

parallel effects processor DP/4+

Reference Manual

Version 2.0

DP/4+ Reference Manual:

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USA

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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.

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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/4+ 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.

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Part # 9310 0177 01 - D Model # MM-86

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Welcome!

Congratulations, and thank you for purchasing the ENSONIQ DP/4+ Parallel Effects Processor. The DP/4+ creates 24-bit digital effects using four independent processors, and features four independent inputs and outputs with full internal mixing capabilities. The DP/4+ is equally at home in a professional recording studio, home studio, guitar rig, MIDI setup, or PA system.

The Effects

The ENSONIQ DP/4+ Parallel Effects Processor has over 50 high fidelity fully programmable digital effect algorithms. Reverb, chorusing, flanging, delay, distortion, pitch shifting and an assortment of other programs are provided with dynamic control over most of the settings. There are 400 effect presets; 200 ROM (Read Only Memory) and 200 additional RAM (Random Access Memory) presets for you to edit or store your own creations.

Parallel Processing

While other multi-effects processors only allow one input signal to be "effected," the DP/4+'s four-in, four-out design permits stereo processing of four parallel channels (multi-processing). There is only one user interface, but up to four different input signals can each go to a separate internal signal processor. Multiple inputs and outputs also allow for special types of effects, like vocoding and ducking.

The DP/4+ can be used as one huge effects box, two stereo-in effects boxes, three effects boxes, or four separate effects boxes. The routing between the four processing units is completely programmable, allowing for any combination of serial and parallel effects. The DP/4+ also offers paths to feedback the signal, and side-chain capability. The variable architecture and rich assortment of algorithms provide for unusual effect structures not found in fixed routing systems. The unique output mixing capability can also save you mixer effect return channels by mixing the stereo outputs of the four effects units down to a single stereo pair (outputs 1 and 2).

The DP/4+ 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/4+, four 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, a CV Pedal, etc.

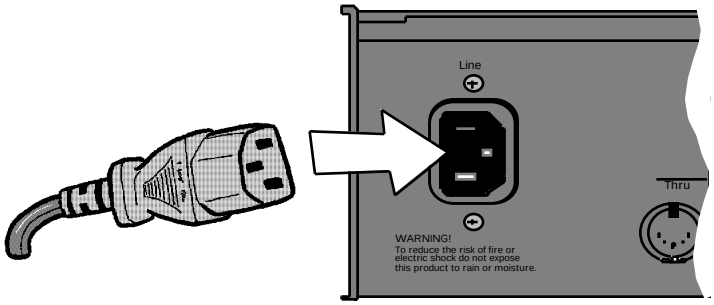
Clean Up and Maintenance

Clean the exterior of your DP/4+ 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/4+ is thoroughly dry before turning on the power. Never use alcohol, benzene, volatile cleaners, solvents, abrasives, polish or rubbing compounds.

Thank you again for choosing ENSONIQ. Enjoy the music!

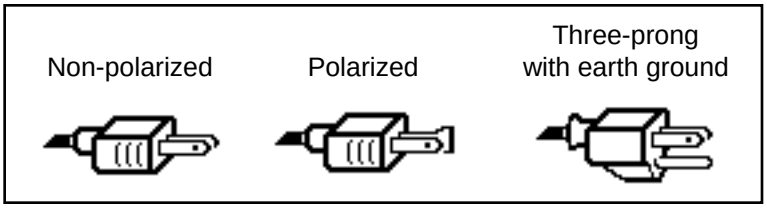
Power

Insert the line cord into the line receptacle on the rear panel of the DP/4+. Plug the other end of the cable into a grounded AC outlet (the proper voltage for your DP/4+ is listed on the Serial Number label on the rear panel). Turn the DP/4+ 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/4+," and then go to Select mode (Select LED on). If you travel, remember the DP/4+ will only operate on the listed voltage.



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 dangerous 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/4+ 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/4+ 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/4+

Temperature Guidelines

The DP/4+ contains a substantial amount of computerized and electronic circuitry that can be susceptible to damage when exposed to extreme temperature changes. When the DP/4+ 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/4+:

- Avoid leaving the DP/4+ in temperatures of less than 50 degrees Fahrenheit or more than 100 degrees Fahrenheit.
- When bringing the DP/4+ 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/4+ inside a vehicle exposed to direct sunlight.

Rack Mounting Guidelines

Because the DP/4+ operates with an internal transformer, there is a certain amount of heat generated by this unit. For better reliability, we recommend that you do not install this unit beneath devices that are sensitive to heat, or above power amps, tube equipment, or other rack-mount units that emit a lot of heat. We recommend leaving a space above the DP/4+, as well.

Amplifying Your DP/4+ Through a Home Stereo System

If you are thinking about amplifying your DP/4+ 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/4+ (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 **Level (dBu)** switch to the -10 dBV position, set the **Output Knobs** to around the 12 o'clock position, and try to keep your levels on the conservative side.

Powering Up Your DP/4+ 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/4+ to receive MIDI information from a keyboard/sequencer, you would turn the keyboard on before the DP/4+. 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.

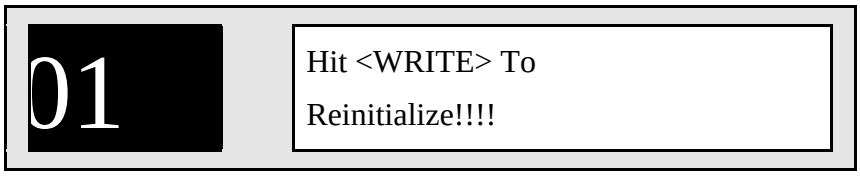
Reinitializing the DP/4+

If your DP/4+ is behaving in peculiar ways (the display is showing alphanumeric characters that shouldn't be there or unexplainable Unexpected Event messages) and a soft reset (or turning the DP/4+ power off and then on again) won't cure the problem, try reinitializing the DP/4+.

> WARNING! THIS PROCESS WILL ERASE ALL RAM PRESETS! The 200 User Presets in the internal memory (RAM) 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/4+, or manually write down the relevant parameters using the provided Preset Parameter Worksheet (or a photo-copy). If you fail to do so, you may accidentally lose the presets you've created.

To reinitialize the DP/4+

1. While holding down the {SYSTEM"MIDI} button,
2. Press the {B} button.
3. Press the {>} button. The display shows:



   Press the {CANCEL} button to quit *without* reinitializing the system, or
4. Press the {WRITE} button to reinitialize the DP/4+. Remember that by doing this you will replace *all* of the RAM Preset data in the DP/4+, and *all* System•MIDI parameters will be reset to their default range!

If reinitializing the DP/4+ does not correct the problem, then contact an Authorized ENSONIQ Repair Station.

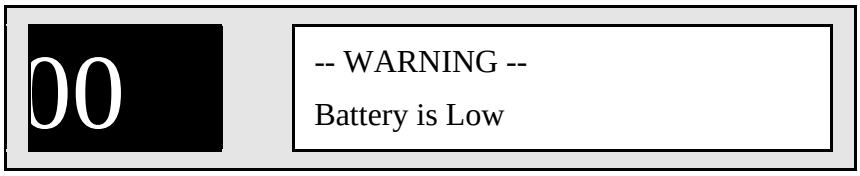
Note: If the DP/4+ is sitting in an infinite loop of system errors (the display is continually cycling through errors), press the {SYSTEM"MIDI} button to escape this state.

Note: In the unlikely event of a system malfunction, 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/4+ "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/4+ memory intact is located inside the DP/4+, and when it becomes discharged, the battery must be replaced by an Authorized ENSONIQ Repair Station.

The battery that came in your DP/4+ is good for up to five years. You will know when it needs replacing, because the DP/4+ 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 will only appear for a short time, and then you can commence with normal operation. Then, make sure that all custom RAM configs, presets and system parameters are saved, and take the DP/4+ to an Authorized ENSONIQ Repair Station as soon as possible to have the battery replaced.

For more information about saving DP/4+ data, see *Section 5 — Storage*.

Available Options for Your DP/4+

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/4+. The CVP-1 Pedal makes a great "wah wah" pedal.
- **SW-10 Dual Damper Foot Switch** — Because the DP/4+ has two stereo Foot Switch jacks, you can use two of these two pedal, piano-type foot switches, for ultimate control! The pedals can be programmed independently to act as a bypass effect switch, offering two separately programmable modulation sources or increase/decrease 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/4+.

If you are considering a foot switch for the DP/4+, 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

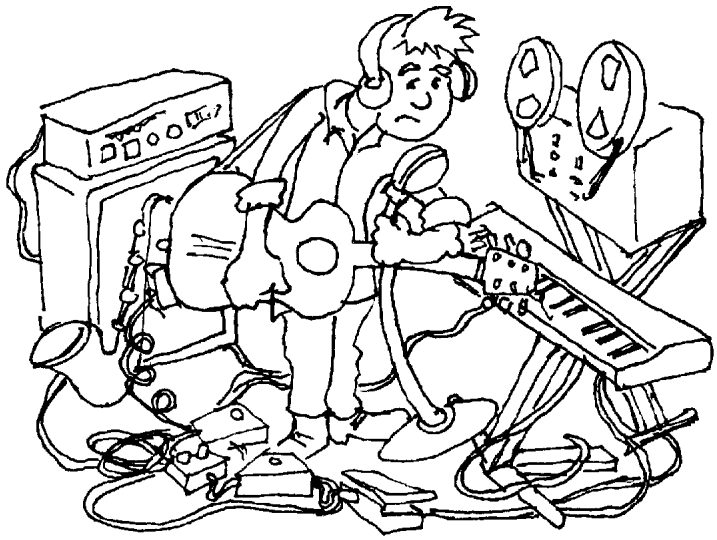
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EQ

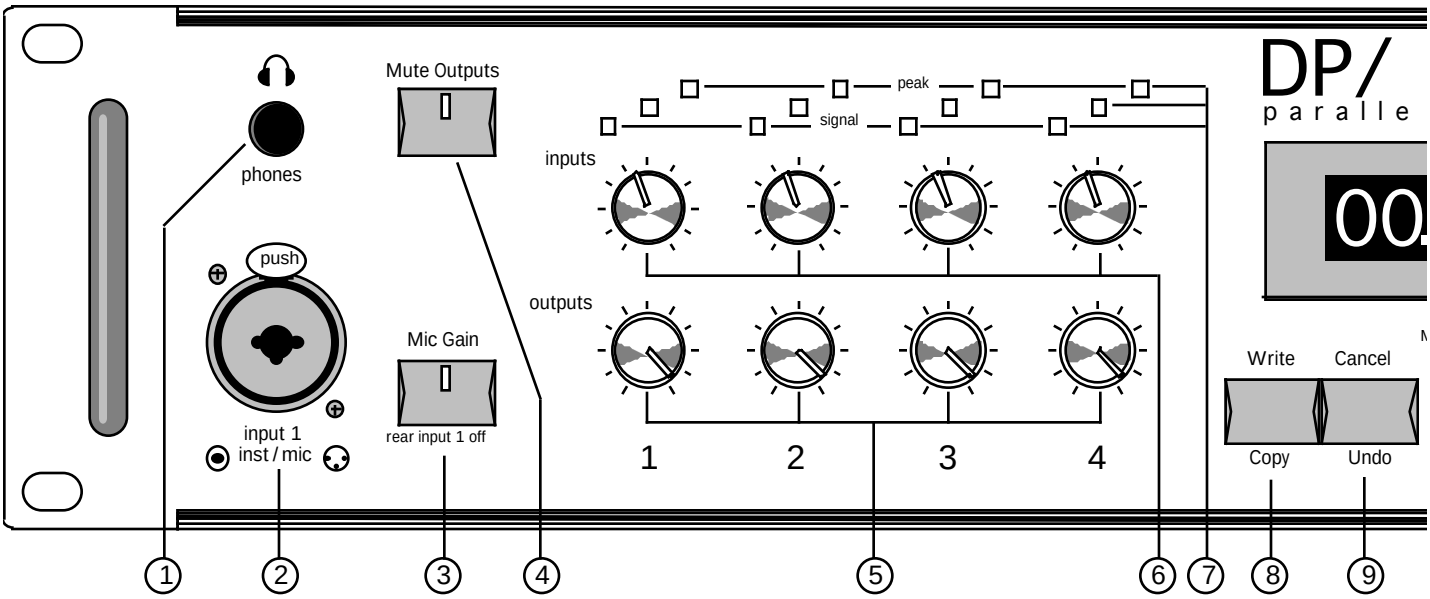
For subscription rates and more information call 1-212-213-3444

Section 1 — Controls & Basic Functions

There must be
something better…



This section provides an introduction to the DP/4+'s many controls and rear panel connections; a conceptual overview of the system; a guide to selecting DP/4+ 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/4+.



Front Panel Controls

1. Phones

Plug headphones into this 1/4" stereo jack to listen to the DP/4+ in stereo. The signal going to this jack is from the sum of all four rear outputs, even if they are not connected. The 4 rear outputs are mapped to the stereo headphone as follows: 1 and 3 are mostly to the left; 2 and 4 are mostly to the right. Headphone volume is controlled by the **Output Knobs**. 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 Knobs** accordingly — high output volume levels could damage your hearing.

2. Input 1 — (inst/mic)

This combination balanced XLR mic/unbalanced 1/4" mono input jack is for connecting a guitar, microphone, 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.

3. Mic Gain — (rear input 1 off)

This activates the XLR Mic (microphone) input and supplies mic gain to both the Mic and Instrument inputs.

When set to Instrument (LED off):
The XLR Mic Input is disabled.
The 1/4" front panel Instrument input is enabled.
Rear panel Input 1 is overridden and disabled when the 1/4" front panel Instrument input is connected.

When set to Mic (LED on):
The XLR Mic Input is enabled.
Mic Gain will be supplied to the XLR Mic input.
The 1/4" front panel Instrument input is enabled.
Mic Gain is not supplied to the 1/4" front panel Instrument input.
Rear panel Input 1 is overridden and disabled.

4. Mute Outputs — (inst/mic)

This button is used to mute the rear panel output jacks. When the LED is on, the output jacks are muted, but the headphone output remains active.

5. Output Knobs

The four **Output Knobs** control the output level of each channel. If separate signals are being processed in the ENSONIQ DP/4+, these knobs will control the "mix-down" volumes. The maximum output level is +19 dBu.

6. Input Knobs

These four input knobs control the gain applied to the input signals. The input circuitry is designed to work with signals ranging from -34.6 dBV to +22 dBu. Use these knobs to set each input to the optimal level for the signal you are feeding into it.

7. Signal/Peak LEDs

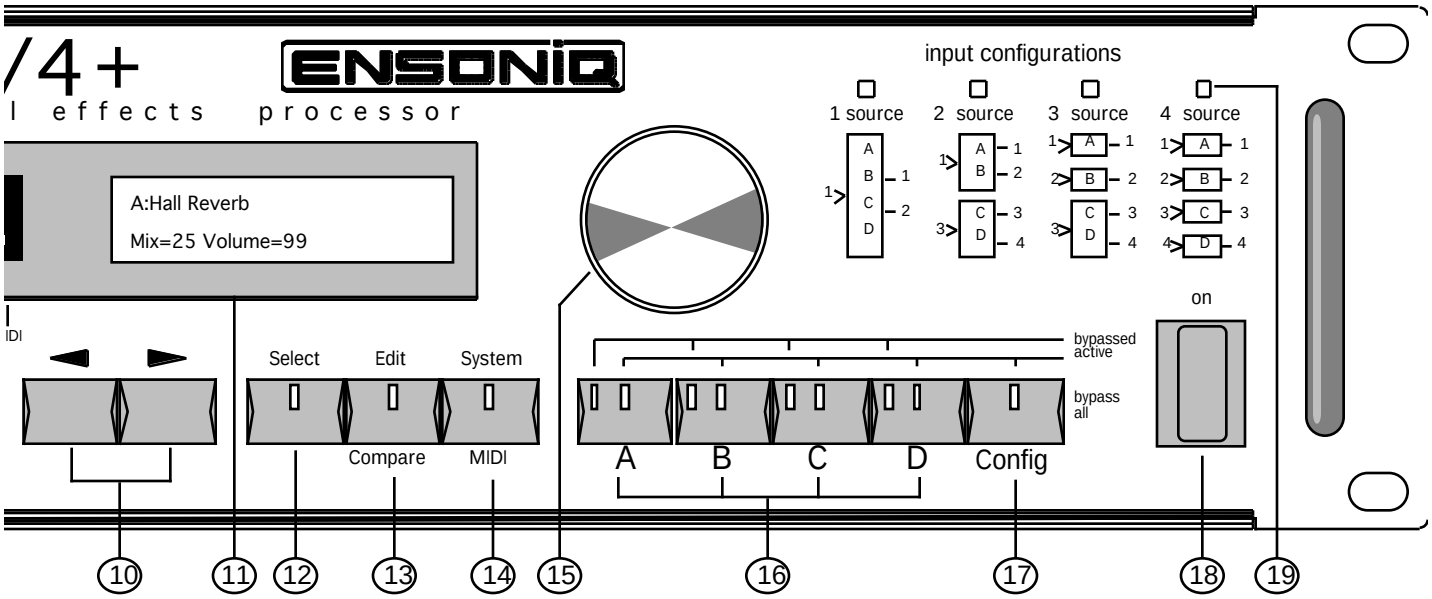
The three LEDs above each knob indicate the level of the input signal being fed into the Analog-to-Digital Converters (ADCs).

- 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 middle LED (yellow) will light at -12dB.
- The Peak LED (red) will light when the incoming signal reaches -6dB below the ADC clipping point.

For optimal level, adjust the **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.

8. Write•Copy Button

The {WRITE"COPY} button is used to save or copy presets to the DP/4+'s internal RAM memory.



9 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.

10. Left and Right Arrow Buttons

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

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

The MIDI Message Indicator (a little red dot in the LED Display) lights when any MIDI events are received; useful for troubleshooting MIDI connections.

12. 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.

13. Edit Button

This is used to edit preset parameters, edit preset titles and save presets.

14. System•MIDI Button

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

15. Data Entry Knob

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

16. Unit Buttons

The four Unit buttons (**A**, **B**, **C**, and **D**) correspond to the four separate signal processors in the DP/4+. Use these buttons to activate a particular Unit for selecting presets or editing parameters. The yellow LED above 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.

17. Config Button

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

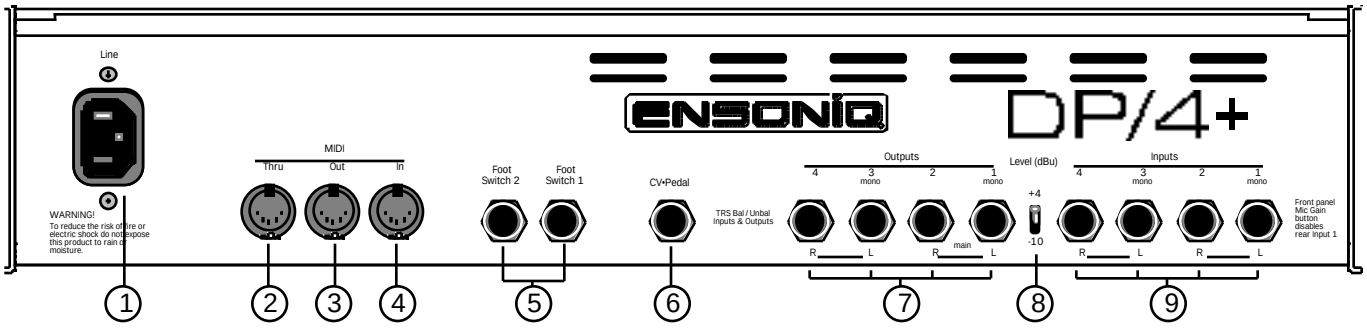
18. Power

The power switch turns the DP/4+ on and off.

19. Input Configuration LEDs

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

Rear Panel Connections



1. Line

The supplied line cord is connected here.

2. MIDI Thru

"Passes on" all MIDI (Musical Instrument Digital Interface) information received by the DP/4+ to other devices. Information generated by the DP/4+ itself does not go to this jack — the Thru jack merely echoes what comes in at the MIDI In jack.

3. MIDI Out

Sends out MIDI information to other instruments and computers when the System•MIDI parameter "63 Send MIDI PrgChg + Controllers" is set to "ON."

4. MIDI In

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

5. 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/4+.

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

6. CV•Pedal

This jack is for connecting an ENSONIQ Model CVP-1 Control Voltage Foot Pedal, which is assignable as a modulator to parameters within the DP/4+.

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.

7. Output Jacks

The four ground compensated output jacks can be configured in numerous ways. Because the DP/4+ offers fully programmable output control, you can have almost any combination ranging from a single mono output to four mixed stereo signals.

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

8. Level Switch

This switch toggles between +4 dBu and -10 dBV (this affects the rear panel input and output jacks only). Because this switch can accommodate a broader range, it allows an improved signal-to-noise ratio.

9. Input Jacks

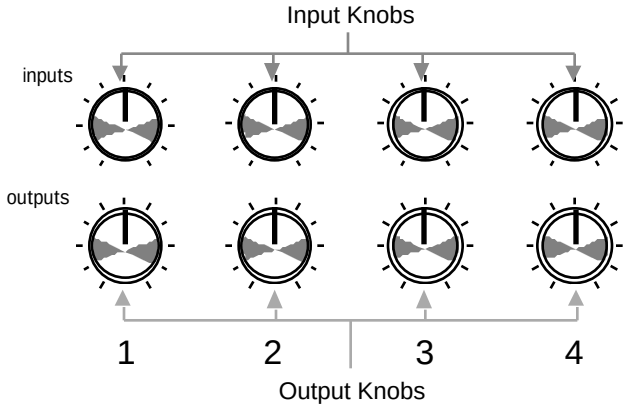
These four balanced input jacks are truly independent inputs and can be used in a 1 source, 2 source, 3 source, or 4 source configuration.

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

DP/4+ RULES

Setting Levels

The input and output levels affect the volume of audio signal going into and coming out of the DP/4+, and are controlled by the two rows of four knobs on the left hand side of the front panel. The top row controls the input levels for Inputs 1 to 4, the bottom row controls the output levels for Outputs 1 to 4.



Setting the Level Switch

The **Level (dBu) switch** (found on the rear panel of the DP/4+) toggles between +4 dBu and -10 dBV (this affects the rear panel input and output jacks only). Because this switch can accommodate a broad range, it allows an improved signal-to-noise ratio. This switch should be set to match the gain structure of your mixer/amplifier, or particular application. Consult your product's manual to determine which is appropriate.

To set the input level(s):

1. With your connections made, send a signal into the DP/4+ 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** 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 any additional inputs you have connected.

To set the output level(s):

1. With your connections made and the input level properly set, send a signal into the DP/4+ and slowly turn the corresponding **Output Knob**(s) clockwise. If you are using a stereo output, use both outputs 1 and 2. You should begin to hear signal coming through the DP/4+ 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/4+ 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.
4. Repeat this process for any additional outputs you have connected.

The DP/4+ circuitry is designed so that if the **Input** and **Output Knobs** are set to 12:00 (the lines in the knobs are pointing up), and you have an input signal of +4 dBu, a +4 dBu signal will go out of the DP/4+. With the knobs at 12:00, and an input signal of -10dBV, the output of the DP/4+ will also be -10 dBV. With these settings, any incoming signals slightly above +4 dBu or -10dBV respectively, will result in clipping.

Using Headphones with the DP/4+

Headphones can be used with the DP/4+ when connected to the front panel 1/4" stereo **Phones** jack to listen to the DP/4+ in stereo. The signal going to this jack is from the sum of all four rear outputs, even if they are not connected. The 4 rear outputs are mapped to the stereo headphone as follows: 1 and 3 are mostly to the left; 2 and 4 are mostly to the right. The outputs are not routed hard left and right, to provide a "mixed stereo" signal:



Headphones

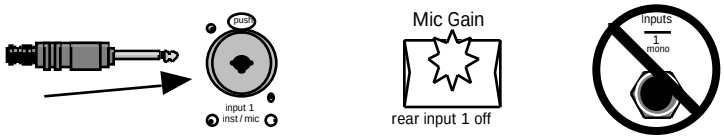
Headphone volume is controlled by the **Output Knobs**. Plugging headphones into the **Phones** jack does not turn off the audio in the outputs. Remember to turn up the **Output Knobs** in pairs (1 and 2, 3 and 4) in order to preserve the stereo image. Depending on the rear panel connections, you may need to turn up Outputs 3 and 4.

☞ **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 Knobs** accordingly — high output volume levels could damage your hearing.

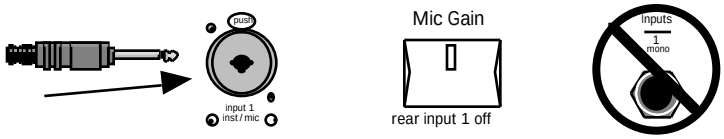
The Input 1 Jack — Front Panel vs. Back Panel

The following diagrams show how the front panel **Input 1** jack (along with the {MIC_GAIN} button) affects the rear panel **Input 1** jack.

