Musician's Manual

parallel effects processor

DP/4

ensoniq®

READ THIS FIRST!

WARNING!!

Grounding Instructions

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

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

CAUTION

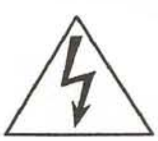
RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION : TO REDUCE THE DANGER OF ELECTRIC SHOCK

DO NOT REMOVE COVER (OR BACK)

NO USER SERVICEABLE PARTS INSIDE

REFER SERVICING TO QUALIFIED SERVICE PERSONNEL



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

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

SEE IMPORTANT SAFETY INSTRUCTIONS ON BACK COVER!

ENSONIQ™

parallel effects processor DP/4

Musician's Manual

Version 1.02

DP/4 Musician's Manual:

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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.

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

IMPORTANT:

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

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

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

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

In order to fulfill warranty requirements, the DP/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.

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

Part # 9310 0104 01 - D

Model # MM-80

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DP/4 List of Tips

Welcome!

Congratulations, and thank you for your purchase of the ENSONIQ DP/4 Parallel Effects Processor. The DP/4 is a great sounding 24-bit digital effects processor combining four independent effects processors, and four independent inputs and outputs featuring full mixdown capabilities. The DP/4 is equally at home in a recording studio, home studio, guitar rig, MIDI setup, or PA system.

The Effects

The ENSONIQ DP/4 Parallel Effects Processor has over 40 high fidelity fully programmable digital effect algorithms. Various reverb, chorusing, flanging, delay, distortion, pitch shifting and an assortment of other programs are provided with dynamic control over many of the settings. There are 400 effect presets; 200 ROM (Read Only Memory) and 200 additional RAM (Random Access Memory) presets for you to create and edit.

Parallel Processing

While other so-called multi-effects processors only allow one input signal to be "effected," the DP/4's four-in, four-out design permits 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 digital effects, like vocoding and ducking.

The DP/4 can be used as one huge effects box, or two stereo-in effects boxes, or 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 return channels by mixing the stereo outputs of the four effects units down to a single stereo pair (outputs 1 and 2).

The Manual

This manual is your guide to unlocking the full power of the DP/4. At this point, you're probably anxious to plug your DP/4 in and use it. *Section 1 — Tutorial* is a quick guide covering two scenarios for plugging in, hooking up, and using the DP/4.

After the initial "I just gotta hear it" phase has passed and you're ready to utilize the full potential of the DP/4, please take the time to read through the rest of the sections on effects, modes, and advanced programming. They'll provide valuable information and tips, as well as speeding up the learning process and enjoyment of the DP/4.

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.

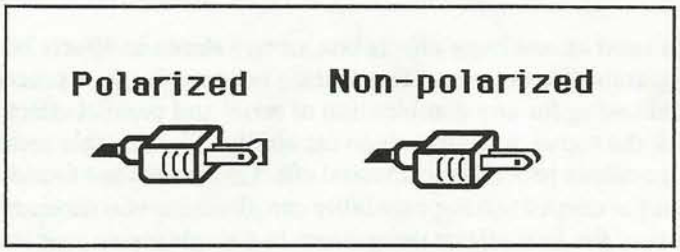
Power — Grounding Information

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 that 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.

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.

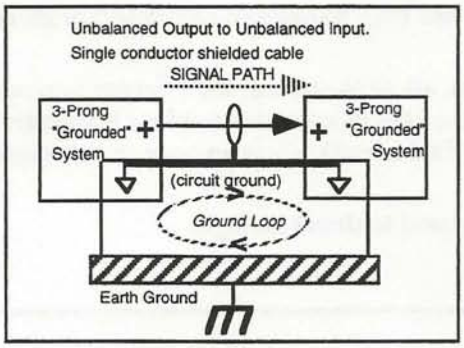


FIG. 1

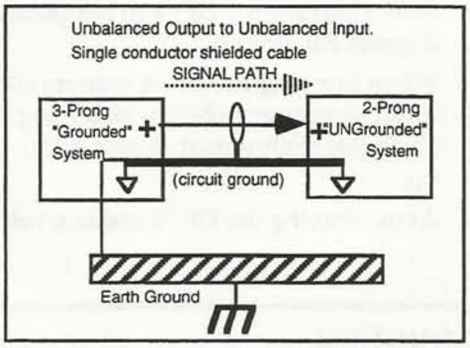


FIG. 2

Fig. 1 depicts a system interconnection where a ground loop can exist. Fig. 2 depicts a system interconnection where a ground loop does NOT exist. When interconnecting 3-prong grounded systems, you can use signal isolation transformers to prevent ground loops. This coupling transformer effectively isolates two interconnected system signal grounds, while allowing the signal to pass through.

AC Line Conditioning

As is the case 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.

Temperature Guidelines

The inner workings of the DP/4 contain 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 twenty 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

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.

Powering Up Your DP/4 In a MIDI Set Up

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

The great power and flexibility of the DP/4 lies in the fact that it is really a computer — a computer disguised as an effects processor, but a computer nonetheless. The software that operates the DP/4 is very sophisticated. In fact, there is a 128K computer program that runs inside the DP/4 (the Operating System code). That's as much as some personal computers. If you have ever used a computer, you should be familiar with the need to occasionally reboot your system when you get an error message, etc. Reinitializing the DP/4 is the equivalent of rebooting your computer.

There are many things that can happen to the DP/4 (or any computer system) that might scramble the system software — voltage surges, power failures, static electricity, etc. As with any computer, very infrequently some unforeseen event or combination of events can cause the software to become confused with strange and unpredictable results. Sometimes computers that appear to be broken have no hardware problem, just corrupted data in the internal RAM (Random Access Memory). Sometimes, simply turning the DP/4 power off and then on again will cure the problem. If that doesn't work, perhaps what is needed is to reinitialize the unit.

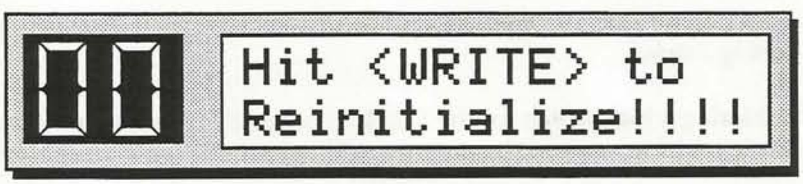
When to Reinitialize:

If your DP/4 begins to behave in peculiar ways, if the display shows words or lines that shouldn't be there, if you start getting Unexpected Event messages, if the edit functions start doing unpredictable things, try reinitializing the DP/4 before you seek factory service.

> **Warning!** When you reinitialize your DP/4 all your current RAM presets and system parameter settings will be lost. (The ROM (Read Only Memory) presets are automatically loaded back into the internal memory after reinitializing.) Therefore good backup habits should be an important part of your routine. Save any important data before reinitializing the DP/4.

To Reinitialize the DP/4:

1. While holding down the **System•MIDI** button, press the **Unit B** button.
2. Press the **Right Arrow** button. The following screen will appear:



3. Pressing the **Write•Copy** button at this point will reinitialize the DP/4.

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

Low Battery Voltage — When to Replace the Battery

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, it must be replaced by an Authorized ENSONIQ Repair Station.

The battery that came in your DP/4 is good for up to five years of life. 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 fixed period of time, and then you can commence with normal operation. Make sure that all 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.

DP/4 Accessories

These optional accessories are available from your **ENSONIQ** Dealer:

- **CVP-1 PEDAL** — A *Control Voltage Foot Pedal* which can be assigned as a modulator to parameters within the DP/4.
- **SW-10 Foot Switch** — The recommended foot switch for use with the DP/4. The SW-10 is a dual (piano-type) foot switch with two separate pedals. When the SW-10 is connected, the pedals can be programmed independently to act as a bypass effect switch, offer two separately programmable modulation sources or increase/decrease presets.
- **SW-2 Foot Switch** — This mono foot switch can increase or decrease presets, or act as a bypass effect switch. It can also function as a modulation controller.

For a full discussion of the differences between these two foot switches and how to use them, see *Section 2 — Getting Started.*

If you are considering a foot switch for the DP/4, we strongly recommend purchasing the SW-10 Dual Foot Switch, which gives you the most versatility with the least compromise.

Section 1 — Tutorial

Decisions, Decisions

The DP/4 is more than just an effects processor. In fact, it's four independent effects processors (labelled **A, B, C** and **D** on the front panel), plus a patch bay for interconnecting those processors, all in one 2-rack high unit!

Having all these elements in one package gives the DP/4 incredible flexibility, allowing you to use it in many different ways; as a multi-effects unit for one input source (such as a guitar or keyboard), as independent effects processors connected to four effects sends from your mixing board, as two independent stereo effects processors, and the list goes on.

This section of the manual is designed to help you plug in, turn-on, and try-out the DP/4, but as you can tell from the above description, there are many ways of doing this. So, you'll first need to determine how you want to use the DP/4, then we'll proceed.

