a special effect.

12 — Mod1 Source
13 — Mod1 Destination
14 — Mod1 Param Range Min
15 — Mod1 Param Range Max
16 — Mod2 Source
17 — Mod2 Destination
18 — Mod2 Param Range Min
19 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, in the beginning of this section.

Rotating Spkr

Rotating Spkr adds the famous classic rotating speaker sound to any instrument. A tunable distortion is added to the input signal and is also passed through the rotors.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. For this algorithm we recommend higher Mix settings.

03 — Rotating Speaker Slow Speed   Range: 01 to 55

Determines the rate of the rotary speaker when in the "Slow" setting (parameter 05). This parameter determines the manual level for the rotary speaker rate when Speed=Slow, or when the selected modulator is at zero output level. The higher the value, the faster the rate. By assigning a modulation controller to this parameter, you can change the slow speed in real time.

04 — Rotating Speaker Fast Speed   Range: 01 to 55

Determines the rate of the rotary speaker when in the "Fast" setting (parameter 05). The higher the value, the faster the rate. By assigning a modulation controller to this parameter, you can change the fast speed in real time.

05 — Rotating Speaker Speed   Range: Slow or Fast

Determines how the rotating speaker will switch between slow and fast speeds. The behavior of this switch accurately reflects an actual rotary speaker, taking time to speed up or slow down, based on the value of the Inertia parameter (06). By assigning a modulation controller to this parameter, you can change between the slow and fast speeds in real time.

06 — Rotating Speaker Inertia   Range: 00 to 99

Determines how long it will take for the rotor effect to speed up or slow down after switching from slow to fast or vice versa. Adjust this parameter to simulate the effect of the rotary speaker gradually picking up speed.

07 — Distortion Level In   Range: -48 to +48 dB

Determines the input signal gain into the amplifier simulation, creating a tube-like overdrive. Higher settings yield more distortion.

08 — Distortion Level Out   Range: 00 to 99

Controls the output of the amplifier distortion. There is a (fixed) clean path in parallel with the distortion). Therefore, to eliminate distortion, set this parameter to 00.

09 — Rotating Speaker Distortion Tone   Range: 000 to 127

This parameter is the distortion tone control. High settings will yield a more raspy distortion tone, whereas mid settings will give that "amp growl." When this parameter is set to 000, there is no distortion.

10 — Rotating Speaker Stereo Spread   Range: 00 to 99

Controls the apparent width of the stereo image created by the rotating speaker effect. A setting of 99 yields a right to left synthetic stereo spread, a setting of 00 yields a left to right synthetic stereo spread, and a setting of 50 yields a mono signal.

11 — Mod1 Source
12 — Mod1 Destination
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

Rumble Filter

Rumble Filter is a high pass filter in cascade with a low pass filter, fourth order (24dB per octave). The high pass filter is good for eliminating turntable rumble. The low pass filter is good for eliminating hiss. Alternatively, these filters can be used in a feedback routing with any other effect.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. For this algorithm, we recommend mid values of the Mix.

03 — HighPass Fc Range: 4 to 8000 Hz

Controls the boost or cut of the high pass filter frequency applied to the input signal.

04 — LowPass Fc Range: 100 Hz to 16 KHz

Controls the boost or cut of the low pass filter frequency applied to the input signal.

05 — Filter Gain Range: -48 to +48 dB

Because the cascade of high pass with low pass causes an insertion loss, this parameter allows you to boost the filtered output signal.

06 — Mod1 Source
07 — Mod1 Destination
08 — Mod1 Param Range Min
09 — Mod1 Param Range Max
10 — Mod2 Source
11 — Mod2 Destination
12 — Mod2 Param Range Min
13 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

Sine/Noise Gen

Sine/Noise Gen is a utility algorithm, but when used with a real time modulator/controller, can provide some interesting musical effects. Filters are provided for the noise, but no filters are provided for the sinusoid.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. We recommend lower Mix settings.

03 — Sine Frequency — Range: 0 to 9999 Hz

This parameter controls the sine wave frequency.

04 — Sine/Noise Gen Balance — Range: 00 to 99

Controls the mix between the sine wave and white noise. A setting of 00 would yield all sine wave; a setting of 99 would yield all white noise.

05 — Noise Filter Low Pass Fc — Range: 100 Hz to 16 KHz

Cuts out the high frequencies and can be used to create pink noise.

06 — Bass Fc — Range: 0 to 1000 Hz

Selects the cutoff frequency of the low shelving filter applied to the noise.

07 — Bass EQ Gain — Range: -48 to +48 dB

Sets the amount of boost or cut applied to the low shelving filter applied to the noise.

08 — Treble Fc — Range: 01 KHz to 16 KHz

Selects the cutoff of the upper frequency band high shelving filter applied to the noise.

09 — Treble EQ Gain — Range: -48 to +24 dB

Sets the amount of boost or cut applied to the high shelving filter applied to the noise.

10 — EQ Input Level Trim — Range: -24 to +00 dB

Adjusts the input volume before the EQs to eliminate the possibility of clipping boosted signals.

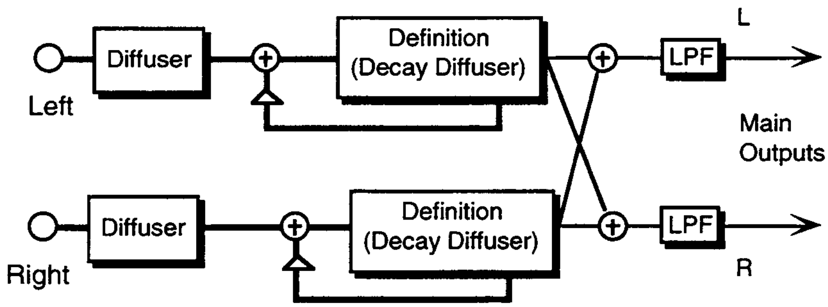
11 — Mod1 Source
12 — Mod1 Destination
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

Small Plate

A plate reverb takes the vibrations from a metal plate and uses them to create a metallic sounding reverb. Small plate reverbs are most often used in the studio for drums and percussion. **Small Plate** is a tight sounding plate reverb

Small Plate Signal Routing



The internal values of the components (not user programmable) differentiate the large and small plate reverbs. The signal goes directly through the diffusers which smear the signal. The signal is then routed to a larger decay diffuser, known as Definition, and is diffused over a period of time (creating a decay). The signal is then routed to the output, and then goes through a low pass filter. There is a parameter that controls the decay time of both the left and right signals (shown as triangles above). This signal is then routed back into the definition. There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section.

03 — Decay

Range: 0.20 to 100.0 sec.

Controls the amount of time it takes for the reverberation to decay. Percussion sounds best using the Small Plate. High values of decay sound good on these algorithms.

04 — Predelay Time

Range: 0 to 500 ms

Controls the amount of time it takes for the input signal to be presented to the plate reverb. A value of 0 would offer no delay.

05 — HF Damping   Range: 00 to 99

Increasing the value of this parameter will gradually filter out increasing amounts of high frequency energy. Higher values yield an abrupt decay. This parameter controls the cut off of a low pass filter in series with the decay within the definition.

06 — HF Bandwidth   Range: 01 to 99

This parameter acts as a low pass filter on the output of the plate reverbs, controlling the amount of high frequencies present. The higher the setting, the more high frequencies are allowed to pass through, offering a brighter ringing sound. Some interesting effects can be created by using a mod controller over a large range.

07 — Diffusion 1   Range: 00 to 99

Smears the input signal to create a smoother sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear, making the echoes less apparent.

08 — Diffusion 2   Range: 00 to 99

This Diffuser, similar to and in series with the previous one, offers control over lower frequency ranges. Plate reverbs tend to sound metallic, and the diffusers help to smear the signal, eliminating the metallic sound.

09 — Decay Definition   Range: 00 to 99

Controls the rate at which echo density increases with time. Higher values can cause the echo density to build at a rate that exceeds the decay rate. For the best results, try to select the highest value that works with your sound source.

10 — Early Ref Level 1
11 — Early Ref Level 2
12 — Early Ref Level 3
13 — Early Ref Level 4   Ranges: -99 to +99

Control four early reflection levels. Setting these levels to lower values will produce a wetter sound. These four reflection levels are close to the input of the Decay Definition.

14 — Left/Right Balance   Range: -99 to +99

Controls the left/right stereo balance of the plate reverb signal. A setting of -99 would offer hard left, whereas a setting of +99 would offer hard right. A setting of +00 would place the reverb in the center of the stereo spectrum.

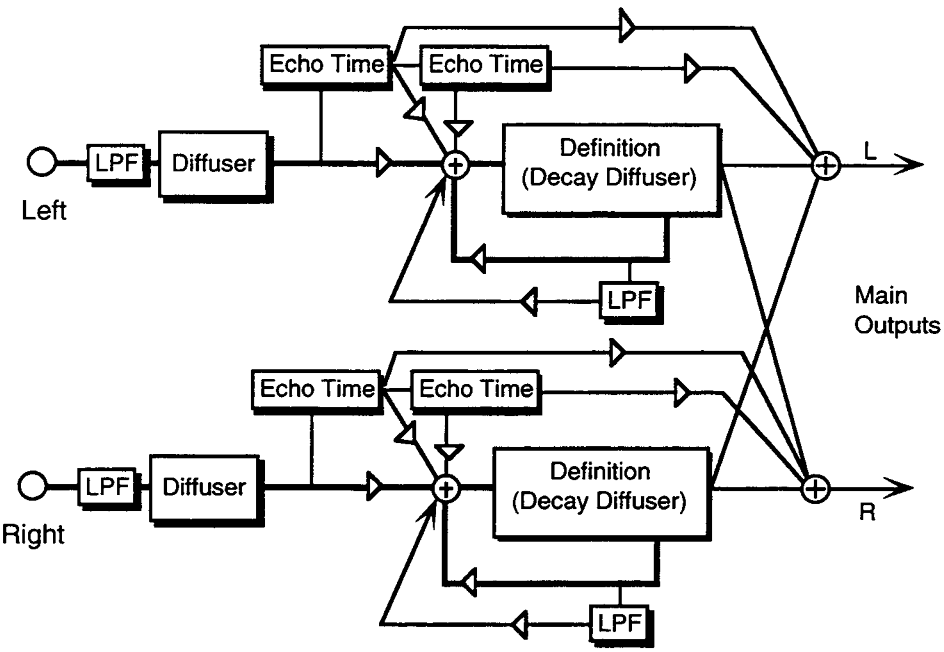
15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Small Room Rev

Small Room Rev provides the ambience of a small room.

Small Room Rev Signal Routing



The signal enters a low pass filter, and goes directly through the diffusers which smear the signal. The signal is then routed to a larger decay diffuser, known as Definition, and is diffused over a period of time (creating a decay). There are taps from both the left and right Definition that are routed to the output to create a synthesized stereo output. A signal from the Definition goes through a low pass filter followed by a low frequency decay parameter, which controls the rate of decay of the low frequencies. There is also a parameter at this stage that controls the decay time of both the left and right signals. The left and right signals are routed back into the Definition. There are two echo times between the diffuser and the definition that can be routed directly to the output, or sent back through the definition. There is also an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

01 — Mix
02 — Volume

These parameters are explained in detail under the Mix and Volume Parameters description, found in the beginning of this section. Reverbs sound best with a Mix of wet and dry.

03 — Decay Range: 0.20 to 100.0 sec.

Controls the amount of time it takes for the reverberation to decay away to a very low level after the input signal stops. In room reverbs we don't recommend higher settings, which tend to create an infinite and unnatural sustain. Since most ambient room reverbs don't naturally have a large decay, set this low for the best sound.

04 — Predelay Time Range: 0 to 450 ms

Controls the amount of time it takes for the original signal to be presented to the reverb. Higher values denote a longer delay.

05 — LF DecayTime Range: -99 to +99

Functions as a tone control and boosts (when set to a positive value) or cuts (when set to a negative value) the rate at which low frequencies will decay.

06 — HF Damping Range: 00 to 99

Controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will gradually filter out (damped) more and more high frequency energy.

07 — HF Bandwidth Range: 01 to 99

The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass. This functions like a tone control on a guitar.

08 — Diffusion1 Range: 00 to 99

This parameter smears the input signal transients, to diffuse and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

09 — Diffusion2 Range: 00 to 99

This parameter, similar to and in series with Diffusion1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffusion parameters to find the settings that are right for your source.

10 — Decay Definition Range: 00 to 99

Controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate. A general rule of thumb is this: Definition should not exceed the LF Decay Time added to the Decay Time.

11 — Detune Rate Range: 00 to 99

Controls the LFO rate of detuning introduced into the reverberation decay. Detuning creates a slight oscillating pitch shift into the decay, giving it a more natural sound by breaking up resonant modes.

12 — Detune Depth   Range: 00 to 99

Controls the depth of the detuning, that is, how much the pitch will change. Low values yield a metallic sound. Some sounds may require very low values, while others sound more natural with higher values.

13 — Primary Send   Range: -99 to +99

Controls the level of the diffused input signal into the reverb definition.

14 — Ref 1 Time   Range: 0 to 120 milliseconds

Controls the delay time for the first pre-echo. Pre-echoes are the first sounds which have been reflected back from the walls or reflective "live" surfaces. Higher values delay the diffused signal more.

15 — Ref 1 Level   Range: 00 to 99

Controls the level of the first pre-echo. This pre-level controls the echo send to the Definition.

16 — Ref 1 Send   Range: 00 to 99

Controls the level of the first pre-echo, with the echo routed directly to the output.

17 — Ref 2 Time   Range: 0 to 120 milliseconds

Controls the delay time for the second pre-echo.

18 — Ref 2 Level   Range: 00 to 99

Controls the level of the second pre-echo. As a signal continues to bounce off the different reflective surfaces (walls), it decreases in volume. Set this parameter to a lower value than Ref 1 Level, in order to create a natural sounding echo.

19 — Ref 2 Send   Range: 00 to 99

Controls the level of the second pre-echo, with the echo routed directly to the output.

20 — Position Balance (1)
21 — Position Balance (2)
22 — Position Balance (3)   Ranges: -99 to +99

These parameters simulate the depth of the room. Think of these parameters as three different microphones placing at various distances within the room (parameter 20 is closest to the front, and parameter 22 is farthest from the front). When the range (volume) is higher for parameter 20, the sound appears closer to the front, whereas a higher setting for parameter 22 appears farther from the front, suggesting a deeper (wetter) room.

23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

These modulation control parameters are identical for all of the algorithms and are explained in detail under the Algorithm Modulators description, found in the beginning of this section.

SpeakerCabinet

SpeakerCabinet simulates the warm sound of an open-back speaker cabinet. Speaker Cabinet is fabulous for a guitar, bass or any other stringed instrument, and will find much use in the studio when recording directly to the console. This algorithm contains the resonances and the nonlinearity of a real musical instrument speaker. Be careful not to overdrive this speaker cabinet by feeding too hot of a signal from the preceding effect; turn down the volume there and make up for it with the output gain here.

For a brighter speaker emulation, try using Tunable Speaker.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Speaker Output Gain Range: -48 to +24 dB

Since speaker cabinets are "lossy," output gain is required to compensate losses in perceived volume. Setting this gain too high will cause clipping of the output signal.

04 — Mod1 Source
05 — Mod1 Destination
06 — Mod1 Param Range Min
07 — Mod1 Param Range Max
08 — Mod2 Source
09 — Mod2 Destination
10 — Mod2 Param Range Min
11 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

Tempo Delay

Tempo Delay features a stereo digital delay (similar to MultiTap) where the tempo is controlled by an assignable modulation source, like a foot switch.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Tempo Delay Time   Range: various

This parameter selects one of twelve different settings to determine the delay rate: 1/32 note, 1/16 triplet, 1/16 note, 1/16 dotted, 1/8 triplet, 1/8 note, 1/8 dotted, 1/4 triplet, 1/4 note, 1/4 dotted, 1/2 triplet and 1/2 note.

04 — Internal Clock Tempo   Range: 050 to 250 bpm

This parameter determines the number of beats per minute (bpm) for the tempo when controlled by the internal clock. If MIDI Clocks or Footswitch1 Tapping is assigned (parameter 06), this parameter does nothing.

05 — TempoDelay Fine Tune   Range: -99 to +99

This parameter allows you to fine tune the delay time. Lower values have a faster speed.

06 — Tempo Control   Range: Internal Clock, MIDI clocks, FtSw1L Tapping

Determines how the tempo will be controlled. In order for Foot Switch 1-L to work as a controller, it must be assigned as a DP/2 Controller in System/MIDI mode (parameter 45). The foot switch is then pressed twice (tapping quarter notes) to set the tempo. Continued tapping on the foot switch will cause the tempo to change, because the DP/2 always reads the sum of the last two presses. This could be a useful technique for songs or arrangements where the tempo is constantly changing.

07 — Tempo Delay Regen   Range: 00 to 99

Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delay.

08 — Tempo Delay Pan   Range: -99 to +99

Sets the location within the stereo spectrum for the delayed signal.

09 — Tempo Delay Smoothing   Range: 50µs to 10.0 s

Controls the average period of incoming MIDI clocks. Longer smoothing times provide more stable results and less clicking; shorter smooth times have less doppler, and track tempo changes faster. We recommend a setting of 200 ms for starters.

10 — Tempo Delay Pan   Range: -99 to +99

Sets the location within the stereo spectrum for the delayed signal.

11 — Mod1 Source
12 — Mod1 Destination
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

Tunable Spkr 1

Tunable Spkr 1 offers an EQ controllable speaker sound which is brighter than Speaker Cabinet. By tuning three parametric filters, you can simulate many different speaker cabinet sounds that are used in all styles of music.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Mid1 Fc
Range: 50 to 9999 Hz

Sets the center of the mid-frequency parametric. Higher values have a brighter sound.

04 — Mid1 Gain
Range: -48 to +24 dB

Sets the amount of cut (negative values) or boost (positive values) applied to this mid-frequency parametric.

05 — Mid1 Q
Range: 01 to 18

This parameter is a bandwidth control that determines the width of the resonant peak at the center of the frequency band. By raising the value, you can produce a narrower bandwidth.

06 — Mid2 Fc
07 — Mid2 Gain
08 — Mid2 Q
09 — Mid3 Fc
10 — Mid3 Gain
11 — Mid3 Q

These parameters are identical to the previous ones, but can be assigned to control different bandwidths within the mid-range.

12 — Speaker Input Attenuation
Range: -24 to +00 dB

This parameter allows you to adjust the input level before the EQs to eliminate the possibility of clipping boosted signals.

13 — Speaker Output Gain
Range: -48 to +24 dB

Since speaker cabinets are "lossy," output gain is required to compensate losses in perceived volume. Setting this gain too high will cause clipping of the output signal.

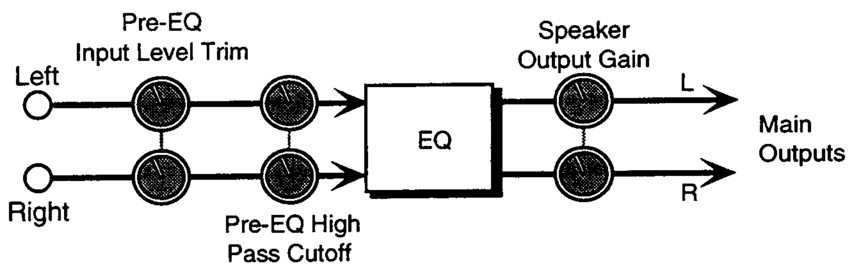
14 — Mod1 Source
15 — Mod1 Destination
16 — Mod1 Param Range Min
17 — Mod1 Param Range Max
18 — Mod2 Source
19 — Mod2 Destination
20 — Mod2 Param Range Min
21 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

Tunable Spkr 2

Tunable Spkr 2 is similar to Tunable speaker 1, offering an EQ controllable speaker sound, but with a warmer, "analog" sound. By tuning three parametric filters, you can simulate many different speaker cabinet sounds that are used in all styles of music.

Tunable Spkr 2 Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section.

03 — Mid1 Fc   Range: 50 to 9999 Hz

Sets the center of the mid-frequency parametric. Higher values have a brighter sound.

04 — Mid1 Gain   Range: -48 to +24 dB

Sets the amount of cut (negative values) or boost (positive values) applied to this mid-frequency parametric.

05 — Mid1 Q   Range: 01 to 18

This parameter is a bandwidth control that determines the width of the resonant peak at the center of the frequency band. By raising the value, you can produce a narrower bandwidth.

06 — Mid2 Fc
07 — Mid2 Gain
08 — Mid2 Q
09 — Mid3 Fc
10 — Mid3 Gain
11 — Mid3 Q

These parameters are identical to the previous ones, but can be assigned to control different bandwidths within the mid-range.

12 — PreEQ InputLevel Trim   Range: -18 to +06 dB

This parameter allows you to adjust the input level before the EQs to eliminate the possibility of clipping boosted signals.

13 — Speaker Output Gain   Range: -48 to +24 dB

Since speaker cabinets are "lossy," output gain is required to compensate losses in perceived volume. Setting this gain too high will cause clipping of the output signal.

14 — Noise Gate Off Below   Range: -96 to +00 dB

This parameter sets the threshold level at which the noise gate shuts off the audio.

15 — Gate Release Time   Range: 1ms to 10.0s

Determines how long it takes for the gate to be fully released after the input signal drops below the threshold level. Lower settings yield a quick gate.

16 — Pre-EQHighPass Cutoff   Range: 4 to 1000 Hz

Filters out the low frequencies. The higher the value, the less low frequencies pass through. This parameter is used to increase brightness.

17 — Mod1 Source
18 — Mod1 Destination
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

VandrPolFilter

VandrPol Filter adds synthetic high harmonics to the input signal, brightening the overall sound. This algorithm is most often used in the studio for vocalists, but feel free to experiment with this algorithm using your favorite instrument as well. This algorithm features prominent transient enhancement which makes it ideal for "plucked" sounds. The filter in this algorithm operates on the signal prior to enhancement. Set the filter to enhance the frequency region that you desire. Then mix the enhanced signal with the dry signal.

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. For this algorithm, we recommend mid values of the Mix.

03 — HighPass Fc

Range: 4 to 8000 Hz

Controls the boost or cut of the high pass filter frequency applied to the input signal.

04 — LowPass Fc

Range: 100 Hz to 16 KHz

Controls the boost or cut of the low pass filter frequency applied to the input signal.

05 — Filter Gain

Range: -48 to +48 dB

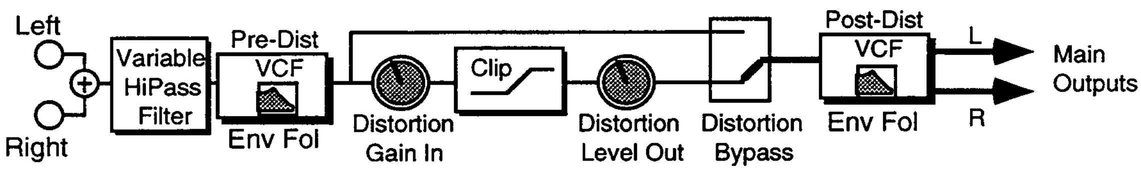
Because the cascade of high pass with low pass causes an insertion loss, this parameter allows you to boost the filtered output signal.

06 — Mod1 Source
07 — Mod1 Destination
08 — Mod1 Param Range Min
09 — Mod1 Param Range Max
10 — Mod2 Source
11 — Mod2 Destination
12 — Mod2 Param Range Min
13 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators in the beginning of this section.

VCF-Distort 1

VCF-Distort 1 combines a voltage control filter and a raspy distortion, and a second voltage control filter. Three effects can be obtained: Distortion, Wah-wah, and Auto-wah. The last two functions use the same VCF. These filters can be disabled or used as EQ if desired. When used for distortion, any speaker cabinet emulation (such as Tunable Speaker) in cascade with this effect is recommended. There is a second VCF that exists after the distortion that can be set to act like a simple speaker simulator, or it can be modulated in parallel with the pre-distortion VCF.



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters earlier in this section. For this algorithm, Volume controls the distortion output level. For high distortion input gains, use lower volumes.

03 — Distortion Level In Range: 00 to 99

Controls the gain going into the distortion effect. Distortion Level In will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the Distortion Level Out (04) down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.

04 — Distortion Level Out Range: 00 to 99

Controls the gain coming out of the distortion effect. Generally, if the Distortion Level In (03) is set high, set this parameter lower to control the volume.

05 — Pre-Distortion VCF Fc Range: 01 to 99

Determines the filter cut off frequency before the distortion. Higher values have a brighter sound. This parameter can be modulated, using a CV Pedal for a wah wah pedal effect. To disable the distortion filter, set this parameter to 99. To use as an EQ, set the desired value and make sure envelope follower (parameter 07) is 00. To use as the auto-wah, set this parameter close to 01 (lower values) and turn on parameter 07.

06 — Pre-Distortion VCF Q Range: 01 to 25

Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the *presence* of the peak. This setting is important for the auto-wah effect.

07 — Envelope Follower to Pre VCF Range: -99 to +99

Determines how much the amplitude of the incoming signal will modify the distortion filter cutoff frequency. When set to +00, no modification will occur. When set to mid positive values, the Pre-Distortion VCF Fc will go high, but then come down to its nominal setting. When set to negative mid values, the Pre-Distortion VCF Fc will go low, and then go back up to its nominal setting. How quickly it does so is determined by parameters 11 and 12. This sound is the auto-wah; positive values will boost the high frequencies, offering an "oww-oww" sound, and negative values will cut the high frequencies, producing a "dweep-dweep" sound.

08 — Post-Distortion VCF Fc

09 — Post-Distortion VCF Q

10 — Envelope Follower to Post VCF

These three parameters are identical to the previous parameters, and are used to control the second VCF that exists after the distortion.

11 — Envelope Follower Attack Range: 50µs to 10.0s

Sets the attack of the envelope follower (i.e. determines how closely the attack is followed) once the incoming signal has been detected. Generally the attack should be short.

12 — Envelope Follower Release Range: 1ms to 10.0s

Sets the amount of time after the incoming signal has ceased for the envelope follower to shut down. Generally these times are longer than the attack times.

13 — Distortion Bypass Range: Off or On

This parameter allows you to bypass the distortion (as shown on the signal routing diagram).

14 — Pre-EQ High Pass Cutoff Range: 0 to 1000 Hz

Filters out the low frequencies before the EQ. The higher the value, the less low frequencies will pass through.

15 — Mod1 Source

16 — Mod1 Destination

17 — Mod1 Param Range Min

18 — Mod1 Param Range Max

19 — Mod2 Source

20 — Mod2 Destination

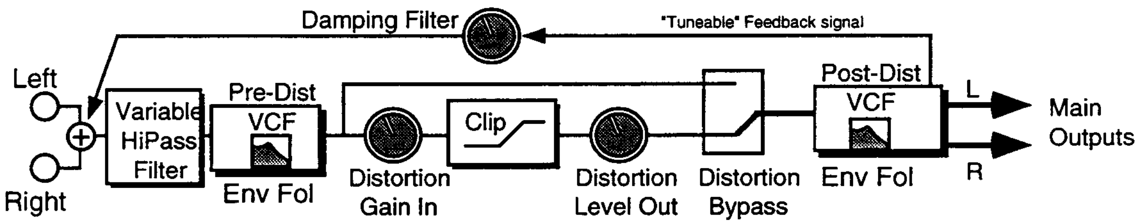
21 — Mod2 Param Range Min

22 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

VCF-Distort 2

VCF-Distort 2 combines a voltage control filter and a raspy distortion, and a second voltage control filter. This algorithm is identical to the VCF - Distort 1 algorithm, with the addition of a "tunable" feedback signal.



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters earlier in this section. For this algorithm, Volume controls the distortion output level. For high distortion input gains, use lower volumes.

03 — Distortion Level In Range: 00 to 99

Controls the gain going into the distortion effect. Distortion Level In will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the Distortion Level Out (04) down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.

04 — Distortion Level Out Range: 00 to 99

Controls the gain coming out of the distortion effect. Generally, if the Distortion Level In (03) is set high, set this parameter lower to control the volume.

05 — Pre-Distortion VCF Fc Range: 01 to 99

Determines the filter cut off frequency before the distortion. Higher values have a brighter sound. This parameter can be modulated, using a CV Pedal for a wah wah pedal effect. To disable the distortion filter, set this parameter to 99. To use as an EQ, set the desired value and make sure envelope follower (parameter 07) is 00. To use as the auto-wah, set this parameter close to 01 (lower values) and turn on parameter 07.

06 — Pre-Distortion VCF Q Range: 01 to 25

Determines the level and width of the resonant peak at the filter cutoff point. While the Fc (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the *presence* of the peak. This setting is important for the auto-wah effect.

07 — Envelope Follower to Pre VCF Range: -99 to +99

Determines how much the amplitude of the incoming signal will modify the distortion filter cutoff frequency. When set to +00, no modification will occur. When set to mid positive values, the Pre-Distortion VCF Fc will go high, but then come down to its nominal setting. When set to negative mid values, the Pre-Distortion VCF Fc will go low, and then go back up to its nominal setting. How quickly it does so is determined by parameters 11 and 12. This sound is the auto-wah; positive values will boost the high frequencies, offering an "oww-oww" sound, and negative values will cut the high frequencies, producing a "dweep-dweep" sound.

08 — Post-Distortion VCF Fc
09 — Post-Distortion VCF Q
10 — Envelope Follower to Post VCF

These three parameters are identical to the previous parameters, and are used to control the second VCF that exists after the distortion.

11 — Envelope Follower Attack   Range: 50µs to 10.0s

Sets the attack of the envelope follower (i.e. determines how closely the attack is followed) once the incoming signal has been detected. Generally the attack should be short.

12 — Envelope Follower Release   Range: 1ms to 10.0s

Sets the amount of time after the incoming signal has ceased for the envelope follower to shut down. Generally these times are longer than the attack times.

13 — Distortion Bypass   Range: Off or On

This parameter allows you to bypass the distortion (as shown on the signal routing diagram).

14 — Pre-EQ High Pass Cutoff   Range: 0 to 1000 Hz

Filters out the low frequencies before the EQ. The higher the value, the less low frequencies will pass through.

15 — Speaker HighPass Cutoff   Range: 4 to 1000 Hz

This parameter filters out the low frequencies of the main amp prior to the speaker. The higher the value, the less low frequencies pass through.

16 — Amp Feedback Amount Range: -99 to +99

Controls the amount of feedback allowed to pass from the post-distortion VCF Envelope follower to in front of the pre-EQ high-pass cutoff filter. The sign of the value determines the polarity of the regen.

17 — Amp Feedback HF Damping   Range: 00 to 99

This filter controls the rate of attenuation of high frequencies in the feedback signal. Increasing the value of this parameter will gradually filter out increasing amounts of high frequency energy.

18 — Amp Feedback Delay   Range: 000 to 127

This is a very fast delay and is used to "tune" the feedback signal.

Mod1 Source
Mod1 Destination
Mod1 Param Range Min
Mod1 Param Range Max
Mod2 Source
Mod2 Destination
Mod2 Param Range Min
Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Vocal Remover

Vocal Remover is the "instant karaoke" algorithm; it removes from a stereo source any signal that is present in both the left and the right channels. The vocal track is usually common to both channels of a stereo recording, and thus can be removed by subtracting one channel from the other.

This algorithm uses a Vocal Position control to compensate for panning. There is also an L/R Delay control to make up for any slight delay that may exist between the left and right channels due to idiosyncrasies of the recording or playback process.

Bandpass (Mid) filters extract the vocal range from the stereo source for further processing by the cancellation circuit. Lowpass and highpass (Bass and Treble) filters restore the high and low end after processing.

☞ **Important:** This special algorithm is only made available in the DP/2 as a ROM Config Preset (location #54 in Bank 1), because it requires special input signal routing.

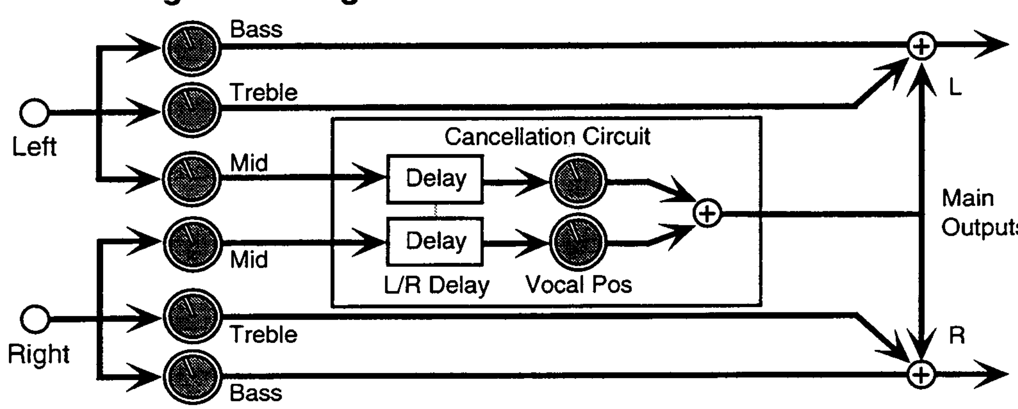
How to use the Vocal Remover:

1. Feed a stereo program source, and set the Bass Level and the Treble Level to 00.
2. Set the Mid Level to 99.
3. Set the Mid Fc to about 2000 Hz.
4. Set the Mid BW to about 4000 Hz or higher.
5. Start with a Vocal Position setting of +000, and an L/R Delay setting of +000.
6. Adjust these two parameters until the vocal level has been reduced satisfactorily (note that vocals that have been treated with reverb or other effects usually cannot be removed completely).
7. Set the Bass Fc to about 100 Hz.
8. Set the Treble Fc to about 10000 Hz.
9. Gradually adjust the Bass level, Bass Fc, Treble level, and Treble Fc to achieve a pleasing sound.

   Note: If the Bass Fc is set too high, or if the Treble Fc is set too low, some vocal components may begin to leak into the output of the effect.
10. The Mid Level, Mid Fc, and Mid BW may be adjusted.

Experiment with steps 9 and 10 for the best result.

Vocal Remover Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, in the beginning of this section. For this algorithm, we recommend a mix setting of 99.

03 — Vocal Pos
Range: -127 to +127

This is used to compensate for panning in the original recorded vocal.

04 — L/R Delay
Range: -127 to +127

This makes up for any slight delay that may exist between the left and right channels due to idiosyncrasies of the recording or playback process.

05 — Bass Level
Range: 00 to 99

Controls the output level of the lowpass filters. This signal goes directly to the main outputs.

06 — Treble Level
Range: 00 to 99

Controls the output level of the highpass filters. This signal goes directly to the main outputs.

07 — Mid Level
Range: 00 to 99

Controls the output level of the bandpass filters. As shown in the diagram, the output of the bandpass filters goes to the cancellation circuit.

08 — Bass Fc
Range: 80 to 1000 Hz

Determines the cutoff frequency of the lowpass filters. If this parameter is set too high, some vocal components may begin to leak into the output of the effect.

09 — Trebl Fc
Range: 1000 to 16000 Hz

Determines the cutoff frequency of the highpass filters. If this parameter is set too low, some vocal components may begin to leak into the output of the effect.

10 — Mid Fc
Range: 80 to 16000 Hz

Determines the cutoff frequency of the bandpass filters.

11 — BW
Range: 80 to 16000 Hz

This determines the bandwidth of the bandpass filters.

12 — Mod1 Source
13 — Mod1 Destination
14 — Mod1 Param Range Min
15 — Mod1 Param Range Max
16 — Mod2 Source
17 — Mod2 Destination
18 — Mod2 Param Range Min
19 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators earlier in this section.

Vocoder (2 Unit)

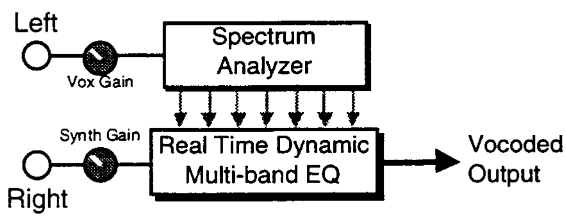
The DP/2 features a vocoder. A vocoder analyzes the frequency spectrum from an incoming source (most commonly speech from a microphone) and applies that analysis to the pitched sounds from the output of another source (like a synthesizer).

☞ **Important:** The vocoder, though made up of two 1-Unit algorithms, is only made available in the DP/2 as a Config Preset. Don't go looking for the two 1-Unit algorithms that make up the vocoder in the list of 1-Unit Presets -- they are not there. Each Vocoder (Part 1 and Part 2) can be used seperately. They both cover the same frequency range, but the frequency bands of the two vocoders are interleaved.