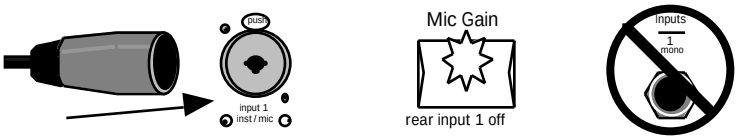
With a 1/4" phone plug connected to the front panel **Input 1** jack, and the Mic Gain LED on:
The front panel **Input 1** works — the rear panel **Input 1** doesn't work:



With a 1/4" phone plug connected to the front panel **Input 1** jack, and the Mic Gain LED off:
The front panel **Input 1** works — the rear panel **Input 1** doesn't work:



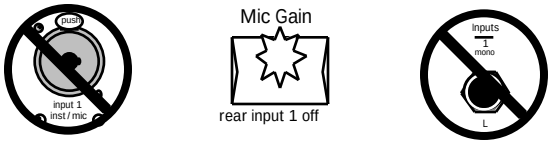
With an XLR cable connected to the front panel **Input 1** jack, and the Mic Gain LED on:
The front panel **Input 1** works — the rear panel **Input 1** doesn't work:



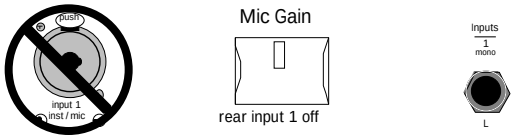
With an XLR cable connected to the front panel **Input 1** jack, and the Mic Gain LED off:
The front panel **Input 1** doesn't work — the rear panel **Input 1** works:



With nothing connected to the front panel **Input 1** jack, and the Mic Gain LED on:
The front panel **Input 1** doesn't work (nothing's plugged into it) — the rear panel **Input 1** doesn't work:

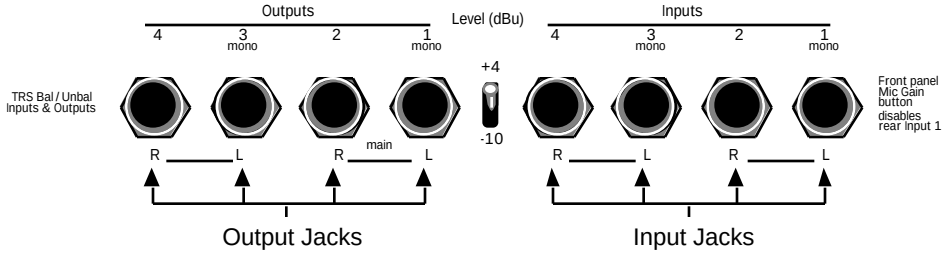


With nothing connected to the front panel **Input 1** jack, and the Mic Gain LED off:
The front panel **Input 1** doesn't work (nothing's plugged into it) — the rear panel **Input 1** works:



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/4+ employs extensive *automatic switching* on each stereo pair of inputs and outputs. That is:

- Normally, Inputs 1 and 2, and Inputs 3 and 4 are treated as stereo inputs. However, if nothing is plugged into 2 or 4, Inputs 1 and 3 will work as mono inputs and will also provide signal to Inputs 2 and 4 respectively.

  Note: In some cases, you may not want to have the mono signal plugged into Inputs 1 and/or 3 sent to Inputs 2 and/or 4. To send a discrete mono signal to Input 1 and/or 3, connect a "dummy" cable into the Input 2 and/or 4 jack (a dummy cable is just a standard balanced/unbalanced cable that is not connected to any external device).

- Similarly, Outputs 1 and 2, and Outputs 3 and 4 are normally stereo outputs. If nothing is plugged into Outputs 2 or 4, however, the stereo signal will be summed to mono and sent to Outputs 1 and 3 respectively.
- If nothing is plugged into Output 3, the stereo signals from outputs 3 and 4 will be summed with the stereo signal from outputs 1 and 2 *before* the automatic switching circuit described above.

  Note: In some studio applications (such as when using a patch bay), you may want to have outputs 3 and 4 connected, and still have the stereo signals from outputs 3 and 4 summed with the stereo signal from outputs 1 and 2. There is a parameter in System•MIDI mode that allows you to accomplish this (parameter #60, Mix Output 3/4 onto 1/2). See *Section 4 — System•MIDI* for more information about this parameter.

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.

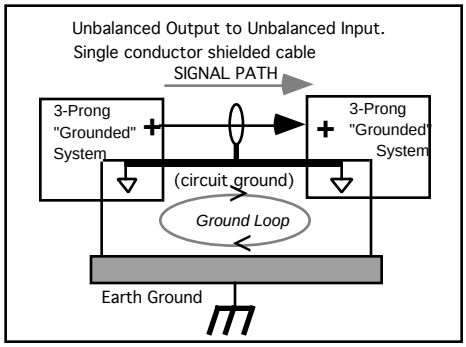


FIG. 1

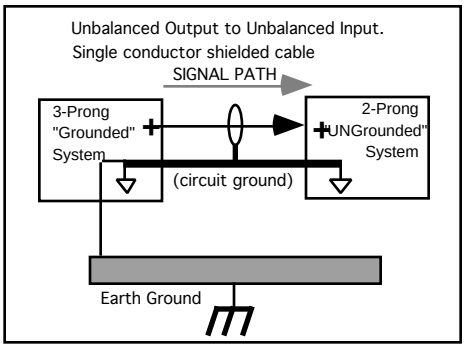


FIG. 2

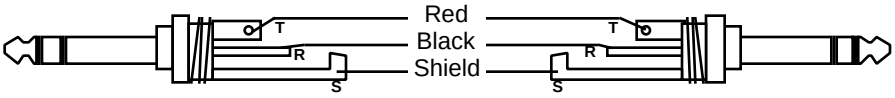
Fig. 1 shows a system interconnection where a ground loop can exist. Fig. 2 shows a system interconnection where a ground loop does NOT exist.

The DP/4+ 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/4+ 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:

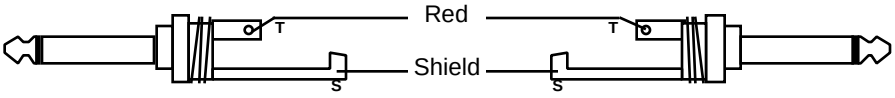
1. When a standard balanced cable is used from the preceeding piece of equipment (i.e., a standard stereo cable).

Standard Balanced Cable



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

Standard Unbalanced Cable

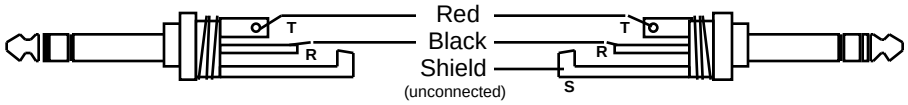


This does not mean there will always be an input ground loop problem, just the possibility.

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.

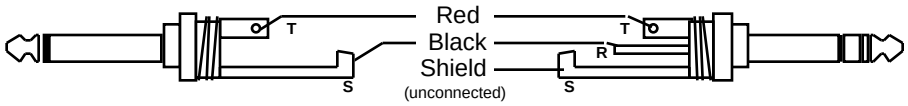
Custom Balanced Cable (to eliminate input ground loop)



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/4+ input connector. The two tips connect normally.

Custom Unbalanced Cable (to eliminate input ground loop)



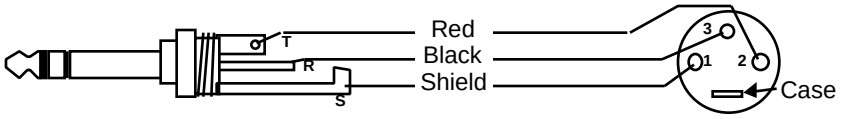
OTHER 1/4" output (unbalanced) ENSONIQ 1/4" input (balanced)

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/4+

The DP/4+ 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

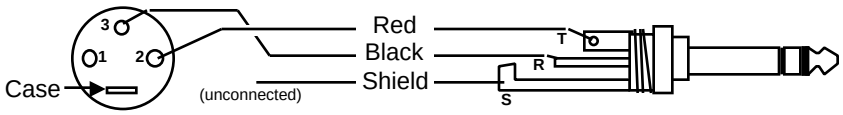


ENSONIQ 1/4" output (ground compensated) 3-Pin XLR male input (balanced)

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:

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

Custom XLR to 1/4" Balanced Cable

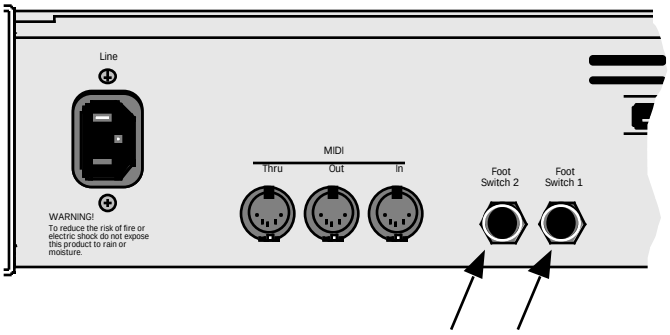


3-Pin XLR female output (balanced) ENSONIQ 1/4" input (balanced)

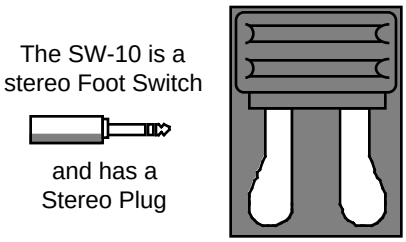
2. Use an audio isolation transformer.

If all audio equipment adopted this 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/4+ is the ENSONIQ Model 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.



SW-10

Warning!
The use of single (mono) foot switches is not recommended, and can affect the operation and performance of the DP/4+.

If you are considering using a foot switch, we strongly recommend the ENSONIQ SW-10 Dual Foot Switch.

About Mono Foot Switches

The DP/4+ 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 will act 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/4+ require two simultaneous button presses, and will not work properly because the DP/4+ 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/4+ will assume that *two* button presses (the left sides for each foot switch) are continually engaged, and *the DP/4+ 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/4+ 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/4+ power is switched on, you will briefly see "Button #15" in the display.

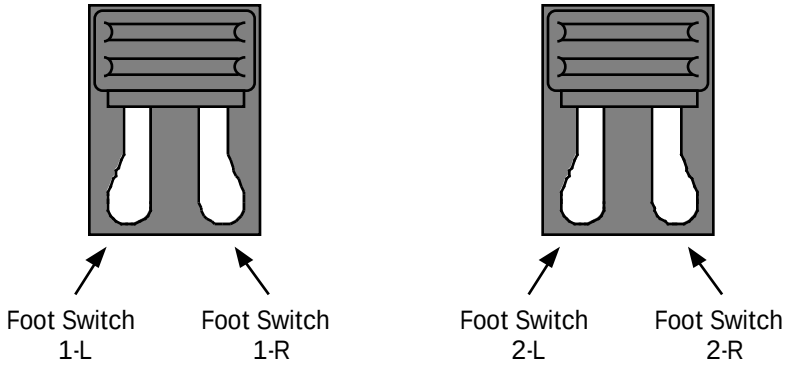
If you must use a mono foot switch, please consider performing one of the two modifications explained in "HOT MODS," found later in this section.

Four-On-The-Floor

An Application For Using Two Stereo Foot Switches to Bypass Effects

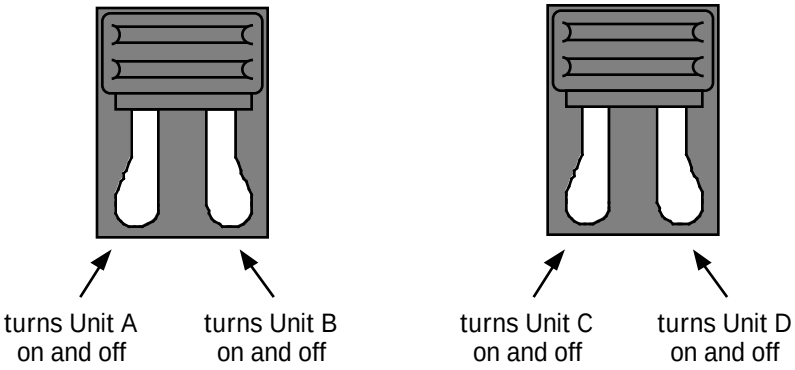
The DP/4+ allows you to employ two stereo foot switches (such as the ENSONIQ SW-10) to turn the DP/4+'s four effects processors on and off. To set up your foot pedals:

1. Connect one foot pedal to the **Foot Switch 1** jack on the DP/4+'s back panel.
2. Connect the other foot pedal to the **Foot Switch 2** jack.
3. Physically place the two foot pedals side-by-side, with the first to the left of the other.



4. Press the {SYSTEM"MIDI} button on the DP/4+'s front panel.
5. Press the {>} or {<} button until the large red number reads "06" and the top line of the display shows "Unit A Bypass=."
6. Turn the large silver **Data Entry Knob** to dial in "Ftsw 1-L Toggle."
7. Press {>} until the red number shows "13" and the display reads "Unit B Bypass=." Dial in "Ftsw 1-R Toggle."
8. Press {>} until the red number shows "20" and the display reads "Unit C Bypass=." Dial in "Ftsw 2-L Toggle."
9. Press {>} until the red number shows "27" and the display reads "Unit D Bypass=." Dial in "Ftsw 2-R Toggle."

Each foot switch is now assigned to its own processor:



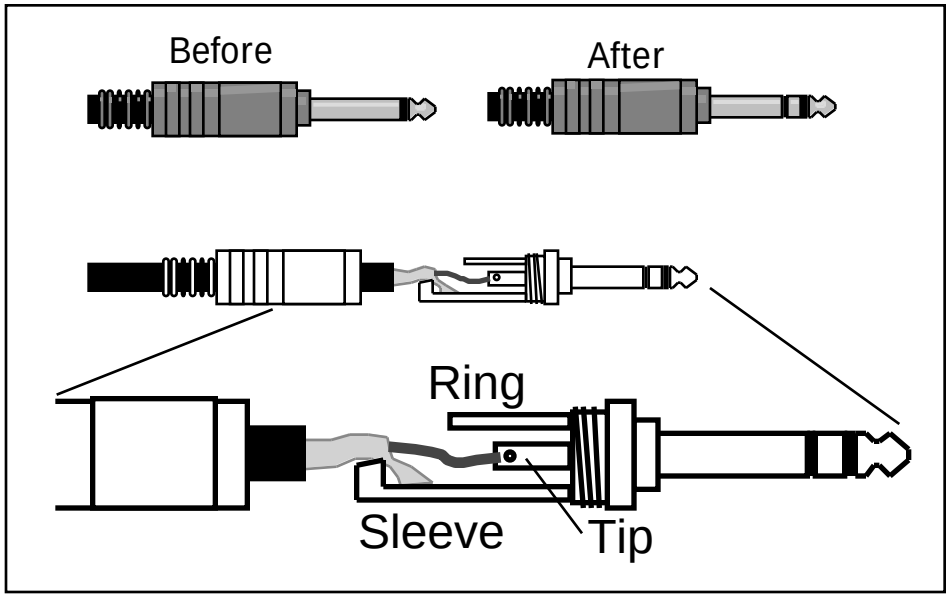
Note: Whether it's the effect being added to a sound (bypass/unbypass) — or the sound itself (kill/unkill) —is determined by each Unit's "(b)ypass and (k)ill" setting. See the description of the Bypass/Kill parameter in *Section 3—Config Parameters* for more information.

HOT MODS!

Although they are not recommended, mono foot switches such as the ENSONIQ Models 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:

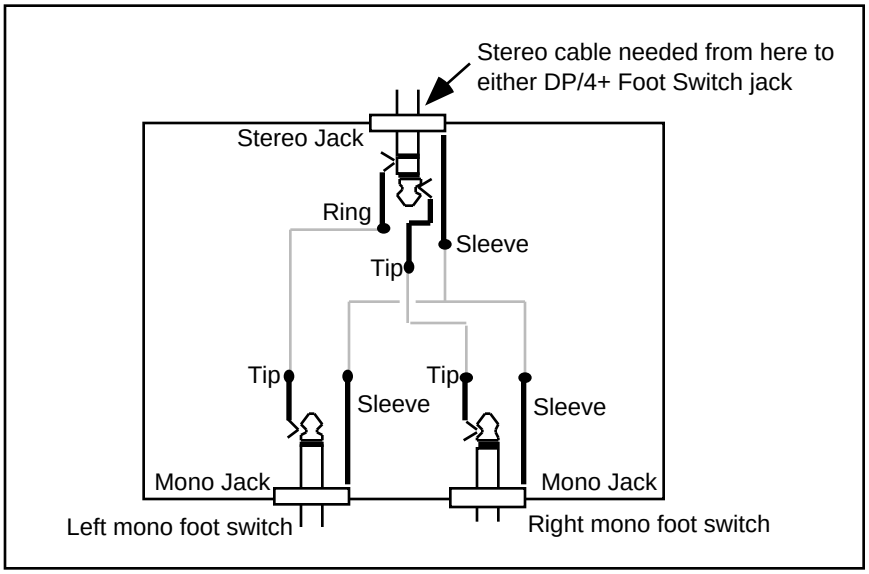
- soldering iron
- solder
- wire cutters
- 1/4" stereo plug

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 you it will allow you to make two mono foot switches function as one stereo foot switch.



Tools / supplies required:

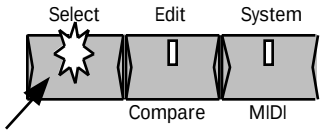
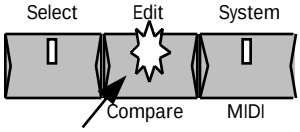
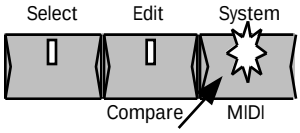
soldering iron
solder
wire cutters
drill and drill bits
one plastic housing assembly (must be large enough to mount three jacks)
one stereo jack
two mono jacks
shielded wire
1 / 4" stereo-to-stereo cable

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 / 4+ 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 / 4+ control!

DP/4+ Modes

The DP/4+ 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.

Select Mode  	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, 4 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*.
Edit Mode  	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 four Units ({A}, {B}, {C}, or {D}) 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*.
System•MIDI Mode  	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. For specific information about the System•MIDI parameters, see *Section 4 — System•MIDI*. In System•MIDI mode, pressing the {WRITE} button will access the MIDI System Exclusive (SysEx) data transfer function for storage of DP/4+ presets and system parameters. For information about using SysEx to store DP/4+ data, see *Section 5 — Storage*.

Button Names

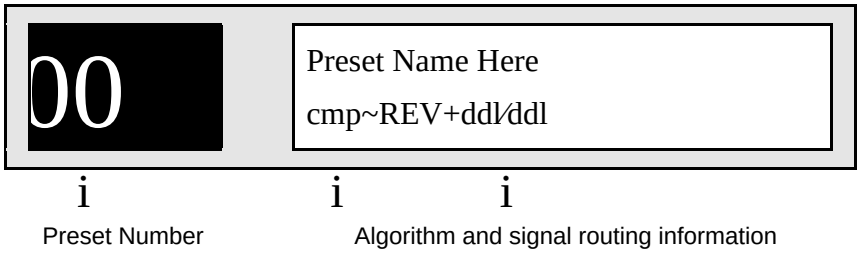
Throughout this manual, when we refer to an actual button, it will appear as a "button" in the text. For example, if the text read something like "press the Edit button," it would appear as "press the {EDIT} button." This will help you to quickly skim through familiar sections of the manual as you verify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

About Select Mode

In Select mode, you select Presets. These can be 1 Unit, 2 Unit, 4 Unit, or Config Presets, depending on the current config and on which unit buttons ({A}, {B}, {C}, or {D}) have been pressed after pressing the {SELECT} button.

Input Configuration type:	What type of presets you can select:
1 source configuration	4-Unit Presets
2 source configuration	2-Unit Presets
3 source configuration	two 1-Unit Presets and one 2-Unit Preset
4 source configuration	four 1-Unit Presets

In this 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/4+ memory. If any parameters have been edited (changed) within this 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, or 4 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 different Unit buttons ({A}, {B}, {C}, and/or {D}) to see the abbreviations change between lower case and upper case. When you press the {CONFIG} button, there are no capitalized algorithm abbreviations.

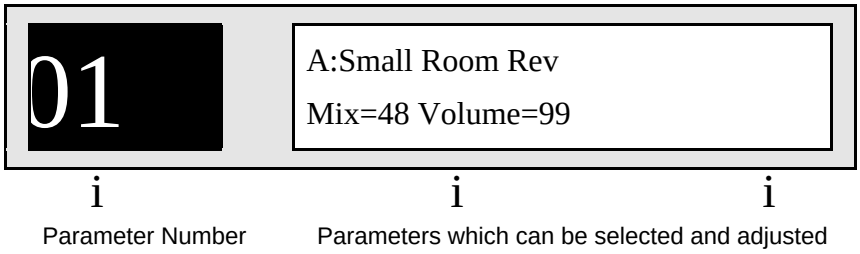
When a unit is selected by pressing its Unit button ({A}, {B}, {C}, or {D}):

- 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 four Units ({A}, {B}, {C}, or {D}) 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}, {C}, or {D}) or {CONFIG} determines what you will be editing. The display shows:



- 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 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.

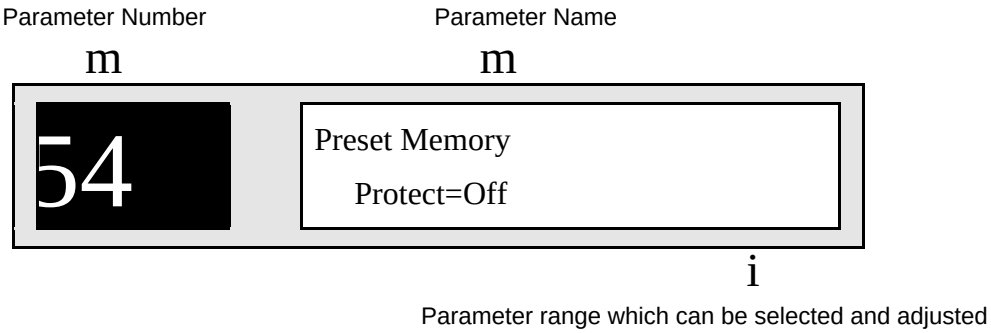
By pressing the {EDIT} button you can toggle between the original setting (LED solidly lit) and the newly edited version (LED flashing) of that algorithm. What is visible on the display is the parameter settings that you hear.

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:



- The red LED display (on the left) shows the *number* of the currently selected parameter. There are 67 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 67 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}, {C}, {D}, 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.

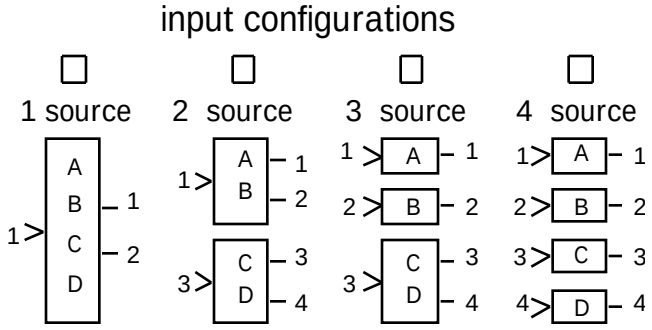
About Presets

The DP/4+ has 400 presets in its memory, but you can't get to all of them at any one time. This is because:

- 100 of the presets are Config presets which change the input configuration and routing as well as load new effects algorithms; and
- of the remaining 300 presets, the type(s) of presets available for viewing and selecting depends on the current config type.

Input Configurations

The diagram in the upper right corner of the DP/4+ shows the input configurations. All of the DP/4+ input configurations are based on this diagram:



One Source Input Configuration

In a 1 Source Config, the LED above the 1 source input configuration diagram will light. Use Input 1 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. 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.

Two Source Input Configuration

In a 2 Source Config, the LED above the 2 source input configuration diagram will light. For your first source, use Input 1 for a mono signal, or Inputs 1 and 2 if your source is a stereo signal. For your second source, use Input 3 for a mono signal, or Inputs 3 and 4 if your second source is a stereo signal. You can choose a stereo or mono input selection for Inputs 1 and 2 and/or Inputs 3 and 4 using the appropriate 2 Source Config parameters.

Three Source Input Configuration

In a 3 Source Config, the LED above the 3 source input configuration diagram will light. Use Inputs 1 and 2 for two independent mono signals. For your third source, use Input 3 for a mono signal (such as a guitar or a microphone), or Inputs 3 and 4 if your source is a stereo signal. The choice of stereo or mono for Inputs 3 and 4 is a 3 Source Config parameter.

Four Source Input Configuration

In a four source config, the LED above the 4 source input configuration diagram will light. Four separate mono sources are plugged into Inputs 1, 2, 3, and 4.

Selecting Config Presets

Of the four Preset types, the most powerful is the *Config Preset*. The Config preset lets you save, and later recall, the current state of the DP/4+, including all algorithm, signal routing and mixing information.

Selecting a Config preset will

- Reconfigure the DP/4+ inputs and outputs;
- Change the signal routing between units; and
- Load a new algorithm into each of the four Units.

To select a Config preset

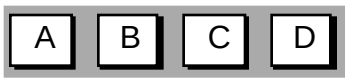
1. Press {SELECT}.
2. If the Config LED is not already on, press {CONFIG}.
3. Move the large silver **Data Entry Knob**, or press the {<} and {>} buttons. The Select LED flashes, indicating that you are previewing presets. The display shows the available Config presets.
4. When the display is showing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

Note: The first 10 ROM Config locations (presets #50 to 60) can be used as "starting places" for creating your own configurations, and cover common signal routing set-ups.