In this tutorial, we will start with "Quick Steps to Hear the Presets," then focus on the two most common ways the DP/4 can be used. They are:

- a guitar or one source configuration, applying four different effects to one incoming signal,
- a mixing or four source configuration, with the four independent effects processors inside the DP/4 being fed by four effects sends (or aux. busses) from a mixing board.

Other configurations are covered later in this manual, but these two commonly used configurations will allow you to experience and understand all of the major components and features of the DP/4. However, please read on into *Section 2 — Getting Started*, and the rest of the manual to fully comprehend all the complexities and features of the DP/4.

And now we'll begin. The tutorial will go over initial hook-up and preset selection for a one source configuration first, then cover the same ground for a four source configuration. After that, some information that is common to both configurations (and for that matter, all the various configuration set-ups in which the DP/4 can be used) regarding simple editing and saving of presets will be discussed.

Let's plug in.

Quick Steps to Hear Presets

This page of the Tutorial will take you step-by-step through the DP/4 so that you can *quickly* select and hear all of the presets. Plug your mono sound source into the front panel jack labeled Input 1, or for stereo into the Inputs labeled 1, 2 on the back of the DP/4. Connect Output 1 (and Output 2 for stereo) to an audio source (amplifier, mixing board, etc). If you don't hear any sound, set all Input and Output knobs to a 12 o'clock position, and adjust the Input levels as needed.

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
    cmp→pit→rev+ddl
```

3. Press the **Select** button again.
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 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
    ddl→rev dry+dry
```