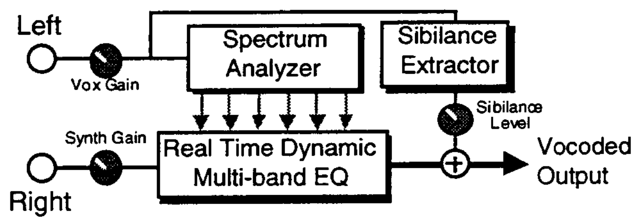
How the Vocoder Works

In the DP/2, the vocoder uses both units to perform one function. They are connected in parallel so that they receive the same two inputs. The vocoder algorithm analyzes the incoming signal (Input 1) and applies it to another source (Input 2). The vocoder config preset joins the two different algorithms (Vocoder Part 1 and Vocoder Part 2) which work together to create the vocoder effect.

Vocoder (Part 1) Routing

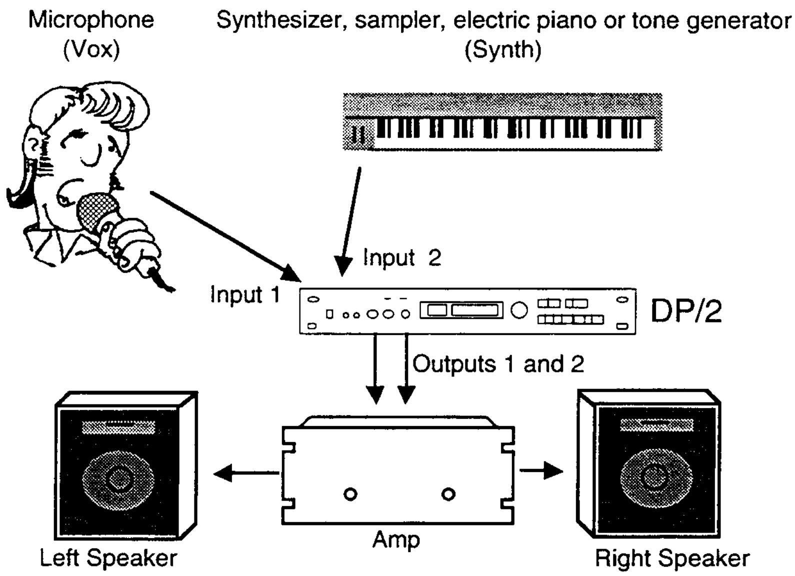


Vocoder (Part 2) Routing



The incoming voice signal (vox), connected to Input 1, is sent to the Spectrum Analyzer. The bandpass filters within the Analyzer divide the voice signal into separate frequency bands. The Analyzer then measures the signal level in each of these bands and supplies this information to the Real Time Dynamic Multi-band EQ. This EQ section divides the carrier signal (Input 2) into separate frequency bands. The output level of each of these bands is controlled by the signal level measured in the corresponding band of the analyzer. The result is that the frequency spectrum of the Synth signal is forced to match the spectrum of the Vox signal. In Vocoder Part 2, there is also an internal signal from the Vox input that bypasses the spectrum analyzer and sends the high-frequency sibilance sounds (t's, p's, clicks, pops, etc.) directly to the output for improved articulation.

Setting Up the Vocoder



Making the Right Connections
The vocoder will not function if the connections are not right. Plug the incoming signal (vox) into Input 1 (front or back of the DP/2). Connect the synthesis signal (Synth) to Input 2 on the back panel. This signal should be harmonically rich and have a wide bandwidth for optimal performance. Connect Outputs 1 and 2 to your audio system as shown in the diagram.

Selecting the Vocoder Preset

1. Press (SELECT), then the (CONFIG) button.
2. Turn the **Data Entry Knob**, or use the ◁ and ▷ buttons to select preset #53 Vocoder 2U (Bank 1). The display looks like this:

3. Press (SELECT) again to confirm the selection.

Using the Vocoder

As you speak into the mic, play appropriate notes on the keyboard (or other controller source) at the same time and listen to the results. Using a vocoder may require a little practice, but can provide some rewarding musical effects. Some common effects are to produce "robot-speech" by talking into the mic while playing a single note, or to create choir sounds by singing "aah" or "ooh" into the mic while playing chords on the keyboard.

Note that the pitch of the output signal is entirely determined by the pitch of the synth input, and is not affected by the pitch you sing into the microphone. The characteristics of the synth input signal also affect the vocoder quality. The synth signal must not only contain sufficient harmonics to cover the frequency range of the vocoder, it must be played in a pitch range that roughly corresponds to the pitch of the microphone (vox) input. For example, it would be hard to get good results if you are talking in a low pitched voice, but are playing high notes on the keyboard.

Although the Input 1 (vox) is optimized for speech, any signal source can be used. The vocoder will apply the spectrum of any Input 1 signal to the synth signal, which can produce some interesting timbres.

Vocoder Part 1
Vocoder Part 2

01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section. For this algorithm, we recommend a Mix setting of 99.

Vocoder Gain Vox — Range: 00 to 99

Adjusts the boost or cut applied to the Input 1 (vox) source. Experiment with this level until it sounds right.

Vocoder Gain Synth — Range: 00 to 99

Adjusts the boost or cut applied to the Input 2 (synth) source. Experiment with this level until it sounds right.

Vocoder Response Time — Range: 1 ms to 10.0 sec

Selects the rate at which the synth will track the vox signal. A faster response time will analyze and synthesize the signal quickly. A slower response time will analyze and synthesize the signal more accurately.

Vocoder Pan — Range: Centr, Left, or Right

Allows you to choose the location of the vocoded signal in the output. For a pseudo-stereo effect, pan the first vocoder left and the second vocoder right (in a 2-Unit vocoder preset).

Vocoder Sibilance Level — Range: 00 to 99

(Part 2 only) Controls the level of high frequency sibilance sounds passed to the output. This filter will add all vox frequencies above approximately 3500 Hz (see earlier diagram) directly to the synthesized output. In general, higher values offer improved articulation. We recommend a setting of approximately 20.

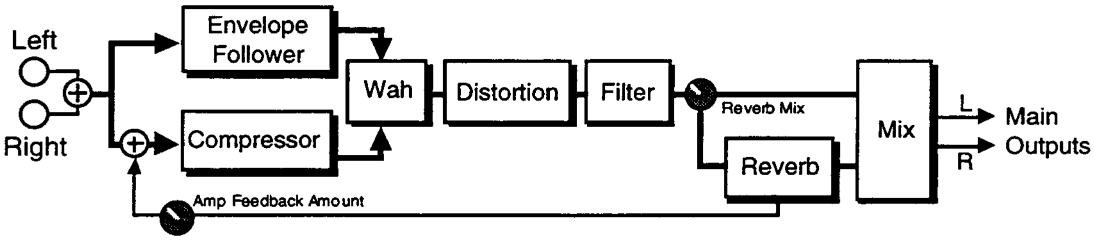
Mod1 Source / Mod2 Source
Mod1 Destination / Mod2 Destination
Mod1 Param Range Min / Mod2 Param Range Min
Mod1 Param Range Max / Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Wah-Dist-Revrb

A guitar-effects chain-amp simulator, featuring distortion, compression, reverb, and a resonant filter following the amplitude of the signal.

Wah-Dist-Revrb Signal Routing



01 — Mix
02 — Volume

See the descriptions under the Mix and Volume Parameters, found earlier in this section.

03 — Reverb Mix — Range: 00 to 99

Controls the mix between the processed signal and the reverb. Setting this parameter to 00 will allow only the processed dry signal to be heard (no reverb), while a setting of 99 will send all of the processed signal to the reverb.

04 — Compressor Threshold — Range: -96 to +00 dB

Controls the threshold level for the compressor. As the input signal dies away below the threshold, the compressor will increase the gain of the signal, causing feedback to increase as well. To turn off, set the threshold to -96dB.

05 — Comp Attack — Range: 50µs to 100ms

Determines the attack rate after the initial signal has been detected and before the compression takes affect.

06 — Comp Release — Range: 1ms to 10.0s

Determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally set longer than the attack time (parameter 05).

07 — Compressor Gain — Range: -48 to +48 dB

This parameter boosts the compressed signal level.

08 — Wah Center — Range: 00 to 99

Determines where the resonant peak of the wah will occur. Higher values create a higher pitched wah.

09 — Wah Range — Range: -99 to +99

Controls how much depth the wah will have. This is analogous to how far you move a wah-wah pedal back and forth.

10 — Wah Attack — Range: 50µs to 10.0s

Sets the attack of the envelope follower (i.e. determines how closely the attack is followed) once the incoming signal has been detected. Generally the attack should be short.

11 — Wah Release   Range: 50µs to 10.0s

Sets the amount of time after the incoming signal has ceased for the envelope follower to shut down. The Wah Attack and Wah Release parameters are used to get different "wah" effects.

12 — Distortion Level In
13 — Distortion Level Out   Ranges: 00 to 99

These two parameters control the levels going into and coming out of the distortion effect.

14 — Post-Distortion VCF Fc   Range: 01 to 99

Determines the filter cut off frequency after the distortion. Higher values have a brighter sound. This parameter can be used to emulate a speaker cabinet. To disable the distortion filter, set this parameter to 99, and set the Q parameter to 01.

15 — Post-Distortion VCF Q   Range: 01 to 25

Determines the level and width of the resonant peak at the filter cutoff point. While the Post-Dist VCF Fc parameter determines where (at what-frequency) this peak will occur, this parameter controls the *sharpness* of the peak.

16 — Distortion Mix   Range: 00 to 99

Determines the Dry/Wet, or in this case, dirty/clean mix of the signal. A value of 00 yields a clean signal; 99 yields an all distortion signal.

17 — Amp Feedback Amount   Range: -99 to +99

Controls the amount of signal applied from the output of the reverb back into the input of the compressor. The sign of the value determines the polarity of the feedback.

18 — Reverb Decay   Range: 0.20 to 100.0 sec

Controls the amount of time it takes for the reverberation to decay to a very low level (-60dB) after the input signal stops.

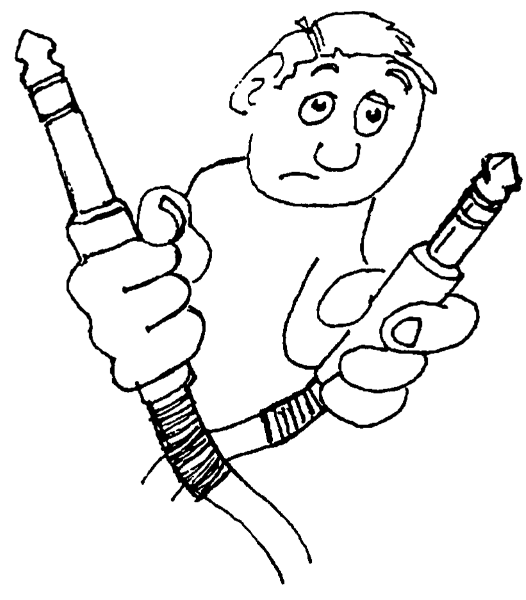
19 — Reverb HF Damping   Range: 00 to 99

Shapes the tone of the reverb decay. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Higher settings cause the high frequency components to decay more rapidly.

20 — Mod1 Source
21 — Mod1 Destination
22 — Mod1 Param Range Min
23 — Mod1 Param Range Max
25 — Mod2 Source
26 — Mod2 Destination
27 — Mod2 Param Range Min
28 — Mod2 Param Range Max

See the descriptions under the Algorithm Modulators, found in the beginning of this section.

Section 3 — Config Parameters



This section will teach you about configs, how the effects (called algorithms) are routed in the DP/2, and define all of the parameters relating to the possible DP/2 configurations.

Section 3 — Config Parameters

What is a Config?

The Config (short for CONFIGuration) determines the number of input sources that are to be processed by the DP/2, and how the units and their inputs and outputs are connected. A "1 source config" means that one signal *source* (stereo or mono) is going into the DP/2. A "2 source config" means that two mono sources are going into the DP/2.

Config Presets

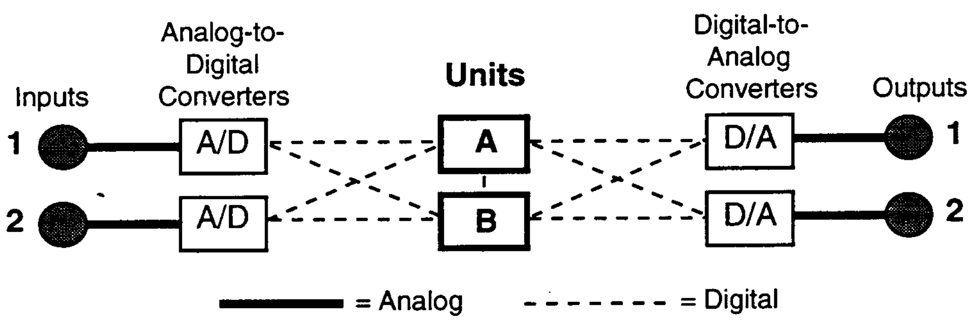
Of the three DP/2 preset types (1 Unit, 2 Unit, and Config), the most powerful is the *Config Preset.* The Config preset lets you save, and later recall the current state of the DP/2, including all algorithms, signal routing, and mixing information. There are 200 Config Presets within the DP/2 and are divided into the following four locations:

Memory Location	Bank Location	Preset Location
RAM	Bank 1	00 to 49
ROM	Bank 1	50 to 99
RAM	Bank 2	00. to 49.
ROM	Bank 2	50. to 99.

You can create/write your own presets in RAM; the ROM presets cannot be changed.

About Signal Routing

The two audio inputs are analog signals which are fed to two analog-to-digital converters. The two units are digital audio signal processors which have digital inputs and outputs. Routing between the units is digital. The output of a unit is converted back to analog audio for the output jack.

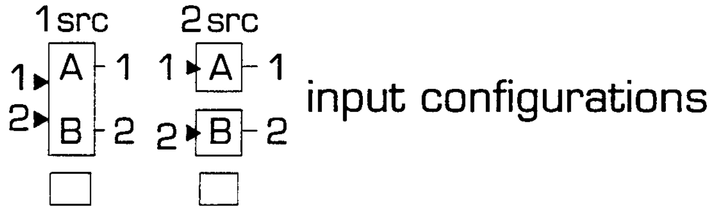


(Not all of the possible signal routings are shown.)

All of the above elements are under complete software control.

Input Configurations

The diagram in the upper right corner of the DP/2 shows the input configurations:



1 Source Input Configuration

In a 1 Source Config, the LED beneath the 1 source input configuration diagram will light. Use Input 1 (front or back panel) for a mono signal (such as a guitar), or Inputs 1 and 2 if your source is a stereo signal (such as a keyboard). The choice of stereo or mono for an input is a 1 Source Config parameter, and will be covered later in this section. Remember, any mono signal (high or low impedance) can be plugged into the jack on the front panel. The Input 1 jack (front panel) will always override the Input 1 jack on the rear panel.

2 Source Input Configuration

In a 2 Source Config, the LED beneath the 2 source input configuration diagram will light. For your first source, use Input 1 (front or back panel) for a mono signal. For your second source, use Input 2 for a mono signal. Even though the input signals to the units must be mono, the effect processing can generate two *stereo* output signals.

Selecting Config Presets

Of the three Preset types (1 Unit, 2 Unit, and Config), the most powerful is the *Config Preset*. The Config preset lets you save, and later recall, the current state of the DP/2, including all algorithm, signal routing and mixing information.

Selecting a Config preset will

- Reconfigure the DP/2 inputs and outputs;
- Change the signal routing between units; and
- Load a new algorithm into each of the Units.

To select a Config preset

1. Press (SELECT).
2. If the Config LED is not already on, press (CONFIG).
3. Move the **Data Entry Knob**, or press the (◁) and (▷) buttons. The Select LED flashes, indicating that you are previewing presets. The display shows the available Config presets.
4. When the display is showing the preset you want to load, press (SELECT) again. This selects the preset, and the Select LED stops flashing.

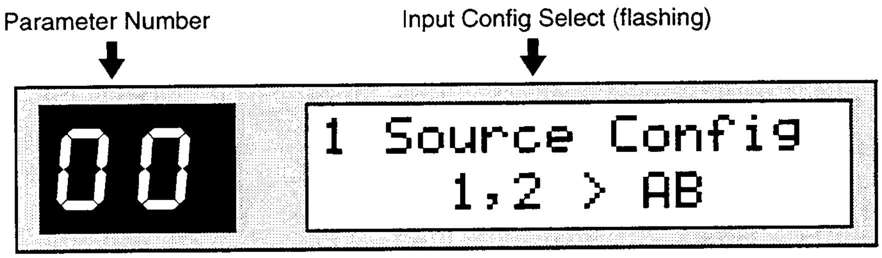
Note: The first three ROM Config locations in Bank 1 (presets #50 to 52) and the first two ROM Config locations in Bank 2 (presets #50. and 51.) can be used as "starting places" for creating your own configurations, and cover common signal routing set-ups. Note that presets numbers in Bank 2 will appear with a dot in the display.

Editing a Config Preset

In Edit mode, you can select between the input configurations and edit their related parameters (which contain other signal routing information) using the ◁ and ▷ buttons to select parameters, and the **Data Entry Knob** to change the value of the active (flashing) parameter.

To edit a Config Preset:

1. Press the EDIT button.
2. If the yellow Config LED is not already on, press CONFIG. The display shows:

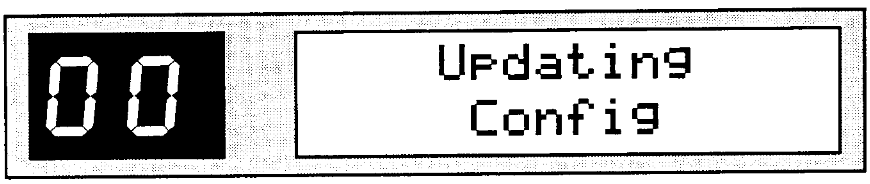


The red LED display should indicate parameter 00, which is the *Input Config Select* parameter, and the current config type should be flashing in the upper line of the LCD display (if not, press ◁ until this is the case).

3. If you move the **Data Entry Knob** now, you will change the config type. There are two different types of configs that can be edited:

1 Source Config	2 Source Config

When you stop moving the knob, the display momentarily shows:

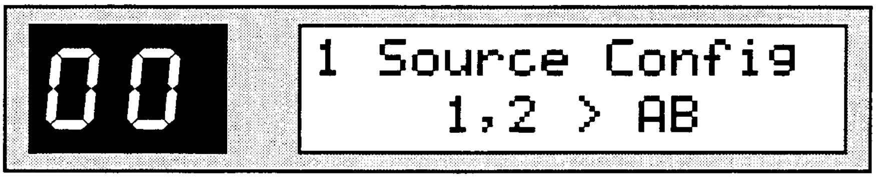


The DP/2 is now updated into the config showing on the display.

4. To edit the remaining config parameters, press the ▷ button to scroll to other parameters, and move the **Data Entry Knob** to change their values.

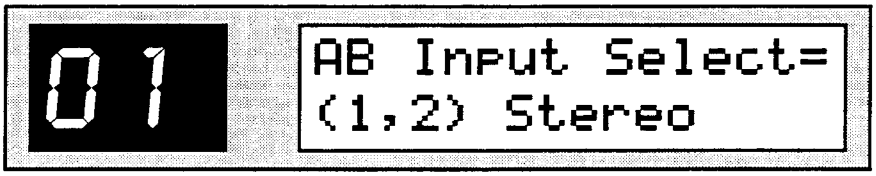
Note: Changing the config type, or editing some config parameters may cause a brief interruption in the audio output. This will happen if you change the number of sources, or if the mono/stereo output routing is different. This interruption is normal and is required for the system to reconfigure its signal routing.

1 Source Config



00 — 1 Source Config

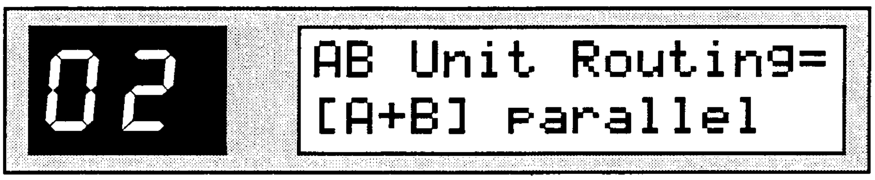
The 1 Source Config arranges the DP/2 as one big multi-effects processor, using both units to process the same input signal. 1 Source Configs have two input select options, mono or stereo (refer to parameter 01).



01 — AB Input Select

Range: (1,2) Stereo or (1) Mono

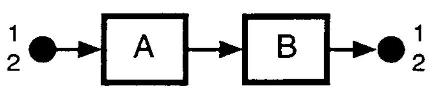
Selects either a mono (Input 1) or stereo (Inputs 1 and 2) signal.



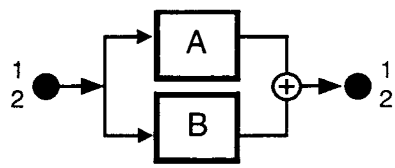
02 — AB Unit Routing

Range: serial, parallel, feedback1 or feedback2

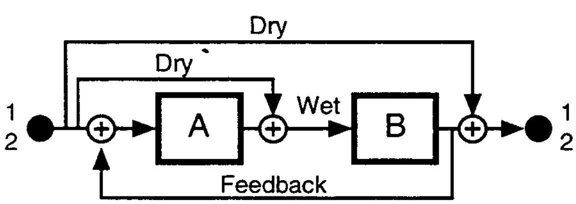
Units A and B can be routed together in one of four different ways:



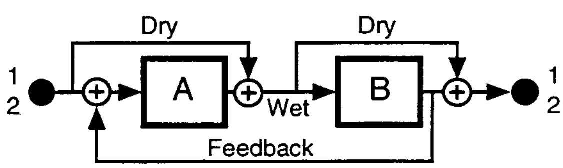
Serial



Parallel



Feedback 1

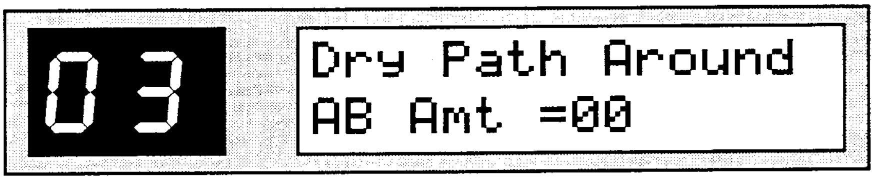


Feedback 2

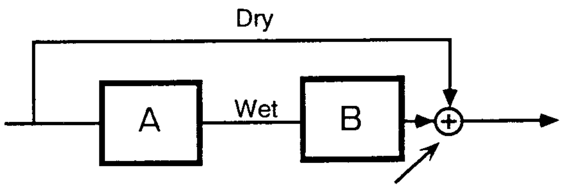
The feedback routings are similar to the serial routing, with the addition of a feedback signal. The difference between Feedback 1 and Feedback 2 is how the dry signal is mixed into the wet signal (as shown above). Note that the feedback signal is all wet, and that it is tapped before the dry signal.

03 — (Config Dependent)

This parameter is dependent on how Units A and B are routed (determined by parameter 02). If parameter 02 is serial, this screen shows:

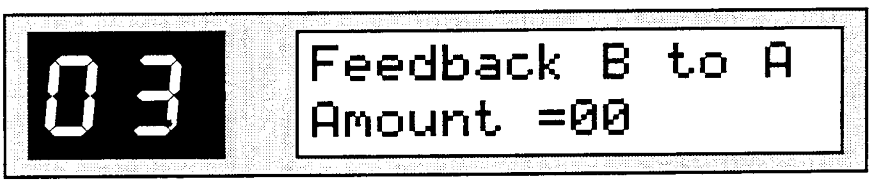


This screen allows you to control an external dry signal around Units A and B. A setting of 00 would not allow a dry signal around the units, whereas a setting of 99 would permit a full signal around the units.

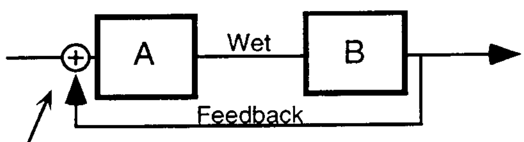


This parameter controls this amount.

If parameter 02 is feedback 1 or 2, this screen shows:

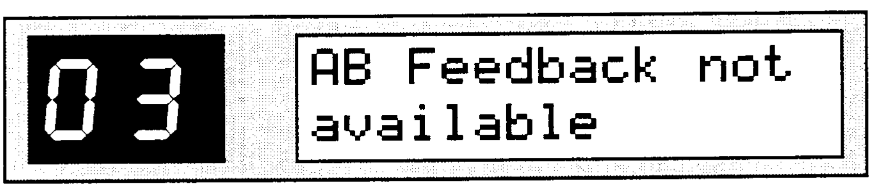


This screen allows you to control a feedback signal from the output of Unit B to the input of Unit A. A setting of 00 would not allow the feedback signal back into the units, whereas a setting of 99 would permit a full feedback signal.

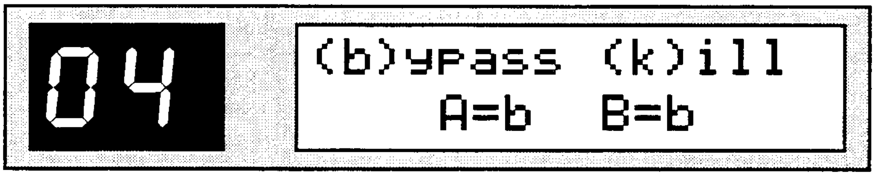


This parameter controls this amount.

If parameter 02 is parallel, this screen shows:



This parameter does nothing because of the nature of a parallel connection, there is no dry path (or feedback) around the units.



04 — Bypass Kill (Unit) A

05 — Bypass Kill (Unit) B

These parameters determine what happens when you bypass a unit (red LED lit). When set to bypass (b), the red LED is solidly lit, and only the dry signal passes through the unit. When set to kill (k), the red LED is flashing, and no signal passes through the unit.

When the units are set to bypass (b), it's like setting the Mix to 00. When the units are set to kill (k), it's like setting the Volume to 00, although your preset values are not affected.

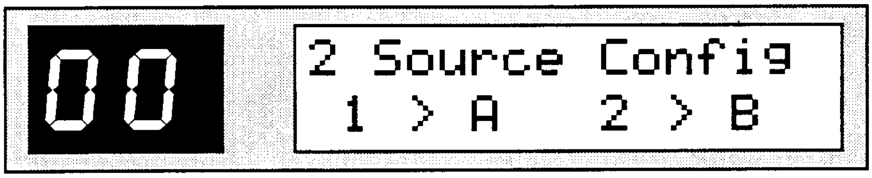
In order to use a foot switch to bypass a unit, it must be set to function as a DP/2 controller (see the description of System/MIDI parameters 31 to 34 in *Section 4 — System/MIDI*).

Notes

When using Kill in a configuration involving unit feedback pairs (for example, AB Unit Routing=feedback1):

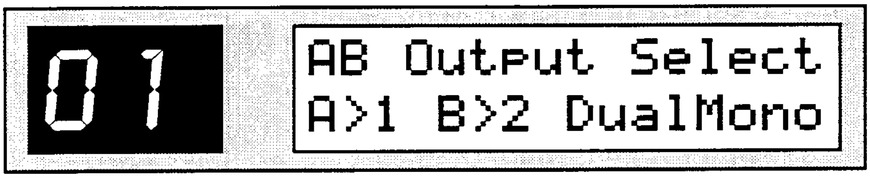
- Killing Unit B in a feedback pair mutes the signal.
- Killing Unit A does NOT mute the signal. Unit B can still pass dry signal.

2 Source Config



00 — 2 Source Config

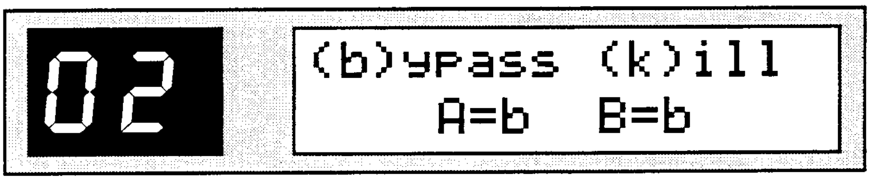
In a 2 Source Config, each unit functions as an independent 1 Unit effects processor.



01 — AB Output Select

Range: Dual Mono or Mixed Stereo

This parameter allows you to assign Units A and B as two independent mono signals to Outputs 1 and 2 respectively, or mix Units A and B into a stereo configuration.



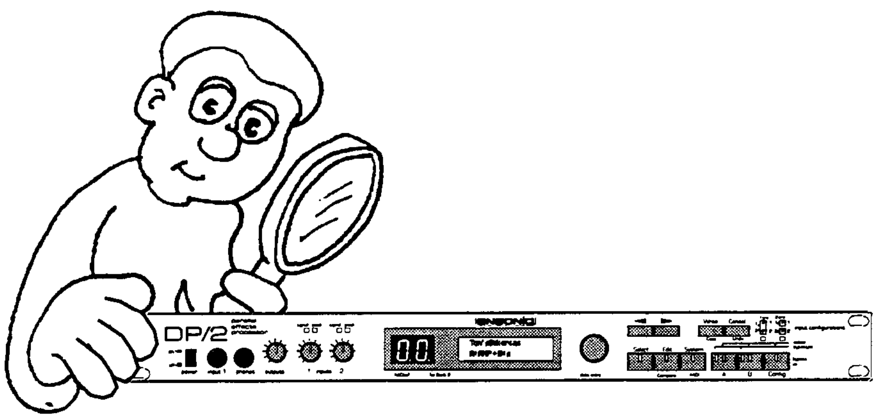
02 — Bypass Kill (Unit) A

03 — Bypass Kill (Unit) B

These parameters determine what happens when you bypass a unit (the red LED is lit). When set to bypass (b), only the dry signal passes through the unit. When set to kill (k), no signal passes through the unit.

> **Note:** In order to use a foot switch to bypass a unit, it must be set to function as a DP/2 controller (see the description of System/MIDI parameters 31 to 34 in *Section 4 — System/MIDI*).

Section 4 — System/MIDI

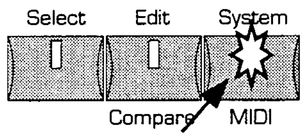


This section offers complete descriptions of the system (global) and MIDI parameters found within the DP/2, and provides an overview about how to set them to your liking.

Section 4 — System/MIDI

About System/MIDI

Press the (SYSTEM/MIDI) button to enter System/MIDI mode. Its LED will light:



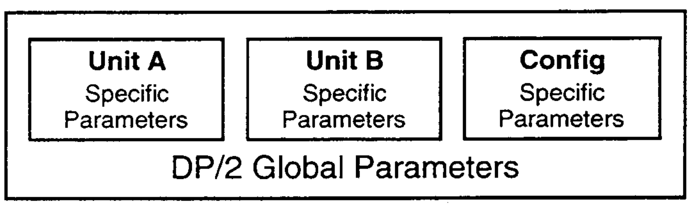
In this mode, you will find the system and MIDI parameters which control how the DP/2 responds to MIDI, the foot switch, and CV Pedal control inputs. You will also find user preference parameters, such as parameter wrapping and preset Auto-Load which allow you to tailor the user interface to your liking. System parameters are not affected by preset changes.

Furthermore, you will find a flexible system exclusive facility which allows presets and system parameters to be dumped to and loaded from an external MIDI System Exclusive recorder. See *Section 5 — Storage* for more information about System Exclusive dumps.

All parameters in System/MIDI mode belong to one of two categories:

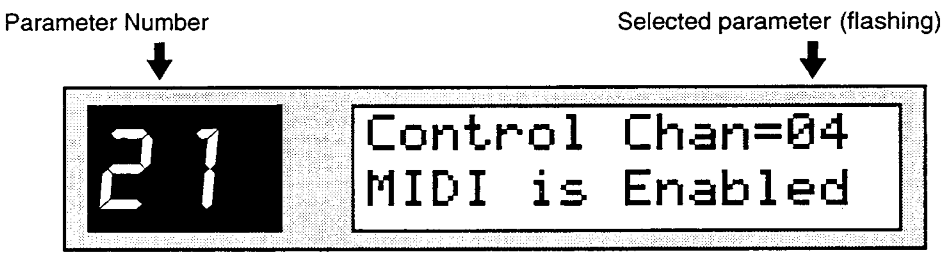
1. Parameters specific to units (MIDI channels, program change maps, etc.)
2. Parameters that affect the operation of the system globally, such as user preference switches and the DP/2 system controllers used as algorithm modulation sources.

System/MIDI Parameters



To set the System parameters

1. Press (SYSTEM/MIDI). The display shows the selected parameter. For example:



The red LED display (on the left) indicates the parameter number, and the currently selected parameter is flashing in the LCD display.

2. To edit the System/MIDI parameters, press the (◁) and (▷) buttons to scroll to the one you want to modify, and move the **Data Entry Knob** to change its value.

Shortcuts for Selecting System/MIDI Parameters

In addition to selecting System/MIDI parameters with the ◁ and ▷ buttons, you are provided with the following shortcuts to help you get quickly to the parameters you wish to modify:

- Press the SYSTEM/MIDI button, then press a Unit button (A or B) or CONFIG to get to the beginning of that unit's specific parameters.
- Press the SYSTEM/MIDI button repeatedly to cycle through groups of system global parameters.

To make it easier to get through the large number of System parameters, they have been organized into logical groups and the first parameter of each group can be accessed through the following button presses:

Param#	Parameter function	Press:
00-06	Unit A MIDI setup	SYSTEM/MIDI, then A for #00
07-13	Unit B MIDI setup	SYSTEM/MIDI, then B for #07
14-20	Config MIDI setup	SYSTEM/MIDI, then CONFIG for #14
21-22	MIDI chan for controllers	SYSTEM/MIDI repeatedly, until #21 is displayed
23-30	Defining the 8 DP/2 controllers	SYSTEM/MIDI again for #23
31-34	Footswitch functions	SYSTEM/MIDI again for #31
35-37	Song feature for creating preset chains	SYSTEM/MIDI again for #35
38-39	MIDI Sys-Ex enable and ID number	SYSTEM/MIDI again for #38
40-51	User Preference parameters	SYSTEM/MIDI again for #40
52	Software Version number	SYSTEM/MIDI again for #52

To go to a parameter far up in the list, use the above button presses to get to the parameter nearest the one you want to edit, then scroll to it using the ◁ and ▷ buttons.

Note: If you are already in System/MIDI mode (the System/MIDI LED is lit), you don't have to press the SYSTEM/MIDI button again each time to go to the MIDI parameters for Unit A, B or Config. Just press A, B, or CONFIG.

Unit Specific Parameters

The first 21 System/MIDI parameters (00 to 20) are the unit specific MIDI setup parameters. Each unit (A, B) and Config has seven MIDI setup parameters available. In this context only, the Config may be considered as a "virtual unit" because it has its own set of MIDI parameters which allow program changes to select config presets and external controllers to bypass all of the units together as is the case when the (CONFIG) button is toggled.

Note: The following screen displays are all shown for Unit A. The Unit B and Config screens are similar and can be reached by pushing their respective buttons and then scrolling with the (▷) button. The parameter numbers for Units A, B, and Config are listed in a box following each of the descriptions.

00 — MIDI Channel

Range: 01 to 16

Selects the MIDI channel to which the unit responds, if MIDI reception is enabled for the unit.

This is:	For:
parameter 00	Unit A
parameter 07	Unit B
parameter 14	Config

01 — MIDI Enable

Range: Disabled or Enabled

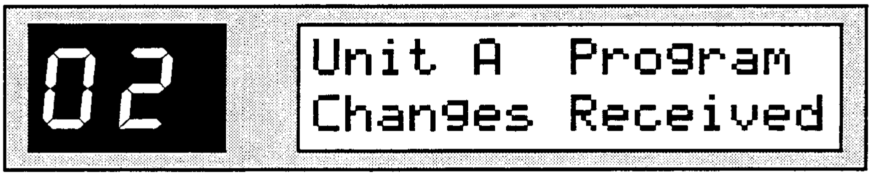
Determines whether the selected unit's MIDI reception will be enabled or disabled.

This is:	For:
parameter 01	Unit A
parameter 08	Unit B
parameter 15	Config

How the DP/2 Uses MIDI Channels

The DP/2 is capable of responding to a maximum of four MIDI channels at once. Each Unit (A, B) and the Config can have its own MIDI channel on which to receive program changes (parameters 02, 09, and 16) and MIDI Controller 7 (volume) information (parameter 47). Additionally, there is a separate controller channel used to receive controllers, pitch bends, channel aftertouch, key pressure, note events and velocity (parameter 21).