How the Config Type Affects Selecting Presets

The four different config types effectively turn the DP/4+ into 1, 2, 3, or 4 independent effects processors, with a varying number of DP/4+ effects units per "processor":

Configuration type:

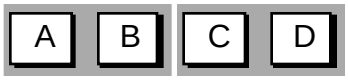


A 1 source config turns the DP/4+ into one giant multi-effects processor, with four effects units all processing the same input signal. These are called 4-Unit Presets.

Preset Example (s):

Crystal Cave CMP~fla+pit/rev

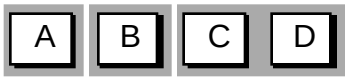
In a 4-Unit Preset, the lower line of the display shows the 3-letter abbreviations for the algorithms in all 4 units. A 4-Unit Preset will load new algorithms into all four units.



A 2 source config turns the DP/4+ into two multi-effects processors, with two effects units applied to each input signal. These are called 2-Unit Presets.

Airplane Hangar A:REV ~ B:rev

In a 2-Unit Preset, the lower line shows the 3-letter abbreviations for the algorithms in both units, and indicates which 2 units the preset will be loaded into (A&B or C&D).



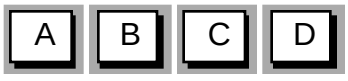
A 3 source config turns the DP/4+ into three effects processors; two with one effects unit a piece (A and B), and one with two effects units (C and D). Units A and B select 1-Unit Presets, and Units C and D select 2-Unit Presets.

Vocal Plate 1 A:Large Plate

In 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 will be loaded into (A or B).

Airplane Hangar A:REV ~ B:rev

In a 2-Unit Preset, the lower line shows the 3-letter abbreviations for the algorithms in both units, and indicates which 2 units the preset will be loaded into (A&B or C&D).



A 4 source config turns the DP/4+ into four independent effects processors; each of the four Units processes a different input signal and is treated as a separate device.

Vocal Plate 1 A:Large Plate

In 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 will be loaded into (A, B, C, or D).

If you think of the outer grey boxes shown above as different "processors," you'll notice that each of these configs creates a number of "processors" that are either 1, 2, or 4 Units combined. This determines what presets you can select at any given time. The DP/4+ has 1 Unit, 2 Unit, and 4 Unit presets, and you can only select presets of the type(s) allowed by the current config.

There are two ways that the DP/4+ shows you which type of preset you are selecting:

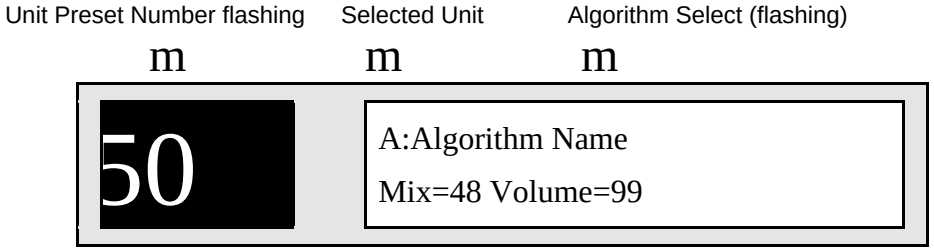
1. In Select mode, when you press any of the Unit buttons ({A}, {B}, {C}, or {D}), either 1, 2, or 4 of the yellow Unit LEDs will light. The number of lighted LEDs corresponds to the type of preset you will be selecting if you move the **Data Entry Knob**. The yellow Unit LEDs also tell you which unit(s) will be loaded with a new algorithm if you select a new preset.
2. The display gives you constant feedback. For all 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.

Replacing the Algorithm in a Single Unit

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

To replace the algorithm in a unit:

1. Press {EDIT}.
2. Press Unit {A}, {B}, {C}, or {D} to select that unit for editing. The active 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 100 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 another 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.

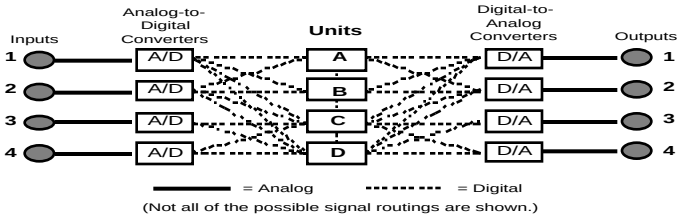
Loading a 2 Unit Preset While in a 1 Source Config

When you are using the DP/4+ in a 1 Source Config (where you can only select 4 Unit Presets) it is possible to load the effects combination from a 2 Unit preset (or a 2 Unit algorithm) into A & B or C & D. Here's how:

1. Press {EDIT} (if you are not already in edit mode).
2. Press either the {A} *and* {B}, or {C} *and* {D} buttons at the same time. Both LEDs will light up.
3. Move the **Data Entry Knob** to display the 2 Unit preset you desire, wait a moment and it will load automatically.

About Signal Routing

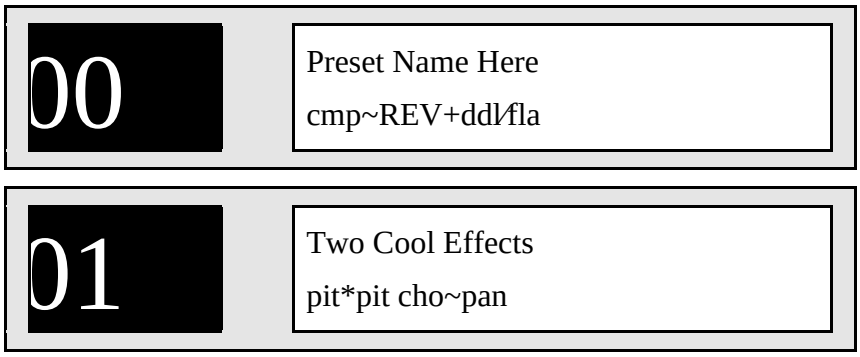
The four audio inputs are analog signals which are fed to four analog-to-digital converters. The four units are digital audio signal processors which have digital inputs and outputs. Routing between the units is digital. The output of a unit is converted back to analog audio for the output jack.



All of the above elements are under complete software control.

Signal Routing Between Units

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



~	Indicates a serial connection from the unit on the left into the unit on the right. The top example shows a serial connection from Unit A into Unit B.
+	Indicates a parallel connection between the unit on the left and the unit on the right. The top example shows this between Unit B and Unit C.
/	Indicates a feedback 1 connection from the unit on the right back into the unit on the left. The top example shows a feedback 1 connection from Unit D back into Unit C.
€	Indicates a feedback 2 connection from the unit on the right back into the unit on the left.
*	Indicates that the two units are “ganged together” using an algorithm that requires more than one unit of processing power (for instance the PitchShift 2U algorithm). The routing between units cannot be modified as long as such a “ganged” 2 Unit algorithm is active. The bottom example shows this between Units A and B.
(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 all four units are always connected together). The bottom example shows that there is no connection between Unit B and Unit C; signifying that this is a 2 source configuration.

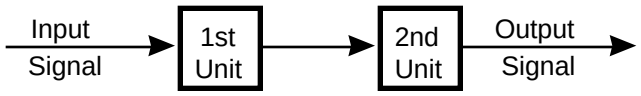
Understanding Serial, Parallel and Feedback Signal Routing

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 the first unit *before* being sent to the input of the second unit.

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

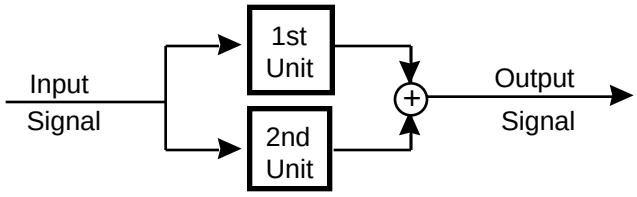


If, for example, the first unit is a chorus, and the second a reverb, you have the signal first going through the chorus, then into the reverb. As a result, you would hear the chorused sound with reverb applied to it.

Parallel Routing

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

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

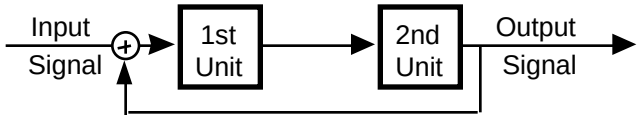


In this example, if the first unit is a chorus, and the second a reverb, you would hear the chorused sound *and* a sound with reverb, but the chorused sound would *not* have reverb on it, and the sound coming out of the reverb would not have chorusing.

Feedback Routing

Feedback routing (shown by a / symbol) is similar to serial routing, with the addition of a feedback signal. See *Section 3 — Config Parameters* for more on feedback routing.

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



In this example, if the first unit is a chorus, and the second 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 the first unit (the chorus).

Bypassing Units

At some point while you're selecting various 4 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 three units.

To bypass a single unit's algorithm:

1. Press the unit button ({A}, {B}, {C}, or {D}) that you want to bypass.
2. Press the same unit button again. The red LED above 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 completely bypass all of the unit algorithms:

1. Press the {CONFIG} button.
2. Press the {CONFIG} button again. All of the red LEDs above the units will be lit, and the units will be bypassed.
3. Further presses of the {CONFIG} button will toggle all four 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/4+.

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).

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.

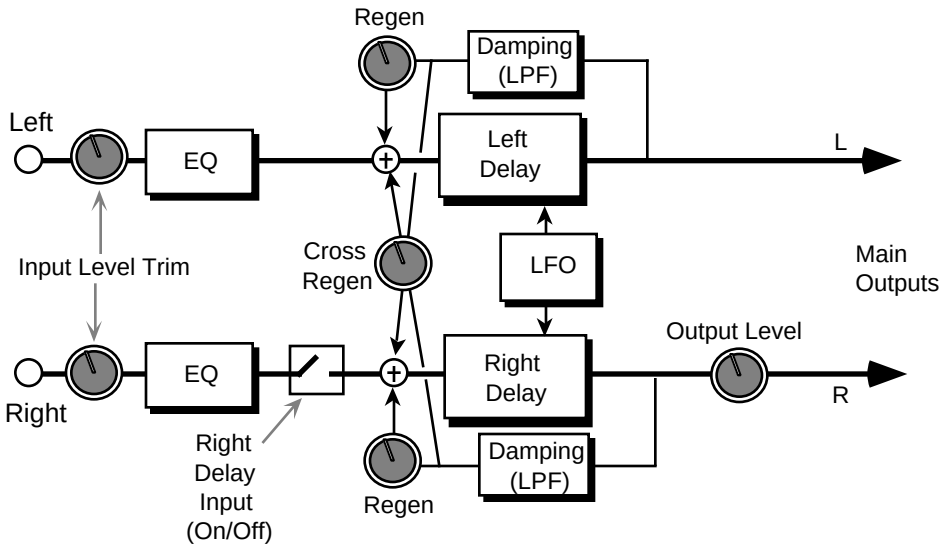
17 — Mod1 Source
18 — Mod1 Destination
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

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

EQ - DDL - WITH LFO

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.

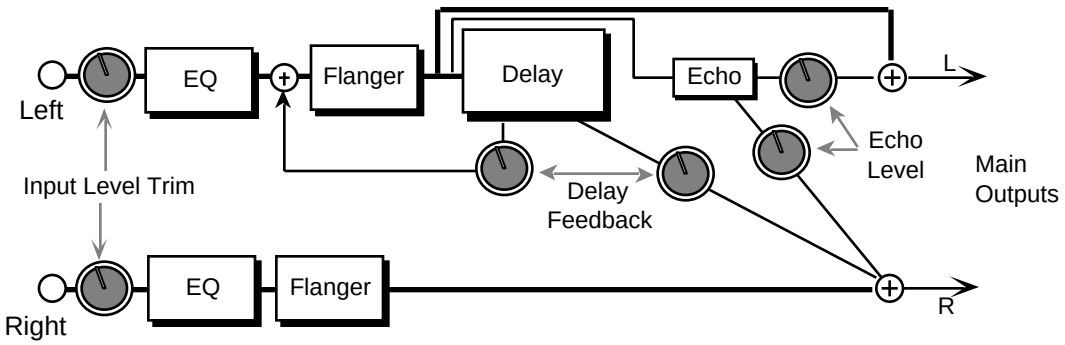
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

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.

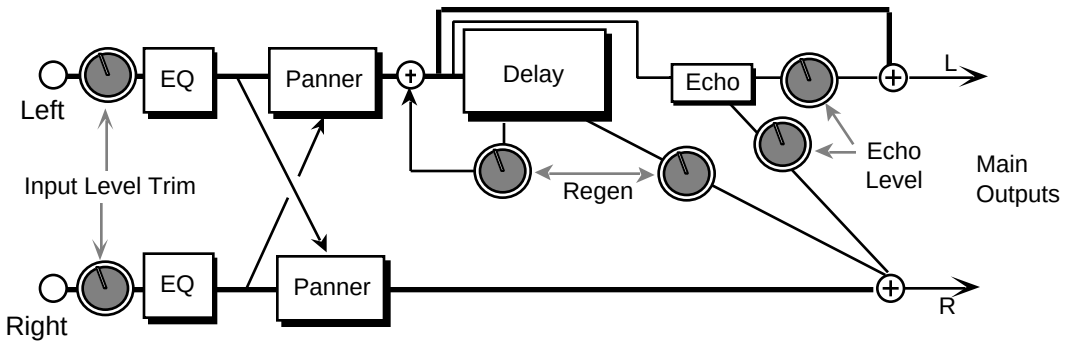
21 — Mod1 Source
25 — Mod2 Source
22 — Mod1 Destination
26 — Mod2 Destination
23 — Mod1 Param Range Min
27 — Mod2 Param Range Min
24 — Mod1 Param Range Max
28 — Mod2 Param Range Max

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.

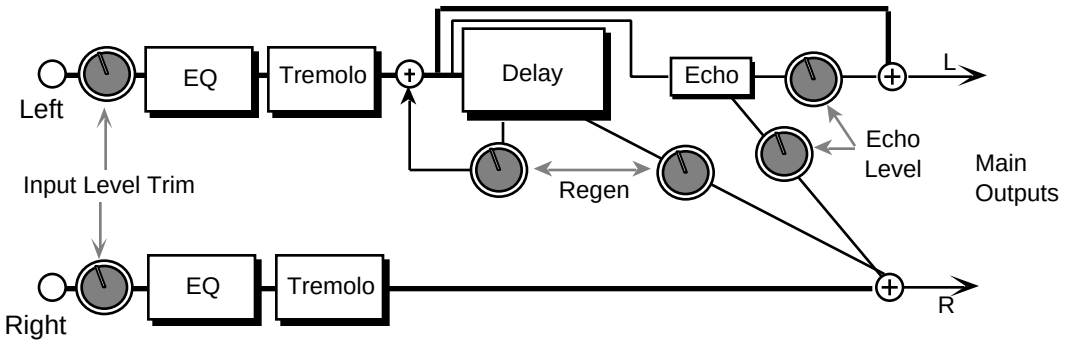
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

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.

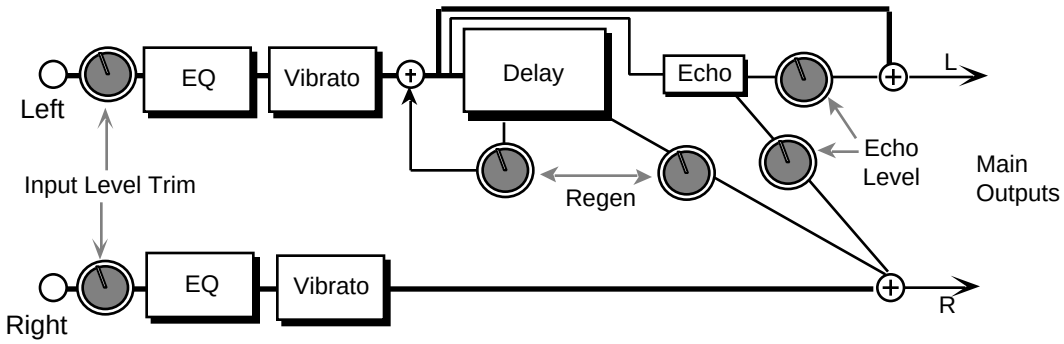
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

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.

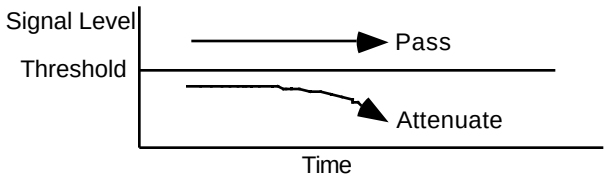
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

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.

FAST PITCH SHIFT

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).

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/4+ 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.

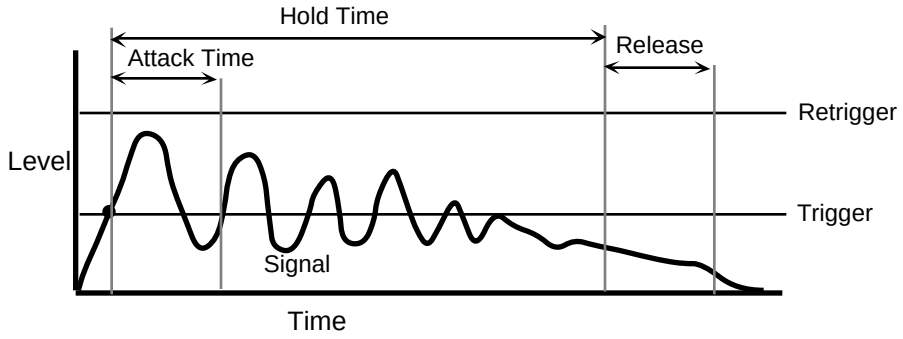
07 — Mod1 Source
08 — Mod1 Destination
09 — Mod1 Param Range Min
10 — Mod1 Param Range Max
11 — Mod2 Source
12 — Mod2 Destination
13 — Mod2 Param Range Min
14 — Mod2 Param Range Max

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

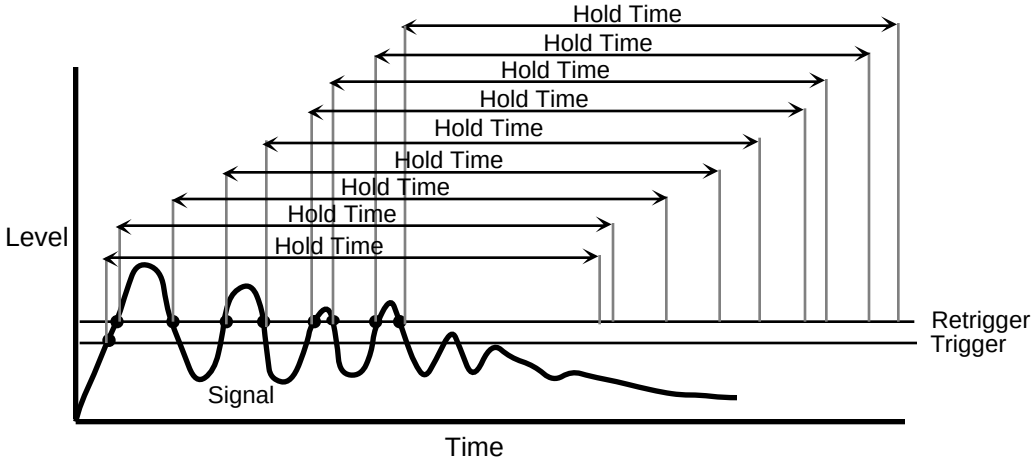
GATED REVERB

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/4+ 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/4+ 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: 100 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: 100 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 **26 — Mod2 Source**

23 — Mod1 Destination **27 — Mod2 Destination**

24 — Mod1 Param Range Min **28 — Mod2 Param Range Min**

25 — Mod1 Param Range Max **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: 100 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: 100 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.

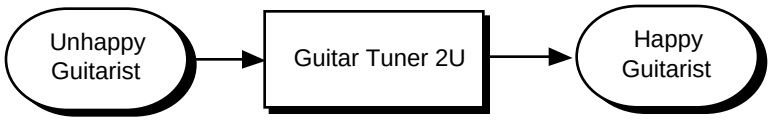
23 — Mod1 Source
24 — Mod1 Destination
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

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

GUITAR TUNER 2U

GuitarTuner2U 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.

Guitar Tuner 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

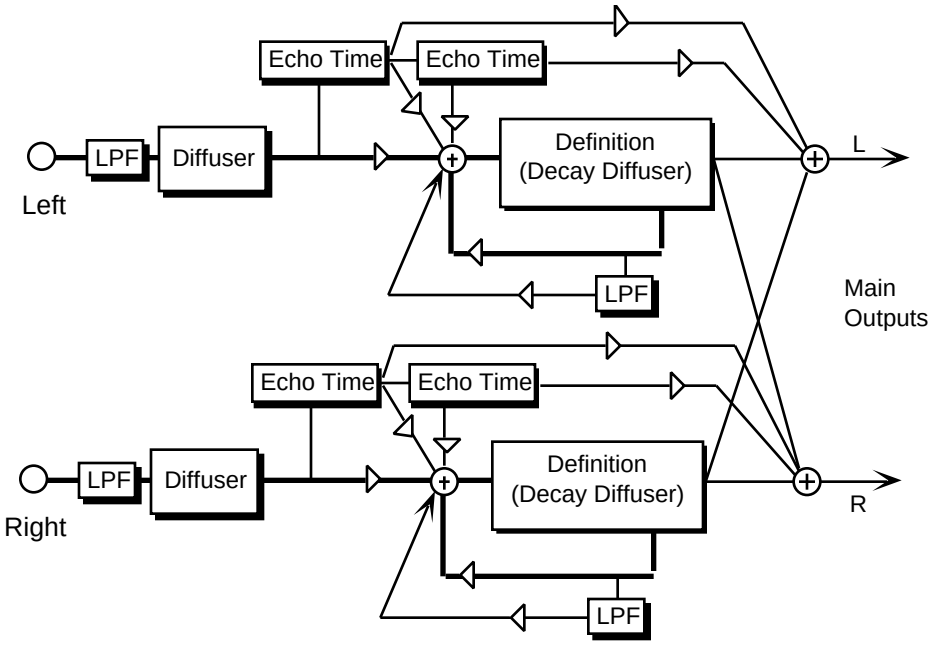
Range: A438 to A445

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

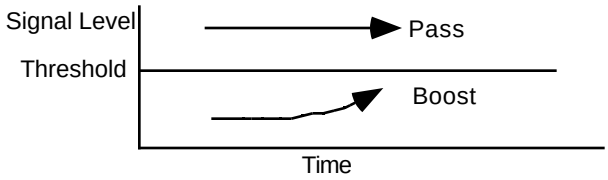
These parameters simulate the depth of the hall. Think of these parameters as three different microphones placing at various distances within the hall (parameter 20 is closest to the front, and parameter 22 is farthest from the front). When the range (volume) is higher for parameter 20, the sound appears 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.

INVERSE EXPANDER

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

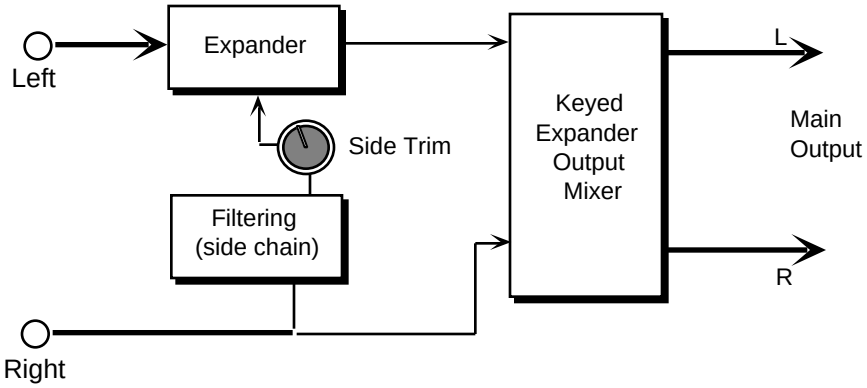
See the descriptions under the Algorithm Modulators at the beginning of this section.

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/4+ as a 2-Unit ROM Preset (location #86), 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

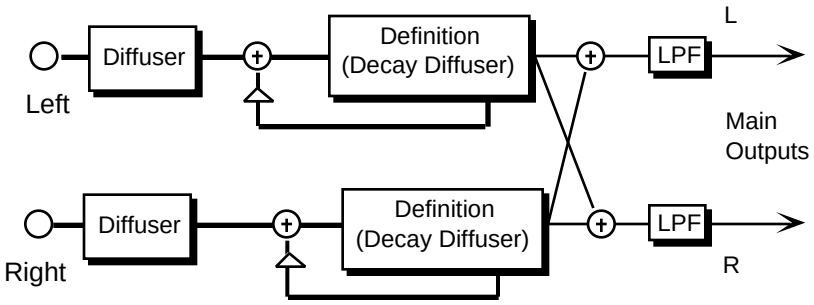
See the descriptions under the Algorithm Modulators in the beginning of this section.

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.