3. Press the **Select** button again.
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 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
    rev dry dry dry
```

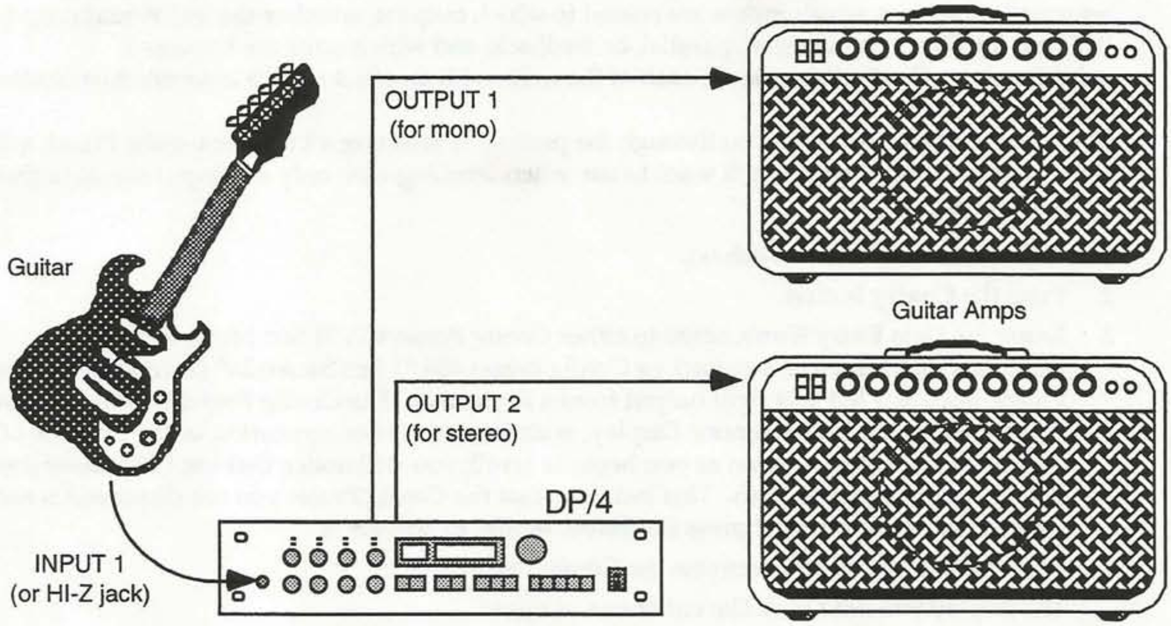
3. Press the **Select** button again.
4. Turn the **Data Entry Knob** to display the names of the different 1 Unit presets.
5. Press the **Select** button to hear the preset on the display.
6. To select other 1 Unit Presets, turn the **Data Entry Knob** and press **Select** whenever you see a Preset you'd like to hear.

Plugging In for a 1 Source Config

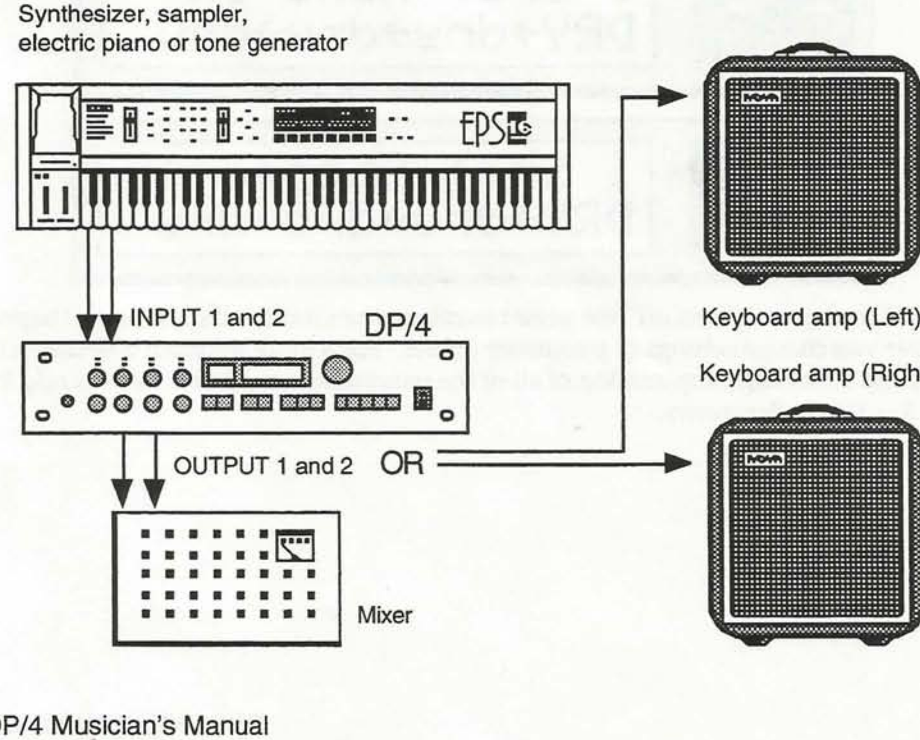
The following examples illustrate how to make the necessary connections with the DP/4 if you intend to use it in a 1 source configuration. A 1 source configuration will allow you to apply up to four separate effects to one incoming audio signal.

> **Note:** If you desire to use the DP/4 in a four source configuration with a mixing board (to apply different effects to as many as four separate signals), see the plugging in for 4 source diagram later in this section.

For an Electric Guitar or Bass:



For a Keyboard (1 Source Configuration):



Plugging In

Using the diagrams on the previous page, make the necessary connections. A single input, such as a guitar or bass can be plugged into the Input 1 jack on the front panel of the DP/4, instead of the Input 1 jack on the rear panel.

Selecting a 1 Source Config

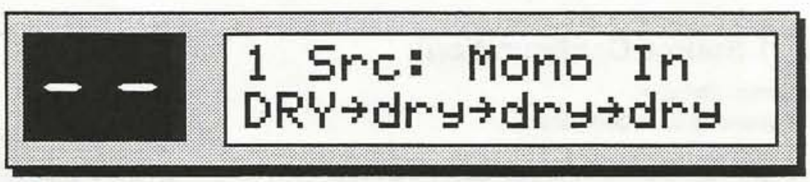
In order to determine how the DP/4 routes its signal paths between the various inputs and outputs, you must select a *Config Preset*.

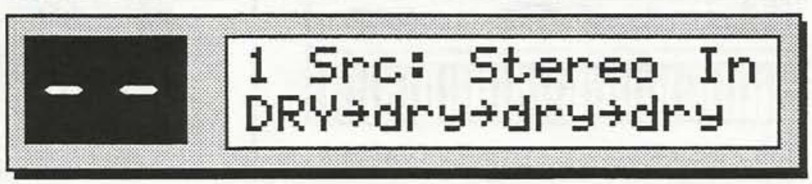
A Config Preset is the largest type of preset in the DP/4. It contains all the signal routing information, such as which inputs are routed to which outputs, whether the individual units (A, B, C and/or D) are run in serial, parallel, or feedback, and which units are bypassed. Additionally, a Config Preset loads each of the units with an effect and its associated parameters.

The following steps will take you through the process of selecting a 1 Source Config Preset, which is the type of Config Preset you'll want to use when working with only one input signal (either mono or stereo):

1. Turn the DP/4 power switch on.
2. Press the **Config** button.
3. Using the **Data Entry Knob**, scroll to either Config Preset #53 "1 Src: Mono In" (if you're using one input, such as a guitar), or Config Preset #54 "1 Src: Stereo In" (if you're using two inputs, such as a left and right output from a keyboard). The Config Preset location number will appear in the LED Numeric Display, with the description appearing in the two-line LCD display to the right. As soon as you begin to scroll, you will notice that the LED above the **Select** button begins to flash. This indicates that the Config Preset you see displayed is not yet selected, and you must press the **Select** button to activate it.
4. Press the **Select** button to activate the Config Preset.

   The display will now look like either one of these:





The LED Display will "turn off" the preset number when it is invalid. This will happen whenever you change settings or parameter values. For a more in-depth discussion on Config Presets, and a thorough explanation of all of the parameters associated with Config Presets, see *Section 3 — Config Parameters*.

Setting Levels

After you've selected the proper Config Preset for your setup, the next thing you'll need to do is set your input and output levels. These 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 1-4, the bottom row controls the output levels for Outputs 1-4. Separate controls exist within the DP/4 for setting the Mix and Volume of the individual units (see Simple Editing later in this section).

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 LED above that Input Knob(s) will begin flashing as soon as a signal is detected.
2. Continue turning the knob(s) clockwise until the red LED above the Input Knob begins to flash. This red LED starts to flash 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 knob clockwise as far as you can until you begin to 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 back down (counterclockwise) just enough so that distortion is no longer present.
4. Repeat this process for any additional outputs you have connected.

Selecting Presets

After a 1 Source Config has been selected, the DP/4 automatically takes you to Unit A, and all four unit LEDs (yellow) will be lit. At this point you can select different 4 Unit Presets.

A 4 Unit Preset is a collection of 4 algorithms and their associated parameters which are placed in the four units (**A**, **B**, **C** and **D**) in a particular order. Think of a 4 Unit Preset as a way to load new effects that are designed to work well together into each of the 4 effects processors in the DP/4.

To do this, after having selected a 1 Source Config Preset as described above:

1. Turn the **Data Entry Knob** to view the different 4 Unit presets (there are 50 RAM and 50 ROM locations). The bottom line of the display shows abbreviations for the algorithms included in that 4 Unit preset and how they are routed to each other. Again, notice that the LED above the **Select** button begins to flash as soon as you begin turning the **Data Entry Knob**, indicating that the 4 Unit Preset you see displayed is not yet selected, and you must press the **Select** button to activate it.
2. Press the **Select** button to activate the new 4 Unit Preset.
3. Repeat the above process, continuing to select various 4 Unit Presets, to get a sampling of the variety of high-quality digital effects that the DP/4 has to offer.

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 unit simply means that the signal will "go around" that particular unit, and the signal will not be affected by that unit's algorithm. 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.

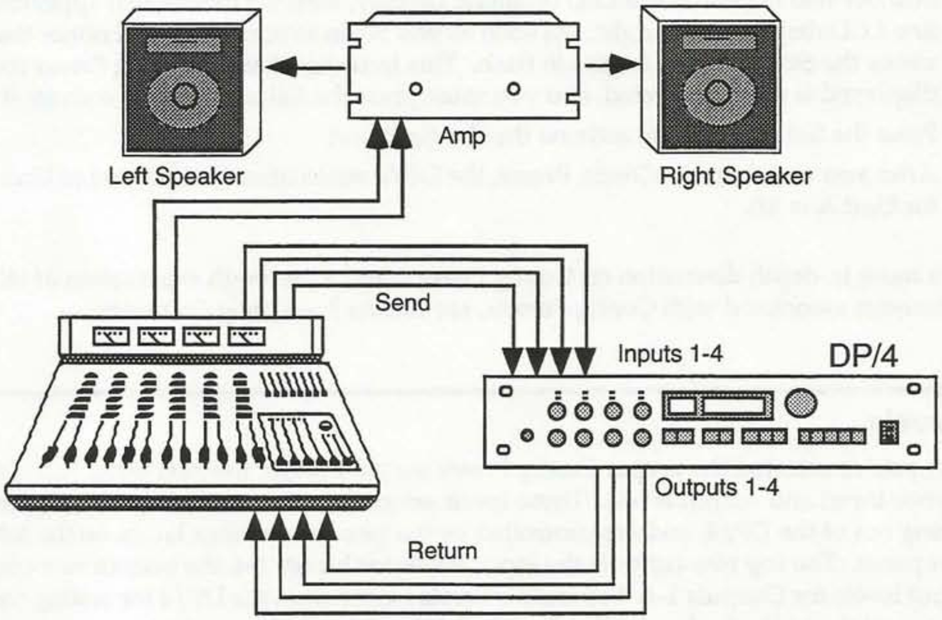
In Select mode, if the Config button has been pressed at least twice and a new Config Preset was not selected, after a few seconds the DP/4 will exit Config mode returning you to Unit A (the yellow Config LED will not be lit). For more information, refer to the description of parameter 59 in *Section 6 — System•MIDI Mode*.

Plugging In for a 4 Source Config

The following examples illustrate how to make the necessary connections with the DP/4 if you intend to use it in a four source configuration. A four source configuration will allow you to apply a completely independent effect to each of four incoming audio signals.