Each of these channels can be enabled and/or disabled individually. The only restriction is that the Config channel has to be different than the Unit channels. The Unit channels may be the same, and the controller channel can be the same as any Unit or Config channel.



02 — Program Change

Range: Ignored or Received

The DP/2 can receive MIDI program change messages to select presets and bypass/kill units (see description of parameter #5). This parameter determines whether you want to receive or ignore MIDI program changes for the selected unit. Program changes received on the Config channel select config presets. Program changes received on the Unit channels select 1 or 2 unit presets, depending on the current config type (see chart below). Program change reception can be received/ignored separately for each unit and the config. There is also a MIDI program change master switch that has to be enabled in order to receive *any* program changes (see description of parameter #41).

This is:	For:
parameter 02	Unit A
parameter 09	Unit B
parameter 16	Config

Program changes received on any unit call up the appropriate type of preset for the current config type:

1 Source Configuration

MIDI Channel:	Selects:
Unit A	2 Unit Presets (A & B)
Unit B	2 Unit Presets (A & B)

2 Source Configuration

MIDI Channel:	Selects:
Unit A	1 Unit Presets (Unit A)
Unit B	1 Unit Presets (Unit B)

- The Config MIDI Channel is always active.

Note: The DP/2 can be set-up to override this chart and always select 1 Unit Presets on unit MIDI channels regardless of the unit config type (see parameter 42).

03 Unit A Program
Change Map Off

03 — Program Change Map   Range: Off or On

Each Unit and the Config has a user-programmable "program-change-to-preset" map. This parameter determines whether the user-programmable maps will be enabled or disabled separately for each unit and the config. The next two parameters, used in conjunction with this one, allow you to define which DP/2 preset is selected by each MIDI program change number received. The map may also be programmed to ignore specific program change numbers, or to control the bypass status of the unit(s).

When set to "Off," the DP/2 will respond to MIDI Bank Select and Program Changes as follows:

MIDI Bank Select and Program Change Map
Bank Select LSB 000, Program Changes 000-099 select presets 00-99 (Bank 1).
Bank Select LSB 001, Program Changes 000-099 select presets 00.-99. (Bank 2).
Bank Select LSB 002 to 127 will be ignored, and will have no effect on Program Change reception.
Bank Select MSB values 000 to 127 will be ignored, and will have no effect on Program Change reception.

If this parameter is set to "On," MIDI program changes received by the unit or config are translated into DP/2 presets using the unit's programmable map. Whenever the system is reinitialized, the user-programmable maps are reset to their default settings using the mapping listed below:

Default Program Change Map
All MIDI Bank Select messages (LSB and MSB) will be ignored.
MIDI program changes 000 to 099 select presets 00 through 99 (Bank 1). Presets 00. to 99. (Bank 2) cannot be selected.
Program change 100 bypasses the affected unit(s).
Program change 101 kills the affected unit(s).
Program change 102 activates (un-bypasses) the affected unit(s).
Program changes 103 to 127 are ignored.

The program change maps for each of the units and the config are located at the following parameter numbers:

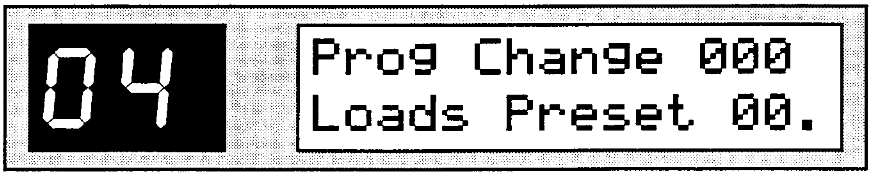
This is:	For:
parameter 03	Unit A
parameter 10	Unit B
parameter 17	Config

If it does not seem to be working:

1. Verify that your synthesizer (or other device) is really sending MIDI program change events and the DP/2 is receiving those messages. The DP/2's MIDI message indicator (located at the bottom of the red LED display) will light up when the DP/2 is receiving MIDI program changes (or other MIDI messages).
2. Make sure that the MIDI transmit channel (of the synthesizer) and the MIDI receive channel of the current unit (in the DP/2) match, and that program changes are enabled on the DP/2 in both the unit/config setup (parameter 02, 09, or 16) *and* the global switch (parameter 41).

Program Change-to-Preset Map Editor

This two-parameter screen is where you edit the program "change-to-preset" map.



04 — Program Change

Range: 000 to 127

The first parameter on this page selects the MIDI program change numbers used in the map.

This is:	For:
parameter 04	Unit A
parameter 11	Unit B
parameter 18	Config

05 — Loads Preset

Range: Loads Preset 00 to 99 (Bank 1), 00. to 99. (Bank 2), Bypasses Unit, Un-bypasses Unit, Kills Unit, Is Ignored

The second parameter defines which DP/2 preset the displayed MIDI program change number will select. Bank 2 preset numbers will be displayed with a decimal point after the preset numer (as shown in the display above). Program changes received on a unit or the config call up the appropriate type of preset for the current config type.

This parameter also allows you to bypass, kill, or un-bypass (un-kill) units using the displayed MIDI program change number (parameter 04).

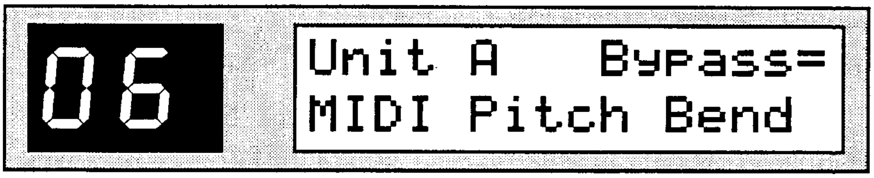
This is:	For:
parameter 05	Unit A
parameter 12	Unit B
parameter 19	Config

You can now go back and forth between these two parameters and define which DP/2 preset is selected by each MIDI program change number. This process is called mapping. Multiple program change numbers can map onto the same preset number. A common application is to map your synthesizer's presets onto effects presets so each sound on the synth has an associated external effect.

Tip: There is a quick way to edit the program change map if you have a keyboard or other device which can send program changes connected to the DP/2 MIDI Input:

- Select the second parameter (preset number).
- Send a program change (select a sound on your synthesizer) and notice the first parameter change. It should show the number of the program change received.
- Turn the **Data Entry Knob** to select the preset to be assigned to this program change.

You've just defined one entry in the program change-to-preset map. Send other program changes and use this process to map all 128 locations without ever changing parameters.



06 — Unit Bypass

Range: various

This parameter allows you to choose the controller source which will function as a bypass switch for the unit.

The available bypass controller sources are:

MIDI Pitch Bend	Footswitch 1-L	Ftsw2-L Toggle
MIDI Note Number	Ftsw1-L Toggle	Footswitch 2-R
MIDI Note Veloc	Footswitch 1-R	Ftsw2-R Toggle
MIDI Aftertouch	Ftsw1-R Toggle	MIDI Control #000 to #127
DP2 Analog CV In	Footswitch 2-L	Unassigned

The same controller source can be used to modulate *and* bypass an effect. However, to avoid unexpected side effects, the modulation and bypass functions should usually be assigned to different controllers. Sending Controller value 0-63 will un-bypass the unit, and sending Controller value 64-127 will bypass the unit.

Note that MIDI Controllers used for bypass are only received on the DP/2's Control channel.

This is:	For:
parameter 06	Unit A
parameter 13	Unit B
parameter 20	Config

List of MIDI Controller Names

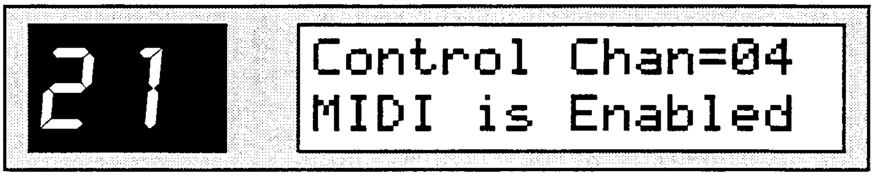
This list of MIDI Controller names (as found in the DP/2) represents the current state-of-the-art MIDI controller assignments as defined in the MIDI Detailed Specification, version 95.1:

Bank Select #000 - Bank Select	Expression #043 - Expression LSB	MIDI Control#086 - UNDEFINED
Mod Wheel #001 - Mod Wheel or Lever	FX Control1 #044 - Effect Control 1 LSB	MIDI Control#087 - UNDEFINED
Breath #002 - Breath Controller	FX Control2 #045 - Effect Control 2 LSB	MIDI Control#088 - UNDEFINED
MIDI Control#003 - UNDEFINED	MIDI Control#046 - UNDEFINED	MIDI Control#089 - UNDEFINED
Foot Control#004 - Foot Controller	MIDI Control#047 - UNDEFINED	MIDI Control#090 - UNDEFINED
Glide Time #005 - Portamento Time	GenPurpose1 #048 - UNDEFINED	FX Depth 1 #091 - Effects Depth 1
Data Entry #006 - Data Entry MSB	GenPurpose2 #049 - General Purpose 1 LSB	FX Depth 2 #092 - Effects Depth 2
Volume #007 - Volume	GenPurpose3 #050 - General Purpose 2 LSB	FX Depth 3 #093 - Effects Depth 3
Balance #008 - Balance	GenPurpose4 #051 - General Purpose 3 LSB	FX Depth 4 #094 - Effects Depth 4
MIDI Control#009 - UNDEFINED	MIDI Control#052 - General Purpose 4 LSB	FX Depth 5 #095 - Effects Depth 5
Pan #010 - Pan	MIDI Control#053 - UNDEFINED	Data Inc #096 - Data Inc
Expression #011 - Expression	MIDI Control#054 - UNDEFINED	Data Dec #097 - Data Dec
FX Control1 #012 - Effect Control 1	MIDI Control#055 - UNDEFINED	NonRegPmLSB #098 - Non-Reg param Num LSB
FX Control2 #013 - Effect Control 2	MIDI Control#056 - UNDEFINED	NonRegPmMSB #099 - Non-Reg param Num MSB
MIDI Control#014 - UNDEFINED	MIDI Control#057 - UNDEFINED	RegParamLSB #100 - Reg param Num LSB
MIDI Control#015 - UNDEFINED	MIDI Control#058 - UNDEFINED	RegParamMSB #101 - Reg param Num MSB
GenPurpose1 #016 - General Purpose 1	MIDI Control#059 - UNDEFINED	MIDI Control#102 - UNDEFINED
GenPurpose2 #017 - General Purpose 2	MIDI Control#060 - UNDEFINED	MIDI Control#103 - UNDEFINED
GenPurpose3 #018 - General Purpose 3	MIDI Control#061 - UNDEFINED	MIDI Control#104 - UNDEFINED
GenPurpose4 #019 - General Purpose 4	MIDI Control#062 - UNDEFINED	MIDI Control#105 - UNDEFINED
MIDI Control#020 - UNDEFINED	MIDI Control#063 - UNDEFINED	MIDI Control#106 - UNDEFINED
MIDI Control#021 - UNDEFINED	Sustain #064 - Sustain	MIDI Control#107 - UNDEFINED
MIDI Control#022 - UNDEFINED	PortamentoSw#065 - Portamento On/Off	MIDI Control#108 - UNDEFINED
MIDI Control#023 - UNDEFINED	Sostenuto #066 - Sostenuto	MIDI Control#109 - UNDEFINED
MIDI Control#024 - UNDEFINED	Soft Pedal #067 - Soft Pedal	MIDI Control#110 - UNDEFINED
MIDI Control#025 - UNDEFINED	Legato Ftsw #068 - Legato Ftsw	MIDI Control#111 - UNDEFINED
MIDI Control#026 - UNDEFINED	Hold 2 #069 - Hold 2	MIDI Control#112 - UNDEFINED
MIDI Control#027 - UNDEFINED	PatchSelect #070 - Sound Variation (Patch Select)	MIDI Control#113 - UNDEFINED
MIDI Control#028 - UNDEFINED	Timbre #071 - Harmonic Content (Timbre)	MIDI Control#114 - UNDEFINED
MIDI Control#029 - UNDEFINED	Release #072 - Release	MIDI Control#115 - UNDEFINED
MIDI Control#030 - UNDEFINED	Attack #073 - Attack	MIDI Control#116 - UNDEFINED
MIDI Control#031 - UNDEFINED	Brightness #074 - Brightness	MIDI Control#117 - UNDEFINED
Bank Select #032 - Bank Select LSB	SoundCntl 6 #075 - Sound Controller 6	MIDI Control#118 - UNDEFINED
Mod Wheel #033 - Mod Wheel LSB	SoundCntl 7 #076 - Sound Controller 7	MIDI Control#119 - UNDEFINED
Breath #034 - Breath Controller LSB	SoundCntl 8 #077 - Sound Controller 8	AllSoundOff #120 - All Sound Off
MIDI Control#035 - UNDEFINED	SoundCntl 9 #078 - Sound Controller 9	ResetCntrls #121 - Reset All Controllers
Foot Control#036 - Foot Controller LSB	SoundCtl 10 #079 - Sound Controller 10	LocalCntrlSw#122 - Local Control
MIDI Control#037 - Portamento Time LSB	GenPurpose5 #080 - General Purpose 5	AllNotesOff #123 - All Notes Off
Data Entry #038 - Data Entry LSB	GenPurpose6 #081 - General Purpose 6	OmniModeOff #124 - Omni Mode Off
Volume #039 - Volume LSB	GenPurpose7 #082 - General Purpose 7	OmniModeOn #125 - Omni Mode On
Balance #040 - Balance LSB	GenPurpose8 #083 - General Purpose 8	MonoModeOn #126 - Mono Mode On
MIDI Control#041 - UNDEFINED	Portamento #084 - Portamento Control	PolyModeOn #127 - Poly Mode On
Pan #042 - Pan LSB	MIDI Control#085 - UNDEFINED	Unassigned

Note: Controllers #000-031 are the MSBs and #032-063 are the LSBs for controllers with 14 bit resolution, and their names are displayed identically in the list of values.

System Global Parameters

There are some system global parameters (beginning with parameter 21) that can be reached by pressing (SYSTEM/MIDI) repeatedly. Parameter 21 is the first page of a sub-group. The following system global parameters are available:



21 — Control Chan   Range: 01 to 16

The first parameter on this page allows you to select the MIDI channel (01 through 16) on which MIDI controller messages (modulation sources) are received by the DP/2. This is the *only* channel on which the DP/2 can receive modulation and bypass controllers.

22 — Control Channel Reception   Range: Disabled or Enabled

This parameter allows you to enable or disable the receiving of MIDI controllers for the entire system (with the exception of MIDI Controller#7, which is used for Volume; see System/MIDI parameter #47).

23 DP2 Controller1=
DP2 Analog CV In

23 — DP2 Controller 1
24 — DP2 Controller 2
25 — DP2 Controller 3
26 — DP2 Controller 4
27 — DP2 Controller 5
28 — DP2 Controller 6
29 — DP2 Controller 7
30 — DP2 Controller 8

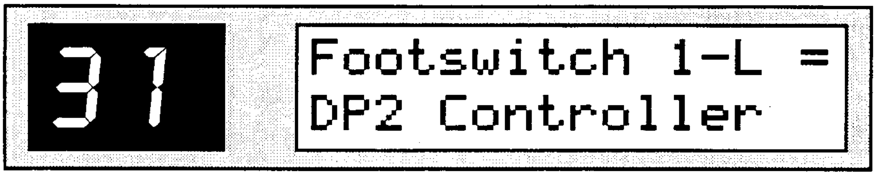
Ranges: (see Source List which follows)

Parameters 23 to 30 allow you to define eight system controllers to be used as modulation sources. Every algorithm (except GuitarTuner 2U) has parameters which allow you to use two of these controllers to modulate any two parameters in the algorithm (except the algorithm name–which cannot be modulated).

Source List

Source	Notes and Examples of modulation application
MIDI Controller Sources	
MIDI Pitch Bend	Can be used to control such things as left/right panning and rotor speed.
MIDI Note Number	Can be used to enable higher MIDI note numbers from a piano to shorten the decay on the reverb.
MIDI Note Veloc	Can raise or lower the effect mix. Higher velocities from drums can increase detune amount.
MIDI Aftertouch	Both channel (mono) and key (poly) pressure are recognized and are combined into a single mono source.
MIDI Control Numbers	The full range of controllers (0 to 127) is supported.
Additional (non-MIDI) Controller Sources	
DP2 Analog CV In	Usually a CV Pedal (like the ENSONIQ CVP-1), but could be generated by anything that produces a 0 - 5 volt control voltage, such as an analog synthesizer's CV out.
Footswitch 1-L Ftsw 1-L Toggle Footswitch 1-R Ftsw 1-R Toggle Footswitch 2-L Ftsw 2-L Toggle Footswitch 2-R Ftsw 2-R Toggle	The down position generates maximum modulation and the up position sets modulation to minimum. The **Foot Switch 1** and **2** jacks are designed exclusively for stereo (dual) foot switches like the ENSONIQ SW-10 Dual Foot Switch, allowing two discrete controllers per foot switch.
Unassigned	No system controllers are defined.

Tip: You can quickly change the modulation sources globally instead of individually for each unit preset, simply by reassigning the modulator sources for the eight system controller parameters.



31 — Foot Switch 1-L
32 — Foot Switch 1-R
33 — Foot Switch 2-L
34 — Foot Switch 2-R

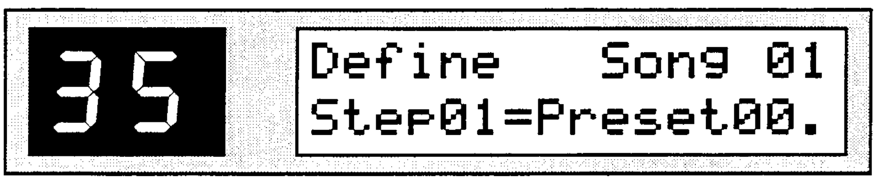
Ranges: (see the following table)

Parameters 31 to 34 define how foot switch 1-L and 1-R and 2-L and 1-R will be used. These are the possible ranges:

DP2 Controller	Allows the foot switch to be used as a modulation source. This is the only setting which allows the foot switch to be used for the tap-tempo feature in the Tempo Delay algorithm.
Increment Preset	Selects next higher preset of the current preset type on the current unit(s).
Decrement Preset	Selects next lower preset of the current preset type on the current unit(s).
Increment Song	Selects next higher song.
Decrement Song	Selects next lower song.
Song Preset Up	Selects next higher song step.
Song Preset Down	Selects next lower song step.
Unassigned (Off)	Ignores foot switch events.

Note: In order for the foot switches to be used to bypass units, they must be set to "DP2 Controller."

Song Editor



35 — Define Song
Range: 01 to 20

36 — Step
Range: 01 to 05

37 — Preset
Range: 00 to 99 (Bank 1), 00. to 99. (Bank 2), Goto Step 1

Parameters 35 (Song), 36 (Step) and 37 (Preset) allow you to set up lists of presets that differ from the order they are stored in memory. This list is meant primarily for live performance and can only be accessed by the foot switches.

There are 20 songs available, each containing 5 steps. You can assign any preset (00-99 in Bank 1, 00. to 99. in Bank2) in the current mode (in the current setting) to these steps. Bank 2 preset numbers will be displayed with a decimal point after the preset number. Assign your foot switch(es) to increment/decrement through songs/steps to make use of this feature. We recommend the SW-10 Dual Foot Switch for this purpose, as its two pedals let you change from song to song using one pedal and select steps within a song using the other.

> **Tip:** By creating a Config preset, copying it to different locations, and assigning different combinations of bypassed/unbypassed to each copy, you can create a song that uses the same "basic preset," but with different combinations of active and bypassed units (Config Presets store what's bypassed/un-bypassed).

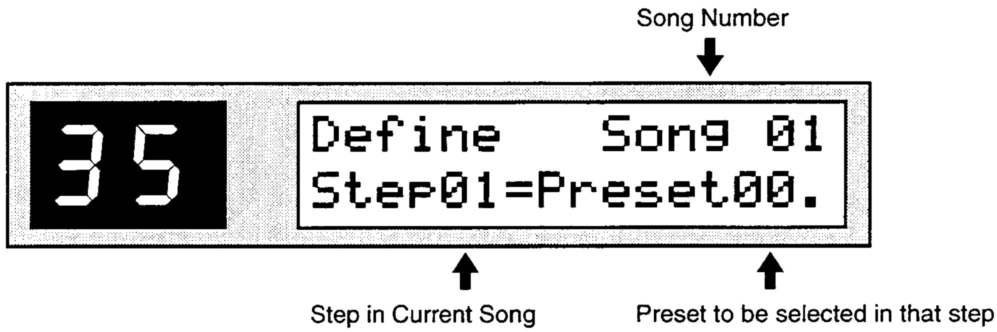
Using the Song Editor Feature

Here's how to define a series of presets to be selected in a specified order with the DP/2 Song Editor Feature:

1. Plug a stereo foot switch into the **Foot Switch 1** jack on the rear panel of the DP/2.
2. Press (SYSTEM/MIDI), until parameter #31 is showing on the display. Set the Foot Switch 1-L and 1-R parameters to the following values:

Param #:	Parameter:	Set to:	This will:
31	Foot Switch 1-L =	Song Preset Up	Advance to the next preset defined in the current song
32	Foot Switch 1-R =	Increment Song	Advance to the next song

3. Press the (▷) button to get to the Song Editor (params #35-37). The display shows:



4. Use the three Song Editor parameters to define up to 20 songs as you will need. When you are done editing, go back to parameter #35 and reset it to Song #01.
5. Now, press Foot Switch 1-L (on the left). Each time you press it, the DP/2 will select the preset that is the next step in the song, returning to step 01 after step 05. Pressing Foot Switch 1-R (on the right) will advance to the next song, returning to song 01 after song 20.

Note: This function is not limited to Foot Switch 1 only, Foot Switch 2-L and 2-R could just as easily be used. If you use Foot Switch 2, then make sure you set System/MIDI parameters #33 and 34 to the above listed values.

Using a Foot Switch to Alternate Between Two Presets
The DP/2 Song feature can also be used to simply alternate between two presets. For example, a guitarist might want to switch between a lead tone and a rhythm tone by pressing a single foot switch. In this case, we will need just one song, and a feature that lets you limit a song to less than five steps:

1. Press (SYSTEM/MIDI), until parameter #31 is showing on the display. Using the **Data Entry Knob**, set this parameter to "Foot Switch 1-L = Song Preset Up."
2. Press the (▷) button four times to get to the Song Editor (params #35-37). Select Song 01.
3. Let's suppose you want to alternate between Preset 82 (Bank 1) and Preset 55. (Bank 2). Set parameter #36 to Step 01, and set the preset selector (param #37) to "Preset 82."
4. Press (◁) to go back to the Step parameter, and select Step 02.
5. Press (▷), and set the preset selector (param #37) to "Preset 55. (Bank 2)."

   Note: Make sure the point appears in the display for selecting presets from Bank 2.

6. Press (◁) to go back to the Step parameter, select Step 03, and then set param #37 to "GotoStep1." You will find this choice at the top of the list.

Now, each time you press Foot Switch 1-L, it will toggle between the two presets you defined. The other foot switches can just as easily perform this function, or be programmed for other purposes, if you wish (System/MIDI parameters 31 to 34 allow this). Bear in mind that the "GotoStep1" option could also be entered as Step 04 or Step 05 to create songs of 3 or 4 steps instead of 2.

38 — MIDI SysEx ID   Range: 01 to 16

This parameter sets the system exclusive ID. This number is not a MIDI channel. It is simply an ID number imbedded within the system exclusive message. This allows usage of multiple DP/2's with a universal editor librarian, etc. All outgoing dumps will contain this ID number and incoming dumps will only be received if the ID in the message matches this parameter value.

39 — SysEx Reception   Range: Disabled or Enabled

This parameter (second line) defines whether system exclusive messages can be *received* by the DP/2. Outgoing dumps can always be initiated by pressing the (WRITE) button while in System/MIDI mode.

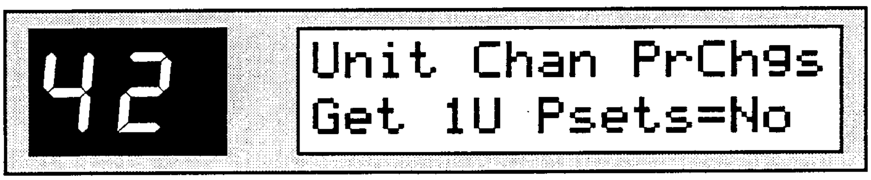
Set this parameter to "Disabled" to protect preset memory against being unintentionally changed by incoming dumps.

40 — Preset Memory Protect Range: Off or On

This parameter will prevent the RAM presets in both Bank 1 and Bank 2 from being changed or erased when set to "On." It must be set to the "Off" position in order to write an edited preset or copy a selected preset to a new location. It must also be "Off" in order to use the system utility command to initialize the RAM presets.

41 — MIDI Prog Change MasterSwitch Range: Off or On

This is the main program change receive switch. If it is set to "Off," all program change messages are ignored regardless of how the individual units are set up. If it is "On," the range in the individual unit MIDI setups will determine whether or not program changes are recognized.

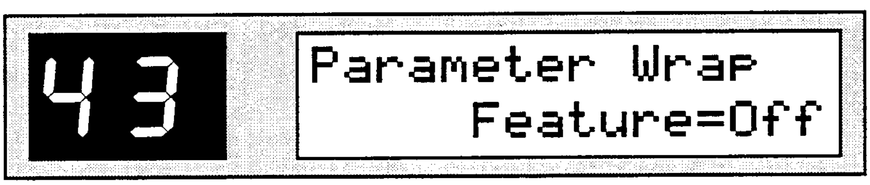


42 — Unit Chan PrChgs Get 1U Psets Range: No or Yes

This parameter controls how the DP/2 will respond to program changes on unit MIDI channels.

- When set to No, the DP/2 will respond to incoming program changes on unit MIDI channels, selecting the appropriate type of preset for the current config type (see the description of parameter 2 for more information).
- When set to Yes, program changes received on Unit Channels will always select 1U presets, regardless of the current config type.

  Note: When set to Yes, program changes will be ignored if a ganged multi-unit algorithm is installed in a unit.



43 — Parameter Wrap Feature Range: Off or On

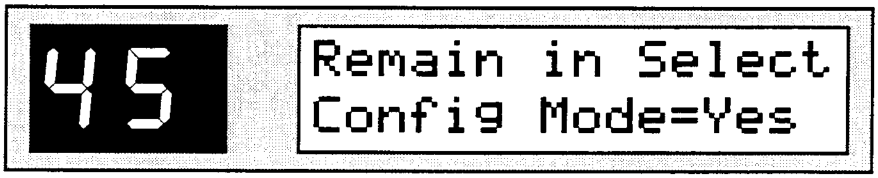
In the "Off" position, when ◁ is pressed, the display will stop at the lowest parameter. Likewise, when ▷ is pressed, it will stop at the highest parameter.

In the "On" position, when you are on the highest numbered parameter, pressing ▷ will wrap around to parameter 00. Likewise, to get to the highest number parameter from parameter 00, press ◁.

44 — Auto-Load Preset Range: No or Yes

In Select mode, when this parameter is set to "Yes," unit and config presets are automatically loaded one second after being selected with the **Data Entry Knob**, and without having to press the (SELECT) button. This allows you to select presets quickly.

When this switch is set to "No," you must press (SELECT) to actually load the displayed preset. For example, in a live context, a sound engineer may pre-dial an effect preset without activating it. It is now at the engineer's (or your) fingertip to bring in the preset at exactly the correct time simply by pressing (SELECT).



45 — Remain in Select Config Mode Range: No or Yes

This parameter determines whether or not the system will remain in Select Config mode after a config preset is selected.

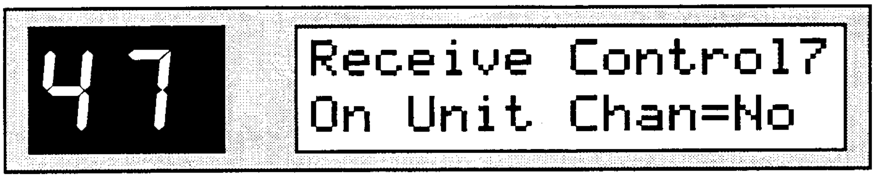
- When set to Yes, the DP/2 will stay in Select Config mode after a config preset is selected.
- When set to No, the DP/2 will behave as though the Unit A button was pressed after a config preset is selected, and will enter Select Unit mode.

  Note: When Auto-Load Preset (SelectMode)=Yes, the system ignores the setting of this parameter, and behaves as though it is set to Yes. (If it didn't do this, every time a Config was selected the screen would jump to the Select Unit A page -- that might be confusing).

46 — Set All 1U Pset Mixes To Wet Range: No or Yes

When the DP/2 is used in conjunction with a mixing board's auxiliary paths or in any other application where it's desirable to have only the effected (wet) signal appear in the DP/2's audio outputs, set this switch to "Yes." It will automatically set all 1 Unit Preset mix levels to 99 (all wet) when they are selected—or installed in Edit mode—without altering the actual values saved in your presets. When you alternate between using the DP/2 for mixing and as a dedicated instrument effects box, this mix override switch saves you from laborious reprogramming of all of your preset mix ranges. This parameter *will not* change the mix settings on 2 Unit and Config presets selected, only 1 Unit presets.

Note: This parameter works for 1 Unit presets selected *after* this parameter has been set to "Yes," not for the preset currently installed when this parameter is set.

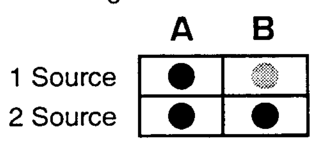


47 — Receive Control7 On Unit Chan Range: No or Yes

When this switch is set to "Yes," the DP/2 will listen to MIDI Controller 7 (volume) messages on unit MIDI channels. This feature allows you to have MIDI controlled effects mixing. This is a "smart" parameter and will only control the volumes of units which are not feeding other units (i.e., units at the end of a serial signal chain; see the examples below). This preserves the gain structure between serial units by only controlling the volume of the unit at the end of the chain. You may need to adjust the Modulation Response Rate parameter (system parameter 50) for best results.

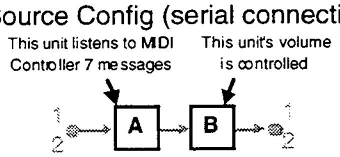
When Receive Control7 On Unit Chan=Yes

MIDI Controller messages will control the volumes of:

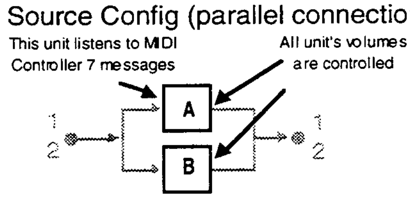


Here are some examples of how this "smart" parameter works:

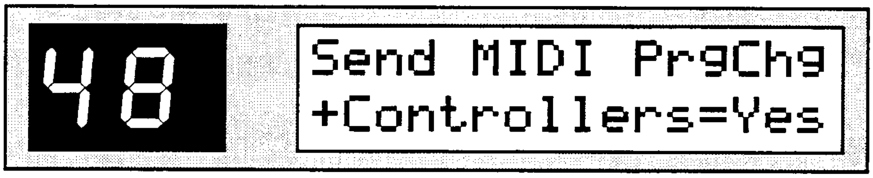
1 Source Config (serial connection)



1 Source Config (parallel connection)



Note: MIDI Controller 7 messages received on the config MIDI channel are ignored.



48 — Send MIDI PrgChg +Controllers Range: No or Yes

When this switch is set to "Yes," the DP/2 will act as a MIDI controller source by generating MIDI messages. These messages are sent out via the DP/2's MIDI Out jack. The DP/2 can send program changes as well as certain controller events, depending on the context. This is useful with a sequencer that will record MIDI events for later playback.

Every time you select a new preset, the DP/2 will send out a Bank Select message and a program change message corresponding to the preset number. The Bank Select value corresponds to the bank in which the preset resides. The program change will be sent on the preset's primary unit channel, following the same conventions for receiving events (refer to the Active MIDI Channel chart earlier in this section). Program changes are not sent out in response to inbound MIDI program changes.

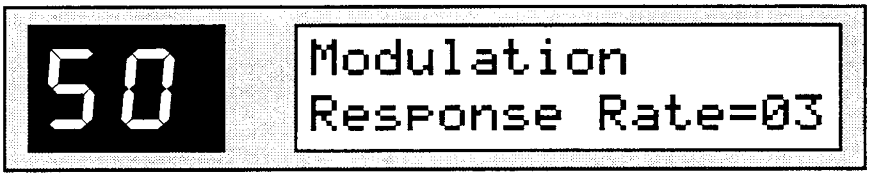
MIDI Channel Controller messages sent out by the DP/2:

Source	Controller number	Standard Function
CV Pedal	04	Foot control
Foot Switch 1-L	70	Sound Variation (Sound Controllers)
Foot Switch 1-R	71	Harmonic Content (Sound Controllers)
Foot Switch 2-L	72	Release Time (Sound Controllers)
Foot Switch 2-R	73	Attack Time (Sound Controllers)

These assignments cannot be changed. All controller messages will be send on the DP/2 controller channel. A foot switch event will be transmitted as a controller only when the Foot Switch Function (system parameters 31 to 34) is set to "DP2 Controller."

49 — Data Entry Knob Response Range: Fast, Normal, Slow

Controls the acceleration rate of the **Data Entry Knob** when spinning quickly through parameter values. The "Slow" setting will reduce the size of the jumps taken by the values as the knob is rotated quickly, whereas the "Fast" setting will make the jumps larger. The default is "Normal."



50 — Modulation Response Rate Range: 01 to 30

Controls the rate at which any modulation is applied to the modulation destinations in the DP/2. As parameters are modulated, their values are changed. This parameter controls how quickly the parameters are changed by setting the size of the change. A setting of 01 would have the slowest response with the finest resolution (small changes), whereas a setting of 30 is fast (larger changes), but the modulation is not as smooth.

51 — Use Alternate ROM Presets Range: No or Yes

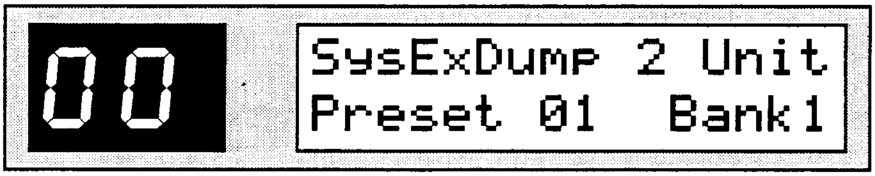
Allows you to replace the ROM presets (locations 50 to 99 in Bank 1, and 50. to 99. in Bank 2) with the factory RAM presets from the corresponding bank. This parameter defaults to "No" at reset, or power-on.

52 — Operating System Version

This read-only page shows the version number of the operating system EPROMs installed in the unit. This same page is displayed when you power on the DP/2.

System Exclusive Dump

Push the (WRITE) button at any time while in System/MIDI mode to engage the system exclusive dump facility.



SysExDump Range: 1 Unit, 2 Unit, Config, System, or All

This two-parameter page allows you to select and send various kinds of MIDI System Exclusive dump messages from the DP/2. The DP/2 is capable of receiving 1 Unit and 2 Unit preset SysEx dumps from the DP/4 and DP/4+.

The first parameter allows you to select which type of preset you want to dump (1, 2 Unit and Config). You can also dump system and MIDI parameters (such as preset maps and user preference switch ranges).

The second parameter is only available when the first parameter is set to 1 Unit, 2 Unit, or Config. It allows you to use the **Data Entry Knob** to select an individual preset (numbers 00 to 49 Bank 1, and 00. to 49. Bank 2) to transmit. If you rotate past "49. Bank 2," you can set the parameter to "Bank 1" or "Bank 2" which will dump the entire bank of the indicated preset type.

Once the screen shows you what you want, press (WRITE) once more to start MIDI transmission of data. You can also press (CANCEL) to exit this screen without sending any data.

Notes

1. At entry, the dump type defaults to the preset belonging to the currently active unit, whose preset type and number are displayed.

2. ROM presets and ROM preset banks may not be dumped from the front panel. System Exclusive Dump Request commands are provided for external devices which need to extract those banks. Refer to the System Exclusive documentation for more details.

3. System Exclusive message reception is "automatic" and does not have to be enabled by any action other than making sure that System Exclusive reception is enabled and that the ID number setting matches the ID embedded in the dump to be received (see System/MIDI parameters 38 and 39). A confirmation message is displayed to indicate what type of dump has been received when the dump reception is complete, or an error message is displayed if there is a problem with the incoming data.