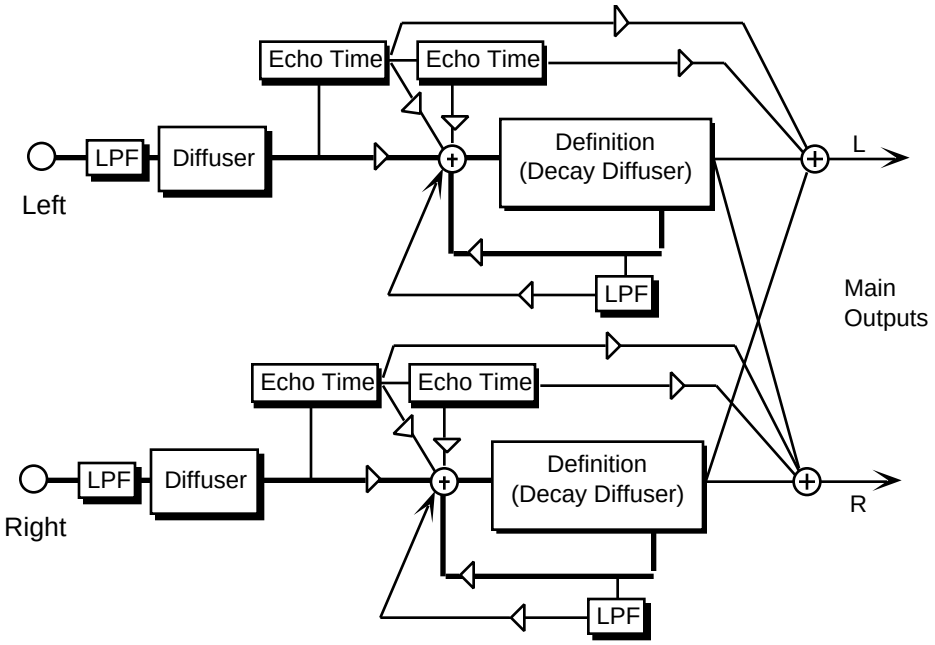
15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

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.

MULTI TAP 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 — Regen Damping

Ranges: 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

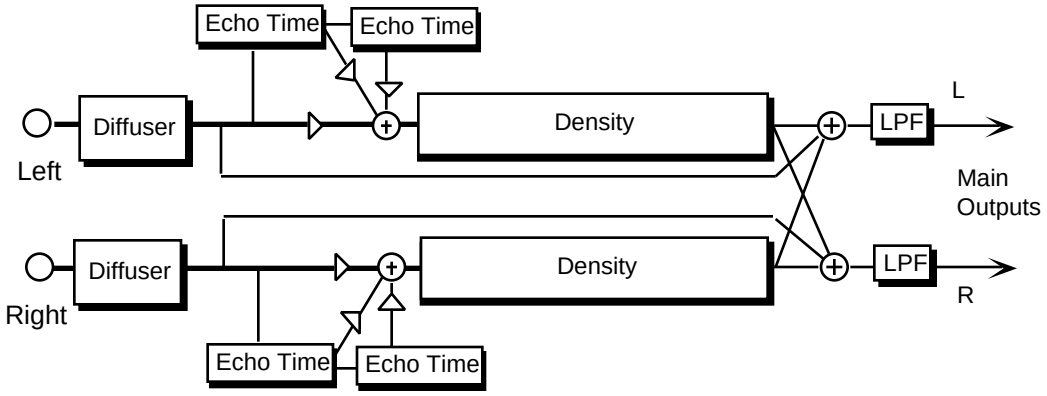
See the descriptions under the Algorithm Modulators in the beginning of this section.

NON LIN REVERB 1, 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.

Non Lin 1 is optimized for shorter duration effects (approx. 0.5 sec.).
Non Lin 2 offers approx. a 1.5 sec. duration.
Non Lin 3 is sonically similar to Non Lin 1, but there is less stereo movement, making it better suited for drum tracks.

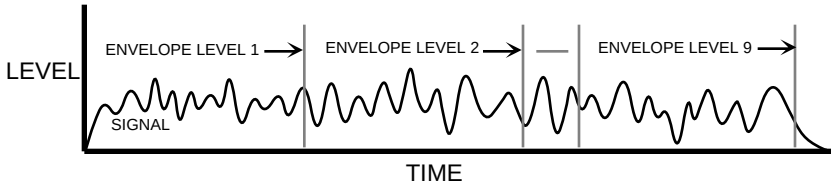
Non Lin Reverb Signal Routing



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: 100 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.

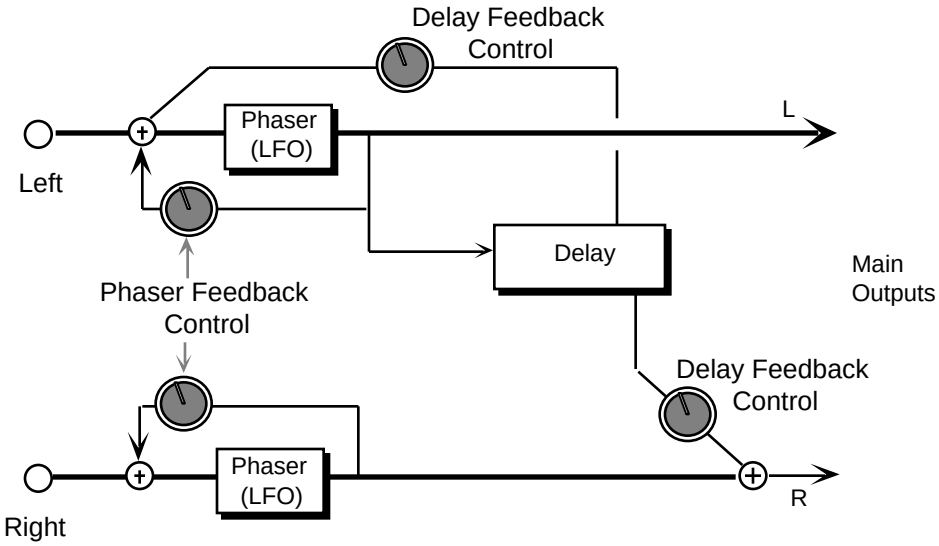
14 — Mod1 Source
15 — Mod1 Destination
16 — Mod1 Param Range Min
17 — Mod1 Param Range Max
18 — Mod2 Source
19 — Mod2 Destination
20 — Mod2 Param Range Min
21 — Mod2 Param Range Max

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

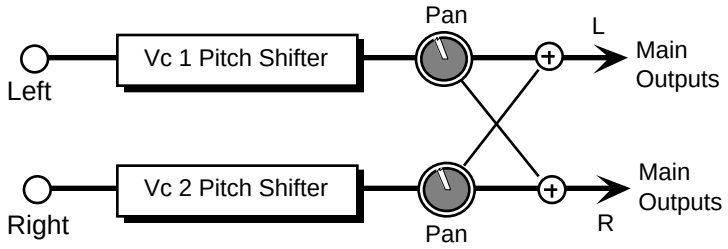
See the descriptions under the Algorithm Modulators, found earlier in this section.

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

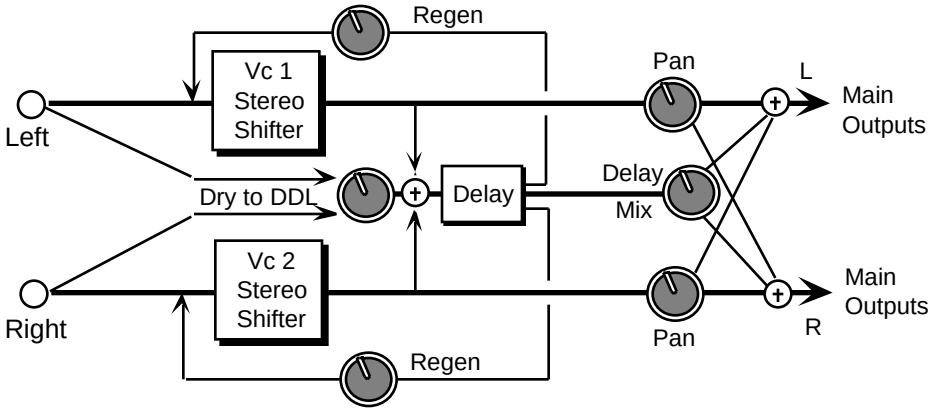
Mod2 Source
Mod2 Destination
Mod2 Param Range Min
Mod2 Param Range Max

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. This is actually a balance control which determines how much of the left and right inputs will be fed to Vc1.

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. This is actually a balance control which determines how much of the left and right inputs will be fed to Vc2.

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

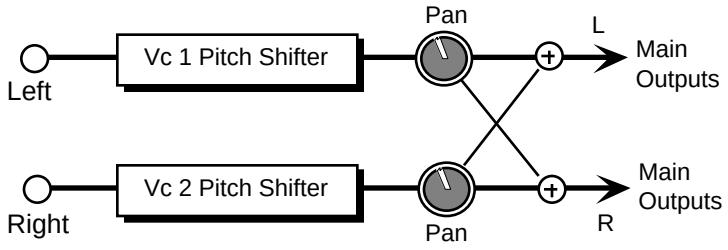
See the descriptions under the Algorithm Modulators earlier in this section.

PITCH SHIFTER

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. **Pitch Shifter** 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. **Pitch Shifter** 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.

Pitch Shifter 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

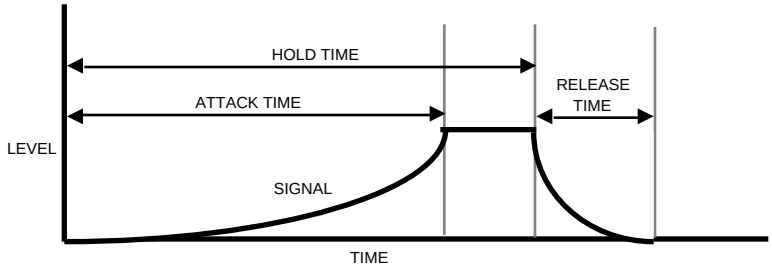
See the descriptions under the Algorithm Modulators earlier in this section.

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.

REVERSE REVERB 2

Reverse Reverb 2 is identical to Reverse Reverb, except this algorithm will retrigger by an assigned input signal level (threshold) determined by the user. 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 SPEAKER

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.

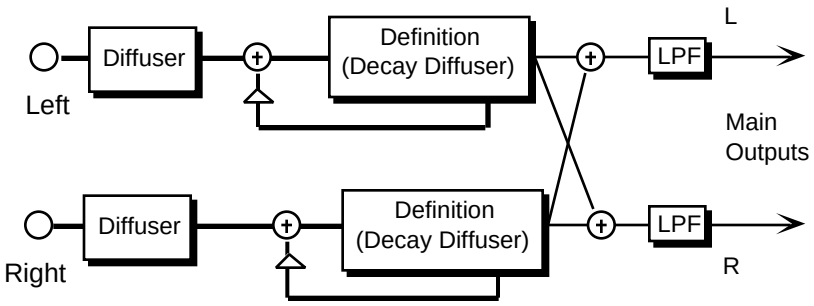
11 — Mod1 Source
12 — Mod1 Destination
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

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.

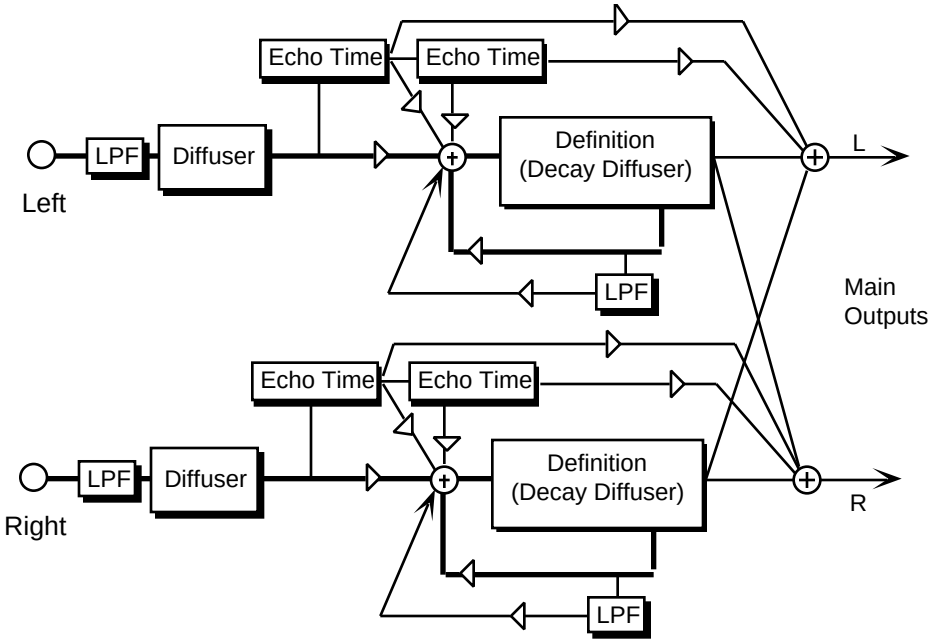
15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

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.

SPEAKER CABINET

Speaker Cabinet 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/4+ 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/4+ 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: 100 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.

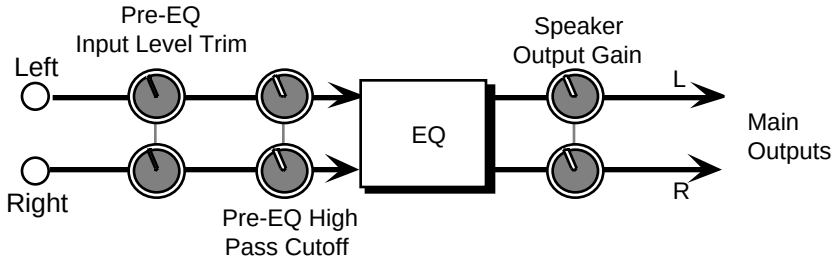
14 — Mod1 Source
15 — Mod1 Destination
16 — Mod1 Param Range Min
17 — Mod1 Param Range Max
18 — Mod2 Source
19 — Mod2 Destination
20 — Mod2 Param Range Min
21 — Mod2 Param Range Max

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

TUNABLE SPKR 2

Tunable Spkr 2 is similar to Tunable Spkr 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: 100 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
21 — Mod2 Source
18 — Mod1 Destination
22 — Mod2 Destination
19 — Mod1 Param Range Min
23 — Mod2 Param Range Min
20 — Mod1 Param Range Max
24 — Mod2 Param Range Max

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

VAN DER POL FILTER

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

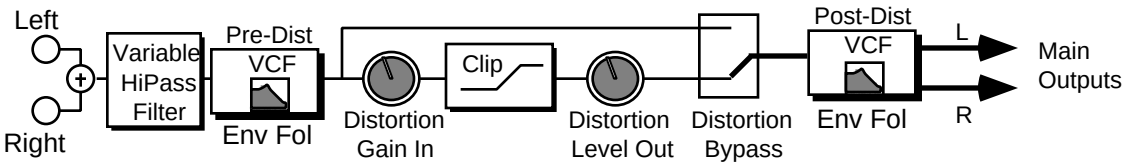
Because the cascade of high pass with low pass causes an insertion loss, this parameter allows you to boost the filtered output signal.

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

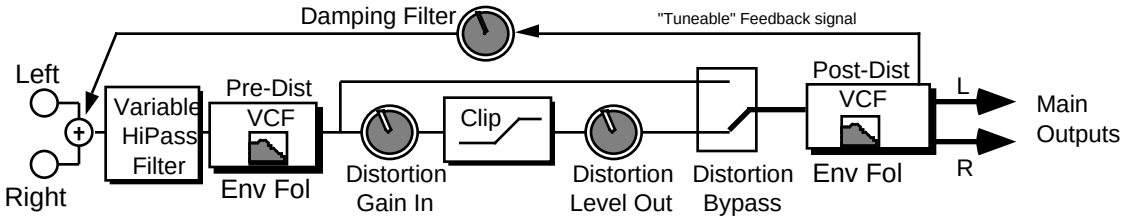
Filters out the low frequencies before the EQ. The higher the value, the less low frequencies will pass through.

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	**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.

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/4+ as a ROM Config Preset (location #99), because it requires special input signal routing. In this preset, Units A and B are used to process a new vocal, and Units C and D are used to remove the original vocal from the recording.

How to use the Vocal Remover:

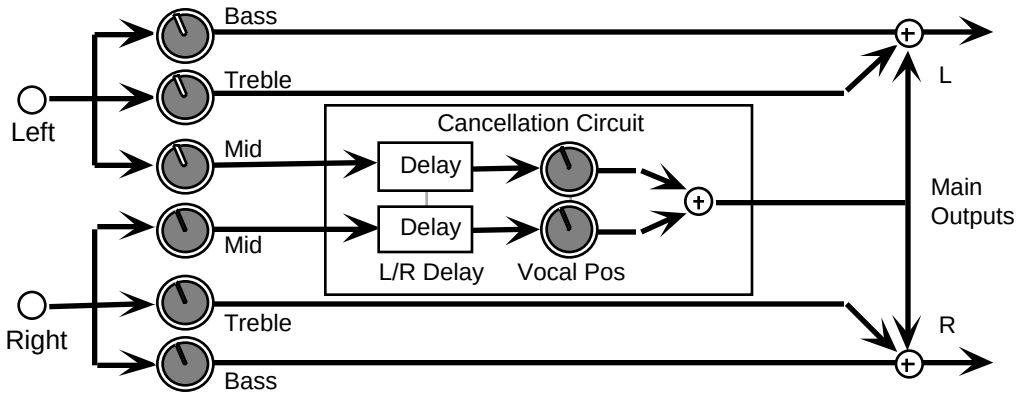
1. Feed a stereo program source into Inputs 3 and 4, and set the Bass and Treble Level to 00.
2. Set the Mid Level to 99.
3. Set the Mid Fc to about 2000 Hz.
4. Set the Mid BW to about 4000 Hz or higher.
5. Start with a Vocal Position setting of +000, and an L/R Delay setting of +000.
6. Adjust these 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

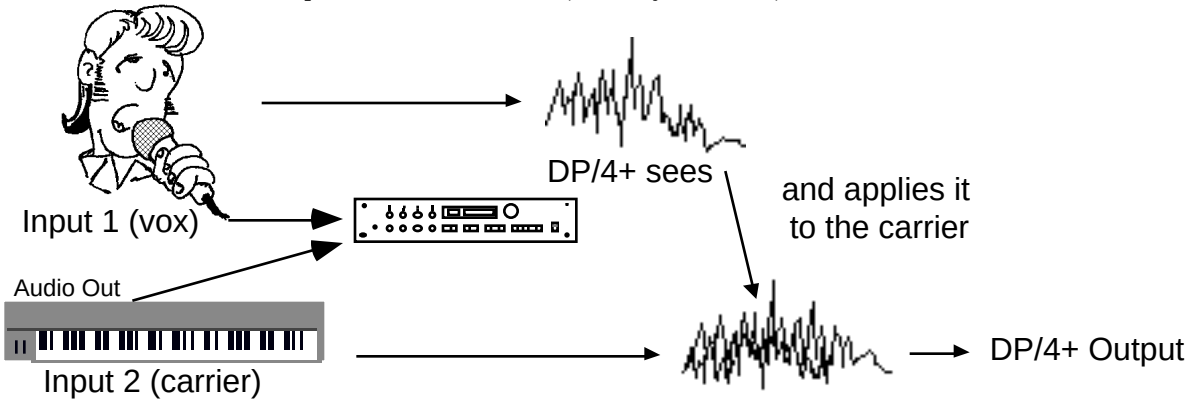
This determines the bandwidth of the bandpass filters.

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

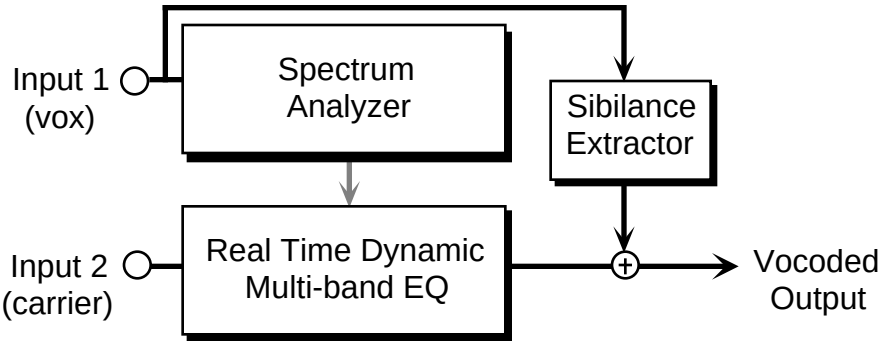
The DP/4+ features a vocoder. A vocoder analyzes the frequency spectrum from an incoming source (most commonly speech from a microphone) and applies that analysis to the pitched sounds from the output of another source (like a synthesizer).



☞ **Important:** The vocoder, though made up of four 1-Unit algorithms, is only made available in the DP/4+ as a Config Preset . There are two different vocoder config presets in the DP/4+ (Config ROM Presets #61 and 62). Don't go looking for the four 1-Unit algorithms that make up the vocoder in the list of 1-Unit Presets -- they are not there. If you accidentally change 1-Unit presets when editing the vocoder, press {CANCEL"NO} to abort the edit.

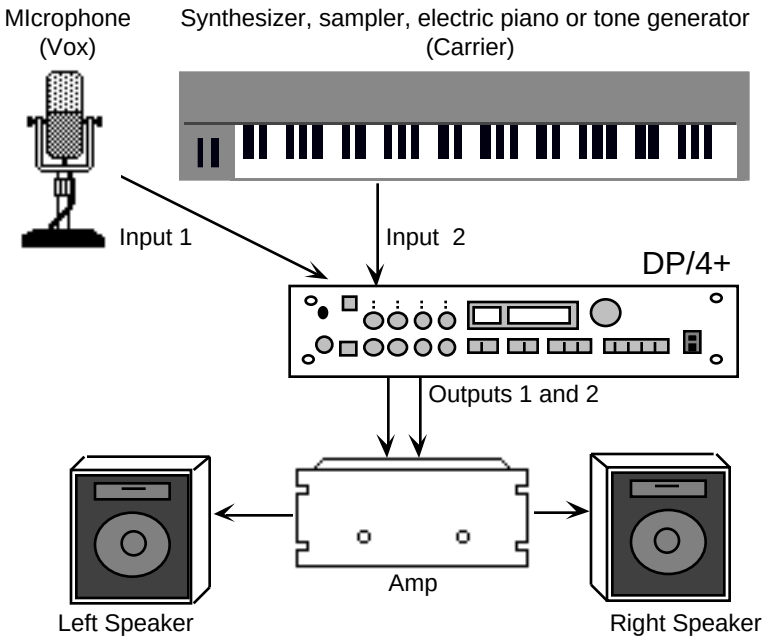
How the Vocoder Works

In the DP/4+, the vocoder uses all four units to perform one function. The four algorithms that make up the vocoder each cover a different frequency band. They are connected in parallel so that they all receive the same two inputs. The vocoder algorithms analyze the incoming signal (Input 1) and apply it to another source (Input 2). The vocoder config preset joins the four different algorithms (Vocoder Low, Vocoder Mid1, Vocoder Mid2, and Vocoder High) which work together to create the vocoder effect.



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 Carrier signal is forced to match the spectrum of the Vox signal. 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.

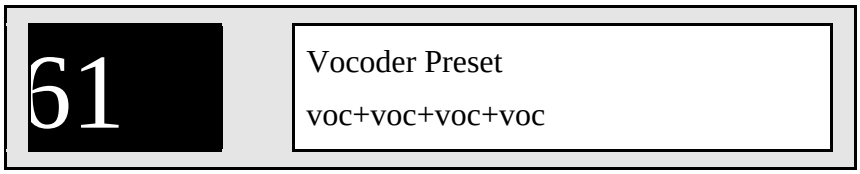
Setting Up the Vocoder



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/4+). Connect the synthesis signal (carrier) 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 #61 Vocoder Preset. 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 sources) 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 carrier input, and is not affected by the pitch you sing into the microphone. The characteristics of the carrier input signal also affect the vocoder quality. The carrier 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 carrier signal, which can produce some interesting timbres.

Vocoder Low
Vocoder Mid 1
Vocoder Mid 2
Vocoder High

Mix Range: 00 to 99

Controls the mix between the dry signal and the 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 (vocoded) portion remaining. We recommend high settings for this parameter.

Volume Range: 00 to 99

Adjusts the output volume. Setting this to 00 will eliminate the signal, producing no sound.

Speech Gain Range: -48 to +48 dB

Adjusts the boost or cut applied to the Input 1 (vox) source, after the pre-emphasis. Higher levels of pre-emphasis require higher speech gains, in general. Experiment with this level until it sounds right.

Vocoder Sibilance Level Range: 00 to 99

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 on one unit (A, B, C, or D), or a setting of 5 on each of the four units.

Vocoder Response Time Range: Slow, Normal or Fast

Selects the rate at which the carrier will track the vox signal. A fast response time will analyze and synthesize the signal quickly. A slow response time will analyze and synthesize the signal more accurately. This parameter defaults to Normal.

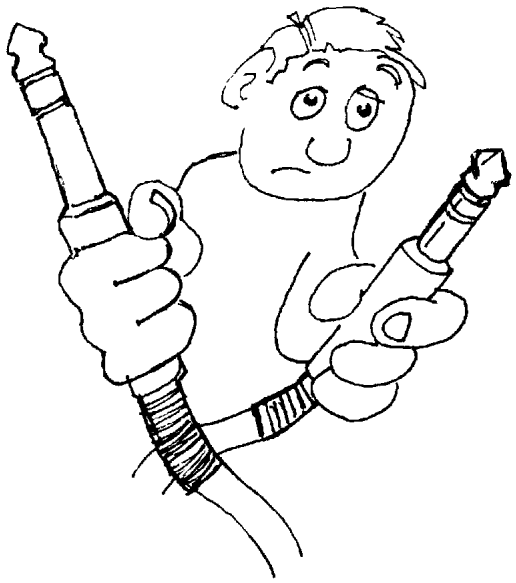
Vocoder Pre-emphasis Range: 00 to 99

This parameter emphasizes the high frequencies of the vox signal (Input 1), and de-emphasizes low frequencies. A setting of 99 gives the most emphasis, whereas a setting of 00 offers no emphasis.