Note: If you desire to use the DP/4 in a one source configuration with a guitar, bass, or keyboard (to apply up to four effects to a single incoming audio signal), see the plugging in a 1 source config diagram earlier in this section.

For PA, Studio, or Mixer (4 Source Config):



Plugging In

Using the diagram shown above, make the necessary connections.

Selecting a 4 Source Config

In order to determine how the DP/4 routes its signal paths between the various inputs and outputs, you must select a *Config Preset*.

A Config Preset is 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-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.

The following steps will take you through the process of selecting a 4 Source Config Preset, which is the type of Config Preset you'll want to use when working with four independent input signals (such as the four aux. sends from a mixing board):

1. Turn the DP/4 power switch on.
2. Press the **Config** button.
3. Using the **Data Entry Knob**, scroll to either Config Preset #59 "4 Src: Stereo Out" (if you're using four separate mono effects sends, but returning to a single stereo effect return bus), or Config Preset #60 "4 Src: 4 Mono Out" (if you're using four separate effects sends and returning to four independent [mono] effect return busses). The Config Preset location number will appear in the LED Numeric Display, with the description appearing in the two-line LCD display to the right. As soon as you begin to scroll, you will notice that the LED above the **Select** button begins to flash. This indicates that the Config Preset you see displayed is not yet selected, and you must press the **Select** button to activate it.
4. Press the **Select** button to activate the Config Preset.
5. After you've selected a Config Preset, the DP/4 automatically takes you to Unit A (the LED for Unit A is lit).

For a more in-depth discussion on Config Presets, and a thorough explanation of all of the parameters associated with Config Presets, see *Section 3 — Config Parameters*.

Setting Levels

After you've selected the proper Config Preset for your setup, the next thing you'll need to do is set your input and output levels. These 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 side of the front panel. The top row controls the input levels for Inputs 1-4, the bottom row controls the output levels for Outputs 1-4. Separate controls exist within the DP/4 for setting the Mix and Volume of the individual units (See Simple Editing later in this section).

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 clockwise. The green LED above that Input Knob will begin flashing as soon as signal is detected.
2. Continue turning the knob clockwise until the red LED above the Input Knob begins to flash. This red LED starts to flash 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.
2. If set to Config Preset #59 "4 Src: Stereo Out," slowly turn Output Knobs 1 and 2 clockwise. You should begin to hear signal coming through the DP/4 into your amplifier, mixer, etc.
3. If set to Config Preset #60 "4 Src: 4 Mono Out," slowly turn each corresponding Output Knob clockwise. You should begin to hear signal coming through the DP/4 into your amplifier, mixer, etc.
4. Continue turning the knob(s) clockwise as far as you can until you begin to 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.
5. Turn the Output Knob(s) back down (counterclockwise) just enough so that distortion is no longer present.

Selecting a Unit

Because each unit is routed independently, you select a different algorithm for each unit. Each unit then processes one of the four audio sources coming into the DP/4. To select a unit:

- Press the individual unit buttons (**A**, **B**, **C**, or **D**). Each units yellow LED should light individually.

Selecting 1 Unit Presets

After an individual unit has been selected within a 4 Source Config Preset, you can select different 1 Unit Presets for the selected unit.

A 1 Unit Preset consists of an algorithm and discrete settings for each of its parameters. The 1 Unit Preset is the smallest of the preset types in the DP/4, and the basic building block for all others (2 Unit, 4 Unit, and Config Preset).

To do this, after having selected an individual unit within a 4 Source Config Preset as described above:

1. Turn the **Data Entry Knob** to view the different 1 Unit presets (there are 50 RAM and 50 ROM locations). The top line of the display shows the name of the 1 Unit Preset, while the bottom line of the display tells you the name of the algorithm being used within the 1 Unit Preset. Again, notice that the LED above the **Select** button begins to flash as soon as you begin turning the **Data Entry Knob**, indicating that the 1 Unit Preset you see displayed is not yet selected, and you must press the **Select** button to activate it.
2. Press the **Select** button to activate the new 1 Unit Preset.
3. Repeat the above process for each of the four units, continuing to select various 1 Unit Presets, to get a sampling of the variety of high-quality digital effects that the DP/4 has to offer.

Bypassing Units

At some point while you're selecting various 1 Unit presets, you may want to listen to how that individual units effect is processing the incoming audio signal. In this case, you would need to *bypass* the unit, in order to hear the signal in its original, unprocessed state. To bypass a unit simply means that the signal will "go around" that particular unit, and the signal will not be affected by that unit's algorithm. 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 all of the units will be bypassed.
3. Further presses of the **Config** button will toggle out of and into bypass.
4. In Select mode, after a few seconds, the DP/4 will exit Config mode returning you to Unit A. For more information, refer to the description of parameter 59 in *Section 6 — System•MIDI Mode*.

Simple Editing

Once you have selected a preset, you can edit it. Algorithms, parameters, routings, and configurations can all be edited and are covered in more detail in later sections of the manual. Here are a few simple editing steps to begin understanding the DP/4's full potential.

Choosing Units to Edit

Before you can edit the algorithms, you need to select each unit individually:

1. Press **Edit**.
2. Press Unit **A**, **B**, **C**, or **D** to select that unit's algorithm (effect) for editing. The active unit's LED should be lit. The LED display should say "00," if not,
3. Press the **Left Arrow** button until the display looks similar to this:

Choosing Different Algorithms