Refer to *Section 5 — Storage* for more information on using System Exclusive messages with the DP/2.

System Utility Functions

The DP/2 has some useful utility functions that are initiated using special button combinations. These include:

Soft Reset

The DP/2 allows you to reset the system without erasing the internal memory. To accomplish this:

1. While holding down the (SYSTEM/MIDI) button,
2. Press the (A) button.

The system will automatically reset itself as if the power switch had been turned off and back on, with the advantage of not unnecessarily stressing the internal components. The data in the units should not be affected by this procedure, and the system should come back on with the same effects loaded. After the DP/2 has been reset, it defaults to Select mode.

Initializing the RAM Presets

If you want to restore the factory default presets into *all* of the 300 internal RAM memory preset locations (Bank 1 and Bank 2), there is a convenient command that does so without affecting the System parameter settings.

> WARNING! THIS PROCESS WILL ERASE ALL RAM PRESETS! The 300 user presets in the internal RAM memory (locations 00-49 in Bank 1 and 00.-49. in Bank 2) are automatically loaded with the factory defaults after reinitialization. Good backup habits should be an important part of your routine. Save any important information by using the MIDI System Exclusive Dump feature of the DP/2 (see *Section 5 — Storage*), or manually write down the relevant parameters using a photocopy of the Preset Parameter Worksheet found at the end of this manual. If you fail to do so, you may accidentally lose the presets you've created.

To initialize the RAM presets:

1. While holding down the (SYSTEM/MIDI) button,
2. Press the (B) button. The following screen will appear:

```
00   Hit <WRITE> To
     Init RAM Presets
```

Press the (CANCEL) button to quit *without* initializing the presets.

Press the (WRITE) button at this point to initialize all of the RAM presets in the DP/2. Remember that by doing this, you will replace *all* of the RAM Preset data in the DP/2 with the factory defaults!

This procedure cannot be completed if the Preset Memory Protect switch (system parameter 40) is set to "On."

Reinitializing the DP/2

If your DP/2 is behaving in peculiar ways (the display is showing alphanumeric characters that shouldn't be there or unexplainable Unexpected Event messages are displayed), and turning the DP/2 power off and then on again won't cure the problem, try reinitializing the DP/2.

> WARNING! THIS PROCESS WILL ERASE ALL RAM PRESETS! The 300 user presets in the internal RAM memory (locations 00-49 in Bank 1 and 00.-49. in Bank 2) are automatically loaded with the factory defaults after reinitialization. Good backup habits should be an important part of your routine. Save any important information by using the MIDI System Exclusive Dump feature of the DP/2 (see *Section 5 — Storage*), or manually write down the relevant parameters using a photocopy of the Preset Parameter Worksheet found at the end of this manual. If you fail to do so, you may accidentally lose the presets you've created.

To reinitialize the DP/2

1. While holding down the (SYSTEM/MIDI) button,
2. Press the (B) button.
3. Press the (▷) button once. The display shows:

```
01   Hit <WRITE> To
     Reinitialize!!!!
```

Press the (CANCEL) button to quit *without* reinitializing the system, or

4. Press the (WRITE) button to reinitialize the DP/2. Remember that by doing this you will replace *all* of the RAM Preset data in both Bank 1 and Bank 2 in the DP/2, and *all* System/MIDI parameters will be reset to their default values!

If reinitializing the DP/2 does not correct the problem, then contact an Authorized ENSONIQ Repair Station.

Note: If the DP/2 is sitting in an infinite loop of system errors (the display is continually cycling through errors), press the (SYSTEM/MIDI) button to escape this state.

Note: In the unlikely event of a system malfunction where the buttons cease to perform their normal functions, you can save your entire set-up (all Preset Banks and System parameters) with a System Exclusive dump by pressing the (WRITE) button. This will help you restore all of the user-defined parameters. For more information about System Exclusive dumps, see *Section 5 — Storage.*

System Diagnostic Parameters

The DP/2 has a number of diagnostic parameters. To access them:

1. While holding down the (SYSTEM/MIDI) button,
2. Press the (CONFIG) button.
3. Use the (◁) and (▷) buttons to scroll through the following parameters:

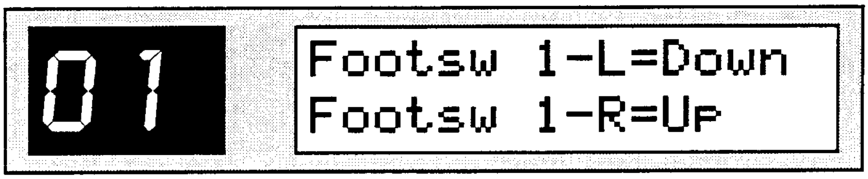
CV Pedal

This read-only parameter displays the current control voltage reading of the CV Pedal input's 8 bit analog-to-digital converter. With a CVP-1 pedal connected, the typical range is 000 to 170, with the fully up position showing 000 and fully down showing approximately 170 (±10%).

If nothing is plugged into the **CV Pedal** jack, the displayed value should between 234 and 255.

Foot Switches 1 and 2

These read-only parameters are used to test the functionality of the foot switch circuitry of the DP/2. They show the current position of Foot Switches 1-L and 1-R. The following page shows the current position of Foot Switches 2-L and 2-R.



- If a dual foot switch with a stereo plug (such as the SW-10) is connected to the **Foot Switch 1** jack, the display will show "Footsw 1-L=Up/Down" for the left foot switch controller, and "Footsw 1-R=Up/Down" for the right foot switch controller.
- If a single foot switch with a mono plug (such as the SW-2) is connected to the **Foot Switch 1** jack, the display will show "Footsw 1-R=Up" when the footswitch is *not* depressed, and "Footsw 1-R=Down" when the foot switch is held down. Foot switch 1-L is disabled and should always indicate the "Down" position (as shown above).
- If nothing is plugged into the **Foot Switch 1** jack, the display should always show "Footsw 1-L=Up" and "Footsw 1-R=Up."

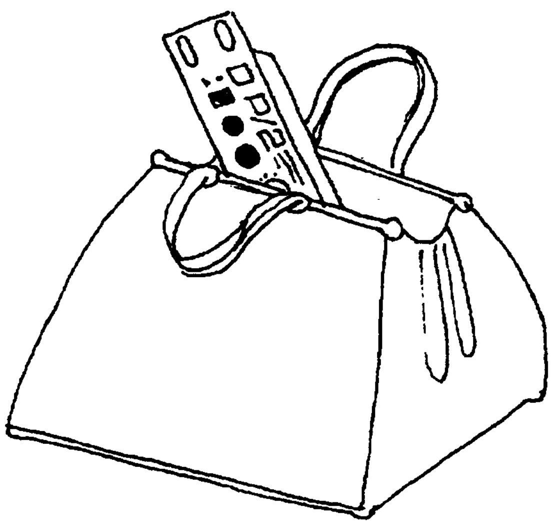
On the following page:

- If a dual foot switch with a stereo plug (such as the SW-10) is connected to the **Foot Switch 2** jack, the display will show "Footsw 2-L=Up/Down" for the left foot switch controller, and "Footsw 2-R=Up/Down" for the right foot switch controller.
- If a single foot switch with a mono plug (such as the SW-2) is connected to the **Foot Switch 2** jack, the display will show "Footsw 2-R=Up" when the footswitch is *not* depressed, and "Footsw 2-R=Down" when the foot switch is held down. Foot switch 2-L is disabled and should always indicate the "Down" position.
- If nothing is plugged into the **Foot Switch 2** jack, the display should always show "Footsw 2-L=Up" and "Footsw 2-R=Up."

Warning!
The use of single (mono) foot switches is not recommended, and can affect the operation and performance of the DP/2.

☞ **Important:** There are a number of additional service and diagnostic parameters in the DP/2 that SHOULD NOT BE ALTERED. If altered, they could cause the DP/2 to reinitialize (erasing the RAM presets), severe volume changes could result that may damage your equipment/hearing, or they could affect the internal circuitry making the DP/2 inoperative.

Section 5 — Storage



This section covers the storage functions on the DP/2, which enable you to copy 1U, 2U, or Config Presets internally to other locations, write (save) edited preset information, and transmit dumps via MIDI system exclusive messages.

Presets may also be manually transcribed using the Preset Parameter Worksheet preset found at the end of this section, and also at the end of this manual.

Internal Storage

The Preset Memory Protect Switch

Before you can copy or write presets, the Preset Memory Protect switch must be set to the "Off" position. If it is not set to "Off" before trying to write or copy a preset, the display will momentarily show "MEMORY PROTECTED."

To set the Preset Memory Protect Switch:

1. Press the (SYSTEM/MIDI) button.
2. Use the (◁) and (▷) buttons to scroll until the display shows:

```
40   Preset Memory
           Protect=On
```

Tip: There is a quick way to get to this display. The System/MIDI parameters are divided into sub-groups. By pressing the (SYSTEM/MIDI) button several times, you can quickly scroll through the sub-groups. Parameter 40 is the first page of one of these sub-groups.

3. If the word "On" is flashing, move the **Data Entry Knob** counterclockwise to the "Off" position. If the word "Off" is flashing, RAM preset data can be changed.
4. Once this switch has been set to the "Off" position, you can save your preset. Press the (EDIT) button to return to Edit mode. Your newly edited preset should still be intact and ready to save.

Saving Presets

Presets can be named and saved into any RAM location (preset numbers 00 to 49 in Bank 1; and 00. to 49. in Bank 2) using the following procedure:

To Name and Save a Preset

1. Press the (EDIT) button (the Edit LED should be on).
2. Press the (WRITE) button. The top line of the display will show one of three different screens:

```
Write to 1U Pset
```

This shows that you are saving a 1 Unit Preset.

```
Write to 2U Pset
```

This shows that you are saving a 2 Unit Preset.

```
Write to Config
```

This shows that you are saving a Config Preset.

The screen that appears and the type of preset that can be written is determined by the current config and the unit that is active when you press (WRITE). The rules which govern this are the same as in Select mode. You can write to the type of preset that you can select in the current config.

3. Using the **Data Entry Knob**, choose a RAM location (preset numbers 00 to 49 in Bank 1; and 00. to 49. in Bank 2) for your new preset. Notice that the LED numeric display shows the *destination* number where your preset will be saved. The old preset in that location will be lost when it is replaced by the new preset. The first 50 storage locations in each Bank for each type of preset are user-programmable (battery backed up). Presets 50 to 99 (Bank 1) and 50. to 99. (Bank 2) are ROM (Read Only Memory) factory presets and cannot be replaced.

   Once you have selected the internal location into which the preset will be written (or saved), you can then edit (change) the name of the new preset.

4. Press the (WRITE) button again. The top line of the display will show one of three different screens, depending on what type of preset you're saving:

```
Edit 1U PsetName
```

This shows that you are naming a 1 Unit Preset.

```
Edit 2U PsetName
```

This shows that you are naming a 2 Unit Preset.

```
Edit Config Name
```

This shows that you are naming a Config Preset.

The name that appears on the bottom line of the display is usually the name of the last selected preset. At this point, you should change the name to better describe the preset that you are saving. The bottom line of the display has 16 spaces to create your own name. The display looks something like this:

```
10   Edit 2U PsetName
     Old Preset Name
```

Cursor (underline) beneath first alpha-numeric character

5. Use the (◁) and (▷) buttons to move the cursor left and right, and the **Data Entry Knob** to change the alpha-numeric characters at the current cursor position.

List of Alpha-Numeric Characters

The following alpha-numeric characters are available in the DP/2 for editing preset names, and appear (in columns) as they would if you were to turn the **Data Entry Knob** clockwise:

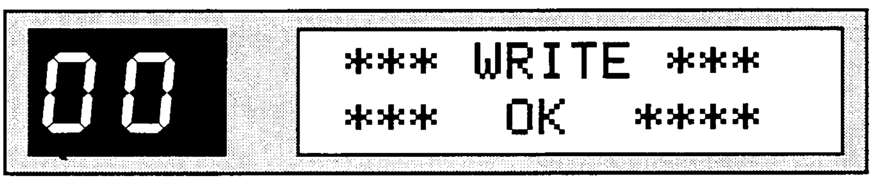
(blank)	)	2	;	D	M	V	_	h	q	z
!	*	3	<	E	N	W	`	i	r	{
"	+	4	=	F	O	X	a	j	s	|
#	,	5	>	G	P	Y	b	k	t	}
$	-	6	?	H	Q	Z	c	l	u	→
%	.	7	@	I	R	[	d	m	v	←
&	/	8	A	J	S	¥	e	n	w	
'	0	9	B	K	T	]	f	o	x	
(	1	:	C	L	U	^	g	p	y	

Tip: There is a quick way to select and scroll through the alpha-numeric characters. While on this page, the (A), (B), and (CONFIG) buttons will act as shortcuts:

To Get:	Press:
Upper case characters A – Z	(A)
Lower case characters a – z	(B)
Numbers 0 – 9	double-click (B) (press two times quickly)
Special Characters (the first one is a blank space)	(CONFIG)
Alternate Special Characters	double-click (CONFIG) (press two times quickly)

6. Once you have named your preset, you can either:

- Press (CANCEL) to return to the Write Preset Location page to confirm that the name and the destination you have chosen are correct, or quit from the writing procedure, or
- Press (WRITE) a third time to save your preset. The display will momentarily read:



The new preset location will automatically be selected after this message disappears.

Tip: After you've saved your preset, you may want to reset the Preset Memory Protect switch (System/MIDI parameter 40) back to the "On" position to eliminate any risk of accidentally erasing or changing your new preset.

Bailing Out

At any point in the saving process, you can press (CANCEL) twice to exit from the writing procedure and return to Edit mode. This may be necessary if the preset type is not what you expected to save. Make sure that the Config LED is not on unless you want to save a config preset.

Advanced Features

Switching Preset Types when Saving

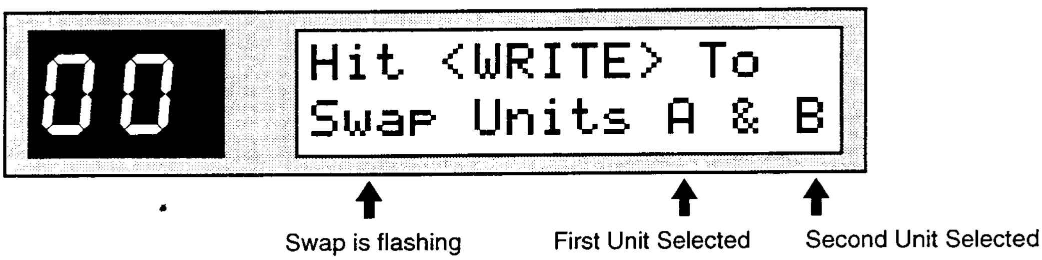
Before selecting a location for your preset, you may press one of the unit buttons (A or B) to force the type of preset being written to be a 1 Unit preset. The **Data Entry Knob** will now select 1 Unit preset locations, and when you press WRITE a third time to confirm the save, the unit data from the indicated unit will be saved as a 1 Unit preset with the name you choose (the default name is the last name of the algorithm). This is useful for saving single units from within a 2 Unit or Config preset where 1 Unit presets are not usually available. Note that ganged 2 Unit algorithms ("3.6 sec DDL 2U," "Pitch Shift 2U," "Guitar Tuner") cannot be saved this way.

Similarly, pressing CONFIG will force the type of preset being written to be a Config preset. The **Data Entry Knob** will now select config preset locations.

Swapping 1 Unit Presets

1 Unit algorithms can be swapped between Units A and B, by using the following procedure:

1. Press EDIT, then WRITE.
2. Press a Unit button (A or B). The top line of the display shows "Write to 1U Pset."
3. While pressing and holding *the same Unit button,* press the other Unit button. The display shows:



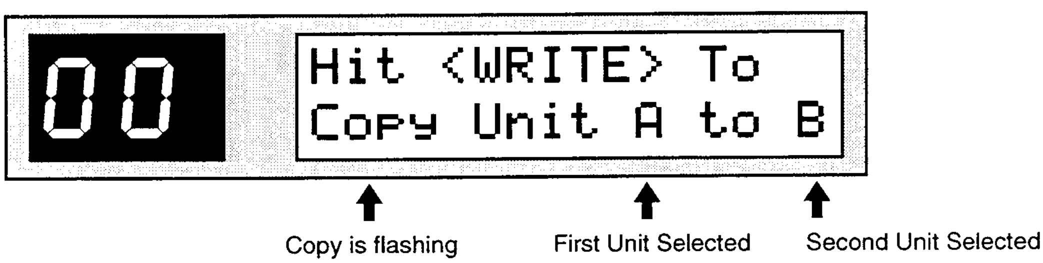
4. Press WRITE to swap the units. The display says "Units Swapped!"

You can use this procedure any time you want to swap a 1 Unit preset with another.

Copying a 1 Unit Preset to the Other Unit

A 1 Unit preset can be copied from one unit to the other, by using the following procedure:

1. Press EDIT, then WRITE.
2. Press the Unit button (A or B) that you want to copy.
3. While pressing and holding *the same Unit button,* press the other Unit button.
4. Turn the **Data Entry Knob** clockwise. The display looks like this:



5. Press WRITE to copy the first preset to the second location. The display says "Unit Copied!"

You can use this procedure any time you want to copy a 1 Unit Preset to the other unit.

Copying Presets

The DP/2 can also copy presets to other RAM locations in either Bank 1 or Bank 2.

To Copy a Preset

1. Press (SELECT) (the Select LED should be on).
2. Press the (COPY) button. The top line of the display shows one of three possible screens:

```
Copy to 1U Pset
```

This shows that you are copying a 1 Unit Preset.

```
Copy to 2U Pset
```

This shows that you are copying a 2 Unit Preset.

```
Copy to Config
```

This shows that you are copying a Config Preset.

The screen that appears and the type of preset that will be copied is determined by the current config and the unit that is active when you press (COPY). The rules which govern this are the same as in Select mode. You can copy what you have most recently selected in the current config.

3. Press (CANCEL) if you wish to exit from the copy procedure and return to Select mode. This may be necessary if the preset type is not what you expected to copy. Make sure that the Config LED is not on unless you wish to copy a config preset.
4. Use the **Data Entry Knob** to choose a new location (preset numbers 00 through 49 in Bank 1; 00. through 49. in Bank 2) to copy your preset. The LED numeric display shows the destination number for your preset.
5. Press the (COPY) button a second time to copy your preset. The display will momentarily show the same "*WRITE OK*" message that appears when saving a preset from Edit mode (see earlier).
6. You have just successfully copied your preset.

MIDI System Exclusive Storage

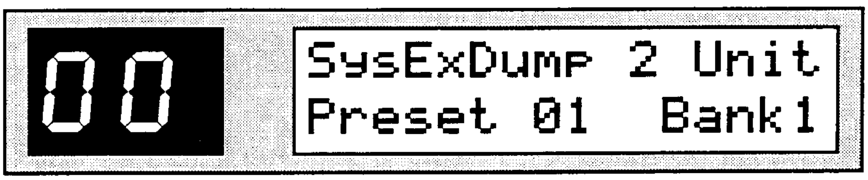
Sending MIDI Sys-Ex Messages to another DP/2 or to a Storage Device

The DP/2 is able to send System Exclusive (Sys-Ex) dumps of presets, either singly or in banks, as well as dumps containing all of the system parameters. These dumps can be directly transmitted to another DP/2, or can be recorded by an external device which has MIDI Sys-Ex Recorder capabilities (such as the ENSONIQ TS-10, TS-12 or ASR-10) to be stored and later retransmitted to the DP/2.

If you need more specific details on the messages, please refer to the *DP/2 MIDI System Exclusive Specification* available from ENSONIQ (see the Appendix for information on how to obtain this document).

To Send DP/2 Data Out via MIDI System Exclusive Dump

1. Press (SYSTEM/MIDI).
2. Press the (WRITE) button at any time while in System/MIDI mode to engage the system exclusive dump utility. The display looks something like this:



This two-parameter page allows you to select and send various kinds of MIDI System Exclusive dump messages from the DP/2. The DP/2 is capable of receiving 1 Unit and 2 Unit preset SysEx dumps from the DP/4 and DP/4+.

The first parameter allows you to select which type of preset you want to dump (1, 2 Unit and Config). You can also dump system and MIDI parameters (such as preset maps and user preference switch ranges).

The second parameter is only available when the first parameter is set to 1 Unit, 2 Unit, or Config. It allows you to use the **Data Entry Knob** to select an individual preset (numbers 00 to 49 Bank 1, and 00. to 49. Bank 2) to transmit. If you rotate past "49. Bank 2," you can set the parameter to "Bank 1" or "Bank 2" which will dump the entire bank of the indicated preset type.

3. When the screen shows what you want to send, make sure that the receiving device is ready to accept data, and press (WRITE) once more to start MIDI transmission of data.

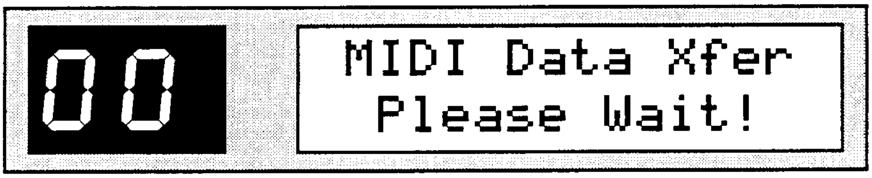
   You can also press (CANCEL) to exit this screen without sending any data.

The Available System Exclusive Dumps in the DP/2 are:

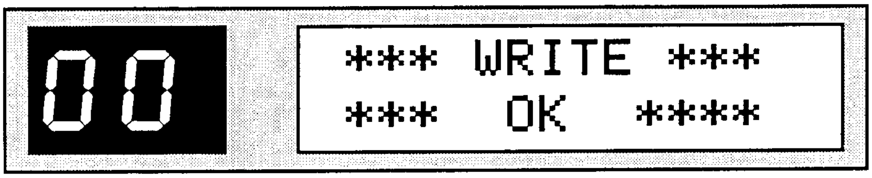
Dump Types:	Display shows:	What's included:
Single 1-Unit Preset	SysExDump 1 Unit Preset 00 Bank1	Single 1-Unit RAM preset from Bank 1 <00 to 49>
Single 1-Unit Preset	SysExDump 1 Unit Preset 00. Bank2	Single 1-Unit RAM preset from Bank 2 <00. to 49.>
1-Unit Preset Bank	SysExDump 1 Unit Preset Bank1	50 1-Unit RAM presets <Bank1 or Bank2>
Single 2-Unit Preset	SysExDump 2 Unit Preset 00 Bank1	Single 2-Unit RAM preset from Bank 1 <00 to 49>
Single 2-Unit Preset	SysExDump 2 Unit Preset 00. Bank2	Single 2-Unit RAM preset from Bank 2 <00. to 49.>
2-Unit Preset Bank	SysExDump 2 Unit Preset Bank1	50 2-Unit RAM presets <Bank1 or Bank2>
Single Config Preset	SysExDump Config Preset 00 Bank1	Single Config RAM preset from Bank 1 <00 to 49>
Single Config Preset	SysExDump Config Preset 00. Bank2	Single Config RAM preset from Bank 2 <00. to 49.>
ConfigPreset Bank	SysExDump Config Preset Bank1	50 Config RAM presets <Bank1 or Bank2>
System	SysExDump System Parameters only	All system parameters
All Preset Banks	SysExDump All Preset Banks	300 RAM presets
All Preset Banks with System Parameters	SysExDump All PsetBanks+System	300 RAM presets and all system parameter settings

Preset data is always transmitted from and received into the internal RAM. The System Parameters dump includes all system and MIDI parameters found in System/MIDI mode (such as program change-to-preset maps and user preference switch settings). It is best to use the smallest single dump type which contains all of the data you wish to reload at one time.

The display will show the following message for a brief time, which depends on the amount of information being transmitted:



When the dump is complete, the following message will appear for a moment to indicate that the transmission occurred without errors:

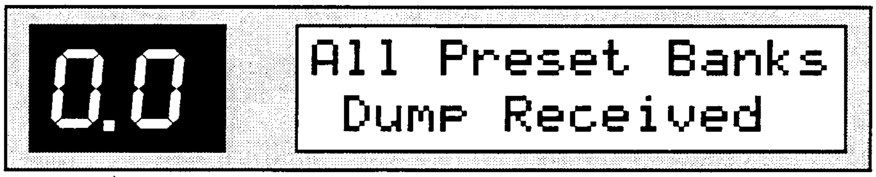


Note: ROM presets and ROM preset banks may not be dumped from the front panel. System Exclusive Dump Request commands are provided for external devices which need to extract those banks. Refer to the *DP/2 MIDI System Exclusive Specification* for more details (see the Appendix).

Remember! The System Exclusive ID number (system parameter 38) is embedded in every message, so it must be set correctly on the transmitting and receiving units if dumps are to be recognized.

Receiving MIDI System Exclusive Dumps with the DP/2

System Exclusive message reception is "automatic" and does not have to be enabled by any actions other than making sure that System Exclusive reception is enabled and that the ID number setting matches the ID embedded in the dump to be received (System/MIDI parameters 38 and 39). The MIDI message indicator will light while the dump is being received. A confirmation message is displayed when the dump reception is complete to indicate what type of dump has been received and where the new data has been stored.



MIDI Message indicator lights when any events are received.

When a SysEx dump is received, one of the following messages will be displayed:

Display shows:	What's received:
1U Preset 00 Dump Received	One 1-Unit RAM preset loaded into Bank 1 <00 to 49>
1U Preset 00. Dump Received	One 1-Unit RAM preset loaded into Bank 2 <00. to 49.>
1U Preset Bank1 Dump Received	50 1-Unit RAM presets loaded into Bank1 or Bank2
2U Preset 00 Dump Received	One 2-Unit RAM preset loaded into Bank 1 <00 to 49>
2U Preset 00. Dump Received	One 2-Unit RAM preset loaded into Bank 2 <00. to 49.>
2U Preset Bank1 Dump Received	50 2-Unit RAM presets loaded into Bank1 or Bank2

Display shows:	What's received:
Cfg Preset 00 Dump Received	One Config RAM preset loaded into Bank 1 <00 to 49>
Cfg Preset 00. Dump Received	One Config RAM preset loaded into Bank 2 <00. to 49.>
Cfg Preset Bank1 Dump Received	50 Config RAM presets loaded into Bank1 or Bank2
System Params Dump Received	All system parameters
All Preset Banks Dump Received	300 RAM presets
All Banks+System Dump Received	300 RAM presets and all system parameter settings

Dumps containing system parameters will have an additional message which follows the confirmation message to indicate that the previous settings of the system parameters have been replaced by new data.

Note: When single presets dumps are received, they will be written to the same memory location from where they were originally stored.

Problems?

An error message will be displayed instead of the confirmation message if there was a problem with the incoming data. If no message appears after the MIDI LED goes off, then the dump was ignored. Make sure the Receive enable is set to "On" and the ID number is set correctly (System/MIDI parameters 38 and 39).

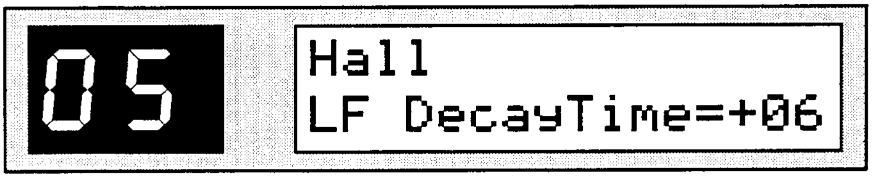
It is possible that some computer interfaces can transmit Sysex dumps faster than the DP/2 can receive it. On many Macintosh™ applications, however, the sys-ex transfer rate can be reduced. You could also try reducing the speed of the interface (e.g., from 2x to 1x).

For more information about error messages, refer to the *DP/2 MIDI System Exclusive Specification* (see the Appendix).

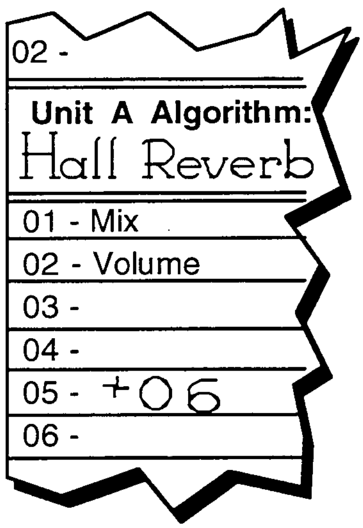
Using the Preset Parameter Worksheet

There is another method for saving presets. You can *manually* record all the parameters of your created preset by using the Preset Parameter Worksheet (or a photo copy) located at the end of this section and at the end of the manual. Although this method is time-consuming and laborious, it is still an accurate method for saving presets if you do not have access to a System Exclusive data recorder, or if you want to use one of your own custom effect creations at another studio without bringing your own DP/2.

Presets consist of a combination of algorithm parameters and config parameters. Although each type of config and algorithm has a different set of parameters, you can still use the worksheet because it is based on the *parameter* number. You can find the parameter numbers by looking at the LED numeric display in Edit mode on your DP/2. For example, in the Hall Reverb algorithm, the LF Decay Time is parameter 05:



In our example, the value for the LF Decay Time (parameter 05) is set at +06. This would be written on the Worksheet like this:



To find the parameter numbers for your config, press the (EDIT) button, then the (CONFIG) button. Use the (◁) and (▷) buttons to scroll through and record the edited parameters.

To find the parameter numbers for your algorithms, press (EDIT), then each Unit button ((A) and/or (B)) that relates to your preset (a 1 Unit preset only uses one algorithm and only requires one column). Use the (◁) and (▷) buttons to scroll through and record the parameter settings.

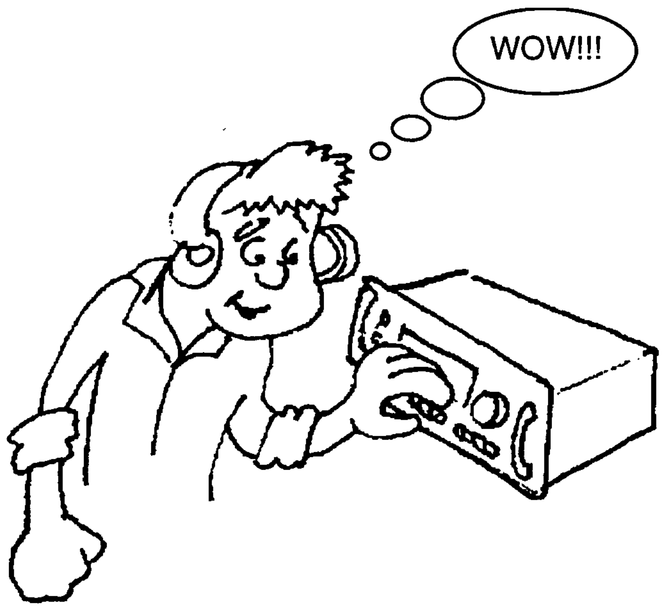
Note: Many of the algorithms and configs do not require all of the spaces provided on the Preset Parameter Worksheet. You should leave these spaces blank.

DP/2 Preset Parameter Worksheet	Preset Name:	
Config Parameters:	1 2 Source Config	01-
02-	03-	04-

Unit A Algorithm:	Unit B Algorithm:
01- Mix	01- Mix
02- Volume	02- Volume
03-	03-
04-	04-
05-	05-
06-	06-
07-	07-
08-	08-
09-	09-
10-	10-
11-	11-
12-	12-
13-	13-
14-	14-
15-	15-
16-	16-
17-	17-
18-	18-
19-	19-
20-	20-
21-	21-
22-	22-
23-	23-
24-	24-
25-	25-
26-	26-
27-	27-
28-	28-
29-	29-
30-	30-
31-	31-
32-	32-
33-	33-
34-	34-

Notes:

Section 6 — Presets



This section shows how to select the presets found in the DP/2, and shows the names, locations, and a brief description of the 1 Unit, 2 Unit, and Config Presets.

Quick Steps to Hear Presets

Plug your mono sound source into the front panel **Input 1** jack, or for stereo into **Inputs 1** and **2** on the back of the DP/2. Connect **Output 1** (and **Output 2** for stereo) to an audio source (amplifier, mixing board, etc.), or plug headphones into the front panel **Phones** jack. If you don't hear any sound, set the **Input** and **Output Knobs** to a 12 o'clock position, and adjust the **Input Knob** levels as needed.

To Select 1 Unit Presets:

1. Press the (SELECT) button, then the (CONFIG) button.
2. Turn the **Data Entry Knob** until the screen shows "52 Select 1U Preset" (in Bank 1).
3. Press the (SELECT) button, then the unit (A) button.
4. Turn the **Data Entry Knob** to display the names of the different 1 Unit presets.
5. Press the (SELECT) button to hear the preset on the display.
6. To select other 1 Unit Presets, turn the **Data Entry Knob** and press (SELECT) whenever you see a preset you'd like to hear.

To Select 2 Unit Presets:

1. Press the (SELECT) button, then the (CONFIG) button.
2. If you are using a mono source, turn the **Data Entry Knob** until the screen shows "50 1 Src: Mono In" (in Bank 1).

   If you are using a stereo source, turn the **Data Entry Knob** until the screen shows "51 1 Src: Stereo In" (in Bank 1).
3. Press the (SELECT) button, then the unit (A) button.
4. Turn the **Data Entry Knob** to display the names of the different 2 Unit Presets.
5. Press the (SELECT) button to hear the preset on the display.
6. To select other 2 Unit Presets, turn the **Data Entry Knob** and press (SELECT) whenever you see a preset you'd like to hear.

To Select Config Presets:

1. Press the (SELECT) button, then the (CONFIG) button.
2. Turn the **Data Entry Knob** to display the names of the different Config Presets.
3. Press the (SELECT) button to hear the preset on the display. Remember that Config Presets dictate how the inputs, outputs, and units are routed together. You may have to rearrange your cables/routings in order to hear certain presets correctly.
4. To select other Config Presets, turn the **Data Entry Knob** and press (SELECT) whenever you see a preset you'd like to hear.

1-Unit RAM Presets (Bank 1)

Reverbs	00	Vocal Plate 1	Bright with lots of early reflections. Try it on backing vocals.
	01	Synth Plate	Long, bright decay.
	02	Vocal Plate 2	Dark, long decay.
	03	Glitter Plate	A bright plate with good spread; try it with drums.
	04	Reflections	Very live plate, with high early reflection levels.
	05	Zobie Plate	High reflection levels and a short decay time. Foot switch 1-L adds pre-delay.
	06	Slam Plate	Tight, with short decay. Foot switch 1-L offers longer variation.
	07	Multi Plate	Foot switch 1-L toggles two decay variations.
	08	Short Plate	Short .71 second decay.
	09	Ballad Reverb	Dark decay, great on vocals.
	10	Close Hall	Second row, center.
	11	Bright Hall	Big space with hard surfaces.
	12	Vocal Hall	Foot switch 1-L toggles two different listening positions within the hall.
	13	Concert Hall 1	Foot switch 1-L will alternate dark and bright variations.
	14	Concert Hall 2	Big hall with lots of reflections. Foot switch 1-L alters decay time.
	15	Summer Hall	Starts with medium decay, foot switch 1-L shortens decay time.
	16	Famous Hall	Smooth, bright decay; 3.47 decay time.
	17	Movie Hall	Hollywood soundtrack reverb; a long decay, bright hall with plenty of sizzle.
	18	Huge Empty Hall	You're in a concert hall all alone... Foot switch 1-L toggles the HF bandwidth.
	19	Dry Room	Tight room sound with quick decay time.
	20	Room 224	Bright room reverb.
	21	Medium Room	Foot switch 1-L will make the room smaller.
	22	Dark Room	Extra rear wall reflections can be added with foot switch 1-L.
	23	Early Reflects	Non linear reverb simulates early echo reflections in a room.
	24	Distant Mics/Rm	Ambience from the back of the room.
	25	Conga Room Rev	Medium size room optimized for percussion.
	26	Live Perc Room	Bright live space; try it with Latin percussion.
	27	Smooth Non Lin	Great for pop dance drum tracks.
	28	Bright Non Lin	Fast decay; try it with drums and percussion.
	29	Fake Linear Verb	Non-linear hall simulation.
	30	Vintage Verb	Sounds like a loosely tuned plate reverb.
	31	Gated Verb	Smooth release, great for toms.
	32	Dark Drum Room	Small room with very absorptive surfaces.
	33	Digital Drm Room	Foot switch 1-L offers two time variations.
	34	Smooth Gated Rev	Gated reverb with a less abrupt decay. Foot Sw 1-L adds the slapback DDL.
	35	DynamicGatedRev	Gated reverb pulses according to the source's dynamic level.
	36	Boom Room	Non linear reverb with plenty of bottom.
	37	Kickdrum NonLin	Medium decay; try it on toms, too.
	38	Doppler Hall	Deep flanging creates the doppler pitch bending effect.
	39	Tiled Room	Short decay time, bight room with hard surfaces.
	40	Gymnasium	A big open room, long decay.
	41	Backstage	Small room, all-purpose.
	42	String Verb	Long pre-delay adds depth.
	43	French Horn Verb	A dark reverb with a long decay for brass.
	44	Slap Room	Small room with an obvious slap echo off the back wall.
	45	Quick Slaps	Small room with hard surfaces and prominent early reflections.
	46	Splatter Plate	Louder early echos thicken & spread. Optimized for synths and lead sounds.
	47	Small Reflection	Small room, short decay, with a pronounced back wall reflection.
	48	Small Room Gtr	Small bright wood room designed for acoustic guitars; also try it with brass.
	49	Bat Cave	Cavernous open space with long decay time.