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.

Section 3 — Config Parameters



This section will teach you about configs, how the effects (called algorithms) are routed in the DP/4+, and define all of the parameters relating to these configurations.

Section 3 — Config Parameters

What is a Config?

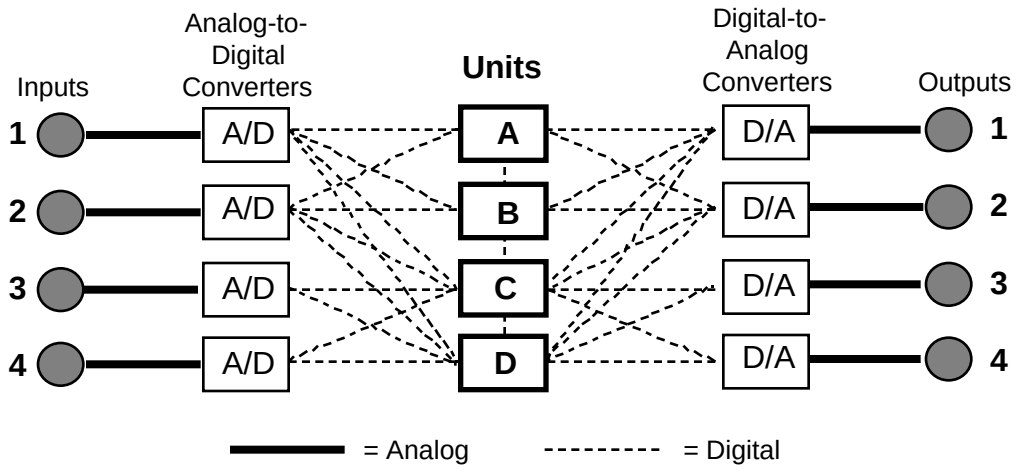
The Config (short for CONFIGuration) determines the number of input sources that are to be processed by the DP/4+, and how the units and their inputs and outputs are connected. A "1 source config" means that one signal *source* (stereo or mono) is going into the DP/4+. Two, three, and four source configs are also available.

Config Presets

Of the four DP/4+ preset types, the most powerful is the *Config Preset*. The Config preset lets you save, and later recall the current state of the DP/4+, including all algorithms, signal routing, and mixing information. There are 100 Config Presets within the DP/4+ (50 ROM and 50 RAM). You can create/write your own presets in RAM; the ROM presets cannot be changed.

About Signal Routing

The four audio inputs are analog signals which are fed to four analog-to-digital converters. The four 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.

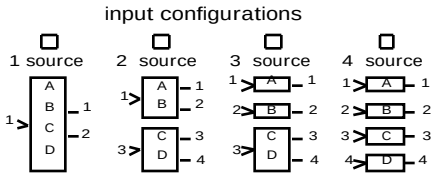


(Not all of the possible signal routings are shown.)

All of the above elements are under complete software control.

Input Configurations

The diagram in the upper right corner of the DP/4+ shows the input configurations. All of the DP/4+ input configurations are based on this diagram:



One Source Input Configuration

In a 1 Source Config, the LED above the 1 source input configuration diagram will light. Use Input 1 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.

Two Source Input Configuration

In a 2 Source Config, the LED above the 2 source input configuration diagram will light. For your first source, use Input 1 for a mono signal, or Inputs 1 and 2 if your source is a stereo signal. For your second source, use Input 3 for a mono signal, or Inputs 3 and 4 if your second source is a stereo signal. You can choose a stereo or mono input selection for Inputs 1 and 2 and/or Inputs 3 and 4 using the appropriate 2 Source Config parameters, covered later in this section.

Three Source Input Configuration

In a 3 Source Config, the LED above the 3 source input configuration diagram will light. Use Inputs 1 and 2 for two independent mono signals. For your third source, use Input 3 for a mono signal (such as a guitar or a microphone), or Inputs 3 and 4 if your source is a stereo signal. The choice of stereo or mono for Inputs 3 and 4 is a 3 Source Config parameter, and will be covered later in this section.

Four Source Input Configuration

In a 4 Source Config, the LED above the 4 source input configuration diagram will light. Four separate mono sources are plugged into Inputs 1, 2, 3, and 4. Even though the input signals to the units must be mono, the effect processing can generate four *stereo* output signals.

Selecting a Config Preset

In Select mode, you can select Config presets which will:

- Reconfigure the DP/4+ inputs and outputs;
- Change the signal routing between units; and
- Load a new algorithm (and its saved parameters) into each of the four Units.

To select a Config preset:

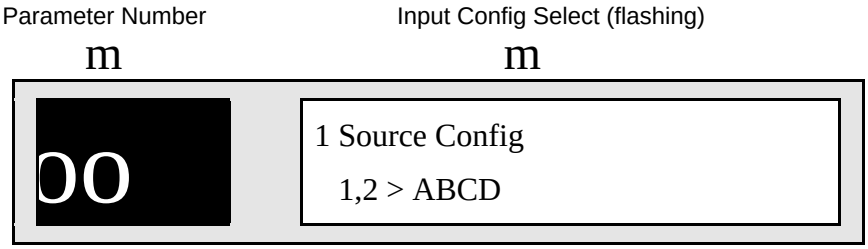
1. Press {SELECT}, then {CONFIG}.
2. Move the **Data Entry Knob**. The Select LED flashes, indicating that you are previewing presets. The display shows the available Config presets.
3. When the display is showing the preset you want to load, press {SELECT} again. This selects the preset, and the Select LED stops flashing (solidly lit).

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:

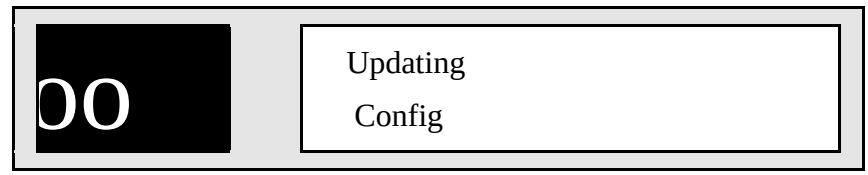


The red LED display should indicate parameter 00, which is the *Input Config Select* parameter, and the current config type should be flashing in the upper line of the LCD display (if not, press {<} until this is the case).

3. If you move the **Data Entry Knob** now, you will select among the config types. There are four different types of configs that can be edited:

1 source Config	2 source Config	3 source Config	4 source Config

When you stop moving the knob, the display momentarily shows:

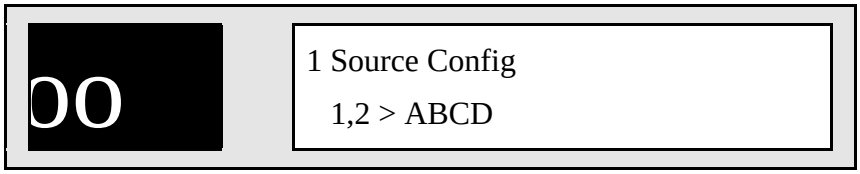


The DP/4+ is now updated into the config showing on the display.

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.

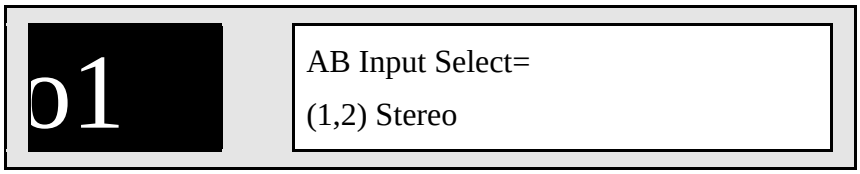
Note: Changing the config type, or editing some config parameters may cause a brief interruption in the audio output. This will happen if you change the number of sources, or if the mono/stereo output routing is different. This interruption is normal and is required for the system to reconfigure its signal routing.

1 Source Config



00 — 1 Source Config

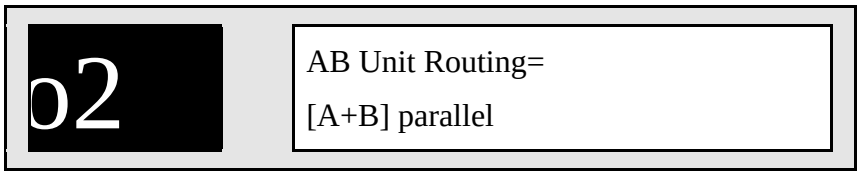
The 1 Source Config arranges the DP/4+ as one giant multi-effects processor, using all four units to process the same input signal. 1 Source Configs have two input select options, mono or stereo (refer to parameter 01).



01 — AB Input Select

Range: (1,2) Stereo or (1) Mono

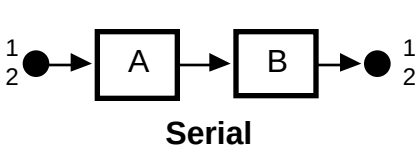
Selects either a mono (Input 1) or stereo (Inputs 1 and 2) signal.



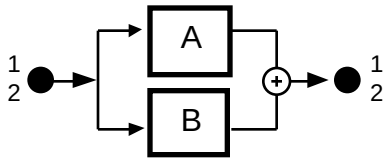
02 — AB Unit Routing

Range: serial, parallel, feedback1 or feedback2

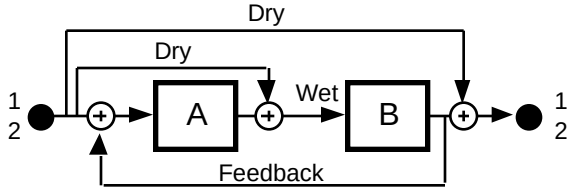
Units A and B can be routed together in one of four different ways:



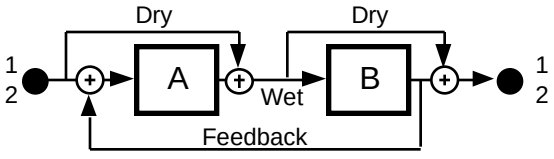
Serial



Parallel

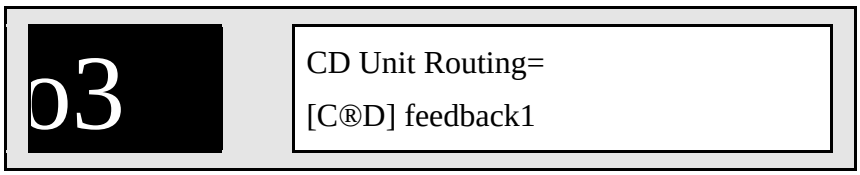


Feedback 1



Feedback 2

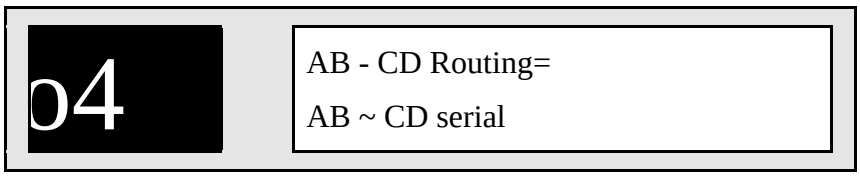
The feedback routings are similar to the serial routing, with the addition of a feedback signal. The difference between Feedback 1 and Feedback 2 is how the dry signal is mixed into the wet signal (as shown above). Note that the feedback signal is all wet, and that it is tapped before the dry signal.



03 — CD Unit Routing

Range: serial, parallel, feedback1 or feedback2

Units C and D can also be routed together in one of four ways.

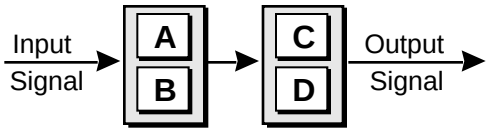


04 — AB - CD Routing

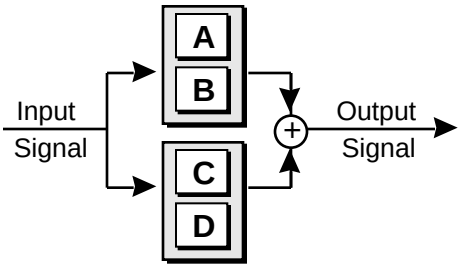
Range: serial or parallel

Units A and B can be routed to Units C and D in one of two different ways:

Serial routing between AB and CD



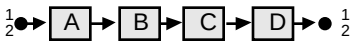
Parallel routing between AB and CD



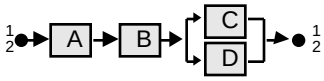
By combining parameters 02, 03, and 04, there are 32 different ABCD routing possibilities.

Note: The difference between Feedback 1 and Feedback 2 is in the *dry path* only (as shown earlier). Because we are not showing the dry path variations, there are only 18 different ABCD routing possibilities shown on the next page.

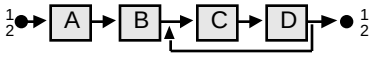
Available ABCD Routings



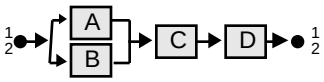
AB Serial → CD Serial



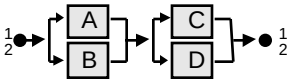
AB Serial → CD Parallel



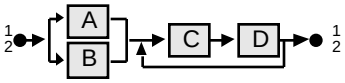
AB Serial → CD Feedback



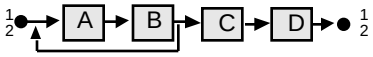
AB Parallel → CD Serial



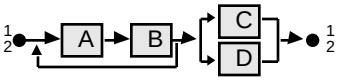
AB Parallel → CD Parallel



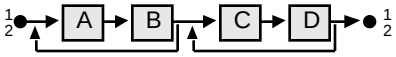
AB Parallel → CD Feedback



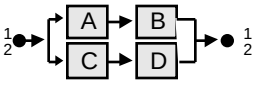
AB Feedback → CD Serial



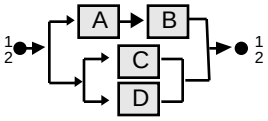
AB Feedback → CD Parallel



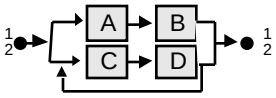
AB Feedback → CD Feedback



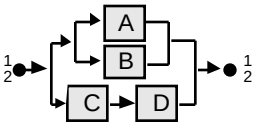
AB Serial + CD Serial



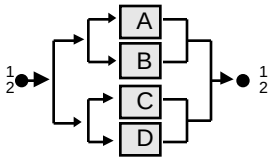
AB Serial + CD Parallel



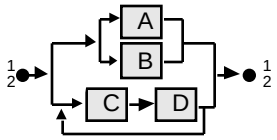
AB Serial + CD Feedback



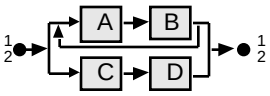
AB Parallel + CD Serial



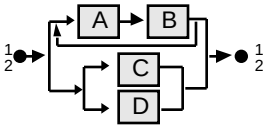
AB Parallel + CD Parallel



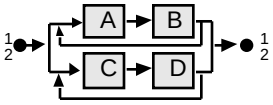
AB Parallel + CD Feedback



AB Feedback + CD Serial



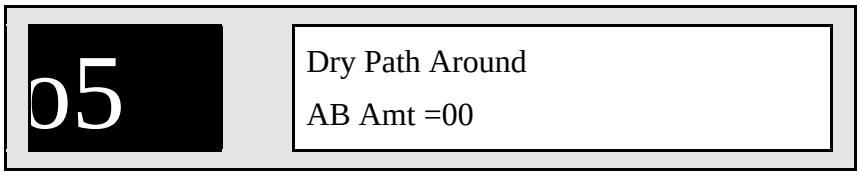
AB Feedback + CD Parallel



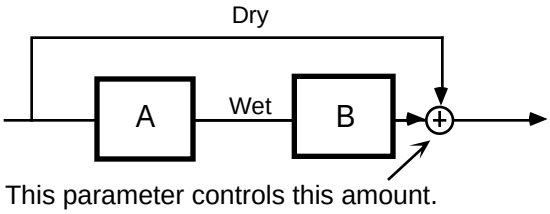
AB Feedback + CD Feedback

05 — (Config Dependent)

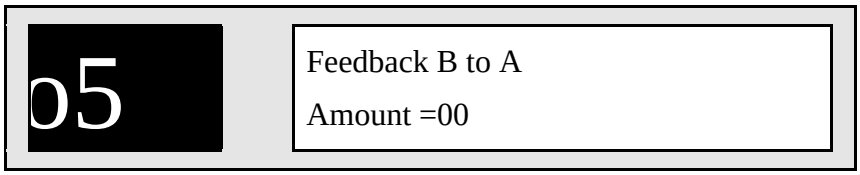
This parameter is dependent on how Units A and B are routed (determined by parameter 02). If parameter 02 is serial, this screen shows:



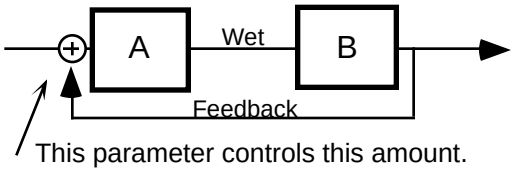
This screen allows you to control an external dry signal around Units A and B. A setting of 00 would not allow a dry signal around the units, whereas a setting of 99 would permit a full signal around the units.



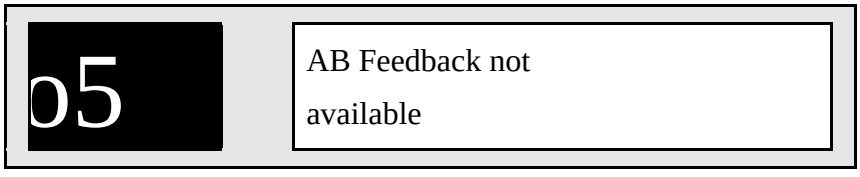
If parameter 02 is feedback 1 or 2, this screen shows:



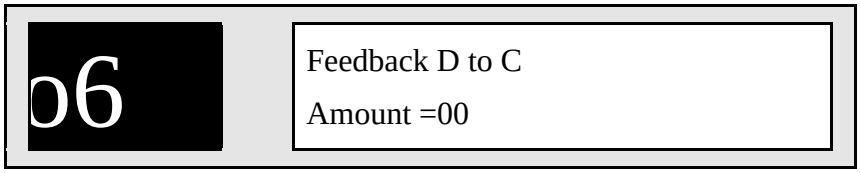
This screen allows you to control a feedback signal from Units B to the front of Unit A. A setting of 00 would not allow the feedback signal back into the units, whereas a setting of 99 would permit a full feedback signal.



If parameter 02 is parallel, this screen shows:

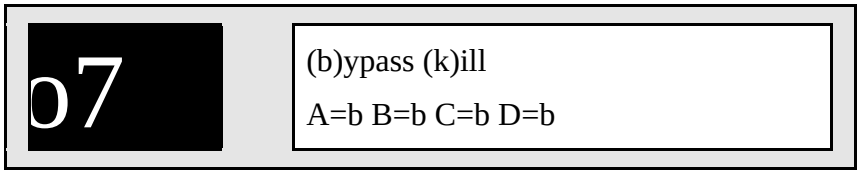


This parameter does nothing because of the nature of a parallel connection, there is no dry path (or feedback) around the units.



06 — (Config Dependent)

This parameter, similar to parameter 05, is dependent on how Units C and D are routed together (determined by parameter 03).



07 — Bypass Kill (Unit) A
08 — Bypass Kill (Unit) B
09 — Bypass Kill (Unit) C
10 — Bypass Kill (Unit) D

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.

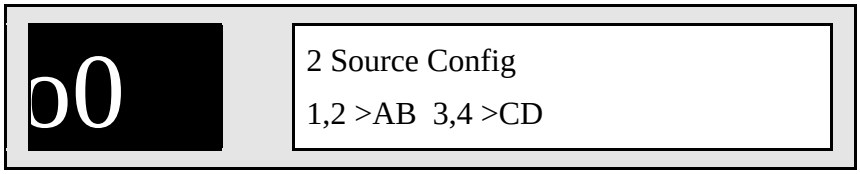
In order to use a foot switch to bypass a unit, it must be set to function as a DP/4+ controller (see the description of System•MIDI parameters 45 to 48 in *Section 4 — System•MIDI*).

Notes

When using Kill in a configuration involving unit feedback pairs (for example, AB Unit Routing=feedback):

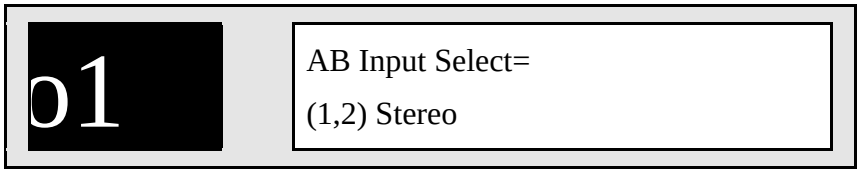
- Killing Unit B in a feedback pair mutes the signal.
- Killing Unit A does NOT mute the signal. Unit B can still pass dry signal.

2 Source Config



00 — 2 Source Config

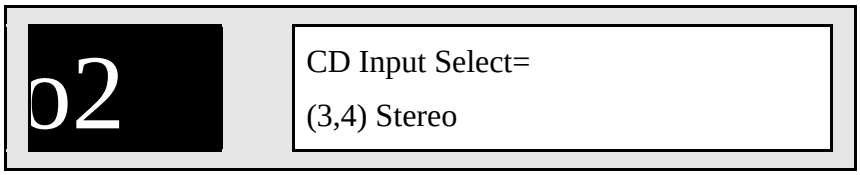
The 2 Source Config divides the DP/4+ into 2 multi-effects processors, each containing 2 units of processing power.



01 — AB Input Select

Range: (1,2) Stereo or (1) Mono

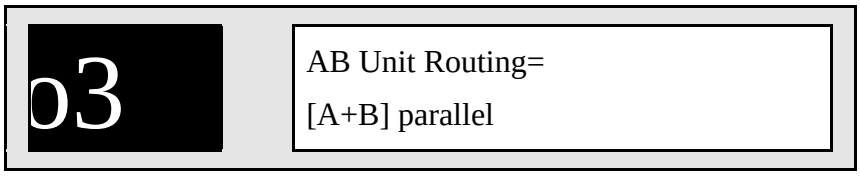
This parameter selects either a mono or stereo input for Units A and B.



02 — CD Input Select

Range: (3,4) Stereo or (3) Mono

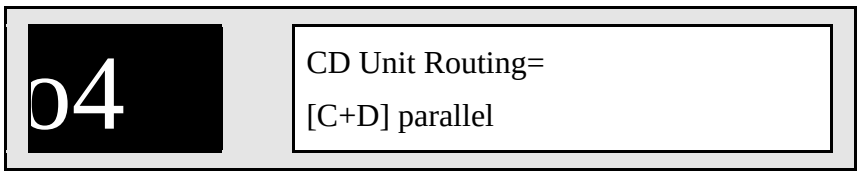
This parameter selects either a mono or stereo input for Units C and D.



03 — AB Unit Routing

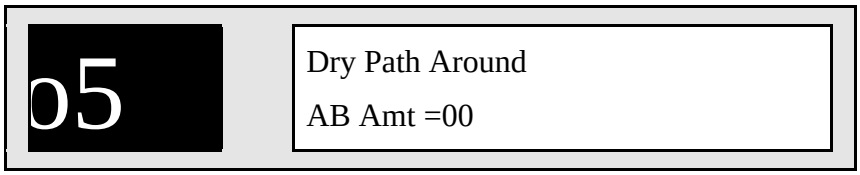
Range: serial, parallel, feedback1, or feedback2

Units A and B can be routed together in either serial, parallel, or two different kinds of feedback (explained in the 1 Source Config description).



04 — CD Unit Routing Range: serial, parallel, feedback1, or feedback2

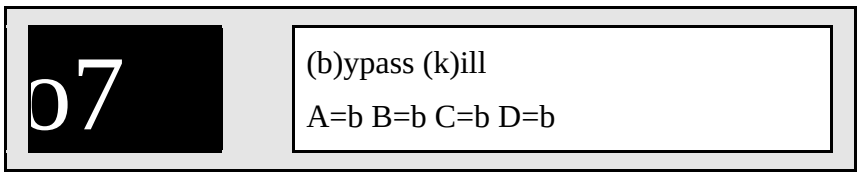
Units A and B can be routed together in either serial, parallel, or two different kinds of feedback (explained in the 1 Source Config description).



05 — AB (Config Dependent)

06 — CD (Config Dependent)

For a complete discussion of config dependent parameters, please refer to 1 Source Config parameters 05 and 06 earlier in this section.



07 — Bypass Kill (Unit) A

08 — Bypass Kill (Unit) B

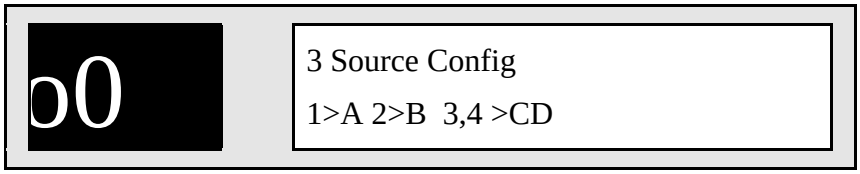
09 — Bypass Kill (Unit) C

10 — Bypass Kill (Unit) D

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.