After you've chosen a unit, the LCD display will show several things. First, the letter in the upper left hand corner shows which unit is active. In this case, it's Unit A. This is followed by the algorithm name that is located in Unit A (name flashing). The second line shows the mix of the effect to the dry signal, and the volume of the algorithm.

- Turn the **Data Entry Knob** to select any of the available effect algorithms. The LED display will show the single effect algorithm preset location, and after one second it will "zero out" to the first parameter of the algorithm. This zeroing out loads that algorithm into the selected unit, replacing the algorithm that was there previously. Try selecting several new algorithms in this manner.

Editing the Mix

- Press the **Right Arrow** button once to select the Mix parameter (01 in the LED Numeric display). The mix amount will be flashing. Move the **Data Entry Knob** clockwise or counter clockwise to change the mix ratio of the effect to the dry signal. As soon as you change the mix value (with the **Data Entry Knob**), the Edit LED flashes. This indicates that you have now changed a parameter from its default value, and the newly edited version is in the EDIT BUFFER. By pressing the **Edit** button you can toggle between the original setting (LED solidly lit) and the newly edited version (LED flashing) of that parameter. The parameter values shown in the display always reflect what is currently being done to process the signal. In other words, what you see is what you hear.

Changing the Parameters

All of the other parameters within the selected unit can be edited in the same fashion:

- Use the **Left** and **Right Arrow** buttons to select the parameters, and the **Data Entry Knob** to change the values of the active parameter.

Saving Your Edited Preset

Setting the Preset Memory Switch

Now that you've created and edited your own preset, you can save it to one of the internal programmable preset locations. To do this, the Preset Memory Protect switch needs to be set to the "Off" position. If this is not set, the display will read "WRITE PROTECTED." The DP/4 defaults to "On" so that you don't accidentally wipe out any previously saved presets. To set the switch to "Off":

1. Press the **System•MIDI** button.
2. Using the **Left** and **Right Arrow** buttons, scroll until the display shows:

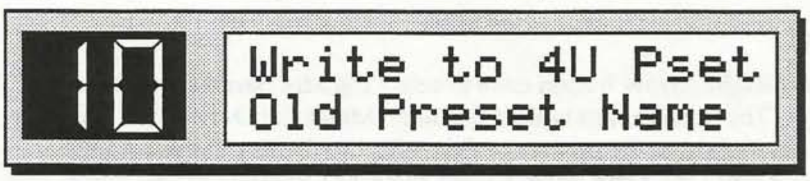
3. If the word "On" is flashing, move the **Data Entry Knob** counterclockwise to the "Off" position. If the word "Off" is flashing, you can save your new preset.

   Tip: There is a quick way to get to this page. The System•MIDI parameters are divided into sub-groups. By pressing the **System•MIDI** button several times, you can quickly scroll through these sub-groups. Parameter #52, is the first page of one of these sub-groups.

Once this protection 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 will still be intact.

Saving a Preset:

In Edit mode, press the **Write•Copy** button. The display looks like this:



The display will show one of four different screens:

- **Write to 1U Pset** — This shows that you are writing a 1 Unit preset.
- **Write to 2U Pset** — This shows that you are writing a 2 Unit preset.
- **Write to 4U Pset** — This shows that you are writing a 4 Unit preset.
- **Write to Cf Pset** — This shows that you are writing a Config preset.

Using the **Data Entry Knob**, you can choose a location (preset numbers 00 through 49) for your new preset. Notice that the LED Numeric display shows the destination number for your preset, while the bottom line of the LCD displays the name of that location's current preset (the one you would replace). The first 50 storage locations are user programmable (battery backed up). Presets 50 to 99 are ROM factory presets and cannot be replaced.

1. Press the **Cancel•Undo** if you wish 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 wish to save a config preset.
2. Press the **Write•Copy** button again. The display will show:

At this point you can name your "ultimate preset." The **Left** and **Right Arrow** buttons will move the cursor left and right, and the **Data Entry Knob** will change the alphanumeric characters. For more about this, refer to *Section 7 — Storage.*

There are 50 RAM (programmable) locations available for each type of preset. Depending on what type of preset you've created, it can only be saved to one of its appropriate locations.

Once you have written your ultimate preset's name...

3. Press the **Write•Copy** button a third time to save your preset. The display will momentarily read:

Tip: After you've saved your preset, you may want to reset the Preset Memory Protect switch back to the "On" position to eliminate any risk of accidentally deleting your new preset.

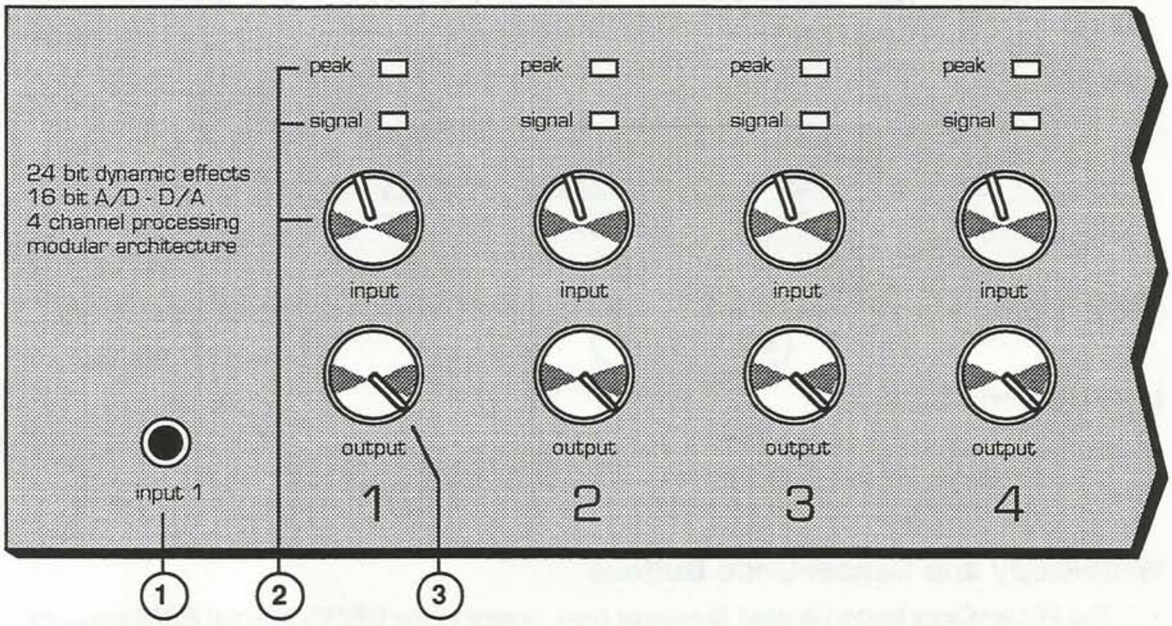
Bailing Out

At any point in this process (prior to the final pressing of the **Write•Copy** button), you can "bail out" if you decide that you no longer want to save your edited preset. To do this, press the **Cancel•Undo** button twice. This will place you back in Edit mode, and you can proceed from there.

Section 2 — Getting Started

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) Input 1 — (Guitar Input)

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

2) Input Knobs — Signal/Peak LEDs

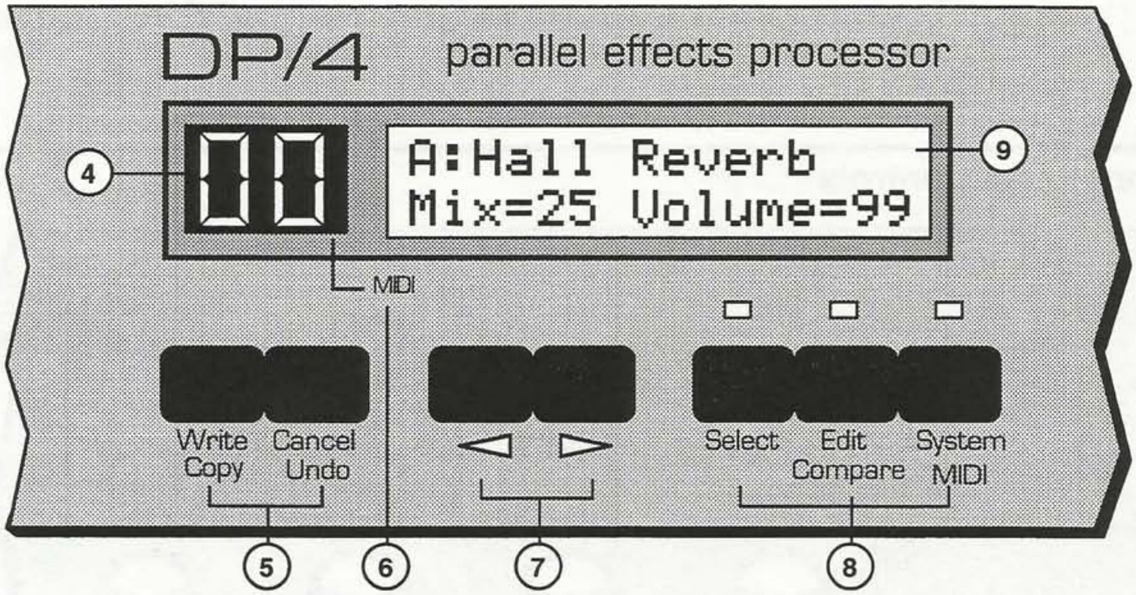
These four input knobs control the gain applied to the input signals. The input circuitry is designed to work with signals ranging from -12.5 dBV to +18 dBV. Use this knob to set each input to the optimal level for the signal you are feeding into it.

The two LEDs above each knob indicate the level of the input signal being fed into the Analog-to-Digital Converters (ADCs). The Signal LED will light when a low level signal is present at the input. Note that extremely low level input signals may not trigger the Signal LED. The Peak LED will light when the incoming signal reaches -6dB below the ADC clipping point. For optimal level, adjust the Input knob so that the red "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.

3) 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 +15 dBV.

Front Panel Controls (cont'd.)



4) LED Numeric Display

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

5) Write•Copy and Cancel•Undo Buttons

The **Write•Copy** button is used to save or copy presets to the DP/4's internal RAM memory. 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.

6) MIDI Message Indicator

The MIDI Message Indicator lights when any MIDI (Musical Instrument Digital Interface) events are received; useful for troubleshooting MIDI connections.

7) 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.

8) Mode Buttons

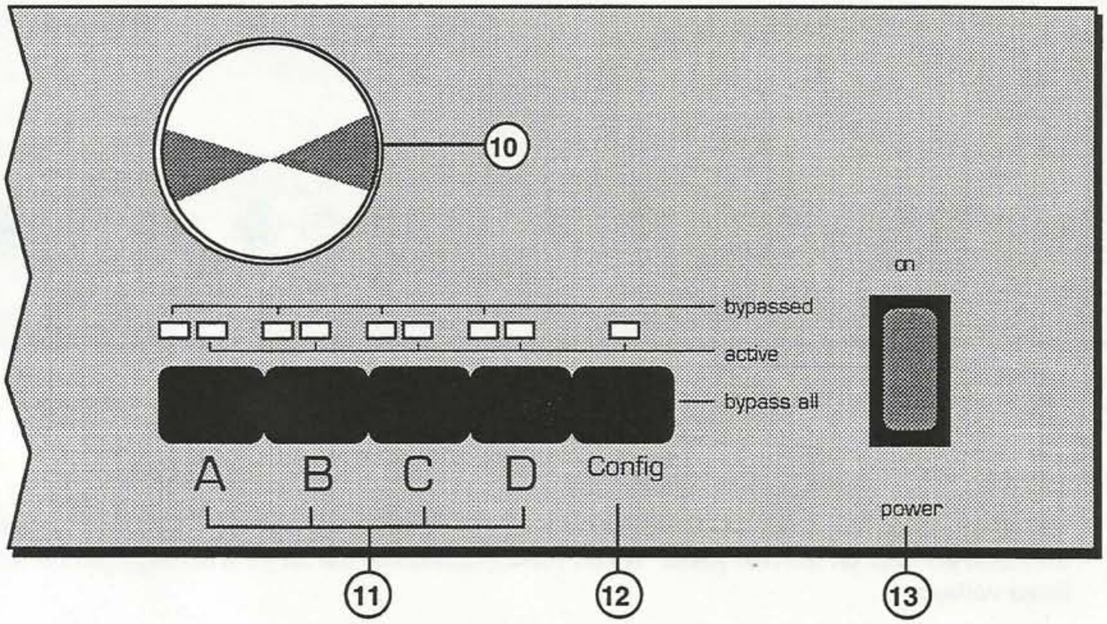