Effect Modulation is based on factory default settings.

1-Unit ROM Presets (Bank 1)

Reverbs	50	Small Room Rev	Foot switch 1-L toggles two decay variations.
	51	Large Room Rev	Foot switch 1-L toggles two decay variations.
	52	Hall Reverb	Large, bright hall with highly reflective walls.
	53	Small Plate Rev	All-purpose plate sound with 1.35 second decay.
	54	Large Plate Rev	Decay time is 2.89 seconds.
	55	Reverse Reverb	FtSw 1-L toggles slapback echo on/off. CV pedal changes trigger threshold.
	56	RetrigReverseRev	Each signal that crosses the threshold will re-start the reverb's envelope.
	57	Gated Reverb	Gated Reverb that has a more abrupt release time than preset #31.
	58	Non Lin Reverb 1	Foot switch 1-L toggles two variations in brightness.
	59	Non Lin Reverb 2	More stereo movement.
	60	Non Lin Reverb 3	Optimized for percussion.
Time Based Processing	61	Plate-Chorus 1	Plate reverb runs through chorusing for an extra smooth decay.
	62	Plate-Chorus 2	Longer plate reverb through chorusing; nice on electric keyboards.
	63	Chorus-Reverb 1	Smooth chorus into bright plate reverb.
	64	Chorus-Reverb 2	Faster chorus rate and shorter decay time.
	65	8 Voice Chorus	Use foot switch 1-L to add more chorus regeneration for a thicker texture.
	66	EQ-Chorus-DDL	Foot switch 1-L turns on the DDL repeats.
	67	Lush Keys	EQ, chorus and delays set up for electric pianos, synths, etc.
	68	Wet Chorus	Chorusing plus delays. Parameters 07 and 08 set delay times.
	69	Fusion Bass	Delays are added to the chorus; great with fretless bass.
	70	Slow Chorus	Slow LFO, less wide, with a slight bit of DDL added.
	71	Fast Chorus	Quicker LFO speed and more width.
	72	NonLin Chorus	Smooth chorusing coupled with a quickly decaying plate.
	73	Sci-Fi Flanger	Deep flanging bends the signal up & down slowly for instant sound effects.
	74	Fast Flanger-Rev	Edit parameter 04 for different rates.
	75	Flanger-Reverb	Flanger plus plate reverb combined in one unit.
	76	EQ-Flanger-DDL	Thick, slow flanging with delays; try it on guitars or synth pads.
	77	Phaser-DDL	Deep phase shifting with long delays.
	78	Flanger	CV pedal sweeps the flanger rate.
	79	Country Keys	Keyboard phase shifter texture.
	80	Medium Flange	Try sweeping the flanging rate with a CV pedal during a solo.
	81	Tight Flange	Deep, slow flanging effect.
	82	Key Funk Flange	Medium flanging along with DDL fatten up rhythm parts.
	83	Key Funk Phaze	Delays with a thick phasing effect; try with clav or percussive keyboard parts.
	84	Phaser-Reverb	Slow moving phase shifter feeds a plate reverb.
	85	Surf Organ	Fast phaser creates 60's cheesy organ sound. FtSw 1-L toggles two rates.
	86	Multi Tap Delay	Four delay taps with separate time, level, pan and regeneration amounts.
	87	Dual Delay	Two delay taps set up for a ping-pong effect.
	88	Tempo Delay	1/2 note delays at 120 BPM. Set parameters 03 and 04 to your needs.
	89	EQ-DDL-with LFO	The LFO moves the delay for a fat delay effect. Try changing parameter 06.
	90	FS Tap-Tempo DDL	Tapping foot switch 1-L sets the delay tempo.
	91	Vocal Spice	Tight slap echos fatten the vocal; set tap 1 time to fit your tempo.
	92	Slap Vocal	Two slapback echoes create a classic vocal treatment.
	93	MIDI Clock Delay	Send MIDI clocks to control delay time. Ft Sw 1-L toggles 1/4 & 1/8 delays.
	94	Multi Slaps	Four quick slap echoes; try it on vocals.
	95	Mono Multi Taps	Four delays in mono; delay times set for 100 BPM.
	96	Ping Pong Delay	Delays slap from one side to the other. Set rate with parameters 03 and 04.
	97	Pitch Shifter	Foot switch 1-L toggles intervals.
	98	Pitch Shift-DDL	Harmonies spiral upward; delay controlled by foot switch 1-L.
	99	Fast Pitch Shift	CV Pedal increases the detuning.

Effect Modulation is based on factory default settings.

1-Unit RAM Presets (Bank 2)

Time Based Processing	00.	Vocal Spreader	Popular de-tune effect with wide panning.
	01.	Tape Stop Effect	Great for sound effects; sounds like the abrupt stop of an analog tape deck.
	02.	Slap in the Face	Four fast delays spread across the stereo field.
	03.	Vintage Delay CV	CV pedal moves the delay time just as you could on an old tape delay.
	04.	Pitch Correction	MIDI pitch bend (DP/2 controllers channel) changes pitch; record into a sequencer.
	05.	Unique Doubler	Short delays, detuning and feedback loop; try it on synths.
	06.	Octave Shift/DDL	An octave harmony through delays.
	07.	Fast LFO Detune	10 cent detuning with fast LFO modulation.
	08.	Phase-0	Slow phasing creates 3-dimensional spacial manipulation and surround FX.
	09.	Dimensional DDL	Unique delay preset created with the reverb pre-delay parameter.
	10.	Sparkle PS-DDL	Swirly vocal effect using a small pitch shift and short delay times.
	11.	1/16 Note DDL	Plays one beat of 1/16th notes. 90 BPM, Internal clock controls tempo.
	12.	1/16 Triplet DDL	90 BPM, Internal clock controls tempo.
	13.	1/8 Note DDL	Internal clock controls tempo. 100 BPM. Set the tempo with parameter 04.
	14.	1/8 Triplet DDL	Internal clock controls tempo. 100 BPM. Set the tempo with parameter 04.
	15.	1/4 Note DDL	Internal clock controls tempo. 100 BPM. Set the tempo with parameter 04.
	16.	1/2 Note DDL	Internal clock controls tempo. 100 BPM. Set the tempo with parameter 04.
Amps, Speakers, Instrument Effects	17.	Slappin' Bass	10:1 compression and bright EQ for slap style bass.
	18.	Orange Phaser	Classic stomp-box. CV Pedal adds echo.
	19.	Recording Bass	EQ and compression set up for recording a bass direct through a console.
	20.	Grungy Bass Amp	Dirty amp in a room; Foot switch 1-L will toggle chorus on/off.
	21.	Contemporary Bs	Bass amp in a room with flanger. Toggle reverb off with foot switch 1-L.
	22.	Bass Speaker	Tunable Speaker set up for electric bass.
	23.	60's Tube Bs Amp	Retro feel and tone.
	24.	Ratty Fuzz Bass	It's that cheesy old fuzz box from up in the attic...
	25.	60's Soul	Classic fuzz solo sound with flanging.
	26.	Roto-Uni-Vibe	Re-creation of a 60's rotary speaker emulator.
	27.	Dead Battry Fuzz	Simulates that cheesy old fuzz box when the 9 volt's on the way out!
	28.	Semi Clean Swirl	Rotary speaker with distortion. Foot switch 1-L toggles speed.
	29.	Super 4-10 Amp	Lightly crunchy all-purpose rock amp tone.
	30.	Tube Mic Preamp	Insert into a mixer channel for vocals or bass.
	31.	It's the Cops!	Wide stereo spread amp tone made famous by the British trio.
	32.	Happy Fuzz Face	Raunchy fuzz box emulation for guitars or bass.
	33.	Muff Fuzz Box	Classic 70's and 'alternative' fuzz sound.
	34.	Tube Saturator	Add warmth and harmonics to anything!
	35.	Octavia	Produces a note an octave above the input.
	36.	SlowAutoWah+Dist	Param 11 controls the speed of the wah attack. Optimized for single notes.
	37.	Meaty Speaker	Dark speaker tone with plenty of bottom.
	38.	2-Speed Vib.Amp	FtSw 1-L toggles 2 vibrato speeds on this somewhat distorted amp tone.
	39.	2-Speed TremAmp	Clean amp tone with 2 tremolo speeds; toggle with foot switch 1-L.
	40.	Shimmer	Fast vibrato effect with delays for guitars or keyboards.
	41.	Dark Jazzy Spkr	Useful when recording single coil pickups direct.
	42.	Nasty Fuzz	Foot switch 1-L toggles 2 different reverb times. Best on single notes.
	43.	Slide Master	For rock slide guitar. The auto-wah adds a little random tone variation.
	44.	Live Clean Amp	Dynamic clean guitar amp with a bit of edge to it.
	45.	'58 Tweed	Cool classic amp sound.
	46.	CV Wah Wah 1	Use the CV pedal to control the wah.
	47.	CV Wah Wah 2	A wah with 2 filters tuned an octave apart, controlled by the CV pedal.
	48.	'66 Car Radio	Makes any signal sent through it sound likes it's played from a 3" speaker.
	49.	Rhythm Amp	Slightly distorted amp tone set up for chords.

Effect Modulation is based on factory default settings.

1-Unit ROM Presets (Bank 2)

Amps, Speakers, Instrument Effects	50.	VCF-Distortion 1	Distorted auto-wah tone for guitars, but try it on bass & keyboards, too.
	51.	VCF-Distortion 2	Parameter 13 will bypass the distortion portion.
	52.	Guitar Amp 1	Rock guitar tone. Experiment with the EQ controls to shape your own preset.
	53.	Guitar Amp 2	Parameters 15-20 offer many tonal variations.
	54.	Guitar Amp 3	High-gain lead sound.
	55.	Guitar Amp 4	Big stack tone.
	56.	Digital Tube Amp	Tonal changes respond to your playing dynamics.
	57.	Dynamic Tube Amp	Versatile touch responsive guitar amp simulation.
	58.	Speaker Cabinet	Try inserting this into a track where you need the sound of a speaker cabinet.
	59.	TunableSpeaker 1	A versatile speaker emulator with tunable EQ.
	60.	TunableSpeaker 2	Combines a tunable speaker emulator with a noise gate.
	61.	FlngCmprDstRev	Flanging, compression, and distorted guitar amp in one unit.
	62.	Wah-Dist-Revrb	Auto-wah into a distorted amp with reverb for guitar.
	63.	Dist-Cho-Revrb	All purpose crunch with chorus and a medium plate reverb.
	64.	Dist-Roto-Revb	Foot switch 1-L toggles rotary speaker speed.
	65.	Rotating Speaker	A clean organ speaker simulation.
	66.	Dist. RotarySpkr	Overdriven rotary organ tone. Parameter 07 adds more distortion.
	67.	Modern Blues	Contemporary blues lead tone. Foot switch 1-L will turn off chorus.
	68.	The Fuzz Box	A versatile vintage distortion/fuzz simulation.
	69.	Grilled Cloth	Rippin' lead tone, circa 1970.
	70.	The Fuzz Wahd CV	Raunchy lead guitar tone with a CV pedal controlled wah wah.
	71.	Fast Vibrato	Great for guitars or electric pianos.
	72.	EQ-Vibrato-DDL	Use a CV pedal to sweep the rate and width of the vibrato.
	73.	Analog Flanger	Retro flanging effect that works well with synths and keyboards.
	74.	EQ-Tremolo-DDL	CV pedal sweeps the rate, foot switch 1-L adds the delays.
	75.	Fast L-R Tremolo	Great for vintage electric piano simulations.
Studio Tools	76.	EQ-Panner-DDL	Three effects in one.
	77.	Wild Panner	Sample & Hold creates a unique effect.
	78.	Slow Panner	Slow stereo panning from left to right.
	79.	Fast Panner	Fast left to right panning.
	80.	Subtle Panner&EQ	Slow panner set for narrow stereo field.
	81.	Staccato	Rhythmic staccato effect; change rate with parameter 03.
	82.	ADSR Env Gen	Simulates the volume pedal swell popularized by guitarists.
	83.	Syn Pad ADSR	Use to re-shape the envelope of a synth, background vocal, etc.
	84.	EQ-Gate	3-band EQ with a noise gate.
	85.	Parametric EQ	Versatile EQ with high and low shelving filters and 2 sweepable mid bands.
	86.	EQ-Compressor	Two effects in one; this variation is set up with 4:1 compression, with no EQ.
	87.	Vocal Compressor	The EQ-Compressor optimized for vocals.
	88.	Snare Compressor	The EQ-Compressor optimized for drums.
	89.	Expander	1:2 expansion exaggerates the dynamics of a track.
	90.	1:10 Expansion	Useful in any situation where you might use a noise gate.
	91.	Inverse Expander	Sound that cross below the threshold will be compressed.
	92.	Noise Gate	Set parameters 09 and 10 for your source material.
	93.	Squish and Gate	2:1 compression and a fast gate.
	94.	VanderPol Filter	Adds high frequency 'exciter' effect. Mix level is critical.
	95.	De-esser	Vary the EQ controls will listening to the processed signal.
	96.	Rumble Filter	Adjust parameter 03 to suit your program material.
	97.	Instant Oldies	Run a loop sample through this to instantly age it.
	98.	Signal Generator	Foot switch 1-L changes between 1k sine wave and white noise.
	99.	No Effect/Silent	Mix parameter set to '99' makes it completely silent.

Effect Modulation is based on factory default settings.

2-Unit RAM Presets (Bank 1)

Reverbs	00	Best Small Space	General-purpose small room reverb.
	01	Luscious Plate	Long plate reverb with a bit of chorus for a smooth decay.
	02	Versatile Hall	Dense hall reverb with medium decay time.
	03	Oldtime Plate	Use anytime you need a slightly 'springy' reverb.
	04	Tube Plate	Dark tone, medium decay.
	05	Trumpet Plate	Versatile plate with a touch of chorusing.
	06	Compressed Plate	Plate reverb into 20:1 compression.
	07	Horn Verb	Try it with any brass solo instrument.
	08	Twisting Hall	Flanged reverb; a cool specialty effect.
	09	EQ/Gate > Hall	Allows you to EQ and gate your console aux send.
	10	In the Room	An all-purpose ambience to add a little air around your instrument(s).
	11	Bloom Canyon	Long decay that builds in intensity.
	12	Dark Verb	A tunable speaker shapes the tone of the reverb component.
	13	Warm Room	Smooth decaying small ambience for guitars, percussion, etc.
	14	Small Space Amb	Very short decay, with obvious early reflections.
	15	Star Gate Reverb	A vintage digital reverb simulation.
	16	Smooth Bloom	A long reverb that slowly builds.
	17	Early & Plate	Short decay, aggressive plate.
	18	Classic 80s Verb	Deep chorusing make this reverb unique.
	19	Cloud Reverb	Creates ever-shifting soundscapes through psycho-acoustic panning.
	20	Town Hall	A small concert hall; try it with piano.
	21	Nice Small Space	Small room tone adds a little air around any track.
	22	Smooth Plate	Long decay; for synths, airy vocals, etc.
	23	Smooth Hall	You're standing on the stage.
	24	Parametric Room	Small room reverb is fed from a parametric EQ. Try other EQ settings.
	25	Pop NonLin Room	Small room texture with a non linear component.
	26	Gated Plate	Plate reverb into a medium length release expander.
	27	Warm Hall	Long smooth decay; Unit A mix parameter offers many tonal variations.
	28	Drum Verb	This one really moves in stereo.
	29	Tight Pop Drums	Feel free to try this on non-pop drums, too.
	30	Drums in Big Rm	Add the sound of Studio 1 to your next mix.
	31	Dark Drum Room	Medium room with a nice, smooth decay.
	32	Compressed Room	A classic engineer's mixdown tool.
	33	Kick Non Lin	Quick decay pop and dance drum sound.
	34	Mega Non Lin	Short decay time; try this one anytime you'd use a room reverb.
	35	Wet Non Lin	Similar to the decay from a gated reverb.
	36	Big Non Lin 1	Pop, dance or techno aggressive reverb.
	37	Big Non Lin 2	A shorter decay time than Big Non Lin 1.
	38	Gated Room 1	Parameter 10 controls the gate threshold.
	39	Gated Room 2	Gate closes slower than in the variation above.
	40	Short Space	Good all-purpose small room ambience. Foot switch toggles two decay times.
	41	Big Rock Dr Rm	You're listening to Studio 'A' of a world famous recording studio.
	42	Rock Toms	Long decay time nonlin and plate.
	43	Percussion Plate	Very quick decay plate.
	44	Rap Boom Room	Extra bottom; try running a loop through this one.
	45	Airplane Hangar	Big reverb blooms slowly; make monstrous rock drum sounds.
	46	Octave Hall Verb	Reverb is pitched up an octave.
	47	Echoing Verbs	Four loud reflections create a novel timbre.
	48	Flanged Reverb 1	Short decay reverb into sweeping flanger.
	49	Flanged Reverb 2	Long decay reverb into flanging; send a percussion track into this one.

Effect Modulation is based on factory default settings.

2-Unit ROM Presets (Bank 1)

Time Based Processing with Reverb	50	Phased Hall	Hall reverb is sent into a slow moving phase shifter.
	51	Feedback Verb	Shows the versatility of the feedback loop routing.
	52	Sample&Phaz Verb	Sample and hold circuit creates a cool special effect.
	53	Spiral Ambience	The signal is pitch shifted and then sent to reverb in a feedback loop.
	54	Coordinates	The flanger is using sample and hold to create this effect.
	55	Jet Reverb	Super-thick flanging applied to reverb.
	56	30th St. Station	That ambience you can only get in a huge open room like a train station.
	57	Parking Garage	The underground sound.
	58	Bend Up Reverb	Reverb is sent to a pitch shifter, bending up, in feedback mode.
	59	Bend Down Reverb	Reverb is sent to a pitch shifter, bending downward, in feedback mode.
	60	Fast Pan Reverb	Source is sent to reverb, which is then panned left to right.
	61	Vocal Magic	Detuning and long delays into reverb.
	62	Chorus & Plate	General purpose chorusing feeding into plate reverb.
	63	Chorus & Room	Try this on guitars or keyboard parts.
	64	Phaser & Plate	For a cool effect, use FtSw 1-L. It's set to extend the decay time 140 seconds.
	65	Phaser & Room	Tight reverb with subtle phaser for keyboards, pop guitar, clav, etc.
	66	Flanger & Plate	Flanger and delays feed into reverb. Great on synths.
	67	Flanger & Room	Slow flanging effect in a tight room.
	68	Delay & Reverb 1	Four multi-taps into plate reverb.
	69	Delay & Reverb 2	Swirling long delays into reverb.
	70	TempoDly & Hall	1/4 note delays. Foot switch tapping will set the tempo for the delays.
	71	TempoDly & Plate	1/4 note delays. Incoming MIDI clocks set the tempo for the delays.
	72	Sparkles & Verbs	Chorus with delays and reverb. For guitars, electric pianos, etc.
	73	Inverse Space	Two reverbs that echo one another.
	74	Backing Vox-Rock	Dense harmonizing into tight reverb.
	75	Parallel Rooms	Two room reverbs in parallel routing create one dense timbre.
	76	Vocal Dbl & Plate	Mono vocal doubling/detuning into plate reverb.
	77	Vocal Non Lin	A pitch shifter spreads out vocals before they're sent to the reverb.
	78	Multi Tap Vocals	Four long multi-taps create a wide stereo image.
	79	Special Sauce	Great lush keyboard texture.
	80	Vocal Swirl	Adds depth to any vocal track with delays and chorusing.
	81	Plate w/ Doubler	Slap back echos great for fattening backing vocals.
	82	Pad Vibe	Slow chorusing reverb for synths and electric piano.
	83	Fat Vocals	Detuned and doubled into a 12.75 second plate.
	84	Big Phased Toms	Special effect; Phased non linear reverb.
	85	Dbl,DDL and Verb	Doubling with long DDL into long reverb.
Time Based Processing	86	Swirling Notch	Fat phase shifter into pitch shifter set to detune the source.
	87	Feedback Phaser	Feedback loop creates unique sci-fi effects on percussive sources.
	88	Laser Flange	Radical flanging with delays.
	89	Phasers on Stun	Super-deep phasing; try guitars, percussion, toms or clavs.
	90	Blazing Phaser	Delays and phasers with sample and hold.
	91	Serial Florus	It's FLangers fed into chORUS, get it?
	92	Phase & Spread	Slow phase shifter feeds into a pitch shifter to spread the sound out.
	93	Panning Delays	1/8 note delays with tempo controlled by incoming MIDI clock into panner.
	94	MultiTap&Flange	Four delay taps fed to a slow flanger.
	95	Ascending Delays	Spacey delay effects that climb forever upward......
	96	Pan-Tapstic	Rich chorus pans in stereo; great for any backing track.
	97	Special Taps	Small room reverb is sent to multi-tap delay.
	98	4 Voice Detune	Two pitch shifter create a fat detune effect.
	99	Old Tape Echo	That old grungy tape echo sound that you used to hate.......

2-Unit RAM Presets (Bank 2)

Time Based Processing	00.	Backing Vox-Lush	A bit of phase shifter makes a cool texture.
	01.	Super Doubler	Use this to fatten background vocals, synths, etc.
	02.	FtSwitchLoop DDL	FtSw 1-L toggles record/playback modes; delay/loop time is set for 2 seconds.
	03.	Air Backgrounds	Panning and chorus designed for vocals.
	04.	Ambient Taps	Tight multi-taps with a small reverb.
	05.	Rhythmic Panner	Delays pan in the stereo field.
	06.	Slap Happy	Slap echos dancing in stereo.
	07.	Chorus-DDL-Pan	Rich chorus sound sweeps across the stereo field.
	08.	Vibrates & Pans	Vibrato that pans in stereo.
	09.	Harmonized Echos	Send single notes into this one for best effect.
	10.	Major Triad Echo	Single notes get made into three note chords.
	11.	Darth	Send a vocal into this to create your own sci-fi voice.
	12.	Detune & Spread	Fattens up keyboards, electric pianos or guitars.
	13.	Slap Station 1	Simulates a classic 8-tap digital delay processor.
	14.	Slap Station 2	A second emulation of a vintage delay processor with longer delay times.
	15.	Ambient Flange	Long delays are slowly flanged.
Amps, Speakers Instruments Effects	16.	Honky Tonk Piano	Play a standard piano through this for an instant bar-room sound.
	17.	RotoSpkr & Hall	Foot switch 1-L will toggle the rotary speed.
	18.	RotoSpkr & Plate	Foot switch 1-L will toggle the rotary speed. Try it on guitar, too.
	19.	Lush El.Piano	A cool R&B treatment for electric pianos or synth.
	20.	Kbd Chorus & Rev	Bright chorus and long reverb.
	21.	R&B El. Piano	A lush pad treatment; great on ballads.
	22.	Sweet El. Piano	Pitch shifted and sent to reverb in a feedback loop.
	23.	Kbd Phaser&Plate	Subtle phaser adds depth to any track.
	24.	E.Pno Phaser&Rev	Deep phaser effect.
	25.	Soft Pad Reverb	Set up for synths; reverb and chorusing.
	26.	Digable Guitar	Fat amp and speaker combo.
	27.	Screamin' Amp	Heavy lead sound.
	28.	Touch Wa Guitar	Auto wah into a distorted amp tone.
	29.	Amp Thru RotoSpk	A favorite 60's guitar sound.
	30.	Vintage Roto Gtr	Cleaner guitar and rotary speaker for rhythm playing.
	31.	Tremolo Amp	Cool rhythm sound with reverb.
	32.	Fretless Bs Solo	Adds a touch of reverb and chorus.
	33.	Dist. Auto-Wah	Crunchy amp tone and auto wah-wah.
	34.	Country Guitar	Slap and tremolo combined.
	35.	Super Mute-ron	Envelope follower effect for guitar or clavs.
	36.	Env Follower &Amp	Stomp box reproduction for funky guitar fills.
	37.	Jangly 60's Gtr	Makes your rhythm 6-string sound like an electric 12.
	38.	Vintage Bass	Warm bass tube amp tone for a retro-60's vibe.
	39.	Gtr Dbl & EQ	Use to fatten rhythm guitar tracks without being muddy.
	40.	Vibrato Amp	Classic amp tone with vibrato circuit.
	41.	Clean Chorus Amp	Chorusing into a dry, clean amp.
	42.	Tube Amp Lite	Lightly crunchy, good for rhythm playing.
	43.	Amp in a Room	Bright clean amp, with a little edge.
	44.	Pop Rhythm Amp	Slightly aggressive big amp rhythm tone.
	45.	GtrDetune&CV Pan	Use the CV pedal to increase the detune amount and to pan the signal.
	46.	CV +1-1 Oct Pedl	The CV pedal shifts the pitch from an octave down to an octave above.
	47.	EQ Chorus Gtr	Super-clean guitar tone with chorus, a classic combination!
	48.	Clean Amp 1	An all-purpose clean amp for guitars, keyboards, etc.
	49.	Clean Amp 2	Fatter clean amp with more bottom.

Effect Modulation is based on factory default settings.

2-Unit ROM Presets (Bank 2)

Amps, Speakers Instruments Effects	50.	Clean Amp 3	Another clean amp variation, with reverb.
	51.	Vibro Amp 1	Vintage practice amp, with medium speed vibrato.
	52.	Vibro Amp 2	A clean, big-speaker vibrato amp.
	53.	Clean Auto-Wah	Input setting is crucial for use with a guitar. Set as high as possible without the peak LED lighting.
	54.	King-Wah / Plexi	Fat wah-wah distorted amp combination.
	55.	Hot Singin Tubes	Texas blues type lead sound.
	56.	Hot Tube Lead	Warm in-yer-face tube sound for lead guitar solos.
	57.	Medium Crunch	Lightly distorted amp tone out in the studio.
	58.	Voxy Lady CV	Wah and amp combo. CV pedal controls wah-wah.
	59.	Ring Mod DDL CV	CV pedal does ring modulation. Foot switch 1-L adds the long delay
	60.	Wah + Echos CV	Deep wah-wah with the CV pedal. Foot switch 1-L toggles delays on/off.
	61.	Cry-Fuz+Echos CV	Crying wah-wah into distortion and plenty of delay.
	62.	Pumpkin Amp	A dynamic close-mic guitar amp.
	63.	Heavy Metal Amp	Crunch guitar tone in a room.
	64.	Vain Helen 90210	Orange phaser into plexi amp. CV pedal adds echo.
	65.	Rocktave Divider	Vintage stomp box reproduction adding slightly distorted lower octave, designed for single note lines.
	66.	Electric Sitar	Convert your guitar into an exotic instrument.
	67.	Metallic Dobro	Makes your guitar brighter and more metallic, like the metal bodied dobro.
	68.	Fuzzy Tube Gtr	Lots of tube amp saturation.
	69.	Fretless	Compression and EQ-Chorus-DDL optimized for fretless bass.
	70.	Clean Brit Amp	A semi-clean amp in a room. A great rhythm tone.
	71.	Comp-Vol+Pitch	CV pedal is set as a volume control; sound is compressed and up an octave.
	72.	Comp-Vol+Vibrato	CV pedal swell volume into a deep vibrato amp sound.
	73.	Touch Wah & Echos	Auto wah with DDL.
	74.	Bridge of Sighs	Classic uni-vibe/amp combo.
	75.	Uni-Wah CV	Subtle phaser with CV pedal wah-wah.
	76.	Room of Blues	Blues guitar tone with dynamic distortion.
	77.	Shred	Dark, fat, heavy metal power chord tone.
	78.	Fast Vibrato Amp	Retro garage band tone.
	79.	Got the Blues	Smooth bright lead sound with a touch of distortion.
	80.	Twang	Slapback echos and a bit of distortion make a great rockabilly tone.
	81.	New Jazzy Amp	Clean, dark amp tone with a bit of chorus.
	82.	Chorus/Pan Amp	Clean amp with chorus and reverb that pans left to right.
	83.	Silver Tone	Small practice amp turned up to '9'. CV pedal functions like a volume pedal.
	84.	Fat 4-12 Amp	Thick, distorted rock tone.
	85.	Trem-O-Amp	Extreme tremolo effect, with a crunchy amp tone. CV controls tremolo speed.
	86.	Spinning Guitar	Pulses with tremolo while spinning around in stereo. CV controls tremolo.
	87.	Fat Flange Amp	Chorus feeds the flanger/amp chain effect.
	88.	Pure Jazz Amp	Just speakers and some reverb. Not just for guitars!
	89.	Bottleneck	Lots of sustain, some distortion and slapback echo.
	90.	Keyboard Amp	Try this to simulate the sound of running keyboards through a small amp.
	91.	70's Keyboard	Phase shifter and a panning speaker cabinet for vintage electric piano tones.
	92.	Chorus & Roto	Suitable for keyboards, organ or guitar; chorus feeds a rotary speaker amp.
	93.	Kbd Phaser w/EQ	Classic stereo electric piano effect.
	94.	Auto Vol & DDL	ADSR generator fades in each note, which is then sent to dual delays.
	95.	Pink Like Floyd	Psychedelic fuzz guitar tone into sweeping phaser and ping pong delays.
	96.	Eleven !	Cranked up amp designed for power chords.
	97.	Xpressive Fusion	ADSR auto volume into crisp lead tone. Ft Sw 1-L toggles two fade in times.
	98.	Guitar Tuner	Setup for guitar or other treble clef instruments.
	99.	Bass Tuner	Setup for bass clef instruments.

Effect Modulation is based on factory default settings.

Config RAM Presets (Bank 1)

One Source Configs	00	Mono Vocal Setup	Use for lead vocals live or when mixing. Compressor and some de-tuning.
	01	Stereo Vox Setup	Stereo input for backing vocals. Compressed and de-tuned.
	02	Mono In Keyboard	Parametric EQ and chorusing. Set up for a mono input from your keyboard.
	03	Stereo In Keybrd	Smooth phaser and chorusing.
	04	Live AcousticGtr	EQ into reverb; try plugging into the front panel Input 1 if you use a pickup.
	05	Rock Gtr Setup	Fat, distorted amp and speaker combo. Plug in to Input 1 on the front panel.
	06	Rhythm Gtr Setup	Full bodied, lightly distorted amp and speaker combination.
	07	Clean Gtr Setup	A clean amp sound with chorus and reverb. Plug into front panel Input 1.
	08	R&B Gtr Setup	A clean amp sound with chorus and reverb. Plug into front panel Input 1.
	09	Bass Setup	Parametric EQ into a compressor for direct recording.
	10	Compresd&Chorusd	3:1 compression into chorus for keyboards or guitars.
	11	Gtr Slap & Chorus	Pop and R&B clean guitar effect. CV pedal alters the chorus blend
	12	Fine Wire Mesh	EQ/Gate into a large, bright, shimmery plate.
	13	Frosty Reverb	EQ/Gate into a large room reverb. Great on toms.
	14	Northern Plain	Huge open spaces. For ambient music.
	15	Diffuse Swell	Slow attack reverb for special effects.
	16	Big, Low Ceiling	Underground parking lot reverb sound.
	17	Spring Reverb	Slightly wobbly old-time reverb.
	18	Springy Slaps	4 fast delays with plenty of regeneration create a cheesy spring reverb effect.
	19	Hall a Looya	Deep chorus, delays and hall for synths, choirs, ethereal effects.
	20	Pan the Halls	Hall reverb pans across the stereo field. CV pedal changes the panning rate.
	21	TakeMeOutToThe..	Voice special effect. You're at the ballpark, listening to the announcer...
	22	B-15 Bass Amp	Simulation of a classic bass amp tone for recording.
	23	Tube Bass Amp	Bass amp with subtle edge.
	24	McPick Bass	For mute style pick bass a la that Liverpool guy.
	25	80's Slap Bass	Adds a touch of DDL for slap solos.
	26	Smooth Bass	Compressed and lightly chorused.
	27	Stick-Man CV-DDL	Progressive rock tone with phasing.
	28	Reggae Bass	Muffled bass tone from the islands. De reel ting!
	29	Octave Bass CV	CV pedal sweeps between lower or upper octave.
	30	Env Filter Bass	A filter that doesn't rob low frequency punch. Input level setting is critical.
	31	Classic 70s Bass	Bright and slightly phase shifted. Try it for jazz.
	32	Z-Box Fat Bass	Punchy bass tone with plenty of compression.
	33	Stereo Grunge CV	Double amps. CV pedal moves from cleaner to more distorted.
	34	Regenerate	Cool sound effect. Thick pitch shifting and phasing with feedback routing.
	35	3-D DDL	Delays that seem to move around your head; great headphone effect.
	36	Chorus & Pan	For direct recording of keyboards. Unit B parameter 03 controls pan rate.
	37	Delays into Comp	Dual delays feed into a 2:1 compressor.
	38	Exciter & DDL	Adjust parameter 01 of Unit A for a more intense effect.
	39	Slap Me Silly	Multiple delays through slow phasing create a unique techno effect.
	40	Cymbal Phaser	A special effect designed for hi-hat or ride cymbal grooves.
	41	TapeflangeVerb	Simulates analog tape machine flanging of 2 reverb sources.
	42	Comb Filtering	Very small delay times create the effect; vary the times to change the filter notches.
	43	Twisted Metal CV	Move the CV pedal while listening to a percussive source; great special effect.
	44	Virtual Dirtbike	No input necessary; sit back and watch the alien dirtbike race!
	45	Get Rappified	Dual speakers add a unique EQ to vocals, loops, etc.
	46	Monster Drum	Killer sound effect. Input 1 level sent to Envelope Follower is critical.
	47	Pitch Down Drums	Non linear reverb through a pitch shifter.
	48	Sick Drums	Warp a drum beat with this unusual timbre.
	49	Science Lab	The DP/2 creates a sound effect for you. No input signal required.

Effect Modulation is based on factory default settings.