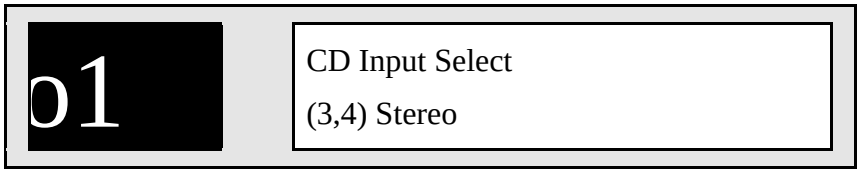
Note: In order to use a foot switch to bypass a unit, it must be set to function as a DP/4+ controller (see the description of System•MIDI parameters 45 to 48 in *Section 4 — System•MIDI*).

3 Source Config



00 — 3 Source Config

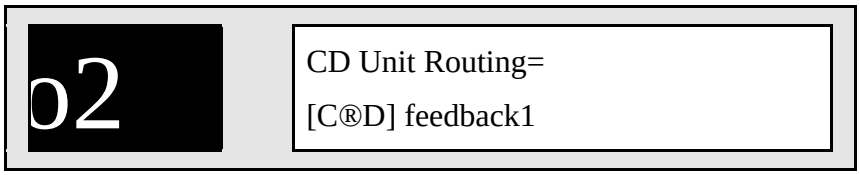
The 3 Source Config divides the DP/4+ into 3 effects processors. Units A and B function independently as 1 Unit processors, while C and D are grouped together as a single 2 Unit effects processor.



01 — CD Input Select

Range: (3,4) Stereo or (4) Mono

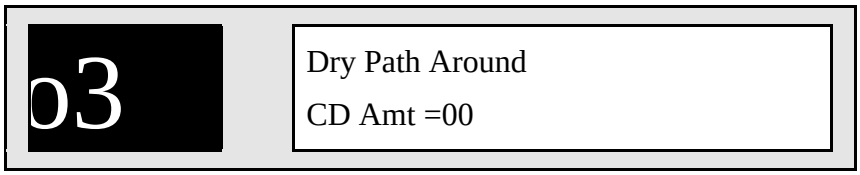
This parameter selects either a mono (Input 3) or stereo (Inputs 3 and 4) signal for Units C and D.



02 — CD Unit Routing

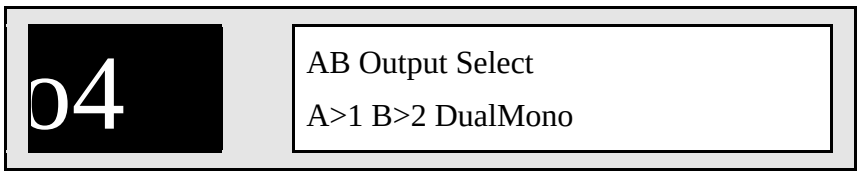
Range: serial, parallel, feedback1, or feedback2

Units C and D can be routed together in one of four different ways, as explained earlier in the 1 Source Config parameters.



03 — (Config Dependent)

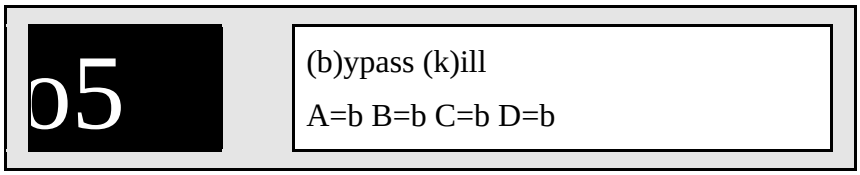
This parameter is dependent on how Units C and D are routed. See the description about config dependent parameters under 1 Source Config (parameters 05 and 06) for more information.



04 — 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 output.

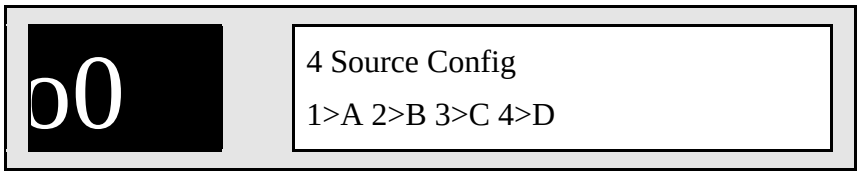


05 — Bypass Kill (Unit) A
06 — Bypass Kill (Unit) B
07 — Bypass Kill (Unit) C
08 — Bypass Kill (Unit) D

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.

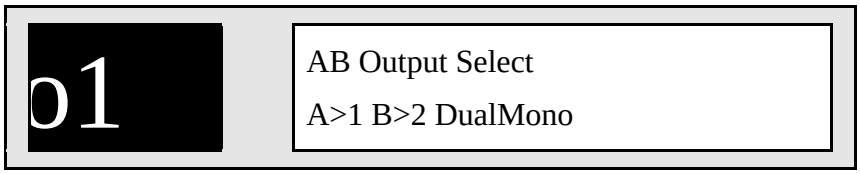
Note: In order to use a foot switch to bypass a unit, it must be set to function as a DP/4+ controller (see the description of System•MIDI parameters 45 to 48 in *Section 4 — System•MIDI*).

4 Source Config



00 — 4 Source Config

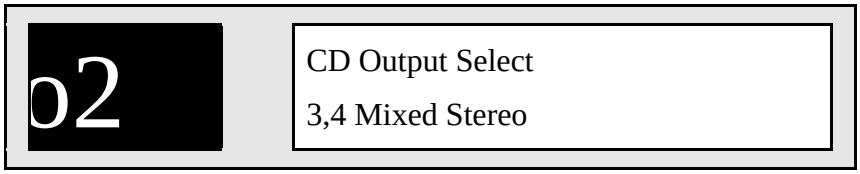
In a 4 Source Config, each unit functions as an independent 1 Unit effects processor.



01 — AB Output Select

Range: Dual Mono or Mixed Stereo

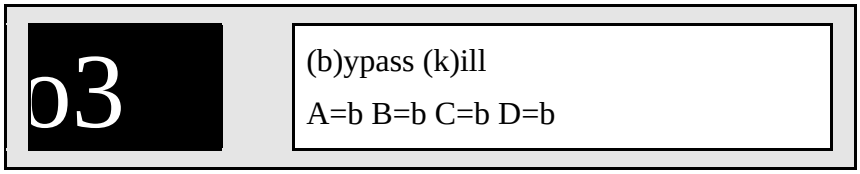
This parameter allows you to assign Units A and B as two independent mono signals to Outputs 1 and 2 respectively, or mix Units A and B into a stereo configuration.



02 — CD Output Select

Range: Dual Mono or Mixed Stereo

This parameter allows you to assign Units C and D as two independent mono signals to Outputs 3 and 4 respectively, or mix Units C and D into a stereo configuration.

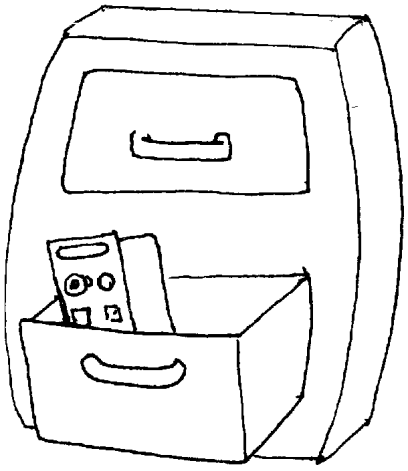


03 — Bypass Kill (Unit) A
04 — Bypass Kill (Unit) B
05 — Bypass Kill (Unit) C
06 — Bypass Kill (Unit) D

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/4+ controller (see the description of System•MIDI parameters 45 to 48 in *Section 4 — System•MIDI*).

Section 5 — Storage



This section covers the storage functions on the DP/4+, which enable you to copy 1U, 2U, 4U, 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.

Section 5 — Storage

The storage functions on the DP/4+ enable you to:

- copy 1U, 2U, 4U, 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 chapter, and also at the end of this manual.

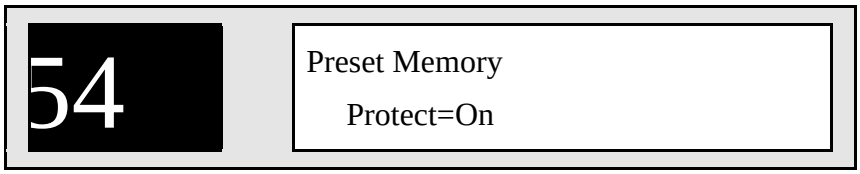
Internal Storage

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 read "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:



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 54 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 through 49) 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 four different screens:

Write to 1U Pset This shows that you are saving a 1 Unit Preset.	Write to 4U Pset This shows that you are saving a 4 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.

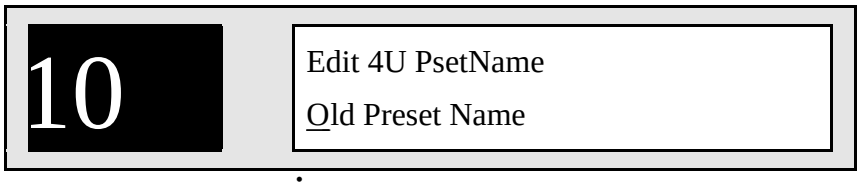
3. Using the **Data Entry Knob**, choose a RAM location (preset numbers 00 through 49) for your new preset. Notice that the LED numeric display shows the *destination* number where your preset will be saved. The old preset in that location will be lost when it is replaced by the new preset. The first 50 storage locations for each type of preset are user-programmable (battery backed up). Presets 50 to 99 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 four 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 4U PsetName This shows that you are naming a 4 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:



i

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/4+ for editing preset names, and appear (in columns) as they would if you were to turn the **Data Entry Knob** clockwise:

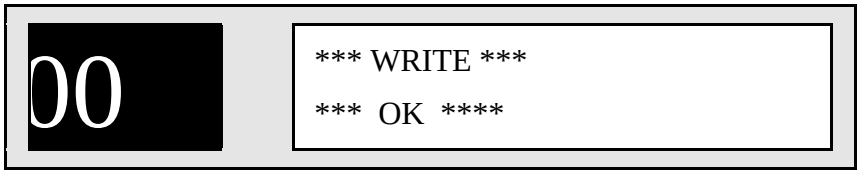
(blank)	)	2	;	D	M	V	_	h	q	z
!	*	3	<	E	N	W	`	i	r	{
"	+	4	=	F	O	X	a	j	s	|
#	,	5	>	G	P	Y	b	k	t	}
$	-	6	?	H	Q	Z	c	l	u	~
%	.	7	@	I	R	[	d	m	v	Ÿ
&	/	8	A	J	S	\	e	n	w	
'	0	9	B	K	T	]	f	o	x	
(	1	:	C	L	U	^	g	p	y	

Tip: There is a quick way to select and scroll through the alpha-numeric characters. While on this page, the Unit buttons ({A}, {B}, {C}, and {D}) will act as shortcuts:

To Get:	Press:
Upper case characters A – Z	{A}
Lower case characters a – z	{B}
Numbers 0 – 9	{C}
Special Characters I (the first one is a blank space)	{D}
Special Characters II	{CONFIG}

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 54) 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}, {B}, {C}, or {D}) 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 2U, 4U, or Config Preset where 1 Unit presets are not usually available. Note that ganged 2 Unit algorithms ("3.3 sec DDL 2U" or "Pitch Shift 2U") or the vocoder algorithm cannot be saved this way.

Similarly, pressing {CONFIG} will force the type of preset being written to be a Config preset. The **Data Entry Kno**b will now select config preset locations.

Saving a 2 Unit Preset While in a 1 Source Config

In a 1 Source config, only 4 or 1 Unit presets can be saved. However, if you have edited the algorithm parameters in two of the four units to create a sound you really like, it is possible to save that as a 2 Unit preset. Temporarily change the config type to 2 Source, save the 2 Unit preset, then change back to a 1 Source config:

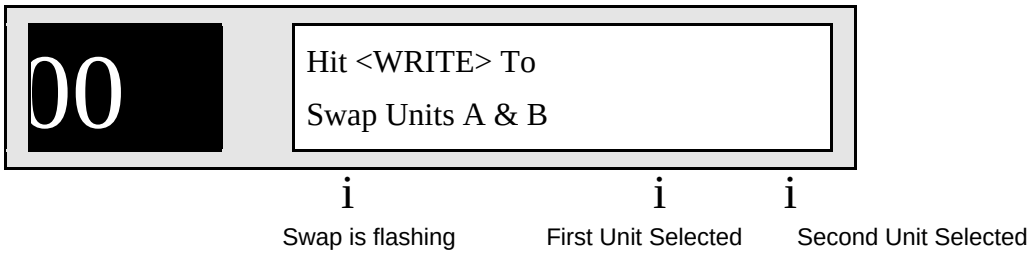
1. Press {EDIT}, then {CONFIG}.
2. Use the {<} button to scroll to the config type page (parameter 00).
3. Use the **Data Entry Knob** to change the config type to "2 Source Config." The display will momentarily show "Updating Config," then change to a 2 source config.
4. Press Unit {A} or {C}, depending on which pair you want to save.
5. Press {WRITE}, and save the 2 Unit preset to one of the RAM locations as described earlier.
6. When the 2 Unit preset has been saved, press {EDIT}, then {CONFIG}, and change the config type back to what it was originally.

You can use a variation on this procedure any time you want to save a preset that is of a type not allowed by the current config.

Swapping 1 Unit Presets

1 Unit algorithms can be swapped around between units, when loaded into either Units A, B, C, or D by using the following procedure:

1. Press {EDIT}, then {WRITE}.
2. Press a Unit button ({A}, {B}, {C}, or {D}) that you want to swap. The top line of the display shows "Write to 1U Pset."
3. While pressing and holding *the same Unit button*, press a Unit button that you want to swap with the first one. The display looks like this:



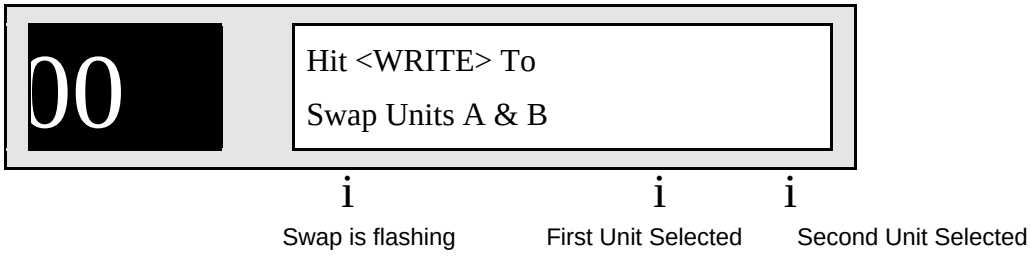
4. Press {WRITE} to swap the two selected units. The display says "Units Swapped!"

You can use this procedure any time you want to swap a 1 Unit Preset with another.

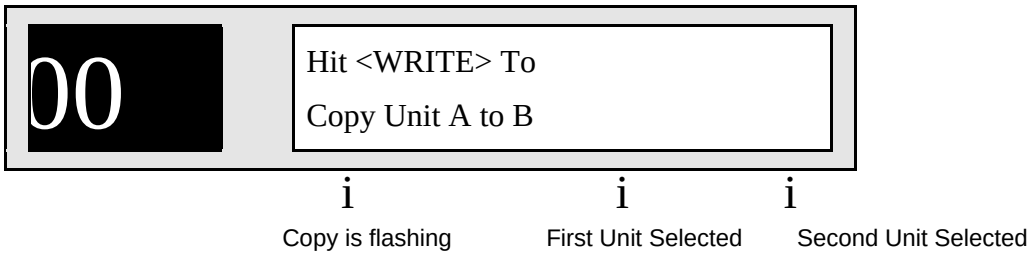
Copying a 1 Unit Preset to Another Unit

A 1 Unit preset can be copied from one unit to another, by using the following procedure:

1. Press {EDIT}, then {WRITE}.
2. Press the Unit button ({A}, {B}, {C}, or {D}) that you want to copy. The top line of the display shows "Write to 1U Pset."
3. While pressing and holding *the same Unit button*, press another Unit button that you want to copy the first preset into. The display looks like this:



3. Turn the **Data Entry Knob** clockwise. The display looks like this:



4. 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 another.

Loading a 2 Unit Preset While in a 1 Source Config

When you are in a 1 Source Config (where you can only select 4 Unit Presets), it is possible to load the effects combination from a 2 Unit preset (or a 2 Unit algorithm) into A & B or C & D:

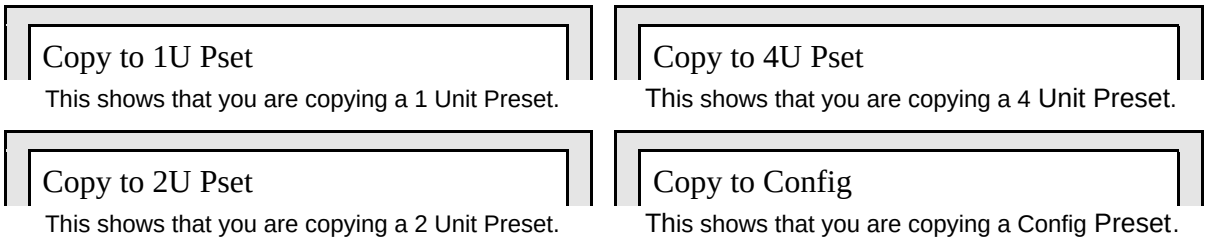
1. Press {EDIT} (if you are not already in Edit mode).
2. Press either the {A} and {B}, or {C} and {D} buttons *at the same time.* Both LEDs will light up.
3. Move the **Data Entry Knob** to display the 2 Unit preset you desire, wait a moment and it will load automatically.

Copying Presets

The DP/4+ can also copy presets to other RAM locations.

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 four possible screens:



   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) 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

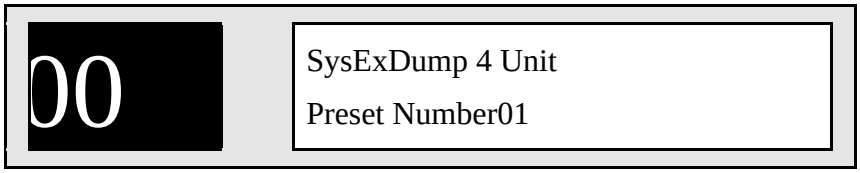
Sending MIDI Sys-Ex Messages to another DP/4+ or to a Storage Device

The DP/4+ 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/4+, 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/4+.

If you need more specific details on the messages, please refer to the *DP/4+ MIDI System Exclusive Specification* available from ENSONIQ (see the Appendix for information on how to obtain this document).

To Send DP/4+ 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/4+. When you first enter this page, the dump type defaults to the single preset belonging to the currently active unit, whose preset type and number are displayed.

The first parameter allows you to select which type of dump you wish to send.

The second parameter on this screen is only available when the first parameter is set to 1 Unit, 2 Unit, 4 Unit, or Config. It allows you to use the **Data Entry Knob** to select an individual preset (numbers 00 through 49) to transmit. If you rotate past 49, you can set the parameter to "Bank" which will dump the entire bank of the indicated preset type.

The Available System Exclusive Dumps in the DP/4+ are:

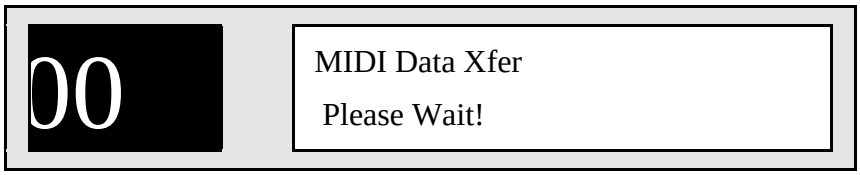
Dump Types:	Display shows:	What's included:
Single 1-Unit Preset	SysExDump 1 Unit Preset Number<00>	Single 1-Unit RAM preset <00 to 49>
1-Unit Preset Bank	SysExDump 1 Unit Preset Bank	50 1-Unit RAM presets
Single 2-Unit Preset	SysExDump 2 Unit Preset Number<00>	Single 2-Unit RAM preset <00 to 49>
2-Unit Preset Bank	SysExDump 2 Unit Preset Bank	50 2-Unit RAM presets
Single 4-Unit Preset	SysExDump 4 Unit Preset Number<00>	Single 4-Unit RAM preset <00 to 49>
4-Unit Preset Bank	SysExDump 4 Unit Preset Bank	50 4-Unit RAM presets

Single Config Preset	SysExDump Config Preset Number<00>	Single Config RAM preset <00 to 49>
ConfigPreset Bank	SysExDump Config Preset Bank	50 Config RAM presets
System	SysExDump System Parameters only	All system parameters
All Preset Banks	SysExDump All Preset Banks	200 RAM presets
All Preset Banks with System Parameters	SysExDump All PsetBanks+System	200 RAM presets and all system params

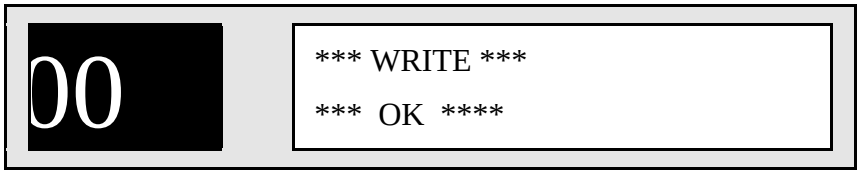
Preset data is always transmitted from and received into the internal RAM. The System Parameters dump includes all system and MIDI parameters found in System • MIDI mode (such as program change-to-preset maps and user preference switch settings). It is best to use the smallest single dump type which contains all of the data you wish to reload at one time.

You can press {CANCEL} to exit this page without sending any data.

Once the screen shows you what you want to send, make sure that the receiving device is ready to accept data, and then press {WRITE} once more to start transmission of MIDI data. 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:

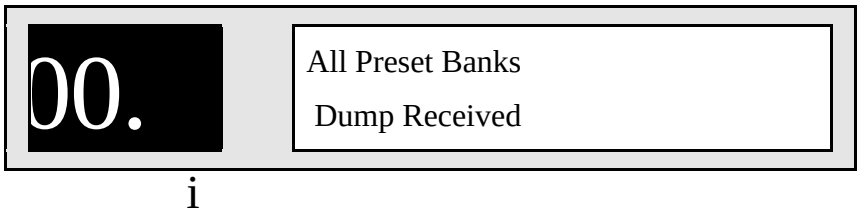


Note: ROM presets and ROM preset banks may not be dumped from the front panel. System Exclusive Dump Request commands are provided for external devices which need to extract those banks. Refer to the *DP/4+ MIDI System Exclusive Specification* for more details (see the Appendix).

Remember! The System Exclusive ID number (system parameter 52) 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/4+

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 52 and 53). 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.

The top line of this message will describe the type of dump received. The preset type and number are shown for single preset dumps. Only the type is shown for preset bank dumps. 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.

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.

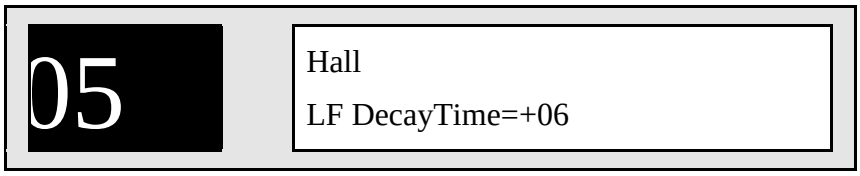
It is possible that some computer interfaces can transmit Sysex dumps faster than the DP/4+ can recieve it. On many Macintosh™ applications, however, the sysex transfer rate can be reduced. You could also try reducing the speed of the interface (e.g., from 2x to 1x).

For more information about error messages, refer to the *DP/4+ MIDI System Exclusive Specification* (see the Appendix).

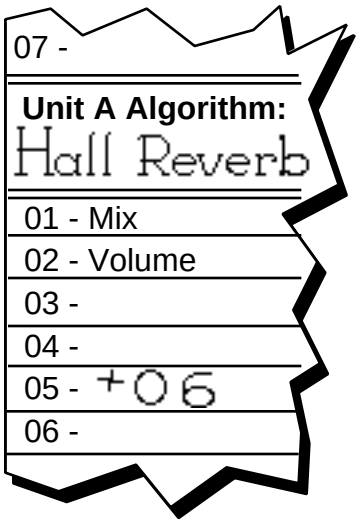
Using the Preset Parameter Worksheet

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/4+.