The DP/4 is always in one of these three *Modes* — SELECT, EDIT, or SYSTEM•MIDI. The current mode is selected by pressing the appropriate mode button. The yellow LED above each button shows the current mode.

- SELECT mode 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.
- EDIT mode is used to edit preset parameters, edit preset titles and save presets.
- SYSTEM•MIDI mode is used to view and modify system (or global) and MIDI parameters.

9) LCD Display

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

Front Panel Controls (cont'd.)



10) Data Entry Knob

In Select mode, rotate the **Data Entry Knob** to 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.

11) 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 solidly lit). Pressing again will reactivate that Unit.

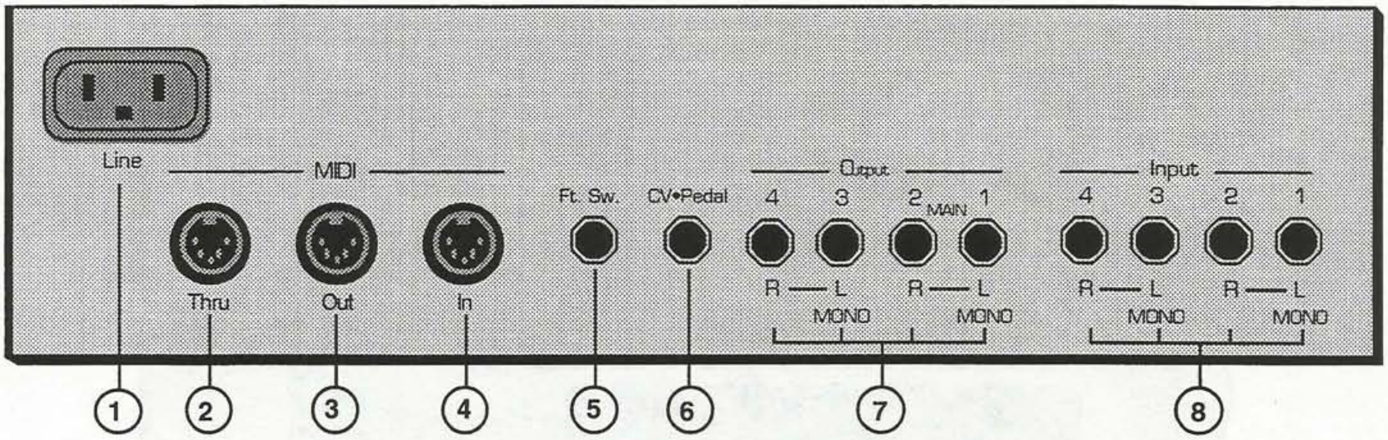
12) Config Button

The **Config** button allows you to select config presets and edit config parameters (e.g. signal routing). When Config is active, the yellow LED above the button will be lit. By pressing the **Config** button a second time, you can bypass all four Units (all red LEDs lit). Pressing **Config** a third time will reactivate the Units.

13) Power

The power switch turns the DP/4 on and off. When you turn the power on, the display will show "ENSONIQ * DP/4," and then go to Select mode.

Rear Panel Connections



1) AC Line In

The supplied line cord should be connected here. The correct voltage for the DP/4 is listed with the serial number on the rear panel. If you travel, remember the DP/4 will only operate on the listed voltage.

2) MIDI Thru

"Passes on" all MIDI 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 "Send MIDI PrgChg + Controllers" is set to "ON."

4) MIDI In

This jack receives MIDI (Musical Instrument Digital Interface) information from other MIDI instruments or computers.

5) Dual Foot Switch Jack

This jack supports either one or two foot switches depending on what is plugged into it:

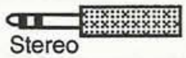
- The recommended foot switch for use with the DP/4 is the ENSONIQ Model SW-10 Dual Foot Switch. The SW-10 is a dual (piano-type) foot switch with two separate pedals. When the SW-10 is connected, the pedals can each be programmed independently to act as effect bypass switches, to provide two separately programmable modulation sources or to select presets.
- You can also use the optional ENSONIQ Model SW-2 Foot Switch. If you plug the SW-2 into this jack, it can increase or decrease presets, act as an effect bypass switch, or function as a modulation controller.

If you are considering using a foot switch with the DP/4, we strongly recommend the SW-10 Dual Foot Switch, which gives you the most versatility with the least compromise.

Rear Panel Controls (cont'd.)

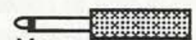
Note: When the SW-2 (or any mono foot switch) is plugged in, it will act as a permanent shut-off switch for the second (stereo) foot switch. Many of the quick steps for getting around on the DP/4 (requiring two simultaneous button presses) will not work properly because the DP/4 reads the second (stereo) foot switch connection as constantly engaged. Also, when a mono foot switch is connected and the DP/4 power is switched on, you will briefly see "Button #15" in the display. This is normal, and does not indicate a problem.

When the SW-10 is connected to the FootSwitch jack:



It offers **two** separately controllable pedals that can select presets, bypass effects, or act as modulation sources

When the SW-2 is connected to the FootSwitch jack:



It offers **one** controllable pedal that can select presets, bypass effects, or act as a modulation source and disables FtSw2

6) Pedal/CV

This jack is for connecting an optional 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 = 470 Ohm resistor to +5 volts, sleeve = ground). 2Meg input impedance, DC coupled. Input voltage range = 0 to 5 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 mono 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.

8) Input Jacks

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

Note: 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.
- 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.

☞ **Important:** You *must* use cables with mono 1/4" phone plugs for both the Input and Output Jacks, in order for the DP/4 to function properly.

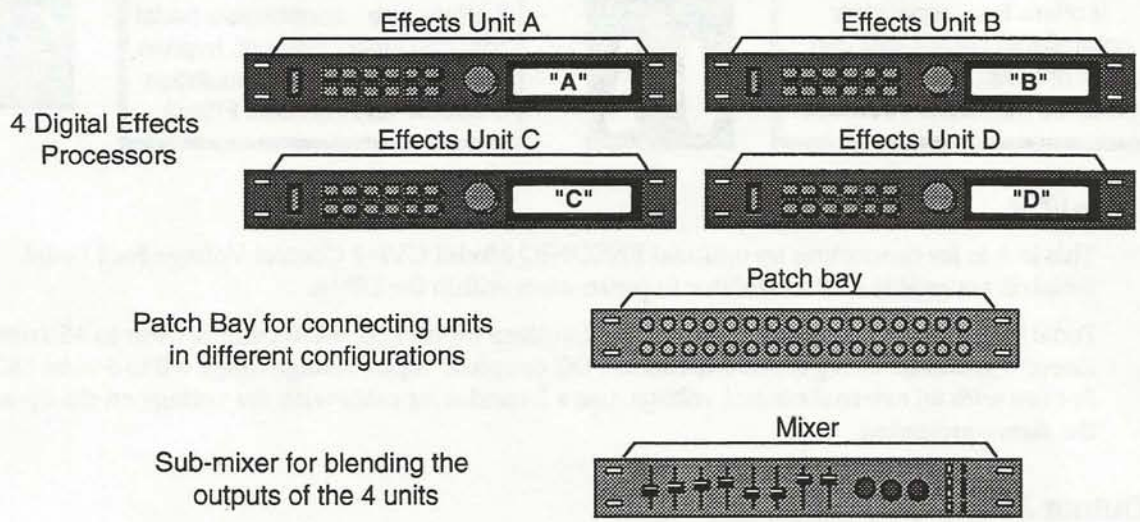
Understanding the DP/4

This part of the manual provides an overview of the theory and operation of the DP/4. We'll start by discussing the make-up of the DP/4 and its component parts; then we'll define some terms you should familiarize yourself with; finally we'll see how they all fit together into a coherent whole.

Let's begin by getting an idea of the component parts that make up the DP/4.

Picture a rack of equipment containing the following:

- 4 state-of-the-art effects processors (we'll call them A, B, C, and D),
- a patch bay to route the input signals to the four units, and to connect them together in almost any *configuration*, and
- a mixer for combining and adjusting the output levels of the four units.



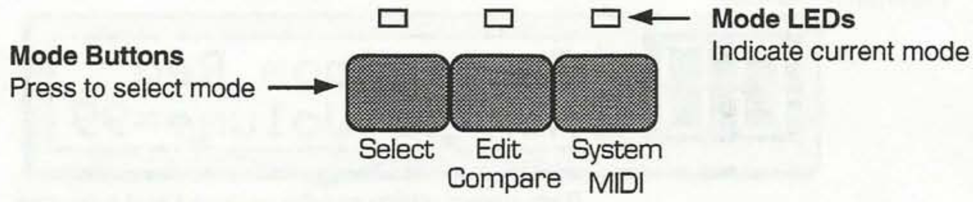
Now suppose you could change the effect in all four units, repatch the patch bay into a whole new setup, and adjust the levels on your sub-mixer, all with the press of a button (or with a MIDI program change, or a press of the foot switch). That's the DP/4.

Of course, the DP/4 does have certain advantages over the rig shown above. For example, all of this is integrated into a single box. And the patching between effects units and the mixing of their output signals is done *digitally*, without the cumulative noise and distortion that goes along with amplifying and mixing analog signals.

Also, unlike the four separate units above, the four signal processors in the DP/4 can be "ganged," two or four at a time, to produce effects which require more processing power than a single unit can handle. The 2 Unit Pitch Shifter is an example of such an effect.

Modes

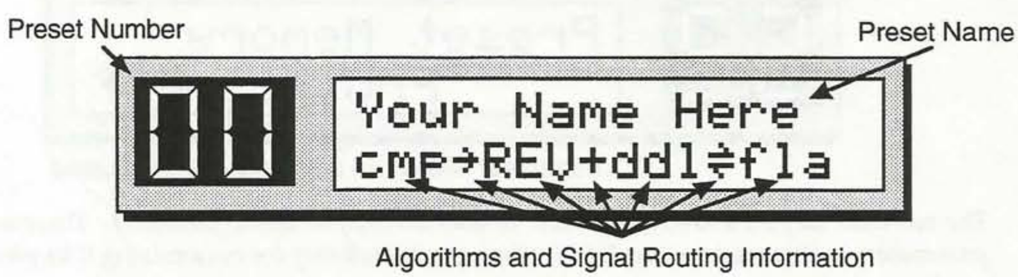
The DP/4 will always be in one of three different modes: Select, Edit, or System•MIDI. You enter one of these modes by pressing the corresponding button on the front panel; the current mode is indicated by the lit LED above one of the mode buttons.



- 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 have been pressed in conjunction with the **Select** button. While in Select mode, presets can also be copied to new locations by pressing the **Write** button.