Config ROM Presets (Bank 1)

One Source Configs	50	1 Src: Mono In	Selects from the list of 2 Unit presets in RAM and ROM.
	51	1 Src: Stereo In	Selects from the list of 2 Unit presets in RAM and ROM. Stereo input.
	52	Select 1U Preset	Selects 1U presets; kills Unit B.
	53	Vocoder 2U	Send trigger signal to Input 1 and signal to be vocoded to Input 2.
	54	Vocal Remover	The settings of parameters 10 & 11 are critical here.
	55	Vocal EQ & Comp	Parametric EQ and a 6:1 compressor.
	56	Live Vocal Chain	Vocals are compressed and then sent to de-tuner and slap echos.
	57	3.6 Sec Delay 2U	Set for the maximum delay time, 3668 ms.
	58	Pitch Shift 2U	Set for detuning; Foot switch 1-L toggles LFO mod on/off.
	59	Detune Chorus 2U	The 2U pitch shifter is used to create a fat chorus sound.
	60	Warm And Fuzzy	An all-purpose smooth decaying, warm space.
	61	Wonderful Space	Tricky serial routing makes a great percussion space.
	62	EQGateChorPlate	For any source that can be chorused. EQ and gate shape the input.
	63	EQGatePlateChor	For keys, percussion, etc. Gate keeps the input quiet, EQ shapes the reverb.
	64	70's Road Rig	A vintage speaker and phase shifter combination
	65	Bkd Vox Spreader	Stereo in; four voices of de-tune and panning.
	66	8 Panning Delays	Two multi-tap delays pan from left to right.
	67	Analog Heaven	Lush ambient space designed for analog synth textures.
	68	Euro Verb	Reverb with a techno edge. Try it with synths or drums.
	69	Echo Synth Verb	Feedback loop creates a super thick long delay and reverb texture for synths.
	70	Dark Taps	Muted delays; great retro feel.
	71	Jazz Vibe	Smooth space designed for vibes or mellow brass solos.
	72	DynGatedSlapVerb	Play a groove into this one; gated reverb pulses to the beat, triggered by the threshold settings, parameters 08 and 09.
	73	Rock Snare 1	Aggressive tight reverb.
	74	Rock Snare 2	Quick slap delays and non linear reverb.
	75	Kick & Sn Crack!	The ADSR is used to shape the reverb. Input level is very important.
	76	EQ & GatedReverb	Signal is EQ'd then sent to a gated reverb.
	77	Plate&Multi Slap	Try it in serial mode, too; an interesting variation.
	78	Schroeder's Room	Big drum room. Cold, hard, but home.
	79	Non Lin Panverb	Non Linear reverb that moves across the stereo field.
	80	Quick Plate	Small plate for brass and guitar.
	81	Little Big Room	An extra large room with a smooth decay.
	82	Brass Ambience	Fatten up that brass sound with some slap echos and tight reverb.
	83	Neat Place	Classy reverb combo.
	84	Chorus Line	Super thick 16-voice chorus; stereo in.
	85	Diffused Delays	Delays with a unique decay character, perfect for solos.
	86	Modulators	Phaser into flanger. Foot switch 1-L toggles delays on/off.
	87	Crystalline	Adds a pair of de-tuned octaves to thick chorus; for synths, guitars, etc.
	88	Frontiers	Gentle phasing into delays that ping-pong and shift slightly in and out of tune.
	89	Ducker & EQ	Input 2 controls the volume of signals sent to Input 1. Remove trigger sound from mix with parameter 04
	90	FS Stereo ADSR	Foot switch 1-L will trigger the beginning of the ADSR envelope.
	91	MIDI Automation	Mod wheel received on controller channel triggers envelope. Record moves into a sequencer for automation.
	92	Compress&De-ess	Set up for stereo input vocals.
	93	De-Ess & Reverb	Vocals are de-essed and then sent to a plate reverb.
	94	De-Essed Delays	Vocals are de-essed and then sent to dual delays.
	95	Drum Squasher	Very aggressive drum tone; signal is limited after passing through reverb.
	96	KeyedExpander&EQ	Input 2 'keys' the expansion for signals sent to Input 1.
	97	Resonant in E!	Run a percussion sequence through this for a unique experience!
	98	EnvelopeShaperCV	CV pedal moves from a staccato envelope to a longer swell envelope.
	99	Gobi Desert	Dry, dry, dry. Utility effect, use in song chains when you need a bypass effect.

Effect Modulation is based on factory default settings.

Config RAM Presets (Bank 2)

Two Source Configs	00.	Basic Mixdown	Plate reverb and a tempo delay. FtSw 1-L will toggle 2 reverb decay times.
	01.	Drum & Perc Mix	Two room ambiences set up for drums and percussion when mixing.
	02.	Vocal Delays	Long dual delays in Unit A and tight slap echos in Unit B.
	03.	Country Rm&Plate	A: Country Room, great on guitar. B: Country Plate, Classic slap/plate effect.
	04.	Radical Drums	Two different non linear reverbs.
	05.	Bright Lush	Unit A: Bright plate reverb. Unit B: Lush plate-chorus for elec. piano, keys, etc.
	06.	Dark Medium	Unit A: Dark large room reverb. Unit B: Thick plate with deep chorus.
	07.	Percs Keys	Unit A: Tight reverb for percussion. Unit B: Plate with chorus for keyboards.
	08.	Short Lush	Unit A: Short decay room for percussion. Unit B: Plate & chorus for keyboards.
	09.	Reflections	Two different highly reflective spaces.
	10.	A Little Air	Room ambiences that are perfect for adding a little air around an instrument.
	11.	Voc Sweetner 1	Unit A: Subtle detune. Unit B: 1/2 note tempo delay, internal clock 120 BPM.
	12.	Voc Sweetner 2	Unit A: More detuning. Unit B: 1/4 note tempo delay, internal clock 120 BPM.
	13.	Voc Sweetner 3	Unit A: Thick detuning. Unit B: 1/4 note tempo delay, internal clock 120 BPM.
	14.	Voc Sweetner 4	Unit A: Detuning. Unit B: 1/4 note tempo delay, MIDI clock controls tempo.
	15.	Vocal Toys 1	Unit A: Fat vocal detuning. Unit B: Slapback echos.
	16.	Vocal Toys 2	Unit A: Quick slaps spread the vocal out. Unit B: Ping pong delay.
	17.	Fat Bass + Harm	A fattener for synth or electric bass plus a de-tune harmonizer.
	18.	MtdGtrChr+Spac	Chorus for electric guitars plus a small space ambience.
	19.	Mix Tools 1	1/4 note delays and plate reverb for vocals. Internal tempo control.
	20.	Mix Tools 2	1/8 note delays and plate reverb. Internal tempo control.
	21.	Mix Tools 3	1/16 note delays and 3-second plate reverb. Internal tempo control.
	22.	Tiled + Nonlin	Two tight, aggressive spaces.
	23.	Orchestra Spaces	Two large hall reverbs for orchestral sections or soloists.
	24.	Short+Long Halls	Unit A is dark with 3 sec. decay, Unit B is brighter with a 4.5 sec decay.
	25.	2 Tight Rooms	Two variations. Small room ambiences that add a little space around a track.
	26.	2 Boom Rooms	Two variations. Small, live rooms with reflective surfaces.
	27.	80's BD/SN Rev	A tighter reverb for the bass drum with a longer ambience for the snare.
	28.	Slick Dance Revb	Two different ambiences for drums, percussion, etc.
	29.	Short Jazzy Amb	Two tight ambiences for adding a bit of room sound.
	30.	Wide Rooms	Different room tones, all-purpose.
	31.	Piano Chambers	A pair of reverbs that are optimized for piano.
	32.	808 Spaces	Two complementary reverbs for drum machine percussion instruments.
	33.	String Spaces	Two spaces that are designed for strings and orchestral soloists.
	34.	Orch Chambers	Two different reverbs for orchestral instruments.
	35.	Twin Speakers	A pair of speakers; insert them into mixer channels to shape guitar sounds.
	36.	Panner Phaser	Unit A is a stereo panner; Unit B is smooth phase shifter. Try them with synths.
	37.	Flanger Phaser	Unit A is a flanger; Unit B is a phase shifter. Set up as console inserts.
	38.	Roto and Dirt	Rotary speaker in Unit A; Unit B is a distorted amp. Great for adding grunge.
	39.	ReverseVariation	A pair of variation on the reverse reverb algorithm.
	40.	* Available *	Save your own custom Config preset here.
	41.	* Available *	Save your own custom Config preset here.
	42.	* Available *	Save your own custom Config preset here.
	43.	* Available *	Save your own custom Config preset here.
	44.	* Available *	Save your own custom Config preset here.
	45.	* Available *	Save your own custom Config preset here.
	46.	* Available *	Save your own custom Config preset here.
	47.	* Available *	Save your own custom Config preset here.
	48.	* Available *	Save your own custom Config preset here.
	49.	* Available *	Save your own custom Config preset here.

Effect Modulation is based on factory default settings.

Config ROM Presets (Bank 2)

Two Source Configs	50.	2 Src: Stereo Out	Both units are processed separately and then mixed to one stereo output.
	51.	2 Src: Mono Out	Each unit is processed independently. Dual mono; useful as channel inserts.
	52.	Chorus + Delay	Stereo mixed output, setup for sending from console aux sends.
	53.	Slaps + Plate	Long plate reverb and slapback delay.
	54.	Vocal Detuners	Two different stereo detune settings for lead or backing vocals.
	55.	Pair of Plates	Diffuse plate with nice depth & a large plate with exceptionally smooth decay.
	56.	2 Digital Halls	Clear, bright, aggressive vintage digital verb simulations.
	57.	Explosive Halls	Two variations. Very slow attack with thunderous buildup.
	58.	Drum Rooms 1	A pair of small spaces; send 2 signals from your console to Inputs 1 and 2.
	59.	Drum Rooms 2	Two more small spaces; send 2 signals from your console to Inputs 1 and 2.
	60.	Dry Kick & Snare	Tight mic'ed kick sound along with a snare reverb.
	61.	Wet Kick & Snare	Big rock kick drum and snare drum reverbs.
	62.	2 Vintage Rooms	Small and large vintage digital reverb simulations.
	63.	Tight + Wet	Chorus and slapback echo in Unit A; Tight plate in Unit B.
	64.	Echos and Hall	Set up in mono for channel inserts. Delays in Unit A, Hall reverb in Unit B.
	65.	2 Bright Plates	Small and large. Mixed stereo output.
	66.	2 Smooth Plates	Small and large. Mixed stereo output.
	67.	2 Country Verbs	Small and large; for guitar, vocal.
	68.	2 Monaural Rooms	2 different small ambiences.
	69.	Mono Plates	Two different mono plate reverbs, long and short.
	70.	2 MIDI ClockDDLs	MIDI clocks appearing at the MIDI IN jack will set the tempo.
	71.	2 MONO Delays	Dual mono mode is used here; useful live or as a channel insert when mixing.
	72.	2 Slapback DDLs	Dual mono mode; setup for channel inserts.
	73.	2 Tempo Delays	Internal clock. Dual mono mode; Unit A is 1/4 notes, Unit B is 1/8 notes.
	74.	2 FS Tap Delays	Use Foot switch 1-L to tap the tempo. Unit A is 1/4 notes; Unit B is 1/8 notes.
	75.	2 Dual Delays	Each unit has different delay times.
	76.	2 MultiTapDelays	Each unit has a different set of four delay times.
	77.	2 Mono Choruses	Use as insert effects when mixing.
	78.	Phaser Flanger	Phaser in Unit A; Flanger in Unit B. Use as insert effects.
	79.	Mono Slap Echos	Two sets of slap echos; Unit B's delays are longer than Unit A's.
	80.	Double Chorus	Two stereo chorus units; try blending some of each for electric piano.
	81.	Phaser Chorus	Phaser in A, Chorus in B for electric keyboards. Stereo output.
	82.	Dbl&Slap-Chorus	Doubling and slap echos in A; Chorus in B.
	83.	Organ Roto Spkrs	Beef up wimpy organ sounds. B is more distorted. Use as console inserts.
	84.	Kick, Snare Comp	Two compressors for drums; use as console inserts.
	85.	EQ'd Drum Comp	Two identical compressors, with EQ, for controlling live snares and toms.
	86.	Mixdown Warmer	Tube saturation emulation. Two identical settings.
	87.	2 Tube Thickners	Insert across mixer channel for warmth.
	88.	Dual EQ&TiteGate	Each channel goes through its own gate and EQ.
	89.	Album EQ w/Gates	EQ with a slight bass and treble boost.
	90.	Stereo Telephone	Extreme EQ. Two identical settings.
	91.	2 Limiters	Two identical settings. Dual mono mode; setup for channel inserts.
	92.	Dual Exciters	Adjust parameters 03 and 04 to suit the source material.
	93.	Dual Expanders	Two identical settings. Use as console inserts when mixing.
	94.	2 Inv. Expanders	Two 15:1 ratio inverse expanders.
	95.	Dual Cmp & Gate	2:1 compression on each channel.
	96.	2 Noise Gates	Parameters 09 and 10 adjust the thresholds.
	97.	2 Fast Expanders	Two 1:30 ratio expanders. Use as console inserts.
	98.	Dual Rumble Fltr	Filters frequencies below 150 Hz on each channel.
	99.	2 Parametric EQs	No boost or cut is applied; try your own settings.

Effect Modulation is based on factory default settings.

Appendix

DP/2 MIDI Implementation

The DP/2 features extensive MIDI (Musical Instrument Digital Interface) implementation. For normal applications, you will find all the information you need regarding the DP/2's MIDI functions in this manual. You can also refer to the MIDI Implementation Chart on the next page for a summary of the DP/2 MIDI implementation.

If you are writing a computer program to communicate with the DP/2 via MIDI, or otherwise require a copy of the full DP/2 MIDI System Exclusive Specification, it is available free of charge by writing to:

ENSONIQ Corp
MIDI Specification Desk
Box 3035
155 Great Valley Parkway
Malvern, PA 19355-0735
USA

Include in your written request your name and address, and indicate that you would like a copy of the "DP/2 MIDI System Exclusive Specification." Please allow 2 to 3 weeks for delivery.

MODEL: DP/2 **MIDI Implementation Chart** **Version: 1.0**

Function...		Transmitted	Recognized	Remarks
Basic Channel	Default Channels	1, 2, 3, 4 * 1-16	1, 2, 3, 4 * 1-16	
Mode	Default Messages Altered	MULTI X X	MULTI X X	
Note Number	True Voice	X	0-127	Modulation Source
Velocity	Note ON Note OFF	X X	O X	Modulation Source
After Touch	Key Channel	X X	O O	Modulation Source
Pitch Bender		X	O	Modulation Source
Control Change		0 Bank Select MSB (always 0) 4 (CV Pedal) 32 Bank Select LSB (values of 0 & 1 only) 70 (Foot Switch 1-L) 71 (Foot Switch 1-R) 72 (Foot Switch 2-L) 73 (Foot Switch 2-R)	0-127	If controller 7 is received on Control channel= Modulation Source. If controller 7 is received on Unit channel= unit volume control. Controller 32 (Bank Select LSB) values of 0 and 1 received select preset banks 1 and 2 when program change maps are off.
Program Change	True Number	0-99	0-99 (map off) 0-127 (map on)	Program changes sent & received on Unit and Config channels
System Exclusive		O	O	
System Common	: Song Pos : Song Sel : Tune	X X X	X X X	
System Real Time	: Clocks : Commands	X X	O X	For tempo sync delays
Aux. Messages	: Local On/Off : All Notes Off : Active Sense : Reset	X X X X	X O[1] X X	[1] Can be assigned to function as a bypass or DP/2 controller.

Notes: * The DP/2 can receive on up to 4 MIDI channels for units A, B, Config and controllers. They may overlap in any way, except unit channels and the config channel have to be different. All modulation sources are received on the control channel.

O= YES
X = NO

Glossary

There are a few terms that you may be unfamiliar with that should be understood before you can unlock the DP/2's full potential as a programmable effects processor. This section defines these terms.

Algorithm An algorithm is a control program for the ESP chips. The ESP chips are digital signal processors that, when programmed, provide the basic sonic building blocks in the DP/2). The word "effect" could be used instead of algorithm, but some algorithms can produce several sonic effects simultaneously. Each algorithm has a set of parameters that control the effect(s) it produces. The values of these parameters are saved with the algorithm in *presets*. Each DP/2 algorithm has a three letter abbreviation which helps to identify it in Select mode. The DP/2 algorithms are:

Algorithm:	abbreviation:	Algorithm:	abbreviation:	Algorithm:	abbreviation:
3.6 sec Delay 2U	ddl	EQ-Vibrato-DDL	vib	Phaser-Reverb	pha
8 Voice Chorus	cho	Expander	exp	Pitch Shift 2U	pit
ADSR Env Gen	env	FastPitchShift	pit	Pitch Shift-DDL	pit
Chorus-Reverb	cho	Flanger	fla	Pitch Shifter	pit
CmprDistFlngRev	chn	Flanger-Reverb	fla	Plate-Chorus	rev
De-esser	ess	Fuzz Box	dst	Reverse Reverb1, 2	rev
DigitalTubeAmp	amp	Gated Reverb	rev	Rotating Spkr	rot
Dist-Cho-Revrb	chn	Guitar Amp 1, 2, 3, 4	amp	Rumble Filter	flt
Dist-Roto-Revb	chn	GuitarTuner2U	tun	Sine/Noise Gen	gen
Dual Delay	ddl	Hall Reverb	rev	Small Plate	rev
Ducker / Gate	gat	InversExpander	exp	Small Room Rev	rev
DynamicTubeAmp	amp	Keyed Expander	key	Speaker Cabinet	spk
EQ-Chorus-DDL	cho	Large Plate	rev	Tempo Delay	ddl
EQ-Compressor	cmp	Large Room Rev	rev	Tunable Spkr 1, 2	spk
EQ-DDL-withLFO	ddl	MultiTap Delay	ddl	VandrPolFilter	flt
EQ-Flanger-DDL	fla	No Effect (Bypass Preset)	dry	VCF-Distort 1, 2	dst
EQ-Gate	equ	NonLin Reverb 1, 2, 3	rev	Vocal Remover	flt
EQ-Panner-DDL	pan	Parametric EQ	equ	Vocoder 2U	voc
EQ-Tremolo-DDL	trm	Phaser - DDL	pha	Wah-Dist-Revrb	chn

Amplify To increase the level or loudness of a signal.

Amplitude The level or loudness of a signal.

Attenuate The process of lowering the level or loudness of a signal.

Balanced-Line Input Three-conductor balanced lines are used to connect various pieces of equipment together, and are often used in professional studios. These balanced-line inputs tend to reject hum and/or radio frequency interference. The DP/2 offers balanced-line inputs and outputs, for connecting with professional studio equipment.

Bandwidth Bandwidth refers to the frequency range of signals that are passed.

Bypass In the DP/2, bypass means that the signal will "go around" that particular unit, and the signal will not be affected by that unit's algorithm.

Bypass Units The last parameter(s) of each Config Preset gives you two choices of how to mute effects. When set to "bypass," pressing two times on a Unit button (red LED lit) will cause the effect processing to be temporarily silenced for that Unit, so all you hear is the dry source signal. It bypasses the algorithm/preset. "Kill" is the other choice (see Kill).

Chorusing An audio effect that takes place when a source signal is pitch modulated and mixed with the original source signal. These modulations create phasing characteristics that make the source signal sound wider/bigger. Usually delays of 10-30 milliseconds will create pleasant choruses.

Compression Ratio The amount by which a signal is compressed. For example, a 4 to 1 compression ratio means that signals above the threshold will cause an increase of 1dB in output level for every 4dB increase in input level. At high ratios (like 20:1 and above), the compressor acts as a limiter.

Compressor A signal conditioning process that reduces a source signal's dynamic range. Loud signals get softer and softer signals get louder.

Config A Config (short for CONFIGuration) controls how the DP/2 handles signals by determining the number of input sources to be processed, how they are to be interconnected, and where the outputs will appear.

There are several uses of the term Config and it is important to understand the distinctions between them. Some terms used in these definitions may not yet be familiar, but they are described later in this glossary.

Config(uration) — This general term refers to the current signal routing arrangement that the system is using. It includes all routing parameters.

Config Parameter — Any one of the parameters which appear in Edit mode when the Config LED is on.

Input Config — The Config parameter which controls how many input signals are to be processed by the DP/2 (equivalent to *Source Config*).

Config Preset — This the largest type of preset in the DP/2. It contains all the signal routing information, such as which inputs are routed to which outputs, whether the units are run in serial, parallel, or feedback, and which units are bypassed. Additionally, a Config Preset loads each of the units with an effect and its associated parameters.

☞ **Important:** Setting up the correct Config is the most important action when using the DP/2. The Config controls how the system operates in many important ways. It is very important to understand this concept clearly in order to avoid later confusion. Please refer to *Section 3 — Config Parameters* for more details on this essential concept.

Damping A parameter in the DP/2 that allows control of high frequency decay in reverb algorithms. You can use damping to customize the perceived size and ambience of an environment (making it wetter/drier or brighter/darker).

The term "damping" is derived from the German word "damphen," which means deaden, muffle, mute.

A bit of trivia: In the early film-making days, when actors needed to read newspapers (or any paper-based document), the turning of pages caused so much noise in the film, that a solution was required to eliminate the noise. Soundmen discovered that if the paper was wet (with water), the crinkling noise of the pages was gone. Although this process is not required in today's film-making standards, it is another example of early "damping" techniques.

De-esser A specialized algorithm that reduces the level of sibilance in a source signal through selective high frequency compression. This sibilance is usually heard as an "s" sound in speech, hence the name De-esser.

Diffusion A parameter commonly found in reverbs that is used to smear the transients, so as to diffuse and smooth the sound. Low diffusion values will cause impulsive sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding).

Digital Delay Line (DDL) An algorithm that causes source signals to be moved later in time relative to the original signal. These "delayed" signals are used to create a myriad of audio effects, such as echo and reverb.

Dual Mono A term used in the DP/2 to describe one signal routing option. Two inputs are treated as separate mono signals rather than as a stereo pair. This option processes the two input sources as two discrete mono outputs. A useful option when more individual effects are needed.

Early Reflections Early reflections are delayed signals that determine how we localize and perceive size of ambient spaces. In the case of a room, where the signal is bounced off all surfaces (walls/ceiling/floor), the perception of the summation of these delayed signals creates what we term ambience. In the DP/2, you can control these delays to create various environments.

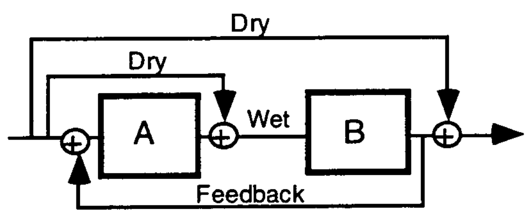
Echo A delay that is perceived as a discrete repeat of the original sound. A classic example of an echo is the effect of shouting into a canyon. You will hear your voice delayed and repeated throughout the canyon. Generally, echoes are created by long delay times.

Expander An algorithm that increases the dynamic range of a source signal by making loud signals louder and soft signals softer. Expansion can be used to lower noise on poorly recorded tracks, or to help control leakage while recording. Signals below threshold are attenuated, signals above threshold are passed with a controllable fixed gain.

Equalization The process of altering the frequency response (tone) of a signal (also called "EQ").

Feedback A signal routing in which the output of an effect is mixed back into the input. Feedback of a delay line is also called regeneration.

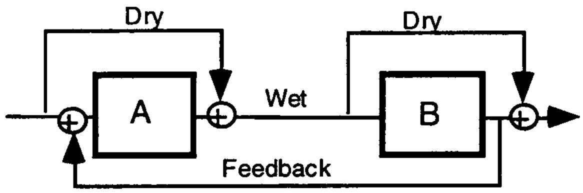
Feedback 1 A two-unit signal routing option in the DP/2 as shown below. The A and B units are in series; the output of the B unit is mixed back into the input of the A unit:



Feedback 1

The feedback amount is available among the config parameters. For example, if A were a delay and B were an EQ, the feedback path would cause the delay to regenerate with the EQ in the regeneration path. In Feedback 1, the wet/dry mix of the B unit combines the dry input to A with the output of B. When set to full dry (0) only the dry input signal is heard.

Feedback 2 A two-unit signal routing option in the DP/2 as shown below. The A and B units are in series; the output of the B unit is mixed back into the input of the A unit:



Feedback 2

The feedback amount is available among the config parameters. In Feedback 2, the wet/dry mix of the B unit combines the output of A with the output of B. When B is set to full dry (0), the output mix of A is heard.

Filter A device that attenuates selected frequencies. For example, a high-pass filter passes all signals higher than a selected frequency, attenuating all those frequencies below it. A low-pass filter passes all signals below a selected frequency, attenuating all those frequencies above it.

Flanger A processor that simulates the effect of two synchronized tape machines in playback of the same signal, where one machine's speed is varied by pressing on the "flange" of the tape reel. The small delay causes a phasing cancellation that produces a comb filter. Changing the delay time causes the "flange" effect. In the DP/2, flanging is achieved using interpolated digital delay lines.

Gate (Noise Gate) A device that completely attenuates a source signal that falls below a pre-determined threshold. A useful tool in eliminating noise, as well as controlling effected signals.

Global Means that it affects all things involved. For example, a global parameter would function on a system-wide basis.

Hysteresis The property of a system whose behaviour is determined by the level, direction, and history of a controlling signal. Used in the DP/2 to provide greater control over gating, triggering, and compression algorithms.

Input Source The signal that is fed into the DP/2 via a balanced/unbalanced cable. It is the signal that gets processed or which controls a side-chain/key.

Inverse Expander An algorithm that forces signals below a control threshold to be raised to that threshold, while signals above that threshold are passed with a controllable fixed gain. This helps create a more even signal. This is more accurately termed upward expansion.

Keyed Expander An expander whose effect is determined by a control signal, as opposed to the input signal. This control signal goes through an EQ side-chain. When the EQ'd control signal meets the requirements for expansion, the expander becomes active. This effect is often used to improve rhythm guitar or drum tracks.

Kill The last parameter of a Config Preset gives you two choices of how to mute effects. When set to "kill," pressing two times on a Unit button (red LED lit) will cause the effect processing and dry signal for that unit to be temporarily silenced so that you hear nothing. "Bypass" is the other choice (see Bypass).

LED LEDs (Light Emitting Diodes) are little lamps that are solid-state devices, and are not like conventional light bulbs. Under normal conditions, they will not burn out, and have a virtually unlimited lifetime.

LFO An LFO (Low Frequency Oscillator) generates very low frequency waves, below the audio spectrum, which can be used to control vibrato, tremolo, and many other effects.

Limiter A device that will prevent a source signal from going above a pre-set level (threshold). A limiter can be thought of as a compressor with an infinite compression ratio.

MIDI Musical Instrument Digital Interface. A communication protocol for musical instruments. MIDI has expanded the ability of the electronic musician by allowing control, editing, and manipulation of products from different manufacturers through a single communication protocol/network.

Mixed Stereo An output routing option of the DP/2 that allows two separate stereo output signals to be digitally combined into a single stereo output. The levels of the two signals are controllable in the DP/2.

Modulation The term used to described a real-time change to a source signal or algorithm parameter. Modulation can be introduced within an algorithm, via MIDI events, or by using external input devices such as the CVP-1 pedal. An important feature in creating new and evocative sounds.

Multi-Effect Algorithm An algorithm that contains more than one effect type. For example, EQ-Chorus-DDL.

Oscillator An oscillator is a device that emits a continuous signal of some kind. The frequency of this signal is measured by the number of cycles that occur in a single second (cycles per second is the same as "Hz," or "Hertz").

Parallel Processing A system with multiple processors working simultaneously to achieve greater speed, efficiency, and reliability. In the DP/2, four units are available to work in parallel, possibly running different algorithms, and perhaps different input sources.

Parameter Any setting of the DP/2 which can be changed or modified is called a parameter. The DP/2 uses a multi-function panel through which parameters of many different types can be selected and controlled. There are four basic types of parameters:

Algorithm parameters	System and MIDI parameters for each unit
Config parameters	System (Global) parameters

Parameters are available to be edited in Edit mode (for Algorithm and Config parameters) and System/MIDI mode (for System and MIDI parameters).

In these two modes, you use the ◁ and ▷ buttons to scroll to the parameter you want to modify, then use the **Data Entry Knob** to change the parameter's value.

Parametric EQ An algorithm that raises or lowers specified frequency regions in program material. A parametric EQ has variable center frequency, gain, and "Q" - the ratio of center frequency to bandwidth.

Phaser Originally conceived as an approximation to the flange effect. Allpass filters are used in place of the delay lines. Allpass filters introduce delay by modifying signal phase, hence the name.

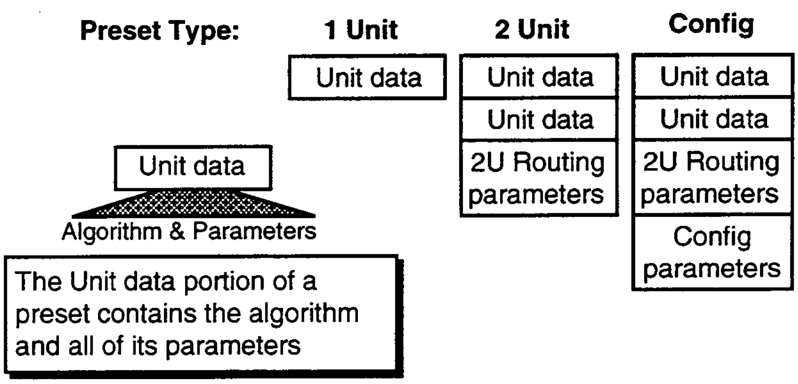
Pre-emphasis This is a noise reduction technique commonly used to control noise in tape recorders. Hiss is most objectionable in the higher frequencies of a signal. Pre-emphasis boosts the treble going into the effect, and a de-emphasis circuit cuts the treble (by an equal negative amount) to its original signal — while at the same time removing hiss from the frequencies where it is most objectionable.

Preset A preset is a combination of an algorithm (or algorithms) and the associated parameter settings. You select a preset to put different effects into the Units. Selecting a preset may load effects into Units A and B. Presets affecting more than one unit also contain signal routing information.

There are four types of presets in the DP/2. They differ in how many units are affected, and how many routing parameters are remembered. The type(s) of presets available for selecting depends on the current Config. The three preset types are:

	Preset Type:	Affects:	Routing Parameters remembered:
(1U)	1 Unit Preset	One Unit	None
(2U)	2 Unit Preset	Two Units	Connections between 2 units
(Config)	Config Preset	Two Units	All routing and configuration params

As the number of Units in a preset increases there are more routing parameters included.



There are 600 presets in the DP/2; 200 presets (storage locations) for each type of preset. The first 50 presets in each Bank (00 to 49, and 00. to 49.) are user programmable (battery backed up RAM). Presets 50 to 99 and 50. to 99. are ROM factory presets:

		Preset Type		
		1 Unit	2 Unit	Config
Preset Number	00 to 49	50 1-Unit RAM Presets (Bank 1)	50 2-Unit RAM Presets (Bank 1)	50 Config RAM Presets (Bank 1)
	50 to 99	50 1-Unit ROM Presets (Bank 1)	50 2-Unit ROM Presets (Bank 1)	50 Config ROM Presets (Bank 1)
	00. to 49.	50 1-Unit RAM Presets (Bank 2)	50 2-Unit RAM Presets (Bank 2)	50 Config RAM Presets (Bank 2)
	50. to 99.	50 1-Unit ROM Presets (Bank 2)	50 2-Unit ROM Presets (Bank 2)	50 Config ROM Presets (Bank 2)

The RAM presets are loaded with factory defaults when the unit is shipped and whenever it is reinitialized. These default presets may be recalled into RAM at any time using a special command described in the System/MIDI section.

Q Another term for resonance. In the DP/2, this is a bandwidth control that determines the width of the resonant peak at the center of the frequency band. This is equal to the cutoff frequency divided by the bandwidth. By raising the Q, you can produce a narrower bandwidth.

Regeneration A signal routing in which some of the output is mixed back into the input. The feedback of a delay line is also called regeneration.

Reverb Multiple echoes and reflections that combine to create an ambient effect. Different devices have been used to simulate these ambiences: springs, plates, tubes, and chambers. The DP/2 uses digital algorithms to create new environments and simulate these classic ambiences.

Rumble Filter An algorithm that attenuates very low frequencies. In the DP/2, the Rumble Filter is created by cascading four first order high pass filters. Originally conceived to eliminate noise in turntables.

Sample and Hold A device that captures a signal and stores it for brief periods of time. In the DP/2, Sample and Hold is often available on the output of the LFO in modulation-type effects, allowing smoothly-swept effects to sweep in a random, chaotic manner, creating interesting sonic effects.

Source Config In Edit mode, this is the Config parameter which controls how many input signals are to be processed by the DP/2 (equivalent to *Input Config*).

Transient A signal that is very short, like the attack of a pick on a guitar string, or the sound of a drumstick hitting a rim. These "transients" are difficult to reproduce, and the ability of a device to respond to these sounds is called the "transient response."

Unbalanced Jack An input jack that has two wires. One carries the positive (+) signal, the other the negative (-) signal and is attached to the ground.

Unit The two independent effects processors in the DP/2 are called Units, and are referred to as A and B. Normally, each of the Units is loaded with a different algorithm, but in some cases both units are combined to create one complex effect, such as the vocoder.

Vandr Pol Filter An algorithm that adds synthetic high harmonics to the input signal, usually brightening the sound. Van Der Pol originally developed the theory for this mathematical model in his study of oscillations caused by the non-linearities in vacuum tube circuits.

VCF-Distortion Voltage Controlled Filter and distortion. Useful for creating Distortion, Wah Wah, and Envelope (Auto) Wah effects.

XLR Connector A type of connector that has three pins. Pin 1 is the ground reference, pin 2 carries the "hot" signal, and pin 3 carries the anti-phase "cold" signal. It is designed to mate with balanced inputs and outputs.