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/4+. 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:



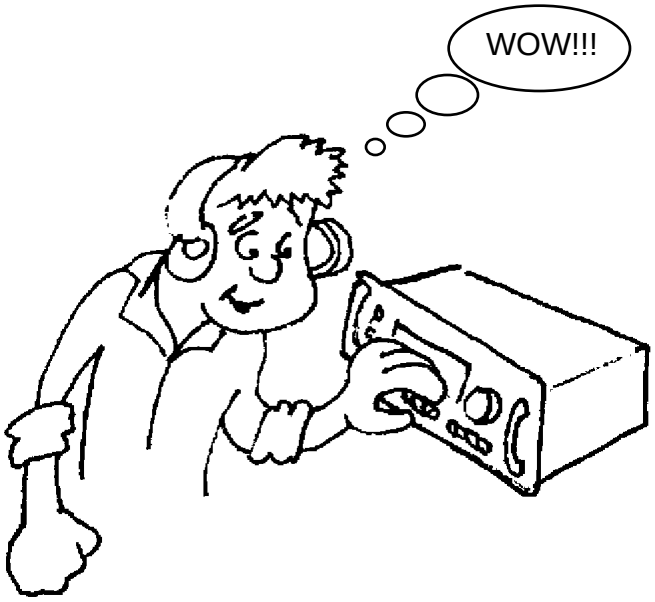
To find the parameter numbers for your config, press the {EDIT} button, then the {CONFIG} button. Use the {<} and {>} buttons to scroll through and record the edited parameters.

To find the parameter numbers for your algorithms, press {EDIT}, then each Unit button ({A}, {B}, {C}, and/or {D}) that relates to your preset (a 2 Unit preset only uses two algorithms and only requires two columns). Use the {<} and {>} buttons to scroll through and record the edited parameters.

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/4+ Preset Parameter Worksheet		**Preset Name:**	
Config Parameters:	1 2 3 4 Source Config	01-	02-
03-	04-	05-	06-
07-	08-	09-	10-
Unit A Algorithm:	**Unit B Algorithm:**	**Unit C Algorithm:**	**Unit D Algorithm:**
01- Mix	01- Mix	01- Mix	01- Mix
02- Volume	02- Volume	02- Volume	02- Volume
03-	03-	03-	03-
04-	04-	04-	04-
05-	05-	05-	05-
06-	06-	06-	06-
07-	07-	07-	07-
08-	08-	08-	08-
09-	09-	09-	09-
10-	10-	10-	10-
11-	11-	11-	11-
12-	12-	12-	12-
13-	13-	13-	13-
14-	14-	14-	14-
15-	15-	15-	15-
16-	16-	16-	16-
17-	17-	17-	17-
18-	18-	18-	18-
19-	19-	19-	19-
20-	20-	20-	20-
21-	21-	21-	21-
22-	22-	22-	22-
23-	23-	23-	23-
24-	24-	24-	24-
25-	25-	25-	25-
26-	26-	26-	26-
27-	27-	27-	27-
28-	28-	28-	28-
29-	29-	29-	29-
30-	30-	30-	30-
31-	31-	31-	31-
32-	32-	32-	32-
33-	33-	33-	33-
34-	34-	34-	34-
Notes:			

Section 6 — Presets



This section shows how to select the RAM and ROM presets found in the DP/4+, and shows the names, algorithms, and routing configurations within the 1 Unit, 2 Unit, 4 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/4+. 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 all **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 Psets."
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. Turn the **Data Entry Knob** until the screen shows "51 Select 2U Psets."
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 4 Unit Presets:

1. Press the {SELECT} button, then the {CONFIG} button.
2. Turn the large silver **Data Entry Knob** until the screen shows "50 Select 4U Psets."
3. Press the {SELECT} button, then the unit {A} button.
4. Turn the **Data Entry Knob** to display the names of the different 4 Unit Presets.
5. Press the {SELECT} button to hear the preset on the display.
6. To select other 4 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

00	Vocal Plate 1 Large Plate
01	Synth Plate Large Plate
02	Vocal Plate 2 Large Plate
03	Zobie Plate Large Plate
04	Slam Plate Large Plate
05	Multi Plate Large Plate
06	Short Plate Small Plate
07	Ballad Reverb Large Plate
08	Bright Hall Hall Reverb
09	Vocal Hall Hall Reverb
10	Concert Hall 1 Large Plate
11	Concert Hall 2 Large Plate
12	Summer Hall Hall Reverb
13	Famous Hall Hall Reverb
14	Small Hall Hall Reverb
15	Room 224 Large Room Rev
16	Medium Room Small Room Rev
17	Dark Room Small Room Rev
18	Early Reflections NonLin Reverb1
19	Smooth Non Lin NonLin Reverb1
20	Gated Verb Gated Reverb
21	Dark Drum Room Small Room Rev
22	Digital Drm Room Large Room Rev
23	Boom Room NonLin Reverb2
24	Kickdrum Nonlin NonLin Reverb3

25	70's Drum Booth Small Room Rev
26	Tiled Room Small Room Rev
27	Gymnasium Hall Reverb
28	Backstage Small Room Rev
29	String Verb Large Plate
30	French Horn Verb Large Plate
31	Country Keys Phaser - DDL
32	Lush Keys EQ-Chorus-DDL
33	Echo Clav Dual Delay
34	Medium Flange Flanger
35	Tight Flange Flanger
36	Wet Chorus 8 Voice Chorus
37	Key Funk Flange EQ-Flanger-DDL
38	Key Funk Phaze Phaser - DDL
39	Dist. RotarySpkr Rotating Spkr
40	Fusion Bass 8 Voice Chorus
41	Clean Guitar Amp Guitar Amp 2
42	Snare Compressor EQ-Compressor
43	Vocal Spice MultiTap Delay
44	Slap Vocal Dual Delay
45	Vocal Spreader FastPitchShift
46	'66 Car Radio Tunable Spkr 1
47	Analog Flanger EQ-Vibrato-DDL
48	Wild Panner EQ-Panner-DDL
49	Tape Stop Effect PitchShift-DDL

1-Unit ROM Presets

50	Small Room Rev Small Room Rev
51	Large Room Rev Large Room Rev
52	Hall Reverb Hall Reverb
53	Small Plate Rev Small Plate
54	Large Plate Rev Large Plate
55	Reverse Reverb Reverse Reverb
56	RetrigReverseRev ReverseReverb2
57	Gated Reverb Gated Reverb
58	Non Lin Reverb 1 NonLin Reverb1
59	Non Lin Reverb 2 NonLin Reverb2
60	Non Lin Reverb 3 NonLin Reverb3
61	Multi Tap Delay MultiTap Delay
62	Dual Delay Dual Delay
63	Tempo Delay Tempo Delay
64	EQ-DDL-with LFO EQ-DDL-withLFO
65	VCF-Distortion 1 VCF-Distort 1
66	VCF-Distortion 2 VCF-Distort 2
67	Guitar Amp 1 Guitar Amp 1
68	Guitar Amp 2 Guitar Amp 2
69	Guitar Amp 3 Guitar Amp 3
70	Guitar Amp 4 Guitar Amp 4
71	Digital Tube Amp DigitalTubeAmp
72	Dynamic Tube Amp DynamicTubeAmp
73	Speaker Cabinet SpeakerCabinet
74	TunableSpeaker 1 Tunable Spkr 1

75	TunableSpeaker 2	Tunable Spkr 2
76	Rotating Speaker	Rotating Spkr
77	EQ-Chorus-DDL	EQ-Chorus-DDL
78	EQ-Vibrato-DDL	EQ-Vibrato-DDL
79	EQ-Panner-DDL	EQ-Panner-DDL
80	EQ-Flanger-DDL	EQ-Flanger-DDL
81	EQ-Tremolo-DDL	EQ-Tremolo-DDL
82	Phaser-DDL	Phaser - DDL
83	8 Voice Chorus	8 Voice Chorus
84	Flanger	Flanger
85	Pitch Shifter	PitchShifter
86	Pitch Shift-DDL	PitchShift-DDL
87	Fast Pitch Shift	FastPitchShift
88	EQ-Compressor	EQ-Compressor
89	Expander	Expander
90	Inverse Expander	InversExpander
91	De-esser	De-esser
92	Rumble Filter	Rumble Filter
93	Parametric EQ	Parametric EQ
94	VanderPol Filter	VandrPolFilter
95	Signal Generator	Sine/Noise Gen
96	FS Tap-Tempo DDL	Tempo Delay
97	Noise Gate	EQ-Compressor
98	Vocal Compressor	EQ-Compressor
99	No Effect/Silent	No Effect

2-Unit RAM Presets

00	Trumpet Plate Large Plate~8 Voice Chorus
01	Oldtime Plate Rotating Spkr~Large Plate
02	Tube Plate SpeakerCabinet~Large Plate
03	Gated Plate Small Plate~Expander
04	Compressed Plate Small Plate~EQ-Compressor
05	Horn Verb Small Plate~Large Plate
06	Twisting Hall Hall Reverb~Flanger
07	Phased Hall Hall Reverb~Phaser - DDL
08	Gated Room 1 Small Room Rev+EQ-Compressor
09	Spring Reverb Rotating Spkr~Large Plate
10	Bloom Canyon NonLin Reverb3~Large Plate
11	Dark Verb Tunable Spkr 1/Large Plate
12	Airplane Hangar NonLin Reverb1~Large Plate
13	Snare Intro Large Plate~EQ-Compressor
14	Drum Verb NonLin Reverb1+Large Plate
15	Drums X NonLin Reverb1/Large Room Rev
16	Drums Y Large Room Rev~Large Plate
17	Drums Z Small Room Rev~Large Plate
18	Kick Non Lin NonLin Reverb1~PitchShift-DDL
19	Mega Non Lin NonLin Reverb1+NonLin Reverb1
20	Wet Non Lin NonLin Reverb1~Gated Reverb
21	Best Small Space Small Room Rev+Small Room Rev
22	Echoing Verbs NonLin Reverb1+NonLin Reverb1
23	Flanged Reverb 1 Large Plate~Flanger
24	Flanged Reverb 2 Large Plate~Flanger

25	Feedback Verb Large Plate/Hall Reverb
26	Sample&Phaz Verb Large Plate~Phaser - DDL
27	Panning Delays Tempo Delay~EQ-Panner-DDL
28	MultiTap&Flange MultiTap Delay~Flanger
29	Ascending Delays PitchShift-DDL/Phaser - DDL
30	Harmoni-Echo PitchShifter/MultiTap Delay
31	Major Triad Echo PitchShifter/MultiTap Delay
32	Pan-Tapstic 8 Voice Chorus~EQ-Panner-DDL
33	Special Taps Small Room Rev~MultiTap Delay
34	Country Guitar EQ-Tremolo-DDL~Large Plate
35	Super Mute-ron VCF-Distort 1~Guitar Amp 3
36	Vocal EQ & Comp Parametric EQ~EQ-Compressor
37	Vocal Magic PitchShift-DDL~Large Plate
38	Darth PitchShifter~Flanger
39	4 Voice Detune PitchShifter+PitchShift-DDL
40	Serial Florus EQ-Flanger-DDL~8 Voice Chorus
41	Detune & Spread 8 Voice Chorus~Dual Delay
42	Vibrates & Pans EQ-Vibrato-DDL~EQ-Panner-DDL
43	Coordinates NonLin Reverb2~EQ-Flanger-DDL
44	Swirling Notch Phaser - DDL~PitchShifter
45	Regenerate PitchShift-DDL/Phaser - DDL
46	Get Rappified Tunable Spkr 1/Tunable Spkr 1
47	Feedback Phaser Phaser - DDL/PitchShifter
48	Big Non Lin 1 NonLin Reverb3~PitchShifter
49	Big Non Lin 2 NonLin Reverb1+NonLin Reverb1

2-Unit ROM Presets

50 3.6 sec Delay 2U
3.6 sec DDL 2U*N/A

51 Pitch Shift 2U
PitchShift 2U*N/A

52 Guitar Tuner 2U
GuitarTuner 2U*N/A

53 Luscious Plate
Large Plate~8 Voice Chorus

54 Versatile Hall
Hall Reverb+Small Plate

55 Warm Room
Large Room Rev+Large Room Rev

56 Small Space Amb
Small Room Rev+Small Room Rev

57 Gated Room 2
Large Room Rev~EQ-Compressor

58 Smooth Bloom
NonLin Reverb1~Large Plate

59 Jet Reverb
Hall Reverb/Flanger

60 30th St. Station
Parametric EQ~Hall Reverb

61 Early & Plate
NonLin Reverb1+Large Plate

62 Parking Garage
Large Plate+Large Room Rev

63 Classic 80s Verb
Large Plate~8 Voice Chorus

64 Bend Up Reverb
PitchShifter/Large Room Rev

65 Bend Down Reverb
PitchShifter/Large Room Rev

66 Laser Flange
Flanger~EQ-Flanger-DDL

67 Blazing Phaser
Phaser - DDL~Phaser - DDL

68 Compresd&Chorusd
EQ-Compressor~8 Voice Chorus

69 Chorus & Room
EQ-Chorus-DDL~Large Room Rev

70 Chorus & Plate
EQ-Chorus-DDL~Large Plate

71 Phaser & Room
Phaser - DDL~Large Room Rev

72 Phaser & Plate
Large Plate~Phaser - DDL

73 Flanger & Room
EQ-Flanger-DDL~Large Room Rev

74 Flanger & Plate
Large Plate~EQ-Flanger-DDL

75	Delay & Reverb 1 MultiTap Delay~Large Plate
76	Delay & Reverb 2 EQ-DDL-withLFO~Large Plate
77	TempoDly & Plate Tempo Delay~Large Plate
78	TempoDly & Hall Tempo Delay~Hall Reverb
79	RotSpkr & Plate Rotating Spkr~Large Plate
80	RotSpkr & Hall Rotating Spkr~Hall Reverb
81	Chorus-DDL-Pan EQ-Chorus-DDL~EQ-Panner-DDL
82	Sparkles & Verbs EQ-Chorus-DDL~Small Plate
83	Backing Vox-Lush Dual Delay~8 Voice Chorus
84	Backing Vox-Rock PitchShifter+NonLin Reverb1
85	Ducker & EQ Ducker / Gate~Parametric EQ
86	KeyedExpander&EQ Keyed Expander~Parametric EQ
87	Exciter & DDL VandrPolFilter~EQ-DDL-withLFO
88	Compress&De-ess EQ-Compressor~De-esser
89	Wolf EQ & Gate Parametric EQ~Gated Reverb
90	Live Vocal Chain EQ-Compressor~PitchShift-DDL
91	Drum Squasher Large Plate~EQ-Compressor
92	Digable Guitar DigitalTubeAmp~Tunable Spkr 2
93	Screamin' Amp Guitar Amp 3~TunableSpkr 1
94	Touch Wa Guitar VCF-Distort 1~Dual Delay
95	Amp Thru RotSpkr Guitar Amp 3~Rotating Spkr
96	FtSwitchLoop DDL 3.6 sec DDL 2U*N/A
97	Owed to Acadia Large Plate~EQ-Panner-DDL
98	Rhythmic Panner Tempo Delay~EQ-Panner-DDL
99	Science Lab Sine/Noise Gen~Phaser - DDL

4-Unit RAM Presets

00 Crystal Cave
EQ-Compressor~Flanger+PitchShift-DDL/NonLin Reverb1

01 Canyon Echo
Small Plate~Small Plate~PitchShifter~Dual Delay

02 Underwater Verb
Small Plate~NonLin Reverb1~Hall Rev~EQ-Vibrato-DDL

03 PhaseFlange Verb
Large Plate~Flanger~EQ-Panner-DDL~Phaser - DDL

04 MassiveVerb 1
Large Plate+Hall Reverb+Large Plate+Hall Reverb

05 MassiveVerb 2
Small Plate+Small Plate+Hall Reverb~Small Room Rev

06 Rhythm Verb
Small Plate+NonLin Reverb1+MultiTap Delay~Large Plate

07 Hugh Bright Verb
Hall Reverb+Hall Reverb+Hall Reverb+Large Plate

08 Hugh Warm Verb
Parametric EQ~Hall Reverb+Parametric EQ~Large Plate

09 Resonant Reverb
EQ-Compressor~Parametric EQ~Hall Reverb~Flanger

10 Studio Vocal 1
EQ-Compressor~PitchShift-DDL~Dual Delay~Large Plate

11 Studio Vocal 2
EQ-Compressor~PitchShift-DDL~Dual Delay~Large Plate

12 Crisp VocalChain
Parametric EQ~EQ-Compresr~Pitch Shifter~VandrPolFilter

13 Up Front Vocal
EQ-Compresr~PitchShifter~NonLin Rev1+Tempo Delay

14 BrightVocalChain
EQ-Compressor~PitchShifter~Large Plate+MultiTap Delay

15 Warm Vocal Chain
EQ-Compressor~PitchShifter~Large Plate+MultiTap Delay

16 Super Spreader
PitchShift 2U*N/A+PitchShift 2U*N/A

17 Time Machine
EQ-Compressor~Dual Delay~8 Voice Chorus~Large Plate

18 Phased Vocals 1
EQ-Compressor~Phaser - DDL~Large Plate+Tempo Delay

19 Jazz Vocal 1
EQ-Compressor~PitchShift-DDL~Large Plate~Dual Delay

20 Rockin' Lead Gtr
GuitarAmp3~TunableSpkr1~FastPitchShift~MultiTap Delay

21 Room Mic Gtr Amp
Guitar Amp 3~TunableSpkr 1~PitchShifter+Large Plate

22 UltraWide Guitar
Guitar Amp 3~Tunable Spkr 1~MultiTap Delay~Hall Rev

23 TuffButNice Gtr
GuitarAmp3~TunableSpkr 1~VandrPolFilter~Dual Delay

24 Vintage Tube Amp
EQ-Vibrato-DDL~GuitarAmp 2~SpeakerCabinet~LrgPlate

25 Rockabilly Gtr 1
Guitar Amp 3~Tunable Spkr 1~Dual Delay~EQ-Chor-DDL

26 Fast Tremolo Amp
EQ-Compressr~Guitar Amp1~EQ-Tremolo-DDL~DualDelay

27 Metal Rhythm Gtr
Guitar Amp 1~PitchShift-DDL~Small Plate~Dual Delay

28 Wah Lead Guitar
VCF-Distort 1~Guitar Amp1~SpkrCabinet+LrgRoomRev

29 SuperFunk Guitar
VCF-Distort 1~Dual Delay~Tunable Spkr 1~PitchShifter

30 Big Acoustic Gtr
EQ-Compresr~EQ-Chor-DDL+RotatingSpkr~LrgRoomRev

31 NY Studio Gtr
EQ-Compressor~Flanger~Hall Reverb+Dual Delay

32 LA Studio Gtr
EQ-Compressor~8 Voice Chorus~PitchShifter~Dual Delay

33 Studio Tremolo
EQ-Compresr~EQ-Tremolo-DDL~SmlRoomRev+DualDelay

34 Rock Bass & Lead
Guitar Amp 3~Tunable Spkr 1~PitchShifter~Param EQ

35 Mellow Jazz Gtr
EQ-Compressor~PitchShift-DDL~Large Plate~Dual Delay

36 Rhythm-DDL Gtr
EQ-Compressr~EQ-Chor-DDL~PitchShifter+MultiTapDelay

37 Comp-DDL Guitar
EQ-Compressor~8 Voice Chorus~Dual Delay~Large Plate

38 Weird Phase Gtr
EQ-Compressor~Phaser - DDL~Dual Delay~Large Plate

39 Rockabilly Gtr 2
EQ-Compresr~GuitarAmp1~EQ-DDL-wLFO~SmlRoomRev

40 Slammin' Bass
EQ-Compressr~EQ-Compressr~8Vc Chorus~VandrPolFilter

41 Super Chorus
Param EQ~8 Voice Chorus+Large Plate~8 Voice Chorus

42 Super Phase-DDL
Parametric EQ~Phaser - DDL+Large Plate~8 Voice Chorus

43 Parallelograms
PitchShifter~MultiTap Delay~Phaser - DDL+EQ-Panner-DDL

44 16-Band EQ
Param EQ~Param EQ+Param EQ~Param EQ

45 Modulation Mania
8 Voice Chorus~Flanger~Phaser - DDL~EQ-Chorus-DDL

46 32 Voice Chorus
8 VoiceChorus+8 Vc Chorus+8 Vc Chorus+8 Vc Chorus

47 Chorus-Phaser
8 Voice Chorus~Phaser - DDL+PitchShifter/Phaser - DDL

48 Twilight Tone
PitchShift-DDL/Phaser - DDL~Gated Rev~Phaser - DDL

49 CV-Wet Thunder!
Sine/NoiseGen~Phaser- DDL~EQ-DDL-withLFO~LrgPlate

4-Unit ROM Presets

50 String Section
Parametric EQ~Hall Reverb~MultiTap Delay~8 Vce Chorus

51 Quartet Chamber
Parametric EQ~Large Room Rev~Dual Delay+PitchShifter

52 MassiveVerb 3
Hall Reverb+Lrg Plate+Lrg Room Rev+Sml Room Rev

53 MassiveVerb 4
Large Plate+Hall Reverb+Small Plate+NonLin Reverb1

54 Mega-Kick Verb
EQ-Compressor~ParamEQ~NonLinRev1~EQ-Compressor

55 Mega-Snare Verb
EQ-Compressor~ParamEQ~LargePlate~EQ-Compressor

56 Mega-Toms Verb
EQ-Compressor~ParamEQ~NonLinRev1~SmallRoomRev

57 Backwards Verb
EQ-Compressor~NonLin Rev1+Sml Plate+MultiTap Delay

58 Wet Vocal Chain
EQ-Compressor~PitchShifter~Large Plate+MultiTap Delay

59 TightVocalChain
EQ-Compressor~Tempo Delay~PitchShifter~Large Plate

60 De-esVocalChain
EQ-Compressor~De-esser~MultiTap Delay~Large Plate

61 Wide Vocal Chain
EQ-Compress~PitchShift-DDL~MultiTapDelay~LargePlate

62 Jazz Vocal Chain
EQ-Compress~PitchShift-DDL~LargePlate~MultiTapDelay

63 '57 Vocal Smooth
Param EQ~EQ-Compressor~LargePlate+PitchShift-DDL

64 '57 Vocal Rap
Parametric EQ~EQ-Compressor~Large Plate+PitchShifter

65 Excited Vocals
ParamEQ~EQ-Compressr~Small Plate+VandrPolFilter

66 Pop Vocal Chain
Parametric EQ~EQ-Compressor~PitchShifter~Dual Delay

67 Phased Vocals
EQ-Compressor~Phaser-DDL~NonLin Rev1+Tempo Delay

68 MetalMaster Gtr
Guitar Amp 3~Tunable Spkr 1~PitchShifter~Large Plate

69 MonsterLead Gtr
Guitar Amp 3~Tunable Spkr 1~PitchShifter~MultiTap Delay

70 CloseMic Gtr Amp
Guitar Amp 3~Tunable Spkr 1~PitchShifter~MultiTap Delay

71 Arena Rock Gtr
Guitar Amp 3~Tunable Spkr 1~PitchShifter~Large Plate

72 Blues Lead Gtr
Guitar Amp 1~Tunable Spkr 1~Dual Delay~Small Room Rev

73 Practice Amp
EQ-Tremolo-DDL~VCF-Distort 1~SpkrCabnt~SmlRoomRev

74 Rhythm Guitar
EQ-Compressor~Guitar Amp 2~MultiTap Delay~Hall Reverb

75	Jimi's Wawa
VCF-Distort 1~TunableSpkr 1~Dual Delay~EQ-Chor-DDL	
76	Ballad Lead Gtr
EQ-Compressor~Guitar Amp 1~Dual Delay+Hall Reverb	
77	Jazzy Rhythm Gtr
Parametric EQ~EQ-Compressor~PitchShifter~Small Plate	
78	Crystal Guitar
Param EQ~EQ-Compressor~8VoiceChorus~LargePlate	
79	Super Clean Gtr
EQ-Compressor~8 Vc Chorus~PitchShifter+MultiTap Delay	
80	Studio Chords
EQ-Compressor~8 Voice Chorus~Hall Reverb+Dual Delay	
81	Studio Tube Amp
DigitalTubeAmp~TunableSpkr2~DualDelay~SmllRoomRev	
82	Super Bass
EQ-Compressor~ParamEQ~8VoiceChorus~VandrPolFilter	
83	Brass FX Chain
ParamEQ~EQ-Compressor~NonLin Reverb1~Small Plate	
84	Piano FX Chain
Param EQ~Large Room Rev+Hall Reverb~8 Voice Chorus	
85	Winds FX Chain
Param EQ~Large Room Rev+Hall Reverb~8 Voice Chorus	
86	Clean EP Chain
Param EQ~8 Voice Chorus+Hall Reverb~8 Voice Chorus	
87	BurningSax Chain
Hall Reverb~InversExpander~PitchShifter~MultiTap Delay	
88	Vibes FX Chain
Param EQ~EQ-Tremolo-DDL+Hall Reverb~8 Voice Chorus	
89	Choir FX Chain
Parametric EQ~8 Voice Chorus~Dual Delay+Hall Reverb	
90	Crystal Tines
Parametric EQ~8 Voice Chorus~MultiTapDelay+Hall Rev	
91	Dirty Organ
Param EQ~EQ-Compressor~Rotating Spkr~Small Plate	
92	Sunday Organ
Param EQ~EQ-Compress~Rotating Spkr~Lrg Room Reverb	
93	Nasty Harmony
PtchShft-DDL+PtchShft-DDL+PtchShft-DDL+PtchShft-DDL	
94	Sweet Harmony
PtchShft-DDL+PtchShft-DDL+PtchShft-DDL+PtchShft-DDL	
95	Mega Flanger
Flanger~EQ-Flanger-DDL+EQ-Flanger-DDL+Flanger	
96	Mega Phaser
Rotating Spkr~Phaser - DDL+Phaser - DDL+Phaser - DDL	
97	Mega Chorus
8 Voice Chorus~8 Vc Chorus+8 Vc Chorus+8 Vc Chorus	
98	Dense Mist
Large Plate~Phaser - DDL+8 Voice Chorus~Large Plate	
99	Guitar Tuner 4U
GuitarTuner 2U*N/A~No Effect~No Effect	