- In *Edit mode* you can edit the parameters of presets, algorithms and configs. Edit Mode is also the easiest place to change the algorithm in a single unit. Presets that have been edited can be saved in Edit mode (by pressing **Write**).
- In *System•MIDI mode* you can edit MIDI parameters, system (global) parameters. Pressing **Write** will access the MIDI SysEx data transfer function for storage of DP/4 presets and system parameters.

Select Mode

In Select mode, the display shows the selected preset's number, name, unit algorithms, the currently selected unit and the signal routing. The **Data Entry Knob** and the **Left/Right Arrow** buttons select new presets.



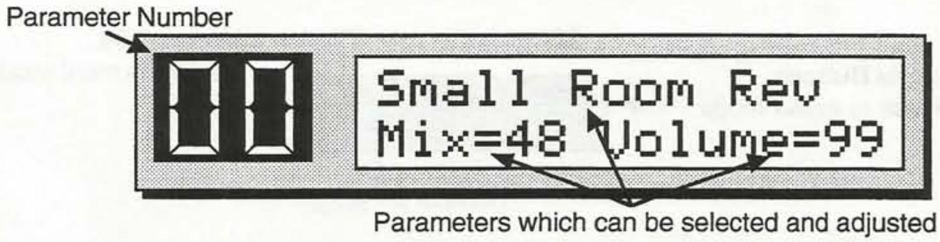
- The red LED display shows the preset's location within the DP/4 memory.
- The top line of the LCD display shows the presets name.
- The bottom line shows which algorithm (effect) is in each unit plus some signal routing information, depending on the current configuration.

In a Config, 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 our preset it's ***Unit B***). If none of the unit abbreviations are capitalized, it means that the Config is activated. Try pressing the different Unit buttons to see the abbreviations change between lower case and upper case. When you press the **Config** button, there are no capitalized letters.

When a unit is selected (capitalized), it means 1) pressing its button again will bypass the unit, and 2) that unit will be selected for editing if you press **Edit**.

Edit Mode

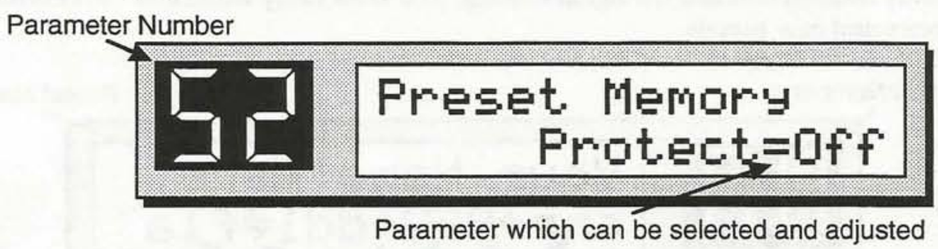
In Edit mode, you edit those parameters having to do with the algorithms in each of the four Units and those having to do with the current Config. After pressing **Edit**, pressing **A**, **B**, **C**, **D**, or **Config** determines what you will be editing. The display shows:



- The red LED display shows the *number* of the currently selected parameter. This will change as you scroll left or right to select different parameters.
- The LCD display shows one or more parameters, which can be selected and adjusted. The currently selected parameter will always be *flashing*.
- The **Left/Right Arrow** 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.

System•MIDI Mode

System•MIDI Mode is like Edit mode, except that it deals only with those parameters which are system-wide, or "global." The System•MIDI parameters are those (such as MIDI channels, Controllers, and program change maps) which do not change as you select different presets and configs. The display shows:

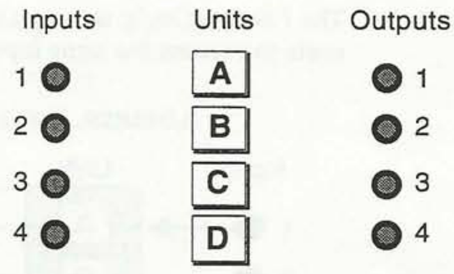


- The red LED display shows the *number* of the currently selected parameter. There are 63 parameters in this mode — the DP/4 offers great flexibility for customizing it to your exact needs.
- The LCD display shows one or more parameters, which can be selected and adjusted. The currently selected parameter will always be *flashing*.
- The **Left/Right Arrow** 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 63 System•MIDI parameters using the **Left/Right Arrow** buttons, this might get tedious; you can use the following shortcuts to get "into the neighborhood" of the parameter(s) you desire:

- After pressing **System•MIDI**, press **A**, **B**, **C**, **D**, or **Config** to go to system 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.

Inputs, Units, and Outputs



The DP/4 has four Audio Inputs, four Effects Units, and four Audio Outputs. To assure maximum flexibility, these can be configured in a variety of ways. The configuration you choose will depend on your current application.

For example, a guitar player might simply want to plug into Input 1, use all four units to simultaneously process his guitar sound, and get a stereo output from Outputs 1 and 2.

An engineer mixing down a tape, on the other hand, might want to route the four inputs independently to the four effects units, and get a stereo output from Outputs 1 and 2. Both of these scenarios, and many other variations, are easily accomplished with the DP/4.

Units, Sources, and Configs

As we mentioned earlier, the DP/4 contains four independent effects processors. We refer to each of these as a *Unit*. The Units are called A, B, C, and D. You will see four buttons on the DP/4 front panel labelled A, B, C, and D which are used for selecting and bypassing the respective Units.

When we start to think about connecting inputs to these units, we think in terms of *Sources*. To be specific, *how many sources* are we going to feed into the DP/4. The DP/4 can be configured to process 1, 2, 3, or 4 Sources simultaneously.

☞ **Important:** It's important not to confuse *sources* with *inputs*. A source can be mono or stereo, depending on the set-up. Two audio inputs, connected to Inputs 1 and 2, for example, might be used as a single stereo source; or they might be used as two independent mono sources. It all depends on the configuration.

The way in which the inputs and outputs are connected to the units is determined by the current *Config* (short for configuration). The Config controls all the connections in the digital "patch bay" we discussed earlier. By changing the Config, you can "repatch" the DP/4 into whatever configuration your current needs require.

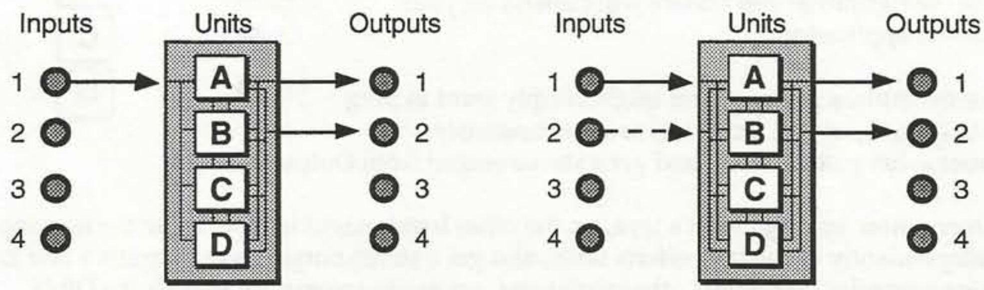
What's more, you can save all the patching information, as well as the effects algorithms loaded into all four units and all their parameter settings, in a special type of preset called a Config Preset, making the process of reconfiguring your DP/4 as easy as selecting a single preset.

What follows is a brief introduction to the various DP/4 Configs and some of their possibilities for input and output signal routing. A more complete discussion of all the Config parameters follows later in this manual.

1 Source Configs

The *1 Source Config* arranges the DP/4 as one *very* powerful multi-effects processor, using all four units to process the same input signal. Two possible 1 Source Configs are shown below:

1 Source, Mono In **1 Source, Stereo In**



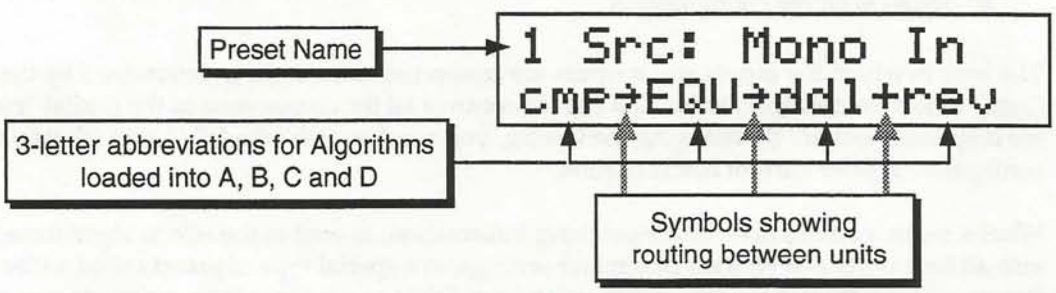
In the first example, the signal present at Input 1 will be fed as a mono signal to all four units. In the second, the signals present at Inputs 1 and 2 will be fed to all four units as a stereo signal. (Remember, a single *source* can be mono or stereo.)

How the signal is distributed among the four units (the internal "patching" between A, B, C, and D) is determined by the setting of the various Config parameters. See "Signal Routing Between Units" later in this discussion for a look at some of the possibilities.

The 1 Source Config is useful for:

- processing a single instrument, such as a guitar or keyboard;