DP/2 Algorithm Parameters
(alphabetical order)

3.6 sec DDL 2U
01 — Mix
02 — Volume
03 — 3.6 Sec Delay Time
04 — 3.6 Sec Delay Regen
05 — 3.6 Sec Delay Pan
06 — 3.6 Sec Delay Regen Damping
07 — 3.6 sec Delay Mode
08 — DelaySet
09 — Mod1 Source
10 — Mod1 Destination Parameter
11 — Mod1 Param Range Min
12 — Mod1 Param Range Max
13 — Mod2 Source
14 — Mod2 Destination Parameter
15 — Mod2 Param Range Min
16 — Mod2 Param Range Max

8 Voice Chorus
01 — Mix
02 — Volume
03 — 8V Chorus LFO Rate
04 — 8V Chorus LFO Width
05 — 8V Chorus Stereo Spread
06 — 8V Chorus Regen
07 — 8V Chorus Left Regen Time
08 — 8V Chorus Right Regen Time
09 — 8V Chorus Delay Regen
10 — Mod1 Source
11 — Mod1 Destination Parameter
12 — Mod1 Param Range Min
13 — Mod1 Param Range Max
14 — Mod2 Source
15 — Mod2 Destination Parameter
16 — Mod2 Param Range Min
17 — Mod2 Param Range Max

ADSR Env Gen
01 — Mix
02 — Volume
03 — Level Detector Off Below
04 — Level Detector On Above
05 — Level Detector Attack
06 — Level Detector Release
07 — Expnd Ratio
08 — Threshold
09 — Exp Attack
10 — Release
11 — Gate Mode
12 — Toggle
13 — A
14 — D
15 — S
16 — R
17 — Mod1 Source
18 — Mod1 Destination
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

Chorus-Reverb
01 — Mix
02 — Volume
03 — Reverb Mix
04 — Chorus LFO Rate
05 — Chorus LFO Width
06 — Chorus Center
07 — Chorus Feedback
08 — Chorus Mix
09 — Large Plate Decay
10 — Plate Predelay Time
11 — Large Plate HF Damping
12 — Large Plate HF Bandwidth
13 — Plate Diffsn1
14 — Diffusion2
15 — Plate Decay Definition
16 — Mod1 Source
17 — Mod1 Destination
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

CmprDstFlngRev
01 — Mix
02 — Volume
03 — Reverb Mix
04 — Compressor Threshold
05 — Comp Attack
06 — Comp Release
07 — Distortion Level In
08 — Distortion Level Out
09 — HighPass Fc
10 — LowPass Fc
11 — Amp Feedback Amount
12 — Flanger LFO Rate
13 — Flanger LFO Width
14 — Flanger Center
15 — Flanger Feedback
16 — Flanger Mix
17 — Reverb Decay
18 — Reverb HF Damping
19 — Mod1 Source
20 — Mod1 Destination
21 — Mod1 Param Range Min
22 — Mod1 Param Range Max
23 — Mod2 Source
24 — Mod2 Destination
25 — Mod2 Param Range Min
26 — Mod2 Param Range Max

De-esser
01 — Mix
02 — Volume
03 — De-esser Output Gain
04 — Comp Ratio
05 — Threshold
06 — Gain Change
07 — Comp Attack
08 — Comp Release
09 — Noise Gate Off Below
10 — Noise Gate On Above
11 — Sidechain EQ HighPass Fc
12 — Bass Fc
13 — Bass Gain (loShv)
14 — Mid1 Fc
15 — Mid1 Gain
16 — Mid1 Q
17 — Mid2 Fc
18 — Mid2 Gain
19 — Mid2 Q
20 — Treble Fc
21 — Treble Gain (HiShv)
22 — Sidechain EQ Input Trim
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

DigitalTubeAmp
01 — Mix
02 — Volume
03 — Pre-EQHighPass Cutoff
04 — PreEQ PreAmp Gain
05 — Pre-EQ1 Fc
06 — Pre-EQ1 Gain
07 — Pre-EQ1 Q
08 — Pre-EQ2 Fc
09 — Pre-EQ2 Gain
10 — Pre-EQ2 Q
11 — Pre-EQ3 Fc
12 — Pre-EQ3 Gain
13 — Pre-EQ3 Q
14 — Amp Drive Gain
15 — Amp Level Detect Attack
16 — Amp Level Detect Release
17 — Amp Waveshaper Onset Level
18 — Amp Waveshaper First Table
19 — Amp Waveshaper Last Table
20 — Amp Waveshaper Table Slope
21 — Amp Tube Bias
22 — Amp Output Level
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

Dist-Cho-Revrb
01 — Mix
02 — Volume
03 — Reverb Mix
04 — Pre-Distortion VCF Fc
05 — Pre-Distortion VCF Q
06 — Distortion Level In
07 — Distortion Level Out
08 — Distortion Mix
09 — Amp Feedback Amount
10 — Post-Distortion VCF Fc
11 — Post-Distortion VCF Q
12 — Chorus LFO Rate
13 — Width
14 — Chorus Center
15 — Chorus Mix
16 — Large Plate Decay
17 — Plate Predelay Time
18 — Large Plate HF Damping
19 — Large Plate HF Bandwidth
20 — Plate Diffsn1
21 — Diffusion2
22 — Plate Decay Definition
23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

Dist-Roto-Revb
01 — Mix
02 — Volume
03 — Reverb Mix
04 — Pre-Distortion LowPass Fc
05 — Distortion Level In
06 — Distortion Level Out
07 — Post-Distortion VCF Fc
08 — Post-Distortion VCF Q
09 — Distortion Mix
10 — Rotor Speed
11 — Slow
12 — Fast
13 — Rotating Speaker Inertia
14 — Tremolo Depth Slow
15 — Tremolo Depth Fast
16 — Vibrato Depth Slow
17 — Vibrato Depth Fast
18 — Rotating Speaker Mix
19 — Rotating Speaker Stereo Spread
20 — Reverb Decay
21 — Reverb HF Damping
22 — Mod1 Source
23 — Mod1 Destination
24 — Mod1 Param Range Min
25 — Mod1 Param Range Max
26 — Mod2 Source
27 — Mod2 Destination
28 — Mod2 Param Range Min
29 — Mod2 Param Range Max

Dual Delay
01 — Mix
02 — Volume
03 — Left Input Delay Time
04 — Left Input Delay Time (fine)
05 — Left Input Delay Regen
06 — Left Input Delay Pan
07 — Right Input Delay Time
08 — Right Input Delay Time (fine)
09 — Right Input Delay Regen
10 — Right Input Delay Pan
11 — Dual Delay CrossRegen
12 — Dual Delay Regen Damping
13 — Mod1 Source
14 — Mod1 Destination Parameter
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination Parameter
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

Ducker / Gate
01 — Mix
02 — Volume
03 — Ducker Output Gain
04 — Ducker Output Mix
05 — Comp Ratio
06 — Threshold
07 — Gain Change
08 — Comp Attack
09 — Comp Release
10 — Noise Gate Off Below
11 — Noise Gate On Above
12 — Bass Fc
13 — Bass Gain (loShv)
14 — Mid1 Fc
15 — Mid1 Gain
16 — Mid1 Q
17 — Mid2 Fc
18 — Mid2 Gain
19 — Mid2 Q
20 — Treble Fc
21 — Treble Gain (HiShv)
22 — Side Chain EQ Input Trim
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

DynamicTubeAmp
01 — Mix
02 — Volume
03 — Pre-EQHighPass Cutoff
04 — PreEQ PreAmp Gain
05 — Pre-EQ1 Fc
06 — Pre-EQ1 Gain
07 — Pre-EQ1 Q
08 — Pre-EQ2 Fc
09 — Pre-EQ2 Gain
10 — Pre-EQ2 Q
11 — Pre-EQ3 Fc
12 — Pre-EQ3 Gain
13 — Pre-EQ3 Q
14 — Amp Drive Gain
15 — Amp Level Detect Attack
16 — Amp Level Detect Release
17 — Amp Waveshaper Onset Level
18 — Amp Waveshaper First Table
19 — Amp Waveshaper Last Table
20 — Amp Waveshaper Table Slope
21 — Amp Tube Bias
22 — Amp Output Level
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

EQ-Chorus-DDL
01 — Mix
02 — Volume
03 — Chorus LFO Rate
04 — Chorus LFO Width
05 — Chorus Center
06 — Left/Right LFO
07 — Chorus Left Delay Time
08 — Chorus Right Delay Time
09 — Chorus Delay Regen
10 — Chorus Left Echo Time
11 — Chorus Right Echo Time
12 — Chorus Echo Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

EQ-Compressor
01 — Mix
02 — Volume
03 — Compressor Gain
04 — Compressor Ratio
05 — Compressor Threshold
06 — GainChange
07 — Comp Attack
08 — Comp Release
09 — Comp Noise Gate Off Below
10 — Comp Noise Gate On Above
11 — Gate Release Time
12 — Bass Fc
13 — Bass EQ Gain
14 — Treble Fc
15 — Treble EQ Gain
16 — EQ Input Level Trim
17 — Mod1 Source
18 — Mod1 Destination Parameter
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination Parameter
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

EQ-DDL-withLFO
01 — Mix
02 — Volume
03 — DDL+LFO Left Delay Time
04 — DDL+LFO Right Delay Time
05 — DDL+LFO LFO Rate
06 — DDL+LFO LFO Width
07 — Left/Right LFO
08 — DDL+LFO Delay Regen
09 — DDL+LFO Delay Cross Regen
10 — DDL+LFO Regen Damping
11 — DDL+LFO Right Delay Input
12 — DDL+LFO Right Output Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

EQ-Flanger-DDL
01 — Mix
02 — Volume
03 — Flanger LFO Rate
04 — Flanger LFO Width
05 — Flanger Center
06 — Flanger Feedback
07 — Flanger Notch Depth
08 — Left/Right LFO
09 — Flanger Sample & Hold Rate
10 — Flanger Left Delay Time
11 — Flanger Right Delay Time
12 — Flanger Delay Feedback
13 — Flanger Left Echo Time
14 — Flanger Right Echo Time
15 — Flanger Echo Level
16 — Bass Fc
17 — EQ Gain
18 — Treble Fc
19 — EQ Gain
20 — EQ Input Level Trim
21 — Mod1 Source
22 — Mod1 Destination Parameter
23 — Mod1 Param Range Min
24 — Mod1 Param Range Max
25 — Mod2 Source
26 — Mod2 Destination Parameter
27 — Mod2 Param Range Min
28 — Mod2 Param Range Max

EQ-Gate
01 — Mix
02 — Volume
03 — Bass Fc
04 — Bass Gain (loShv)
05 — Mid1 Fc
06 — Mid1 Gain
07 — Mid1 Q
08 — Treble Fc
09 — Treble Gain (HiShv)
10 — EQ Input Level Attenuation
11 — Noise Gate Off Below
12 — Noise Gate On Above
13 — Gain Change
14 — Gate Release Time
15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

EQ-Panner-DDL
01 — Mix
02 — Volume
03 — Panner Rate
04 — Panner Width
05 — Left/Right LFO
06 — Panner Sample & Hold Rate
07 — Panner Left Delay Time
08 — Panner Right Delay Time
09 — Panner Delay Regen
10 — Panner Left Echo Time
11 — Panner Right Echo Time
12 — Panner Echo Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

EQ-Tremolo-DDL
01 — Mix
02 — Volume
03 — Tremolo Rate
04 — Tremolo Depth
05 — Left/Right LFO
06 — Tremolo Sample & Hold Rate
07 — Tremolo Left Delay Time
08 — Tremolo Right Delay Time
09 — Tremolo Delay Regen
10 — Tremolo Left Echo Time
11 — Tremolo Right Echo Time
12 — Tremolo Echo Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

EQ-Vibrato-DDL
01 — Mix
02 — Volume
03 — Vibrato Rate
04 — Vibrato Width
05 — Left/Right LFO
06 — Vibrato Sample & Hold Rate
07 — Vibrato Left Delay Time
08 — Vibrato Right Delay Time
09 — Vibrato Delay Regen
10 — Vibrato Left Echo Time
11 — Vibrato Right Echo Time
12 — Vibrato Echo Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

Expander
01 — Mix
02 — Volume
03 — Exp Ratio
04 — Threshold
05 — Gain Change
06 — Exp Attack
07 — Exp Release
08 — Exp Gate Hold Time
09 — Sidechain EQ Gain
10 — HighPass Fc
11 — LowPass Fc
12 — Trigger Mask
13 — Time
14 — Trig Mask Lower Threshold
15 — Expander Output Gain
16 — Mod1 Source
17 — Mod1 Destination Parameter
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination Parameter
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

FastPitchShift
01 — Mix
02 — Volume
03 — PitchShifter Vc 1 Fine
04 — PitchShifter Vc 1 Level
05 — PitchShifter Vc 1 Pan
06 — PitchShifter Vc 2 Fine
07 — PitchShifter Vc 2 Level
08 — PitchShifter Vc 2 Pan
09 — PitchShifter LFO Rate
10 — PitchShifter LFO Width
11 — Mod1 Source
12 — Mod1 Destination Parameter
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination Parameter
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

Flanger
01 — Mix
02 — Volume
03 — Flanger LFO Rate
04 — Flanger LFO Width
05 — Flanger Center
06 — Flanger Regen
07 — Mod1 Source
08 — Mod1 Destination Parameter
09 — Mod1 Param Range Min
10 — Mod1 Param Range Max
11 — Mod2 Source
12 — Mod2 Destination Parameter
13 — Mod2 Param Range Min
14 — Mod2 Param Range Max

Flanger-Reverb
01 — Mix
02 — Volume
03 — Reverb Mix
04 — Flanger LFO Rate
05 — Flanger LFO Width
06 — Flanger Center
07 — Flanger Feedback
08 — Flanger Notch Depth
09 — Left/Right LFO
10 — Large Plate Decay
11 — Plate Predelay Time
12 — Large Plate HF Damping
13 — Large Plate HF Bandwidth
14 — Plate Diffsn1
15 — Diffusion2
16 — Plate Decay Definition
17 — Mod1 Source
18 — Mod1 Destination
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

Fuzz Box
01 — Mix
02 — Volume
03 — HighPass Fc
04 — LowPass Fc
05 — Distortion Level In
06 — Distortion Level Out
07 — Rectifier Mix
08 — Softness
09 — Harmonic Mod
10 — Offset
11 — Gain Bass
12 — Treble
13 — Mid1 Fc
14 — Mid1 Gain
15 — Mid1 Q
16 — Mid2 Fc
17 — Mid2 Gain
18 — Mid2 Q
19 — Post-Distortion VCF Fc
20 — Post-Distortion VCF Q
21 — Noise Gate Off Below
22 — Noise Gate On Above
23 — Gain Change
24 — Gate Release Time
25 — Mod1 Source
26 — Mod1 Destination
27 — Mod1 Param Range Min
28 — Mod1 Param Range Max
29 — Mod2 Source
30 — Mod2 Destination
31 — Mod2 Param Range Min
32 — Mod2 Param Range Max

Gated Reverb
01 — Mix
02 — Volume
03 — Gate Attack
04 — Hold Time
05 — Gate Decay
06 — Release Time
07 — Gate Trigger Threshold
08 — Gated Retrigger Threshold
09 — Gated HF Damping
10 — Gated Diffusion 1
11 — Gated Diffusion 2
12 — Gated Decay Definition
13 — Gated Slapback
14 — Gated Slapback Level
15 — Early Refs (1)
16 — Early Refs (2)
17 — Early Refs (3)
18 — Early Refs (4)
19 — Left/Right Balance
20 — Mod1 Source
21 — Mod1 Destination Parameter
22 — Mod1 Param Range Min
23 — Mod1 Param Range Max
24 — Mod2 Source
25 — Mod2 Destination Parameter
26 — Mod2 Param Range Min
27 — Mod2 Param Range Max

Guitar Amp 1
Guitar Amp 2
01 — Mix
02 — Volume
03 — Amp Preamp Gain
04 — Amp Output Level
05 — Amp Tube Bias
06 — Pre-EQ Input Level Trim
07 — Pre-EQ High Pass Cutoff
08 — Pre-EQ Fc
09 — Pre-EQ Gain
10 — Pre-EQ Q
11 — Noise Gate Off Below
12 — Noise Gate On Above
13 — Gate Release Time
14 — Speaker High Pass Cutoff
15 — OutEQ1 Fc
16 — OutEQ1 Gain
17 — OutEQ1 Q
18 — OutEQ2 Fc
19 — OutEQ2 Gain
20 — OutEQ2 Q
21 — Speaker Low Pass Cutoff
22 — Mod1 Source
23 — Mod1 Destination Parameter
24 — Mod1 Param Range Min
25 — Mod1 Param Range Max
26 — Mod2 Source
27 — Mod2 Destination Parameter
28 — Mod2 Param Range Min
29 — Mod2 Param Range Max

Guitar Amp 3
01 — Mix
02 — Volume
03 — AmpPreamp Gain
04 — Amp Output Level
05 — PreEQ Input Level Trim
06 — Pre-EQ Fc
07 — Pre-EQ Gain
08 — Pre-EQ Q
09 — ExpndRatio
10 — Threshold
11 — Gain Change
12 — Noise Gate Off Below
13 — Noise Gate On Above
14 — Gate Release Time
15 — Speaker High Pass Cutoff
16 — OutEQ1 Fc
17 — OutEQ1 Gain
18 — OutEQ1 Q
19 — OutEQ2 Fc
20 — OutEQ2 Gain
21 — OutEQ2 Q
22 — Speaker Low Pass Cutoff
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

Guitar Amp 4
01 — Mix
02 — Volume
03 — Amp Preamp Gain
04 — Output Level
05 — Amp Level Detect Attack
06 — Amp Level Detect Release
07 — Amp Tube Bias
08 — Pre-EQ InputLevel Trim
09 — Pre-EQHighPass Cutoff
10 — Pre-EQ Fc
11 — Pre-EQ Gain
12 — Pre-EQ Q
13 — Noise Gate Off Below
14 — Gate Release Time
15 — Speaker HighPass Cutoff
16 — OutEQ1 Fc
17 — OutEQ1 Gain
18 — OutEQ1 Q
19 — OutEQ2 Fc
20 — OutEQ2 Gain
21 — OutEQ2 Q
22 — Speaker Low Pass Cutoff
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

GuitarTuner2U
01 — Mix
02 — Volume
03 — Note
04 — Range
05 — Reference

Hall Reverb
01 — Mix
02 — Volume
03 — Room/Hall Decay
04 — Room/Hall Predelay Time
05 — Room/Hall LF DecayTime
06 — Room/Hall HF Damping
07 — Room/Hall HF Bandwidth
08 — Room/Hall Diffusion1
09 — Room/Hall Diffusion2
10 — Room/Hall Decay Definition
11 — Room/Hall Detune Rate
12 — Room/Hall Detune Depth
13 — Room/Hall Primary Send
14 — Room/Hall Ref 1 Time
15 — Room/Hall Ref 1 Level
16 — Room/Hall Ref 1 Send
17 — Room/Hall Ref 2 Time
18 — Room/Hall Ref 2 Level
19 — Room/Hall Ref 2 Send
20 — Position Balance (1)
21 — Position Balance (2)
22 — Position Balance (3)
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

InversExpander
01 — Mix
02 — Volume
03 — Expnd Ratio
04 — Threshold
05 — Gain Change
06 — Exp Attack
07 — Exp Release
08 — Exp Noise Gate Off Below
09 — Comp Noise Gate On Above
10 — Bass Fc
11 — Bass EQ Gain
12 — Treble Fc
13 — Treble EQ Gain
14 — EQ Input Level Trim
15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

Keyed Expander
01 — Mix
02 — Volume
03 — Exp Ratio
04 — Threshold
05 — Gain Change
06 — Exp Attack
07 — Exp Release
08 — Exp Gate Hold Time
09 — Sidechain EQ Gain
10 — HighPass Fc
11 — LowPass Fc
12 — Trigger Mask
13 — TriggeTime
14 — Trigger Mask Threshold
15 — Expander Output Mix
16 — Expander Output Gain
17 — Mod1 Source
18 — Mod1 Destination Parameter
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination Parameter
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

Large Plate
01 — Mix
02 — Volume
03 — Small/Large Plate Decay
04 — Plate Predelay Time
05 — Small/Large Plate HF Damping
06 — Small/Large Plate HF Bandwidth
07 — Plate Diffusion 1
08 — Plate Diffusion 2
09 — Plate Decay Definition
10 — Early Ref Level 1
11 — Early Ref Level 2
12 — Early Ref Level 3
13 — Early Ref Level 4
14 — Left/Right Balance
15 — Mod1 Source
16 — Mod1 Destination Parameter
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination Parameter
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

Large Room Rev

01 — Mix
02 — Volume
03 — Room/Hall Decay
04 — Room/Hall Predelay Time
05 — Room/Hall LF DecayTime
06 — Room/Hall HF Damping
07 — Room/Hall HF Bandwidth
08 — Room/Hall Diffusion1
09 — Room/Hall Diffusion2
10 — Room/Hall Decay Definition
11 — Room/Hall Detune Rate
12 — Room/Hall Detune Depth
13 — Room/Hall Primary Send
14 — Room/Hall Ref 1 Time
15 — Room/Hall Ref 1 Level
16 — Room/Hall Ref 1 Send
17 — Room/Hall Ref 2 Time
18 — Room/Hall Ref 2 Level
19 — Room/Hall Ref 2 Send
20 — Position Balance (1)
21 — Position Balance (2)
22 — Position Balance (3)
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

MultiTap Delay

01 — Mix
02 — Volume
03 — MultiTap 1 Time
04 — MultiTap 1 Level
05 — MultiTap 1 Regen
06 — MultiTap 1 Pan
07 — MultiTap 2 Time
08 — MultiTap 2 Level
09 — MultiTap 2 Regen
10 — MultiTap 2 Pan
11 — MultiTap 3 Time
12 — MultiTap 3 Level
13 — MultiTap 3 Regen
14 — MultiTap 3 Pan
15 — MultiTap 4 Time
16 — MultiTap 4 Level
17 — MultiTap 4 Regen
18 — MultiTap 4 Pan
19 — MultiTap Regen Damping
20 — Mod1 Source
21 — Mod1 Destination Parameter
22 — Mod1 Param Range Min
23 — Mod1 Param Range Max
24 — Mod2 Source
25 — Mod2 Destination Parameter
26 — Mod2 Param Range Min
27 — Mod2 Param Range Max

No Effect

01 — Mix
02 — Volume
03 — Mod1 Source
04 — Mod1 Destination Parameter
05 — Mod1 Param Range Min
06 — Mod1 Param Range Max
07 — Mod2 Source
08 — Mod2 Destination Parameter
09 — Mod2 Param Range Min
10 — Mod2 Param Range Max

NonLin Reverb1
NonLin Reverb2
NonLin Reverb3

01 — Mix
02 — Volume
03 — Envelope Level 1
04 — Envelope Level 2
05 — Envelope Level 3
09 — Envelope Level 4
07 — Envelope Level 5
08 — Envelope Level 6
09 — Envelope Level 7
10 — Envelope Level 8
11 — Envelope Level 9
12 — NonLin HF Damping
13 — NonLin HF Bandwidth
14 — NonLin Diffusion1
15 — NonLin Diffusion2
16 — NonLin Density 1
17 — NonLin Density 2
18 — NonLin Primary Send
19 — Reflection 1 Time
20 — Reflection 1 Send
21 — Reflection 2 Time
22 — Reflection 2 Send
23 — Left/Right Balance
24 — Mod1 Source
25 — Mod1 Destination Parameter
26 — Mod1 Param Range Min
27 — Mod1 Param Range Max
28 — Mod2 Source
29 — Mod2 Destination Parameter
30 — Mod2 Param Range Min
31 — Mod2 Param Range Max

Parametric EQ
01 — Mix
02 — Volume
03 — Bass Fc
04 — Bass Gain (loShv)
05 — Mid1 Fc
06 — Mid1 Gain
07 — Mid1 Q
08 — Mid2 Fc
09 — Mid2 Gain
10 — Mid2 Q
11 — Treble Fc
12 — Treble Gain (HiShv)
13 — EQ Input Level Attenuation
14 — Mod1 Source
15 — Mod1 Destination Parameter
16 — Mod1 Param Range Min
17 — Mod1 Param Range Max
18 — Mod2 Source
19 — Mod2 Destination Parameter
20 — Mod2 Param Range Min
21 — Mod2 Param Range Max

Phaser-DDL
01 — Mix
02 — Volume
03 — Phaser LFO Rate
04 — Phaser LFO Width
05 — Phaser Center
06 — Phaser Feedback
07 — Phaser Notch Depth
08 — Left/Right LFO
09 — Phaser Sample & Hold Rate
10 — Phaser Left Delay Time
11 — Phaser Right Delay Time
12 — Phaser Delay Feedback
13 — Mod1 Source
14 — Mod1 Destination Parameter
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination Parameter
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

Phaser-Reverb
01 — Mix
02 — Volume
03 — Reverb Mix
04 — Phaser LFO Rate
05 — Phaser LFO Width
06 — Phaser Center
07 — Phaser Feedback
08 — Phaser Notch Depth
09 — Large Plate Decay
10 — Plate Predelay Time
11 — Large Plate HF Damping
12 — Large Plate HF Bandwidth
13 — Plate Diffsn1
14 — Diffusion2
15 — Plate Decay Definition
16 — Mod1 Source
17 — Mod1 Destination
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

Pitch Shift 2U
01 — Mix
02 — Volume
03 — PitchShifter Vc 1 Semi
04 — PitchShifter Vc 1 Fine
05 — PitchShifter Vc 1 Level
06 — PitchShifter Vc 1 Pan
07 — PitchShifter Vc 2 Semi
08 — PitchShifter Vc 2 Fine
09 — PitchShifter Vc 2 Level
10 — PitchShifter Vc 2 Pan
11 — PitchShifter LFO Rate
12 — PitchShifter LFO Width
13 — Mod1 Source
14 — Mod1 Destination Parameter
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination Parameter
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

PitchShift-DDL
01 — Mix
02 — Volume
03 — PitchShift Vc 1 Semi
04 — PitchShift Vc 1 Fine
05 — PitchShift Vc 1 Level
06 — PitchShifter Vc 1 Pan
07 — PitchShift Vc 2 Semi
08 — PitchShift Vc 2 Fine
09 — PitchShift Vc 2 Level
10 — PitchShifter Vc 2 Pan
11 — PitchShift Dry Level to DDL
12 — PitchShift Left Delay Time
13 — PitchShift Right Delay Time
14 — PitchShift Delay Mix
15 — PitchShift Delay Regen
16 — Mod1 Source
17 — Mod1 Destination Parameter
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination Parameter
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

Pitch Shifter
01 — Mix
02 — Volume
03 — PitchShifter Vc 1 Semi
04 — PitchShifter Vc 1 Fine
05 — PitchShifter Vc 1 Level
06 — PitchShifter Vc 1 Pan
07 — PitchShifter Vc 2 Semi
08 — PitchShifter Vc 2 Fine
09 — PitchShifter Vc 2 Level
10 — PitchShifter Vc 2 Pan
11 — Delay vs Quality
12 — PitchShifter LFO Rate
13 — PitchShifter LFO Width
14 — Mod1 Source
15 — Mod1 Destination Parameter
16 — Mod1 Param Range Min
17 — Mod1 Param Range Max
18 — Mod2 Source
19 — Mod2 Destination Parameter
20 — Mod2 Param Range Min
21 — Mod2 Param Range Max

Plate-Chorus
01 — Mix
02 — Volume
03 — Large Plate Decay
04 — Plate Predelay Time
05 — Large Plate HF Damping
06 — Large Plate HF Bandwidth
07 — Plate Diffusn1
08 — Diffusion2
09 — Plate Decay Definition
10 — Large Plate Feedback
11 — Chorus LFO Rate
12 — Chorus LFO Width
13 — Chorus Center
14 — Chorus Feedback
15 — Chorus Mix
16 — Mod1 Source
17 — Mod1 Destination
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

ReverseReverb1
01 — Mix
02 — Volume
03 — Reverse Envelope Hold Time
04 — Reverse Attack
05 — Reverse Release
06 — Reverse Trigger Threshold
07 — Reverse HF Damping
08 — Rev Diffusion 1
09 — Rev Diffusion 2
10 — Reverse Decay Definition
11 — Reverse Slapback
12 — Reverse Slapback Level
13 — Mod1 Source
14 — Mod1 Destination Parameter
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination Parameter
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

ReverseReverb2
01 — Mix
02 — Volume
03 — Reverse Envelope Hold Time
04 — Reverse Attack
05 — Reverse Release
06 — Reverse Trigger Threshold
07 — Pre-Trigger Memory
08 — Reverse HF Damping
09 — Rev Diffusion 1
10 — Rev Diffusion 2
11 — Reverse Decay Definition
12 — Mod1 Source
13 — Mod1 Destination Parameter
14 — Mod1 Param Range Min
15 — Mod1 Param Range Max
16 — Mod2 Source
17 — Mod2 Destination Parameter
18 — Mod2 Param Range Min
19 — Mod2 Param Range Max

Rotating Spkr
01 — Mix
02 — Volume
03 — Rotating Speaker Slow Speed
04 — Rotating Speaker Fast Speed
05 — Rotating Speaker Speed
06 — Rotating Speaker Inertia
07 — Distortion Level In
08 — Distortion Level Out
09 — Rotating Speaker Distortion Tone
10 — Rotating Speaker Stereo Spread
11 — Mod1 Source
12 — Mod1 Destination Parameter
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination Parameter
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

Rumble Filter
01 — Mix
02 — Volume
03 — HighPass Fc
04 — LowPass Fc
05 — Filter Gain
06 — Mod1 Source
07 — Mod1 Destination Parameter
08 — Mod1 Param Range Min
09 — Mod1 Param Range Max
10 — Mod2 Source
11 — Mod2 Destination Parameter
12 — Mod2 Param Range Min
13 — Mod2 Param Range Max

Sine/Noise Gen
01 — Mix
02 — Volume
03 — Sine/Noise Gen Sine Freq
04 — Sine/Noise Gen Balance
05 — Noise Filters - Low Pass Fc
06 — Bass Fc
07 — EQ Gain
08 — Treble Fc
09 — EQ Gain
10 — EQ Input Level Trim
11 — Mod1 Source
12 — Mod1 Destination Parameter
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination Parameter
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

Small Plate
01 — Mix
02 — Volume
03 — Small/Large Plate Decay
04 — Plate Predelay Time
05 — Small/Large Plate HF Damping
06 — Small/Large Plate HF Bandwidth
07 — Plate Diffusion 1
08 — Plate Diffusion 2
09 — Plate Decay Definition
10 — Early Ref Level 1
11 — Early Ref Level 2
12 — Early Ref Level 3
13 — Early Ref Level 4
14 — Left/Right Balance
15 — Mod1 Source
16 — Mod1 Destination Parameter
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination Parameter
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

Small Room Rev
01 — Mix
02 — Volume
03 — Room/Hall Decay
04 — Room/Hall Predelay Time
05 — Room/Hall LF DecayTime
06 — Room/Hall HF Damping
07 — Room/Hall HF Bandwidth
08 — Room/Hall Diffusion1
09 — Room/Hall Diffusion2
10 — Room/Hall Decay Definition
11 — Room/Hall Detune Rate
12 — Room/Hall Detune Depth
13 — Room/Hall Primary Send
14 — Room/Hall Ref 1 Time
15 — Room/Hall Ref 1 Level
16 — Room/Hall Ref 1 Send
17 — Room/Hall Ref 2 Time
18 — Room/Hall Ref 2 Level
19 — Room/Hall Ref 2 Send
20 — Position Balance (1)
21 — Position Balance (2)
22 — Position Balance (3)
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

Speaker Cabinet
01 — Mix
02 — Volume
03 — Speaker Output Gain
04 — Mod1 Source
05 — Mod1 Destination Parameter
06 — Mod1 Param Range Min
07 — Mod1 Param Range Max
08 — Mod2 Source
09 — Mod2 Destination Parameter
10 — Mod2 Param Range Min
11 — Mod2 Param Range Max

Tempo Delay
01 — Mix
02 — Volume
03 — Tempo Delay Time
04 — Internal Clock Tempo
05 — TempoDelay Fine Tune
06 — Tempo Control
07 — Tempo Delay Regen
08 — Tempo Delay Pan
09 — Tempo Delay Regen Damping
10 — Tempo Delay Smoothing
11 — Mod1 Source
12 — Mod1 Destination Parameter
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination Parameter
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

Tunable Spkr 1
01 — Mix
02 — Volume
03 — Mid1 Fc
04 — Mid1 Gain
05 — Mid1 Q
06 — Mid2 Fc
07 — Mid2 Gain
08 — Mid2 Q
09 — Mid3 Fc
10 — Mid3 Gain
11 — Mid3 Q
12 — Speaker Input Attenuation
13 — Speaker Output Gain
14 — Mod1 Source
15 — Mod1 Destination Parameter
16 — Mod1 Param Range Min
17 — Mod1 Param Range Max
18 — Mod2 Source
19 — Mod2 Destination Parameter
20 — Mod2 Param Range Min
21 — Mod2 Param Range Max

Tunable Spkr 2
01 — Mix
02 — Volume
03 — Mid1 Fc
04 — Mid1 Gain
05 — Mid1 Q
06 — Mid2 Fc
07 — Mid2 Gain
08 — Mid2 Q
09 — Mid3 Fc
10 — Mid3 Gain
11 — Mid3 Q
12 — PreEQ InputLevel Trim
13 — Speaker Output Gain
14 — Noise Gate Off Below
15 — Gate Release Time
16 — Pre-EQHighPass Cutoff
17 — Mod1 Source
18 — Mod1 Destination Parameter
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination Parameter
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

VanderPolFilter
01 — Mix
02 — Volume
03 — VanderPol Filter HighPass Fc
04 — VanderPol Filter LowPass Fc
05 — Filter Gain
06 — Mod1 Source
07 — Mod1 Destination Parameter
08 — Mod1 Param Range Min
09 — Mod1 Param Range Max
10 — Mod2 Source
11 — Mod2 Destination Parameter
12 — Mod2 Param Range Min
13 — Mod2 Param Range Max

VCF-Distort 1
01 — Mix
02 — Volume
03 — Distortion Level In
04 — Distortion Level Out
05 — Pre-Distortion VCF Fc
06 — Pre-Distortion VCF Q
07 — Envelope Follower to Pre VCF
08 — Post-Distortion VCF Fc
09 — Post-Distortion VCF Q
10 — Envelope Follower to Post VCF
11 — Envelope Follower Attack
12 — Envelope Follower Release
13 — Distortion Bypass
14 — Pre-EQ High Pass Cutoff
15 — Mod1 Source
16 — Mod1 Destination Parameter
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination Parameter
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

VCF - Distort 2
01 — Mix
02 — Volume
03 — Distortion Level In
04 — Distortion Level Out
05 — Pre-Distortion VCF Fc
06 — Pre-Distortion VCF Q
07 — Envelope Follower to Pre VCF
08 — Post-Distortion VCF Fc
09 — Post-Distortion VCF Q
10 — Envelope Follower to Post VCF
11 — Envelope Follower Attack
12 — Envelope Follower Release
13 — Distortion Bypass
14 — Pre-EQ High Pass Cutoff
15 — Speaker HighPass Cutoff
16 — Amp Feedback Amount
17 — Amp Feedback HF Damping
18 — Amp Feedback Delay
19 — Mod1 Source
20 — Mod1 Destination Parameter
21 — Mod1 Param Range Min
22 — Mod1 Param Range Max
23 — Mod2 Source
24 — Mod2 Destination Parameter
25 — Mod2 Param Range Min
26 — Mod2 Param Range Max

Vocal Remover
01 — Mix
02 — Volume
03 — Vocal Pos
04 — L/R Delay
05 — Bass Level
06 — Treble Level
07 — Mid Level
08 — Bass Fc
09 — Trebl Fc
10 — Mid Fc
11 — BW
12 — Mod1 Source
13 — Mod1 Destination Parameter
14 — Mod1 Param Range Min
15 — Mod1 Param Range Max
16 — Mod2 Source
17 — Mod2 Destination Parameter
18 — Mod2 Param Range Min
19 — Mod2 Param Range Max

Vocoder Part1
Vocoder Part2
01 — Mix
02 — Volume
03 — Vocoder Gain Vox
04 — Vocoder Gain Synth
05 — Vocoder Sibilance Lev (Part 1 only)
06 — Vocoder Response Time
07 — Vocoder Pan
08 — Mod1 Source
09 — Mod1 Destination
10 — Mod1 Param Range Min
11 — Mod1 Param Range Max
12 — Mod2 Source
13 — Mod2 Destination
14 — Mod2 Param Range Min
15 — Mod2 Param Range Max

Wah-Dist-Revrb
01 — Mix
02 — Volume
03 — Reverb Mix
04 — Compressor Threshold
05 — Comp Attack
06 — Comp Release
07 — Compressor Gain
08 — Wah Center
09 — Wah Range
10 — Wah Attack
11 — Wah Release
12 — Distortion Level In
13 — Distortion Level Out
14 — Post-Distortion VCF Fc
15 — Post-Distortion VCF Q
16 — Distortion Mix
17 — Amp Feedback Amount
18 — Reverb Decay
19 — Reverb HF Damping
20 — Mod1 Source
21 — Mod1 Destination
22 — Mod1 Param Range Min
23 — Mod1 Param Range Max
24 — Mod2 Source
25 — Mod2 Destination
26 — Mod2 Param Range Min
27 — Mod2 Param Range Max

Edit Config Parameters

1 Source Config

00 — 1 Source Config
01 — AB Input Select
02 — AB Unit Routing
03 — AB (Config Dependent)
04 — Bypass Kill (Unit) A
05 — Bypass Kill (Unit) B

2 Source Config

00 — 2 Source Config
01 — AB Output Select
02 — Bypass Kill (Unit) A
03 — Bypass Kill (Unit) B

System/MIDI Parameters

00 — MIDI Channel
01 — MIDI Enable
02 — Program Change
03 — Program Change Map
04 — Program Change-to-Preset Map Editor
05 — Selects Preset
06 — Unit Bypass

• 07 through 20 are identical to these parameters and control Unit B and Config respectively.

21 — MIDI Control Channel
22 — MIDI Reception

23 — DP/2 Controller 1
24 — DP/2 Controller 2
25 — DP/2 Controller 3
26 — DP/2 Controller 4
27 — DP/2 Controller 5
28 — DP/2 Controller 6
29 — DP/2 Controller 7
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System Exclusive Parameters

Soft Reset (without erasing the internal memory)

- While holding down (SYSTEM/MIDI), press the (A) button.

Initializing the RAM Presets

- While holding down (SYSTEM/MIDI), press the (B) button.
- Press (WRITE) to initialize all of the RAM presets.

Reinitializing the DP/2

- While holding down (SYSTEM/MIDI), press the (B) button.
- Press the (▷) button once.
- Press (WRITE) to reinitialize the DP/2.

Specs

Frequency response (wet and dry) = 2 Hz–16. KHz
Signal-to-noise ("A" weighted) = at +4 dBu
in to out = -90 dB
output only = -92 dB
at -10 dBV
in to out = -87 dB
output only = -90 dB
THD + Noise ("A" weighted) = better than .0032% (-90 dB) at input levels of -6 dB and below
Dynamic range = 96 dB
IM distortion (SMPTE) = 0.05%
Crosstalk between channels better than -80 dB (1 KHz)
Input impedance =
Rear Input = 21.4 KΩ
Front Input= 910. KΩ
Output impedance = 220. Ω
Maximum output level into > 10. KΩ = +17.3 dBu
Maximum output level into 600. Ω = +14.5 dBu
Maximum Input Range= +19.5 dBu (high) to -18.5 dBV (low)

(2) 24/48 bit DSP chips yield 20 MIPS processing power
Digital to Analog conversion = 16 Bit
Analog to Digital conversion = 16 Bit
128K words of delay memory (256 Kbytes)
Max delay time for a single unit algorithm = 1.8 sec.
Max single delay time possible (no regeneration) = 3.6 sec. (using the 3.6 sec DDL 2U algorithm)
Preset Memory = 600, divided between 300 ROM, 300 RAM (user)
Headphone output = 29. mW/channel into 600 Ω, 15. mW/channel into 30 Ω

Physical

2 audio inputs, 2 audio outputs (phone jacks)
"ground compensated" outputs
separate input and output level controls for 2 channels; accommodate -18.5 dBV to +19.5 dBu
For -10 dBV input — input knob setting is at 2:00
For -10 dBV output — output knob setting is at 10:00
For +4 dBu input — input knob setting is at 11:00
For +4 dBu output — output knob setting is at 2:00
two level indicator LEDs per channel (-30 dB, -6 dB)
32 character alpha-numeric LCD display
digital 24 step parameter knob
MIDI in/out and thru
analog control voltage pedal input
2 dual foot switch inputs
external "brick" power supply: use only with PS-1 Power Supply from ENSONIQ
Detachable power cord, internal fuse

Dimensions

19" (48.26 cm) wide x 1 3/4" (4.29 cm) high x 9 3/4" (23.6 cm) deep
19" rack mount standard, 1U high
6.3 lbs. (2.9 kilograms)

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- Vc 2 Level 117
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  Left Delay Time 70
  Left Echo Time 70
  Rate 69
  Right Delay Time 70
  Right Echo Time 70
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Song/Step Worksheet

Feel free to copy this form and use it as a reference when creating programmable preset chains using the DP/2's Song feature.