Config RAM Presets

00 MonoInVocalSetup
EQ-Compressor~PitchShifter~Large Plate+MultiTap Delay

01 StereoVocalSetup
EQ-Compressor~PitchShift-DDL~Dual Delay~Large Plate

02 Rock Gtr Setup
GuitarAmp1~TunableSpkr1~DualDelay~Small Room Rev

03 Stereo In Keybds
Param EQ~8 Voice Chorus+LargePlate~8 Voice Chorus

04 PitchSwept Chain
PitchShifter/Hall Reverb~EQ-Vibrato-DDL~Phaser - DDL

05 Lead Vox BG Vox
PitchShift-DDL~Large Plate PitchShifter+NonLin Reverb1

06 Drums Vocals
NonLin Rev1+NonLin Rev1 EQ-Compressor~FastPitchShift

07 Kik&Snr Toms
NonLin Rev1/PitchShift-DDL NonLinRev1~Phaser-DDL

08 Drums Keys
Sml Room Rev+Sml Room Rev Lrg Plate~8 Voice Chorus

09 Horns Vocals
Small Plate~Large Plate 8VoiceChorus~EQ-Panner-DDL

10 Keybds Vocals
EQ-Chorus-DDL~Large Plate MultiTap Delay~Large Plate

11 Strings Brass
Large Plate~8 Voice Chorus Small Plate~Large Plate

12 Guitar Bass
Guitar Amp 2~Rotating Spkr Phaser - DDL~Phaser - DDL

13 Drums Bass
NonLin Reverb1+Large Plate Phaser - DDL~PitchShifter

14 Bloom Room
NonLin Reverb3~LargePlate Lrg RoomRev+Lrg RoomRev

15 Kik Snr Toms
Small Plate Small Plate NonLin Reverb1+Large Plate

16 Kik Snr Cymbals
NonLin Reverb1 Large Plate Small Plate~Phaser - DDL

17 Kik Snr Vocals
Small Plate Small Plate PitchShift-DDL~Large Plate

18 Drms Bs Keybds
SmlRoomRev 8VoiceChorus EQ-Chorus-DDL~LrgPlate

19 Bas Voc Keybds
8 Voice Chorus MultiTap Delay Phaser - DDL~Large Plate

20 Bas Kys Guitar
EQ-Flangr-DDL EQ-Chor-DDL EQ-Tremolo-DDL~Lrg Plate

21 Rm1 Rm2 Hall
Small Room Rev Small Room Rev Hall Reverb+Small Plate

22 DDL Plt VoxVerb
MultiTap Delay Large Plate Large Plate/Dual Delay

23 Cho Taps LushVox
8 Voice Chorus Dual Delay Large Plate~8 Voice Chorus

24 Pan Phs Harmoniz
EQ-Panner-DDL Phaser-DDL Pitch Shift 2U*N/A

25 LeadGtr Vocals
Guitar Amp 1~Small Plate Large Plate~PitchShifter

26 CleanGtr Vocals
Parametric EQ~8 Voice Chorus Large Plate~PitchShifter

27 E.Piano VocalFX
Large Plate~Phaser - DDL Large Plate+PitchShifter

28 Fretles SoftPad
Large Plate~8 Voice Chorus Large Plate~8 Voice Chorus

29 HornRev Vocals
Small Plate~Large Plate Large Plate+PitchShifter

30 Inverse NonLin
NonLin Rev2~Large Plate NonLin Rev1+NonLin Rev1

31 Ducker Harmoniz
Ducker / Gate~Parametric EQ Pitch Shift 2U*N/A

32 De-essed DDL&Rev
De-esser~Large Plate De-esser~Dual Delay

33 Kik Snr RevDDL
Small Plate Gated Reverb Large Plate+MultiTap Delay

34 Kik Snr Percus
Small Room Rev Large Plate Small Plate~8 Voice Chorus

35 Brs Bels VocalFX
Large Plate EQ-Chorus-DDL PitchShift-DDL~Large Plate

36 DDL Plt Guitar
MultiTap Delay Large Plate VCF-Distort 1~Guitar Amp 3

37 Flg Slap Phaser
EQ-Flanger-DDL Dual Delay Phaser - DDL~Phaser - DDL

38 Phase Hrmo Vrb
Phaser-DDL PitchShifter SmallRoomRev+SmallRoomRev

39 Comp EQ VocalFX
EQ-Compressor Parametric EQ PitchShifter+PitchShifter

40 DDL Plt Flanger
MultiTapDelay LrgPlate MultiTapDelay/EQ-Flanger-DDL

41 Box Room LongDDL
Small Room Rev Small Room Rev 3.3 sec DDL 2U*N/A

42 Albumizer EQs
VandrPolFilter VandrPolFilter VandrPolFilter VandrPolFilter

43 4 Mono Reverbs
Small Room Rev Large Room Rev Small Plate Hall Reverb

44 The Jitters
Reverse Rev+FastPitchShift+Phaser-DDL/NonLin Reverb2

45 The Pad Maker
EQ-DDLwLFO/MultiTapDelay~EQ-VibratoDDL+RotatSpkr

46 Arpeggios
PitchShift-DDL~PitchShift-DDL~PitchShift 2U*N/A

47 Final Frontier
PitchShifter/Hall Reverb~EQ-Tremolo-DDL~Phaser - DDL

48 Outness
PitchShifter/Hall Reverb~EQ-Vibrato-DDL~Phaser - DDL

49 Lunar Self-Gen
PitchShifter/Large Room Rev Sine/Noise Gen~Phaser - DDL

Config ROM Presets

50 Select 4U Psets
EQ-Compressor~PitchShifter~Large Plate+MultiTap Delay

51 Select 2U Psets
EQ-DDL-withLFO~Large Plate No Effect+No Effect

52 Select 1U Psets
Small Room Rev No Effect No Effect No Effect

53 1 Src: Mono In
No Effect~No Effect~No Effect~No Effect

54 1 Src: Stereo In
No Effect~No Effect~No Effect~No Effect

55 2 Src: Stereo In
No Effect~No Effect No Effect~No Effect

56 2 Src: Mono In
No Effect~No Effect No Effect~No Effect

57 3 Src:Stereo Out
No Effect No Effect No Effect~No Effect

58 3 Src:M+M+S Out
No Effect No Effect No Effect~No Effect

59 4 Src:Stereo Out
No Effect No Effect No Effect No Effect

60 4 Src:4 Mono Out
No Effect No Effect No Effect No Effect

61 Vocoder Preset
Vocoder Low+Vocoder Mid1+Vocoder Mid2+Vocoder High

62 Vocoder Preset 2
Vocoder Low+Vocoder Mid1+Vocoder Mid2+Vocoder High

63 All Purp Mixdown
Large Plate PitchShifter NonLin Reverb1 Dual Delay

64 Rock Mixdown
Small Plate NonLin Reverb1 Large Plate Dual Delay

65 Modern Mixdown
Large Plate Gated Reverb PitchShifter Dual Delay

66 Dance Mixdown
Small Plate NonLin Reverb1 PitchShifter Dual Delay

67 Rockabily Mixdwn
Small Plate Small Plate Large Plate Dual Delay

68 R & B Mixdown
8 Voice Chorus PitchShifter Large Plate Dual Delay

69 BigBallad Mixdwn
Large Plate Gated Reverb PitchShifter Dual Delay

70 Jingle Mixdown
Large Plate Large Plate Large Plate MultiTap Delay

71 Jazz Mixdown
Large Plate Hall Reverb Large Room Rev Dual Delay

72 4 Vocal Effects
Sml RoomRev MultiTap Delay Dual Delay EQ-Compressor

73 4 Vocal Compres
EQ-Compresr EQ-Compresr EQ-Compresr EQ-Compresr

74 2 EQs 2 Limiters
Parametric EQ Parametric EQ EQ-Compresr EQ-Compresr

75	4 In-line FX
Tempo Delay	Phaser-DDL Large Plate EQ-Flanger-DDL

76	4 Regenerators
8 Voice Chorus	Flanger Phaser - DDL PitchShift-DDL

77	4 Fast Gates
EQ-Compresr	EQ-Compresr EQ-Compresr EQ-Compresr

78	4 Loudness EQ's
Parametric EQ	Parametric EQ Parametric EQ Parametric EQ

79	4 Mono DDLs
Tempo Delay	MultiTap Delay Tempo Delay Dual Delay

80	Large Halls
Large Plate	Large Plate Large Plate Large Plate

81	Oil Drums
Lrg RoomRev	Lrg RoomRev Lrg RoomRev Lrg RoomRev

82	Small Ambiences
SmlRoomRev	SmlRoomRev SmlRoomRev SmlRoomRev

83	Medium Amb 1
Hall Reverb	Hall Reverb Hall Reverb Hall Reverb

84	Gated Verbs D/B
Gated Reverb	Gated Reverb Gated Reverb Gated Reverb

85	Non Lin Colors
NonLin Rev 3	NonLin Rev 3 NonLin Rev 3 NonLin Rev 3

86	My College Dorm
Small Plate	Small Plate Small Plate Small Plate

87	Rectangle Halls
Lrg RoomRev	Lrg RoomRev Lrg RoomRev Lrg RoomRev

88	Wood Rooms
Small Plate	Small Plate Small Plate Small Plate

89	Elec Drum Rooms
NonLin Rev 1	NonLin Rev 1 NonLin Rev 1 NonLin Rev 1

90	Concrete Rooms
NonLin Rev 2	NonLin Rev 2 NonLin Rev 2 NonLin Rev 2

91	Medium Amb 2
Large Plate	Large Plate Large Plate Large Plate

92	Travel Ambiences
Hall Reverb	Hall Reverb Hall Reverb Hall Reverb

93	Tiled Ambiences
SmlR oomR ev	SmlR oomR ev SmlR oomR ev SmlR oomR ev

94	Big Amp in Room
DigitalTubeAmp~TunablSpkr2~EQ-Comprssr~SmllRmRev	

95	Domino’s Strat
DigitalTubeAmp~Tunable Spkr2~Param EQ~Large Plate	

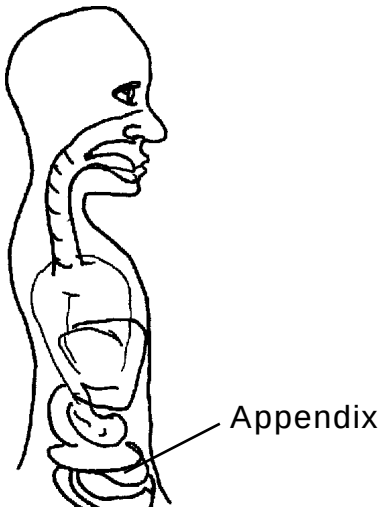
96	Pedal-Wah Guitar
VCF-Distort 1~Guitar Amp 4~EQ-Comprssr~SmllRoomRev	

97	2 Big Ster.Verbs
EQ-Vibrato-DDL/Large PLate	Hall Reverb8 Vce Chorus

98	EchoPlx FlngVerb
VCF-Distort1/EQ-DDL-w/LFO	LrgRoomRev+LrgRoomRev

99	Vocal Remover
EQ-Compressor~Large Plate	Vocal Remover~No Effect

Appendix



DP/4+ MIDI Implementation

The DP/4+ features extensive MIDI (Musical Instrument Digital Interface) implementation. For normal applications, you will find all the information you need regarding the DP/4+'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/4+ MIDI implementation.

If you are writing a computer program to communicate with the DP/4+ via MIDI, or otherwise require a copy of the full DP/4+ 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/4+ MIDI System Exclusive Specification." Please allow 2 to 3 weeks for delivery.

MODEL: DP/4+ MIDI Implementation Chart Version: 1.0

Function…		Transmitted	Recognized	Remarks
Basic Channel	Default Channels	1, 2, 3, 4, 5, 6 * 1-16	1, 2, 3, 4, 5, 6 * 1-16	
Mode	Default Messages Altered	3 X X	3 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		4 (CV Pedal) 70 (Foot Switch 1-L) 71 (Foot Switch 1-R) 72 (Foot Switch 2-L) 73 (Foot Switch 2-R)	0-127	If Control-7 is received on Control channel= Modulation Source. If Control-7 is recieved on Unit channel= algorithm volume control.
Program Change	True Number	O	0-99	Program changes sent & received on Unit channels
System Exclusive		O	O	
System Common	: Song Pos : Song Sel : Tune	X X X	O O 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 X X X	

Notes: * The DP/4+ can receive on up to 6 MIDI channels for units A,B, C, D, or Config and controllers. They may overlap in any way, except units and configs have to be different. All modulation sources are received on the control channel.

Mode 1= OMNI ON, POLY **Mode 2= OMNI ON, MONO** **O= YES**

Mode 3= OMNI OFF, POLY **Mode 4= OMNI OFF, MONO** **X = NO**

Glossary



There are a few terms that you may be unfamiliar with, and need to be understood before you can unlock the DP/4+'s full potential as a programmable effects processor. This section defines these terms.

Algorithm A control program for the digital signal processor (the basic signal processing building block in the DP/4+). 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/4+ algorithm has a three letter abbreviation which helps to identify it in Select mode. The DP/4+ algorithms are:

Algorithm:	abbreviation:	Algorithm:	abbreviation:	Algorithm:	abbreviation:
No Effect (Bypass Preset)	dry	DigitalTubeAmp	amp	EQ-Compressor	cmp
Small Room Rev	rev	DynamicTubeAmp	amp	Expander	exp
Large Room Rev	rev	Speaker Cabinet	spk	InversExpander	exp
Hall Reverb	rev	Tunable Spkr 1, 2	spk	De-esser	ess
Small Plate	rev	Rotating Spkr	rot	Rumble Filter	flt
Large Plate	rev	EQ-Chorus-DDL	cho	Parametric EQ	equ
Reverse Reverb	rev	EQ-Vibrato-DDL	vib	VandrPolFilter	flt
ReverseReverb2	rev	EQ-Panner-DDL	pan	Sine/Noise Gen	gen
Gated Reverb	rev	EQ-Flanger-DDL	fla	3.3 sec Delay 2U	ddl
NonLin Reverb 1, 2, 3	rev	EQ-Tremolo-DDL	trm	Pitch Shift 2U	pit
MultiTap Delay	ddl	Phaser - DDL	pha	GuitarTuner2U	tun
Dual Delay	ddl	8 Voice Chorus	cho	Ducker / Gate	gat
Tempo Delay	ddl	Flanger	fla	Keyed Expander	key
EQ-DDL-withLFO	ddl	Pitch Shifter	pit	Vocoder (4)	voc
VCF-Distort 1, 2	dst	Pitch Shift-DDL	pit	Vocal Remover	flt
Guitar Amp 1, 2, 3, 4	amp	FastPitchShift	pit		

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/4+ 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/4+, 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 / 4+ 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 / 4+ (equivalent to *Source Config*).

Config Preset — This the largest type of preset in the DP / 4+. It contains all the signal routing information, such as which inputs are routed to which outputs, whether the individual units (A, B, C and / or D) 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 / 4+. 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 / 4+ 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.

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/4+ 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/4+, you can control these delays to create various environments.

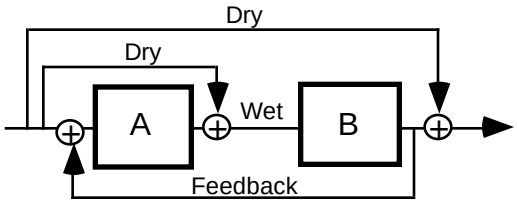
Echo A delay that is perceived as a discrete repeat of the original sound. A classic example of an echo is the effect of shouting into a canyon. You will hear your voice delayed and repeated throughout the canyon. Generally, echoes are created by long delay times.

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.

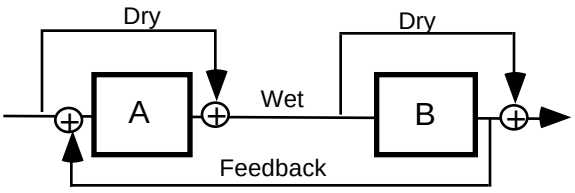
Feedback 1 A two-unit signal routing option in the DP/4+ as shown below. The A and B units are in series; the output of the B unit is mixed back into the input of the A unit:



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/4+ 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/4+, 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/4+ to provide greater control over gating, triggering, and compression algorithms.

Input Source The signal that is fed into the DP/4+ 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 EQed 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/4+ 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/4+.

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/4+, four units are available to work in parallel, possibly running different algorithms, and perhaps different input sources.

Parameter Any setting of the DP/4+ which can be changed or modified is called a parameter. The DP/4+ 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.

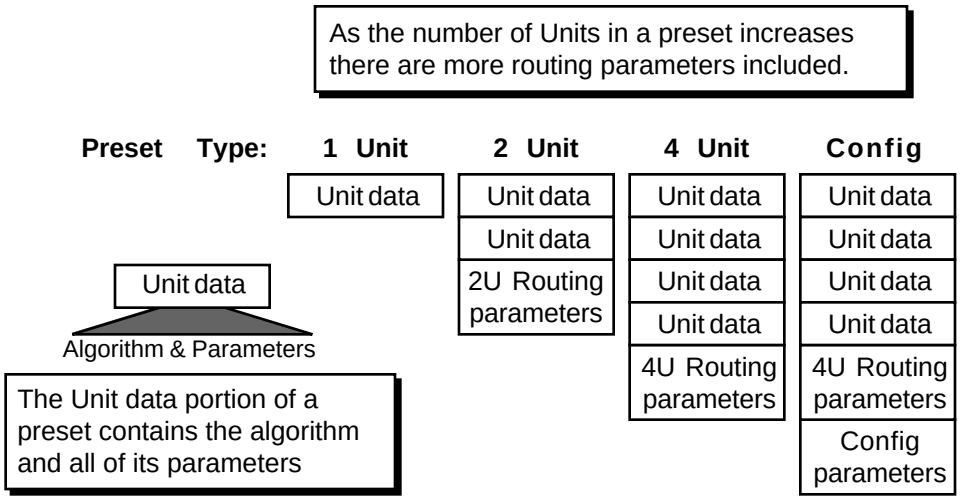
Phaser Originally conceived as an approximation to the flange effect. Allpass filters are used in place of the delay lines. Allpass filters introduce delay by modifying signal phase, hence the name.

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, B, C, and / or D. Presets affecting more than one unit also contain signal routing information.

There are four types of presets in the DP / 4+. 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 four 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
(4U)	4 Unit Preset	Four Units	Connections between all 4 units
(Config)	Config Preset	Four Units	All routing and configuration params



There are 400 presets in the DP / 4+; 100 presets (storage locations) for each type of preset. The first 50 presets (00 to 49) are user programmable (battery backed up RAM). Presets 50 to 99 are ROM factory presets:

		Preset Type			
		1 Unit	2 Unit	4 Unit	Config
Preset Number	99 | 50	50 1 Unit ROM presets	50 2 Unit ROM presets	50 4 Unit ROM presets	50 Config ROM presets
	49 | 00	50 1 Unit RAM presets	50 2 Unit RAM presets	50 4 Unit RAM presets	50 Config RAM presets

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/4+, 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/4+ uses digital algorithms to create new environments and simulate these classic ambiences.

Rumble Filter An algorithm that attenuates very low frequencies. In the DP/4+, 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/4+, 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/4+ (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 four independent effects processors in the DP/4+ are called Units, and are referred to as A, B, C, and D. Normally, each of the four Units is loaded with a different algorithm, but in some cases multiple units are combined to create one complex multi-unit effect, such as the vocoder.

Van Der 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.

Vocoder A device or algorithm that analyzes the frequency spectrum from an incoming source (e.g. speech) and applies that analysis to the sounds of another source, like a sampler/keyboard. Typical examples include: robot voice, talking orchestra, vocal electronic percussion.

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.

VanderPolFilter

The parameters available for this algorithm are:

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

The parameters available for this algorithm are:

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

The parameters available for this algorithm are:

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

The parameters available for this algorithm are:

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 Low
Vocoder Mid 1
Vocoder Mid 2
Vocoder High

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Vocoder Speech Gain
04 — Vocoder Sibilance Lev
05 — Vocoder Response Time
06 — Vocoder Pre-emphasis
07 — Mod1 Source
08 — Mod1 Destination Parameter
09 — Mod1 Param Range Min
10 — Mod1 Param Range Max
11 — Mod1 Source
12 — Mod1 Destination Parameter
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max

Edit Config Parameters

1 Source Config

00 — 1 Source Config
01 — AB Input Select
02 — AB Unit Routing
03 — CD Unit Routing
04 — AB - CD Routing
05 — AB (Config Dependent)
06 — CD (Config Dependent)
07 — Bypass Kill (Unit) A
08 — Bypass Kill (Unit) B
09 — Bypass Kill (Unit) C
10 — Bypass Kill (Unit) D

2 Source Config

00 — 2 Source Config
01 — AB Input Select
02 — CD Input Select
03 — AB Unit Routing
04 — CD Unit Routing
05 — AB (Config Dependent)
06 — CD (Config Dependent)
07 — Bypass Kill (Unit) A
08 — Bypass Kill (Unit) B
09 — Bypass Kill (Unit) C
10 — Bypass Kill (Unit) D

3 Source Config

00 — 3 Source Config
01 — CD Input Select
02 — CD Unit Routing
03 — CD (Config Dependent)
04 — AB Output Select
05 — Bypass Kill (Unit) A
06 — Bypass Kill (Unit) B
07 — Bypass Kill (Unit) C
08 — Bypass Kill (Unit) D

4 Source Config

00 — 4 Source Config
01 — AB Output Select
02 — CD Output Select
03 — Bypass Kill (Unit) A
04 — Bypass Kill (Unit) B
05 — Bypass Kill (Unit) C
06 — Bypass Kill (Unit) D

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 34 are identical to these parameters and control units B, C, D, and Config respectively.

35 — MIDI Control Channel
36 — MIDI Reception

37 — DP/4+ Controller 1
38 — DP/4+ Controller 2
39 — DP/4+ Controller 3
40 — DP/4+ Controller 4
41 — DP/4+ Controller 5
42 — DP/4+ Controller 6
43 — DP/4+ Controller 7
44 — DP/4+ Controller 8

45 — DP/4+ Foot Switch 1-L
46 — DP/4+ Foot Switch 1-R
47 — DP/4+ Foot Switch 2-L
48 — DP/4+ Foot Switch 2-R
49 — Define Song
50 — Define Step
51 — Define Preset

52 — MIDI System Exclusive ID
53 — MIDI Sys Ex Reception

54 — Preset Memory Protect
55 — MIDI Prog Change Master Switch
56 — Unit Channel Program Changes Get 1U Psets
57 — Parameter Wrap Feature
58 — Auto-load Preset (Select Mode)
59 — Remain in Select Config Mode
60 — Mix Outputs 3/4 into 1/2
61 — Set All 1U Preset Mixes To Wet
62 — Receive Control 7 On Unit Chan
63 — Send MIDI PrgChg & Controllers
64 — Data Entry Knob Response
65 — Modulation Response Rate
66 — Use Alternate ROM Presets

67 — Operating System Version

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/4+

- While holding down {SYSTEM"MIDI}, press the {B} button.
- Press the {>} button once.
- Press {WRITE} to reinitialize the DP/4+.

Specs

Frequency response (wet and dry) = 2 Hz–18 KHz

Signal-to-noise (“A” weighted) = at +4 dBu
in to out = -90 dB
output only = -100 dB
at -10 dBV
in to out = -90 dB
output only = -95 dB

THD + Noise (“A” weighted) = better than .0032% (-90 dB)
at input levels of -12 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 = 18.6 KΩ
- Front Input (1/4”)= 910. KΩ
- Front Input (XLR)= 1. KΩ

Output impedance = 300 Ω

Maximum output level into > 10. KΩ = +19.2 dBu

Maximum output level into 600. Ω = +15.7 dBu

(4) 24/48 bit DSP chips yield 40 MIPS processing power

Digital to Analog conversion = 16 Bit

Analog to Digital conversion = 16 Bit

256K words of delay memory (512 Kbytes)

Max delay time per unit = 1.6 sec.

Max single delay time possible (no regeneration) = 6.4 sec.

Preset Memory = 400, divided between 200 ROM, 200 RAM (user)

Headphone output = 37.mW/channel into 600 Ω, 14. mW/channel into 30 Ω

Physical

4 audio inputs, 4 audio outputs (phone jacks)

“ground compensated” outputs

rear panel +4 dBu/-10 dBV switch for pro and semi-pro equipment

separate input and output level controls for 4 channels; accommodate -34.6 dBV to +22 dBu

three level indicator LEDs per channel (-30 dB, -12 dB, -6 dB)

32 character back-lit LCD display

digital 32 step parameter knob

MIDI in/out and thru

analog control voltage pedal input

2 dual foot switch inputs

internal power supply, detachable power cord, internal fuse

Dimensions

19” (48.26cm) wide x 3 1/2” (8.87cm) high x 13” (33.02cm) deep

19” rack mount standard, 2U high

9.5 lbs. (4.30 kilograms)

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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/4+'s Song feature.

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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/4+.

Unit: A B C D	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	