	Step 01	Step 02	Step 03	Step 04	Step 05	Notes
Song 1						
Song 2						
Song 3						
Song 4						
Song 5						
Song 6						
Song 7						
Song 8						
Song 9						
Song 10						
Song 11						
Song 12						
Song 13						
Song 14						
Song 15						
Song 16						
Song 17						
Song 18						
Song 19						
Song 20						

MIDI Program Change Map Worksheet

Feel free to copy this form and use it as a reference when creating programmable MIDI Program Change Maps on the DP/2.

Unit: A B	Additional Information:

Prg Chg #:	Selects Preset	Prg Chg #:	Selects Preset	Prg Chg #:	Selects Preset	Prg Chg #:	Selects Preset
001		033		065		097	
002		034		066		098	
003		035		067		099	
004		036		068		100	
005		037		069		101	
006		038		070		102	
007		039		071		103	
008		040		072		104	
009		041		073		105	
010		042		074		106	
011		043		075		107	
012		044		076		108	
013		045		077		109	
014		046		078		110	
015		047		079		111	
016		048		080		112	
017		049		081		113	
018		050		082		114	
019		051		083		115	
020		052		084		116	
021		053		085		117	
022		054		086		118	
023		055		087		119	
024		056		088		120	
025		057		089		121	
026		058		090		122	
027		059		091		123	
028		060		092		124	
029		061		093		125	
030		062		094		126	
031		063		095		127	
032		064		096		128	

DP/2 Preset Parameter Worksheet	Preset Name:	
Config Parameters:	1 2 Source Config	01-
02-	03-	04-

Unit A Algorithm:	**Unit B Algorithm:**
01- Mix	01- Mix
02- Volume	02- Volume
03-	03-
04-	04-
05-	05-
06-	06-
07-	07-
08-	08-
09-	09-
10-	10-
11-	11-
12-	12-
13-	13-
14-	14-
15-	15-
16-	16-
17-	17-
18-	18-
19-	19-
20-	20-
21-	21-
22-	22-
23-	23-
24-	24-
25-	25-
26-	26-
27-	27-
28-	28-
29-	29-
30-	30-
31-	31-
32-	32-
33-	33-
34-	34-

Notes:

AUTHORIZED ENSONIQ REPAIR STATIONS

Before taking your ENSONIQ product for service or repair, check the troubleshooting sections in this manual. If your ENSONIQ product requires service, first contact the dealer where you purchased it. The following Authorized Repair Stations, listed by state and country, can also perform warranty service.

IN THE UNITED STATES

Excel Electronics	2020 Ivan Street	Anchorage	AK	99507	907-522-4538
Altech	300 Front St	Fairbank	AK	99701	907-456-8324
M & M Music (AK)	9106 Mendenhall Mall Rd	Juneau	AK	99803	907-789-7337
Andys Music	1412 Hillcrest Rd	Mobile	AL	36695	334-633-8744
MMI Inc	4055 Cottage Hill Rd	Mobile	AL	36609	205-660-1277
Boyd Music	5707 W. 12th Street	Little Rock	AR	72704	501-664-3614
Audionyx	605 Mockingbird Ct	Prescott	AZ	86301	602-771-0050
Arizona Organ & Keyboard	11725 N 19th Avenue #6	Phoenix	AZ	85029	602-955-2400
Guitar Etc	3226 E Speedway Blvd	Tucson	AZ	85716	602-748-1111
Guitar Etc	5646 E Speedway	Tucson	AZ	85712	602-748-1111
Synthony Music	3939 East Campbell	Phoenix	AZ	85018	602-955-3590
Pacific Innovative Elec	10840 Van Owen St	N. Hollywood	CA	91605	818-508-9550
MSC Musician Service	647 Tully Rd Suite 6	San Jose	CA	95111	408-297-7532
Electronics Diversified	544 Walter Avenue	Newbury Park	CA	91320	805-499-3982
Studio Maintenance Center	655 Du Bois St Suite F	San Rafael	CA	94901	415-485-6048
Electronic Audio Repair	1030 Folsom St	San Francisco	CA	94103	415-865-0181
Audio Spectrum	1526 Fillmore	San Francisco	CA	94115	415-292-7480
Paul Morte Tech Service	946 North Main Street	Orange	CA	92667	714-532-9540
Alan Robertson Electronic	2274 Norman Court	Eureka	CA	95503	707-444-0128
Zone Music	7884 Old Redwood Hwy	Cotati	CA	94931	707-664-1213
Video Sounds Technology	1270 Lincoln Ave #1000	Pasadena	CA	91103	818-794-8196
Boeman Electronics	411 19th Street	Bakersfield	CA	93301	805-322-6526
A.M.E.	8665 Venice Blvd	Los Angeles	CA	90034	310-559-3157
Caraquin Co	11659 Rocoso Road	Lakeside	CA	92040	619-561-1328
Alectronics (CA)	1355 Lawrence Dr Ste 109	Newbury Park	CA	91320	805-499-0601
Valley Sound	1023 N. La Brea	Hollywood	CA	90038	213-851-3434
Absolute Audio	166 Cohasset Road #2	Chico	CA	95926	916-893-4088
Kinder Musical Instrument Elec	25030 Yucca Dr	Moreno Valley	CA	92388	909-242-5923
Liers Music	452 North E Street	San Bernardino	CA	92401	909-884-8815
Coretronics	120 2nd Street	Eureka	CA	95501	707-444-0237
Bananas At Large	1504 Fourth Street	San Rafael	CA	94901	415-457-7600
East Bay Sound	7017 Village Parkway	Dublin	CA	94568	510-482-4866
Pro Sound & Music	4593 Mission Gorge Pl.	San Diego	CA	92120	619-583-7851
Stanroys Music Center	741 4th Street	Santa Rosa	CA	95404	707-545-4827
Digitron Electronics	7805 E Telegraph Rd Ste#D	Montebello	CA	90640	213-887-0777
Skips Music	2740 Auburn Blvd	Sacramento	CA	95821	916-481-7575
Buley Electronic & Audio	940 B E Main St	Santa Paula	CA	93060	805-933-3992
Sound Zone	27343 Indstrial Blvd-SteA	Hayward	CA	94545	510-786-3745
San Diego Sound Inc	6528 El Cajon Blvd	San Diego	CA	92115	619-582-8511
American Music Co	2597 East Ashlan	Fresno	CA	93726	209-221-0233
Sam Ash Music (CT)	95 Amity Rd	New Haven	CT	06515	203-489-0500
East Coast Music Mall	25 Hampton Rd	Danbury	CT	06811	203-748-2799
Mid Atlantic Music	1702 Kirkwood Hwy	Wilmington	DE	19805	302-995-7170
Mid South Audio	52 Bramhall St	Georgetown	DE	19947	302-856-6993
Guitar Service Center	6 Peddlars Village	Newark	DE	19702	302-368-1104
B & B Educational Music	205 S Dual Highway	Camden	DE	19934	302-697-2155
Harris Music & Sound	707 N Pace Blvd	Pensacola	FL	32505	904-434-6497
Abney's Music Ctr	1033 N Mills Ave	Orlando	FL	32803	407-898-3155
Morning Star Music	5363 Airport Rd N	Naples	FL	33942	941-514-1922
Pro-Tech Services	726 Ohio Avenue	Lynn Haven	FL	32401	904-265-4334
Abe Music	14501 W Dixie Hwy	N Miami	FL	33161	305-944-7429
American Music (FL)	5225 Lenox Ave	Jacksonville	FL	32205	904-781-1080
Johnson Electronics	231 East 54th Street	Hialeah	FL	33013	305-823-1791
Byte Five Inc.	7992 Southside Blvd	Jacksonville	FL	32256	904-641-4455
Donleys Elect. Service	3400 Forsyth Rd- Suite 4	Winter Park	FL	32792	407-677-0861
Wells Electronics Lab	1217 N Mills Ave	Orlando	FL	32803	407-894-3404
Lange Musical Electronics	6355 County Rd 78 West	Alva	FL	33920	813-768-0497
Alura Engineering	207 So Pine Ave	Ocala	FL	34470	904-368-2165
The LeLand Langridge Co.	2687 SE Grand Dr	Port St Lucie	FL	34952	407-337-3509
Thoroughbred Music (CW)	923 McMullen Boot Rd	Clearwater	FL	34619	813-725-8062
Entertainment Support	1003 Broadway	Riviera Beach	FL	33404	407-881-4443
Audio Doctor	1318-C N Monroe St	Tallahassee	FL	32303	904-222-0542
Wizard Electronics (GA)	1434 Tullie Road	Atlanta	GA	30329	404-325-4891
G & S Electronics	2407 Old Flowery Brand Rd	Gainesville	GA	30504	404-534-2374
Soundpost	100 Direct Connection Dr	Rossville	GA	30741	706-891-9404
Normans Electronics	3653 Clairmont Rd NE	Chamblee	GA	30341	404-451-5057

AUTHORIZED ENSONIQ REPAIR STATIONS

Normans Electronics	6115-C Jimmy Carter Blvd	Norcross	GA	30071	404-446-1118
Portmans Music Inc	7650 Abercorn St	Savannah	GA	31406	912-354-1500
Logical Audio Systems	2605 Mountain Ind Blvd Suite 6	Tucker	GA	30084	770-934-4887
TPS Electronics	1530 Makaloa St	Honolulu	HI	96814	808-951-6699
Kepharts Music Center	126 E. Water Street	Decorah	IA	52101	319-382-3684
West Music Co.	1212 5th Street	Coralville	IA	52241	319-351-2000
Wizard Electronics (IA)	1344 23rd Street	Bettendorf	IA	52722	319-359-8815
Noteworty Music	1438 N Tima Marie Ave	Meridian	ID	83642	208-888-5526
Rogers Audio & Design	119 Banner Street	Nampa	ID	83686	208-467-2465
Mikes Musical Instrument	2455 N Yellow Stone	Idaho Falls	ID	83403	208-524-6607
Accutrack Recording	551 N Wolf Road	Wheeling	IL	60090	847-465-8862
ICM Corporation	9050 Helen Lane	Orland Park	IL	60462	708-403-2715
Samuel Music Co	908 W Fayette Ave	Effingham	IL	62401	217-342-9221
Music Lab Inc	17805 Burnham Ave	Lansing	IL	60438	708-895-2218
Audio Pro Service	780 Frontage Rd	Northfield	IL	60093	847-446-4222
Swing City	1312 Vandalia	Collinsville	IL	62234	618-345-6700
Chicago Factory Service	539 W Golf Road	Arlington Hts	IL	60005	847-640-6181
Deltronics	3149 N Halsted	Chicago	IL	60657	312-549-6635
Elmore Musical Warehouse	3611 W Willow Knolls Rd	Peoria	IL	61614	309-692-1253
C.V. Lloyde Sound System	102 South Neil Street	Champaign	IL	61820	217-352-7031
Richards Music & Elec	1020 W Marion St	Joliet	IL	60436	815-729-0182
Midwest Music Menders	1917 Fullerton	Chicago	IL	60614	312-276-3939
Southern Indiana Music Co	109 N Chestnut St	Seymour	IN	47274	812-522-6768
Woodwind & The Brasswind	19880 State Line Rd	Southbend	IN	46637	219-272-8266
Music Today	1325 Meridian Street	Anderson	IN	46016	317-644-3361
Conservatory of Music	3400 South US 41	Terre Haute	IN	47802	812-232-2735
Amtech	7033 Calumet Ave	Hammond	IN	46324	219-937-0248
Opus 1 Music	5420 East Indiana St	Evansville	IN	47712	812-479-6787
I R C Music	5911 E. 82ND ST.	Indianapolis	IN	46250	317-849-7965
Far Out Music	2008 Coopers Lane	Jeffersonville	IN	47130	812-282-1122
Rubinos Music	8623 Louisanna Place	Merrillville	IN	46410	219-736-9344
Brier and Hale Music Co.	424 N. Kansas Avenue	Liberal	KS	67901	316-624-8421
S.M. Hanson Music Inc.	335 South Clark	Salina	KS	67401	913-825-6273
Brier and Hale Music Co.	319 Gunsmoke Avenue	Dodge City	KS	67801	316-225-5333
Thesis Audio	4235 W. Central	Wichita	KS	67212	316-942-7341
Steam Music	3740 Burlingame Circle	Topeka	KS	66609	913-267-3771
Uhlik Music Service	2160 E. Douglas	Witchita	KS	67214	316-262-2840
DBs Music (KY)	1221 Broadway Avenue	Bowling Green	KY	42101	502-782-5973
Owensboro Music Ctr	2350 New Hartford Pike	Owensboro	KY	42301	502-684-2156
Carl's Music Center	1125 Winchester Rd	Lexington	KY	40505	606-254-0324
Audio Video Electronics	805 E K U Bypass #2	Richmond	KY	40475	606-623-4406
Cajun Audio	112 Luke Street	Lafayette	LA	70506	318-269-9974
M.R. Montero Electronics	766 Hickory Avenue	Harahan	LA	70123	504-737-8942
Southern Electronics	1909 Tulane Avenue	New Orleans	LA	70112	504-524-2343
Electro Music Service	2100 Marshall St	Shreveport	LA	71101	318-222-5884
R.M.I.	259 New Boston Rd	Starbridge	MA	01566	508-347-2828
Rockfleet Music Service	175-P New Boston St	Woburn	MA	01801	617-937-0353
Ricks Music World	190 Taunton Ave	Seekonk	MA	02771	508-336-6180
Gordon Music Inc	333 Main St	Southbridge	MA	01550	508-764-2117
B & B Electronics	185 Walnut Street	Leominster	MA	01453	508-534-9242
Ricks Music World	179 Swansea Mall Dr	Swansea	MA	02777	508-672-2500
Syntronics	466 Commonwealth Ave-#103	Boston	MA	02215	617-266-5039
Downtown Sound	21 Pleasant St	North Hampton	MA	01060	413-586-0998
Alactronics Inc. (MA)	192 Worcester Street	Wellesley	MA	02181	617-239-0000
E.U. Wurlitzer Music	65 Bent St	Cambridge	MA	02141	617-738-5455
Washington Music Ctr	2421 Reedie Drive	Wheaton	MD	20902	301-929-2490
Manco Specialty Elec	RR3 Box 191 Old Gray Rd	Newport	ME	04953	207-368-2094
Al Nalli Music	312 S. Ashley	Ann Arbor	MI	48104	313-665-7008
Music Box Studios	42383 Garfield Road	Clinton Township	MI	48038	810-263-1994
Wonderland Music	13519 Michigan Avenue	Dearborn	MI	48126	313-584-8111
Electronic Innovations	21628 Van Dyke	Warren	MI	48089	810-758-6157
Back Stage Audio Ltd.	109 Ann Street	Fenton	MI	48430	810-235-5580
Arnoldt Williams	5701 Canton Center Rd	Canton	MI	48187	313-453-6586
Good News Music Centre	140 E. Front Street	Traverse City	MI	49684	616-946-1230
Bogner Sound & Music	3218 Coronna Rd	Flint	MI	48503	810-238-8777
Good Guys Inc.	1111 Grand Ave	St Paul	MN	55105	612-292-9165
Professional Repair	3448 42nd Ave S	Minneapolis	MN	55406	612-721-3130
On Line Electronics	3817 Broadway	Kansas City	MO	64111	816-753-0077
Crazy Music Sound & Light	201 North Tenth St	Columbia	MO	65201	314-443-2559
Sounds Great	1856 S Stewart St	Springfield	MO	65804	417-883-4543
Morrison Bros.	2233 Hwy - 80 West	Jackson	MS	39204	601-352-0135

AUTHORIZED ENSONIQ REPAIR STATIONS

M & M Electronics	4656 Sharon Rd	Laurel	MS	39440	601-649-3630
M & M Electronics	3106 Audubon Dr North Laurel S	Laurel	MS	39440	601-649-3630
Mississippi Music	4430 Robinson Road	Jackson	MS	39209	601-922-0357
Audio Clinic	3461 Canyon Drive	Billings	MT	59102	406-652-1564
Tritech Electronics Inc	618-H Guilford College	Greensboro	NC	27409	910-292-0330
Bull City Sound & Electr	1001 Broad St	Durham	NC	27705	919-286-1991
Music Tech Service	3021-1 Stoneybrook Dr	Raleigh	NC	27604	919-872-5119
McFadyen Music (Char)	2110 E. Independence Blvd	Charlotte	NC	28205	704-372-3960
McFadyen Music	Po Box 2325	Fayetteville	NC	28302	704-372-3960
Eckroth Music Co	1221 West Divide Avenue	Bismark	ND	58501	701-223-6707
E and J Associates	641 9th Ave SE	Valley City	ND	58072	701-845-3280
Musician Technical Serv	1618 Cass St	Omaha	NE	68102	402-345-4449
Daddys Junky Music	4 Raymond Ave Po Box 1018	Salem	NH	03079	603-893-4057
Triple S	228 Washington Ave	Belleville	NJ	07109	201-751-0481
Dannys Amp Service	6570 Sinkinson Avenue	Pensauken	NJ	08109	609-662-2979
Evans Music	500 Rt 10 West	Ledgewood	NJ	07852	201-584-9049
Grubb Brothers Elec	379 Route 73	West Berlin	NJ	08091	609-767-6627
Russo Music	1989 Arena Drive	Trenton	NJ	08610	609-888-0620
Jacks Musical Instruments	33 Broad Street	Red Bank	NJ	07701	908-747-4315
Sam Ash Music (Chr Hill)	2100 Rt 38	Cherry Hill	NJ	08002	609-667-6696
Audio Technology	Sea Girt Mall- Rt. 35	Sea Girt	NJ	08750	908-223-0274
Daves Sound Repair	622 Rt 10	Whippany	NJ	07981	201-386-5840
Rondo Music	1597 Highway 22	Union	NJ	07083	908-687-2250
Sam Ash Music (PAR)	1 E 50 Rt 4	Paramus	NJ	07652	201-843-0119
Enchantment Electronics	500 Islela SW	Albuquerque	NM	87105	505-873-1010
Grandmas Inc.	800 S-T Juan Tabo NE	Albuquerque	NM	87123	505-292-0341
Mesilla Valley Music	2200 N Main St	Las Cruces	NM	88001	505-526-8777
Starsound Audio	2679 Oddie Blvd.	Reno	NV	89512	702-331-1010
Mahoney's Pro Music & Drum	608 Maryland Pkwy	Las Vegas	NV	89101	702-382-9141
Pro Music & Drum	4972 S Maryland Pkwy	Las Vegas	NV	89119	702-736-1100
TSR	884 State Rt #13Rd	Cortland	NY	13045	800-841-1815
Direct Repair Service	1602 Rt 9	Clifton Park	NY	12065	518-383-0300
S.P.E.C.	684 Sunrise Highway	W Babylon	NY	11704	516-661-2454
McNeil Music	4517 Old Vestal Rd	Vestal	NY	13850	607-729-1548
Big Apple Music	4452 Commercail Dr	New Hartford	NY	13413	315-732-3502
dBm Pro Audio/Music Services	320 W 37 St 5th Floor	New York	NY	10018	212-629-0326
Only Guitar Shop	Route 9	Clifton Pk	NY	12065	518-371-1232
House of Guitars	645 Titus Avenue	Rochester	NY	14617	716-544-9900
Acutone	898A Broadway	Massapequa	NY	11758	516-799-3104
Palomba Music I	974 E Gunhill Rd	N Bronx	NY	10469	718-882-3700
Sam Ash Music	278 Duffy Avenue	Hicksville	NY	11801	516-932-6400
D.B. Musical Elect (NY)	2405 Harlem Road	Buffalo	NY	14225	716-894-9426
Onondaga Music	412 S. Clinton	Syracuse	NY	13202	315-422-8423
Palomba Music II	34 N Main St	Port Chester	NY	10573	914-937-9700
EPR Electronics	505 California Avenue	Middletown	NY	10940	914-343-1237
Triple S Elec Depot	1600 Broadway 8th Floor	New York	NY	10019	212-832-0072
Live Wire Audio	265 Park Ave	Mansfield	OH	44902	419-524-9005
Secret Services	4112 Gordon St	Cincinnati	OH	45223	513-541-2292
Advanced Audio	141 South Main Street	Marion	OH	43302	614-382-9932
Lentines Music	844 N. Main Street	Akron	OH	44310	330-434-3138
Sound Ideas Inc	3671 Karl Road	Columbus	OH	43224	614-263-5742
Coyle Music	915 Schrock Blvd	Columbus	OH	43229	614-842-4823
Buddy Rogers Service	6891 Simpson	Cincinnati	OH	43239	513-729-1950
Advanced Elect. Services	2303 Brookpark Rd	Cleveland	OH	44134	216-741-2230
Dr Music	1569 Chase Ave #5	Cincinnati	OH	45223	513-542-6111
Dayton Musicians Service	1819 Wyoming Street	Dayton	OH	45410	513-253-5377
Reineck Keyboard Service	6219 Sylvan Green	Sylvania	OH	43560	419-885-1075
Professional Technical	1483 W Syluania Ave Ste2	Toledo	OH	43612	419-476-1956
Stage Tech Inc	1166 A Steelwood Ave	Columbus	OH	43212	614-487-1111
Hicks Electronics Corp	3259 S Yale Ave	Tulsa	OK	74135	918-743-7813
Del City Music	2908 Epperly Drive	Del City	OK	73115	405-677-8777
Norman Music	317 W Gray	Norman	OK	73069	405-321-8300
Tulsa Guitar & Electronic	1417 South Harvard	Tulsa	OK	74112	918-742-4912
SureTech Electronic Serv	310 Garfield St. Suite 5	Eugene	OR	97402	503-687-8763
Southbound Sound	278 NW Garden Valley Blvd	Roseburg	OR	97470	503-672-7056
KMA Electronics	617 S.E. Morrison	Portland	OR	97214	503-231-6552
Inner Sound	1818 SE Division	Portland	OR	97202	503-238-1955
Keyboard Associates	1014-B Green Acres Rd	Eugene	OR	97408	503-343-1978
Musicians Elect Service	798 Biddle Street	Ardmore	PA	19003	215-896-7311
Osiecki Bros. Music	2426 Parade Street	Erie	PA	16503	814-453-6565
CB Electronics Inc	300 Wilmington	Chadds Ford	PA	19317	215-358-5675

AUTHORIZED ENSONIQ REPAIR STATIONS

Georges Music	1025 E Woodland Ave	Springfield	PA	19064	610-543-4050
Terrace Music Servicing	1415 Bunting St	Pottsville	PA	17901	717-544-9754
Spectra Sound	Rd #2 Box 2611	Spring Grove	PA	17362	717-229-2086
Cossas Keyboards & Sound	1330-32 Wyoming Avenue	Scranton	PA	18509	717-343-2002
Nelson Mendez	#9 Kilo 1.9 Rd 441	Aguada	PR	00602	809-868-4019
Keyboard Service	Vrb.Sta. Teresita Calle C	Ponce	PR	00731	809-844-8118
MY-Tech	8 Joan Road	Westerly	RI	02891	401-596-5135
Sims Music	1110 Saint Andrews Rd	Columbia	SC	29210	803-772-1185
Doc Tronics	120 Lann Circle	Lexington	SC	29073	803-359-7799
Express Music	159 South Pine St	Spartanburg	SC	29302	803-583-6768
Thomas Marketing Group	1400 E 39 Street N	Sioux Falls	SD	57104	605-332-8156
Ed Lowry Organ Service	223 St. Charles	Rapid City	SD	57701	605-343-1881
Haggertys Music Works	514 St. Joe Street	Rapid City	SD	57701	605-348-4801
Broadway Sound	2830 Broadway N.E.	Knoxville	TN	37917	615-637-1644
Sams Music	7103 B Crosswind Blvd	Brentwood	TN	37027	615-371-5000
Morrell Music Shop (KI)	510 E Center St	Kingsport	TN	37660	423-247-9891
Techstar Services	750 Cowan St Suite 9	Nashville	TN	37207	615-242-9528
Musical Instrument Serv	833 S. Highland Street	Memphis	TN	38111	901-327-0964
Morrell Music	2306 State Street	Bristol	TN	37620	615-764-2171
MUSITECH (TN)	6903 Glen Errol	Chattanooga	TN	37412	423-894-9771
Amro Music	2936 Poplar Avenue	Memphis	TN	38111	901-323-8766
Hermes Trading Co Inc	409 S Broadway	McAllen	TX	78501	210-682-4341
Keyboards of Texas	202 South 31st St	Temple	TX	76504	817-778-3181
Capital Music	6101 Burnett Rd	Austin	TX	78757	512-458-1933
Hermes Trading Co Inc	501 South 11th	McAllen	TX	78501	210-618-3344
BE Goetsch Music Co	222 E Kleberg	Kingsville	TX	78563	512-592-5464
A OK Music Repair	1514 Ahrens Drive	Houston	TX	77017	713-643-5397
Rich's Music	1007 Avenue C	Denton	TX	76201	817-566-3700
Century Music Systems Inc	3515 Sunbelt Drive N.	San Antonio	TX	78218	210-822-7306
Hermes Trading Co Inc	501 S 11th St	McAllen	TX	78501	210-618-5663
Strait Music	908 N. Lamar Street	Austin	TX	78703	512-476-6927
Randy's Music Mart	2600 Paramount Suite H-3	Amarillo	TX	79109	806-358-0131
Sound Vibrations	1638 S. Staples	Corpus Christi	TX	78404	512-884-9308
Musicmakers Austin	517-B S. Lamar	Austin	TX	78704	512-444-6686
Freeman Tuell Speaker Rep	7911 Ferguson Road	Dallas	TX	75228	214-324-1132
Alamo Music	425 N Main	San Antonio	TX	78205	210-224-1010
Crosswind Sound Inc	3501 Dime Circle #113	Austin	TX	78744	512-441-1631
Audio Electronics	9205 Skillman Suite 120	Dallas	TX	75243	214-349-5000
R.L.S. Electronic Service	5523 Richmond Ave	Houston	TX	77056	713-654-9217
Wagstaff Music Inc	206 E 6400 So	Murray	UT	84107	801-261-4555
Guitar City	470 N 1100 West	Centerville	UT	84014	801-292-8461
Stage Sound	103 8th St. SE	Roanoke	VA	24013	540-342-2040
Audio Light and Music	3301 N. Military Hwy.	Norfolk	VA	23518	804-853-2424
Backstage Inc.	310 W. Broad Street	Richmond	VA	23220	804-644-1433
Sound West Audio	2323 Tacoma Avenue South	Tacoma	WA	98402	206-272-1435
Music World Inc	1215 N Division	Spokane	WA	99202	509-328-2853
Bozotronics	525 Dexter Ave N	Seattle	WA	98109	206-622-4968
EFEX Electronixs & Repair	218 SW 153rd Street	Seattle	WA	98166	206-241-4852
Northwest Organ Service	3911 N Monroe	Spokane	WA	99205	509-747-7761
Rons Keyboard & Elec	747 S Fawcett	Tacoma	WA	98402	206-572-8633
Henris Music	511 West College Avenue	Appleton	WI	54911	414-739-9163
Big Music	7910 N 76th St	Milwaukee	WI	53223	414-355-8888
Morgan Music	2405 E Clairmont	Eau Claire	WI	54701	715-834-7177
Music Service Center	900A South Foster St	Merrill	WI	54452	715-536-8283
Music Service Center	Po Box 1	Merrill	WI	54452	715-536-8283
Henris Music	500 S. Military Avenue	Green Bay	WI	54303	414-494-4724
Pied Piper Music	1200 3rd Avenue	Huntington	WV	25701	304-529-3355
Squarewave Audio	245 S Montana	Casper	WY	82609	307-266-1509

IN CANADA

Kaysound Imports	2165 46th Ave	Lachine, QUE	H8T 2P1	514-633-8877
Long & McQuade	1505 17th Ave SW	Calgary, ALB	T2T 0E2	403-244-5555
Ranger Audio	3516 1st Street	Calgary, ALB	T2E 3C9	403-277-1615
Edmonton Audio Works	17310 108th Ave	Edmonton, ALB	T5S 1E8	403-483-2017
Kinetic Sound	4131 Fraser Street	Vancouver, BC	V5V 4E9	604-876-4847
McPherson Micro	416 McDermot Ave	Winnipeg, MAN	R3A 0A9	204-947-9389
Mytronics	2050 Ellesmere Rd #3	Scarborough, ON	M1H 3A9	416-289-0074
Steve's Service	138 Peter Street	Toronto,ON	M5V 2H2	416-593-8889

AUTHORIZED ENSONIQ REPAIR STATIONS

OTHER COUNTRIES

ARGENTINA	Arte Musical S.A.	Talcahuano 218, 1013, Buenos Aires	541-374-8049
AUSTRALIA	Electric Factory	188 Plenty Road, Preston, Victoria	613-9-480-5988
AUSTRIA	Soundware Audio Team	Moostr 123, 5020 Salzburg	43-66-2-824679
BALTIC STATES	A & T Trade Inc.	91 Brivibas Street, Riga, Latvia LV-1047	371-237-1141
BARBADOS	A & B Music Supplies	Prince Alfred Street, Bridgeton	809-427-5384
BENELUX	EBT International	Kapelstraat 12, 5316 BG Delwijnen	31-4185-52106
BERMUDA	Pianos Plus	129 Front St, East Hamilton	809-295-2530
BOLIVIA	Audiomusica	Casilla 299, Correo 22, Santiago	56-2-225-2233
BRAZIL	Pride Internacional	Ave General Ataliba Leonel, 93-cj 65/66, Sao Paolo	55-11-950-1652
CHILE	Audiomusica	Casilla 299, Correo 22, Santiago	56-2-225-2233
COLOMBIA	Sonygraf	Carrera 12 #90-20 Of. 401, AA 59786, Bogota	571-622-8515
COSTA RICA	Albion S.A.	PO Box 666-2150, Moravia, San Jose 2150	506-235-9330
CYPRUS	Empire Music House	Nikis Ave, PO Box 5604, Nicosia	357-249-0472
CZECH	Praha Music Center	Soukenicka 20, 112 27 Praha 1	422-2481-0970
DENMARK	New Musik	Vesterport 8, DK-8000 Aarhus C	4586-190899
EGYPT	Alpha Audio	6 Mahmoud Hafex, Safir Sq, Heliopolis, Cairo	202-243-7119
FINLAND	Musiikki Hellas OY	PL 53, Fin-05201, Rajamaki	358-0-2901021
FRANCE	IML	4 Rue Maurice Audibert, Saint-Priest	33-78-204030
GERMANY	Soundware Audio Team	Paul-Ehrlich St 28-32, Bld G-3, 6074 Rodermark	49-6074-89150
GREECE	Ph. Nakas Music House	147 Skiathou Street, 11255, Athens	301-228-2160
HONG KONG/CHINA	Tom Lee Music Co.	30 Canton Rd, Tsimshatsui, Kowloon	852-2737-7688
ICELAND	RIN Wholesale	Frakkastig 16-101, Reykjavik	354-551-7692
ISRAEL	R.B.X. International	Dizengoff Center, 64 Dizengoff St, Tel-Aviv	972-3-6298251
ITALY	sisme s.p.a.	S. Statale Adriatica 34, 60028 Osimo-Scalo, Ancona	39-71-781-9666
JAPAN	ENSONIQ JAPAN	Nishi-Shinjuku Toyokuni 1F, 2-5-8 Hatsudai, Tokyo	813-5351-1401
KOREA	Han Dok Piano Co.	81-2 Yunhidong, Sudaemoon-Ku, Seoul	822-332-5556
KUWAIT	Easa Husain Al-Yousifi	Abdullah Al Salim St, 13002 Kuwait	965-571-9499
MAURITIUS	Robert Yip Tong Ent.	Yip Tong Centre, 30 Dr. Joseph Riviere St, Port-Louis	230-242-8629
MEXICO	Hermes International	830 N. Cage, Pharr, Texas, USA	210-781-8472
NEW ZEALAND	Electric Factory NZ	163 Archers Road, Glenfield, Auckland	649-443-5916
NORWAY	Norsk Musikk	Bergensgt 26, 0468 Oslo 4	47-22-235680
PAKISTAN	NRH Electronics	#10-11 Naseem Ctr, Sohrab Katrak Rd,Saddar,Karachi	9221-573113
PHILLIPINES	Blue Chip Sales	173 Wilson St, San Juan, Metro Manila	63-2-706138
POLAND	Mega Music Ltd.	48 Gdansk ul Fultona 5, 80-172	48-58-487411
PORTUGAL	Americo Nogueira, LDA	Rua Alto das Torres, 893, 4400 Vila Nova de Gaia	351-2-2004616
RUSSIA	A & T Trade Inc.	B.Gnezdnikovsky Per #10, Moscow 109003	7095-229-7516
SAUDI ARABIA	Halwani Audio	PO Box, 2865, Dammam, 31461	966-3-898-0405
SINGAPORE	Swee Lee Co.	Block 231, Bain St #03-23, Bras Basha Complex 0718	65-336-7886
SLOVENIA	Nova d.o.o.	Cesta v Gorice 4, 61111 Ljubljana	38-661-263-260
SOUTH AFRICA	MidiKing Technology	110 Palliser Rd, Eastleigh Ridge, Edenvale 1610	2711-609-1321
SPAIN	Ventamatic	c/ Corcega 89 entlo., 08029 Barcelona	34-3-430-9790
SWEDEN	Poly-Sonic	Kraketorpsgatan 20, S-431 53 Molndal	46-31-706-9050
SWITZERLAND	SDS Music Factory AG	Hohlstrasse 534, CH-8048 Zurich	41-1-434-2270
SYRIA	Yazigi & Co.	Naser Street, Sati Bldg No. 13, Damascus	963-11-221-5583
TAIWAN	Sea Power	7F. #36 Wo Chung 7 Rd, Wuku Taipei County	886-2298-2688
THAILAND	Music Concepts	4/1-2 World Trade Ctr, 3rd Fl, Rajdemri Rd, Bangkok	662-255-6448
TURKEY	Zuhal Musik	Tunel Gecidi Is Hani, B Block No.11, Tunel, Istanbul	90-212-249-8511
U.A.E.	A.K.M. Music Centre	PO Box 8827, Abu Dhabi	9712-792-734
UNITED KINGDOM	Key Audio Systems	Robjohn's Road, Chelmsford, Essex, CM1 3AG	44-1245-344001
VIETNAM	Vistar Company Ltd.	42 Nha Chung Street, Hoan Kiem District, Hanoi	844-2-43058
WEST INDIES	The Music Shop	105 Frederick Street, Port of Spain, Trinidad	809-622-3060
	Pro Line Music	Rem Brands Plein #8, Amsterdam Ctr, St. Maarten	599-5-25067
	Universal Trading Co.	Colon Shopping Ctr #126, Curacao, Netherlands	599-9-624840

ENSONIQ Corp Worldwide Headquarters 155 Great Valley Parkway,
PO Box 3035 Malvern PA 19355-0735 (610) 647-3930 FAX (610) 647-8908
ENSONIQ products are available through Authorized ENSONIQ Dealers throughout the world.

"INSTRUCTIONS PERTAINING TO A RISK OF FIRE, ELECTRIC SHOCK, OR INJURY TO PERSONS"

IMPORTANT SAFETY INSTRUCTIONS

WARNING—When using electric products, basic precautions should always be followed, including the following:

1. Read all the instructions before using the product.

2. Do not use this product near water - for example, near a bathtub, washbowl, kitchen sink, in a wet basement, or near a swimming pool, or the like.

3. The product should be located away from heat sources such as radiators, heat registers, or other products that produce heat.

4. The product should be connected to a power supply only of the type described in the operating instructions or as marked on the product.

5. The product should be serviced by qualified service personnel when:

    a. Objects have fallen, or liquid has been spilled into the product; or
    b. The product has been exposed to rain; or
    c. The product does not appear to operate normally or exhibits a marked change in performance; or
    d. The product has been dropped, or the enclosure damaged.

6. Do not attempt to service the product beyond that described in the user-maintenance instructions. All other servicing should be referred to qualified service personnel.

SAVE THESE INSTRUCTIONS

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