- creating a chain of high quality effects, such as when processing a vocal or other critical sound.

In Select Mode, when a 1 Source Config Preset is selected, the display shows:

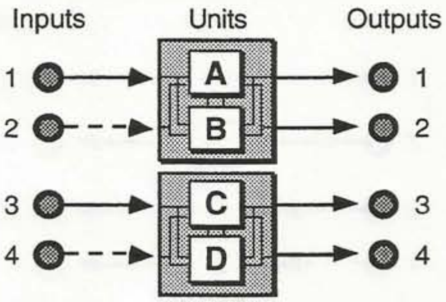


The abbreviations on the lower line tell you what kind of effect is loaded into each of the four units (A, B, C, and D). The symbols between them tell you how those units are interconnected. The meaning of these symbols is discussed later in this section.

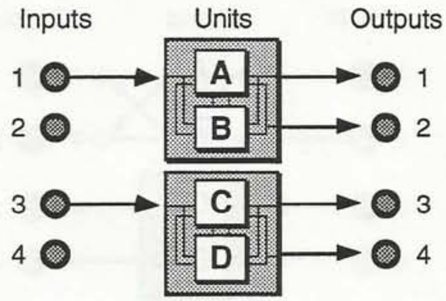
2 Source Configs

The 2 *Source Config* divides the DP/4 into 2 multi-effects processors, each containing 2 units of processing power. One signal is fed to A & B; the other is fed to C & D. The two pairs function as entirely independent, 2 Unit devices.

2 Source, Stereo In



2 Source, Mono In



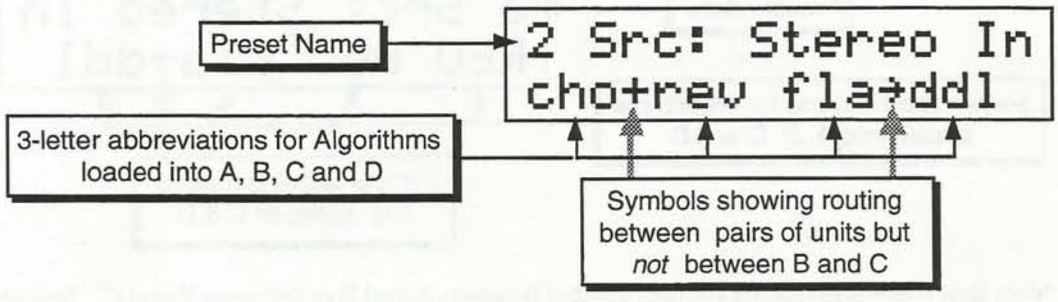
In the first example, the signal present at Inputs 1 and 2 will be fed as a stereo signal to Units A and B. The signal present at Inputs 3 and 4 will be fed as a stereo signal to Units C and D. In the second example, the signal present at Input 1 will be fed as a mono signal to Units A and B. The signal present at Input 3 will be fed as a mono signal to Units C and D.

In both examples, the stereo output of the AB pair goes to Outputs 1 and 2; the stereo output of the CD pair goes to Outputs 3 and 4. (If nothing is plugged into Output jacks 3 and 4, that signal would be mixed into Outputs 1 and 2.) Signal routing between A & B, and between C & D, is determined by the settings of the various Config parameters.

The 2 Source Config is useful for:

- Processing separately the outputs of two keyboards, or a guitar and a keyboard;
- Studio applications where you want two separate multi-effects (such as Chorus & Reverb or Flange & Delay) processing two different signals at the same time;
- Mixed applications, such as processing a guitar plugged into Input 1 with Units A & B, while simultaneously processing the Aux Send of a mixer connected to Input 3 (&4 if stereo) with Units C & D.

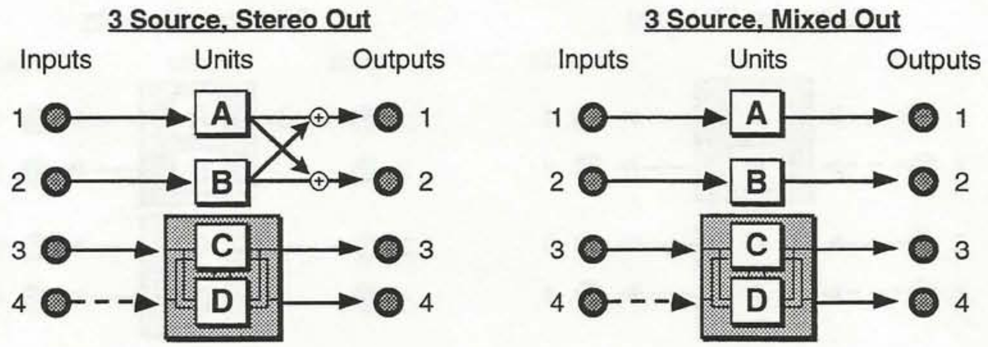
In Select Mode, when a 2 Source Config Preset is selected, the display shows:



Note that there is no signal routing symbol between B and C on the lower line of the display. This shows that there is no connection between Units A & B and Units C & D.

3 Source Configs

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



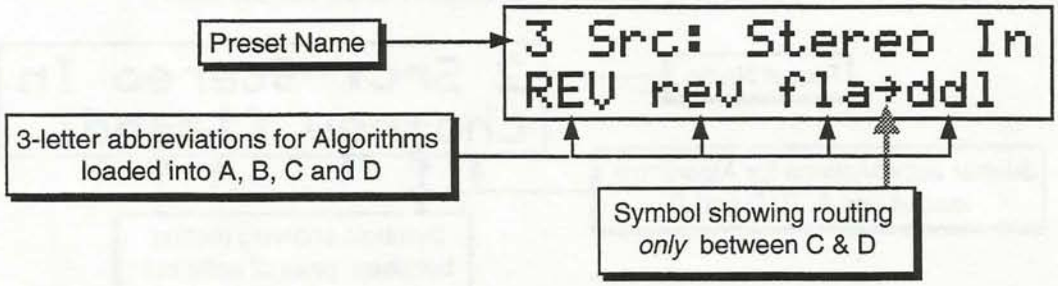
In both examples, the signal at Input 1 goes to Unit A; the signal at Input 2 goes to Unit B. The signal at Inputs 3 and 4 is fed as a stereo signal to Units C & D. In the first example, the stereo outputs of units A and B are digitally mixed together and sent to Outputs 1 and 2. The stereo output of the CD pair goes to Outputs 3 and 4.

In the second example, the output of unit A is summed to mono and sent directly to Output 1; the output of unit B is summed to mono and sent directly to Output 2. As before, the stereo output of the CD pair goes to Outputs 3 and 4. The difference between these examples is the setting of the "AB Output Select" parameter, one of the Config parameters available in a 3 Source Config.

The 3 Source Config is useful for:

- Studio applications where you want two separate single effects (eg. two different reverbs, for drums and vocals) and one multi-effect at the same time;
- Mixed applications, such as processing a guitar plugged into Input 1 with Unit A, while processing one Aux Send of a mixer connected to Input 2 with Unit B, and another Aux Send connected to Input 3 with Units C & D.

In Select Mode, when a 3 Source Config Preset is selected, the display shows:

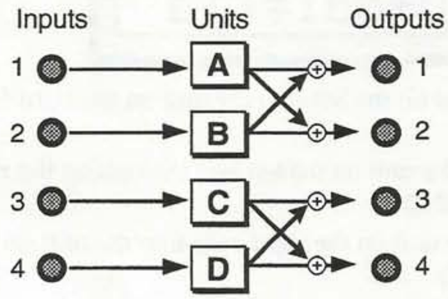


Note that there is no signal routing symbol between A and B or between B and C. This shows that there is no connection between the A, B, and CD Units.

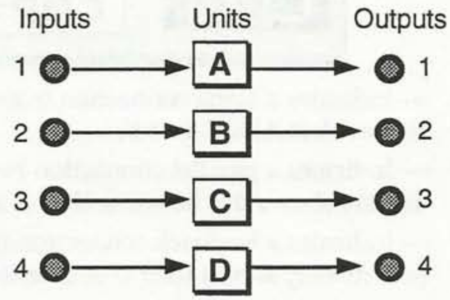
4 Source Configs

The *4 Source Config* is one of the most useful set-ups in the DP/4. In a 4 Source Config, each Unit (A, B, C, and D) functions as an independent, 1 unit, effects processor. The Inputs to the four Units are always mono. The Outputs of the Units can be mixed stereo, or independent mono outs.

4 Source, Stereo Out



4 Source, Mono Out



Input 1 is a mono input to Unit A; Input 2 is a mono input to Unit B; Input 3 is a mono input to Unit C; and Input 4 is a mono input to Unit D.

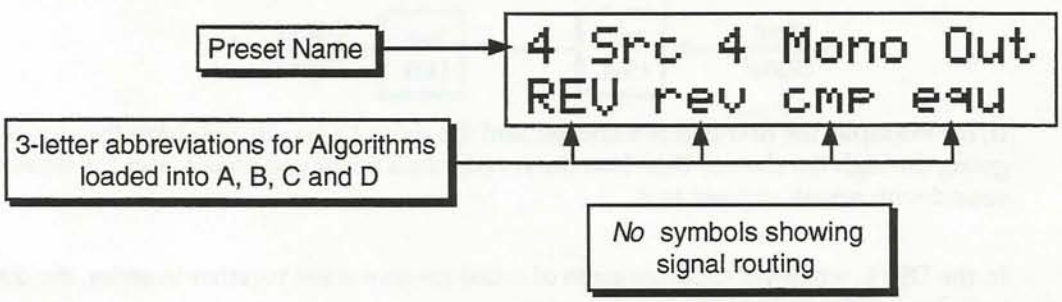
In the first example, the stereo outputs of units A and B are digitally mixed together and sent to Outputs 1 and 2; and the stereo outputs of units C and D are digitally mixed together and sent to Outputs 3 and 4.

In the second example, the output of each unit is summed to mono and sent directly to its respective output: A to 1, B to 2, C to 3, and D to 4. The difference between these examples is in the settings of the "AB Output Select" and "CD Output Select" parameters, Config parameters available in a 4 Source Config (see the section on Config Parameters for more details).

The 4 Source Config is most useful for:

- Studio applications where you want four completely independent single effects at the same time.

In Select Mode, when a 4 Source Config Preset is selected, the display shows:



Note that there are no signal routing symbol between A, B, C, or D. This shows that all four units are functioning independently, not connected in any way.

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 (our display shows ***Unit A*** into ***Unit B***).

\+ — Indicates a parallel connection between the unit on the left and the unit on the right (our display shows this between ***Unit B*** and ***Unit C***).

≑ — Indicates a feedback connection from the unit on the right back into the unit on the left (our display shows ***Unit D*** back into ***Unit C***).

* — 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.

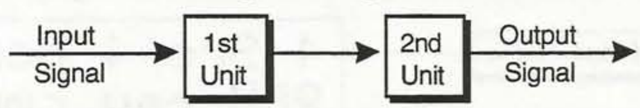
(blank space) — Indicates that there is no connection between units, and that they are routed separately (the blank space will not appear in a 1 Source Config, because all four units are always connected together).

Understanding Serial and Parallel Signal Routing

When we speak of connecting units together using the DP/4's internal digital "patch bay," we are usually referring to one of two types of signal routing, *serial* or *parallel*. To get the most out of the DP/4, it is *very* important to understand the difference between these two concepts.

- **Serial routing** means the input signal is routed *through* the first unit *before* being sent to the input of the second unit.

This is *serial* signal routing between two units:

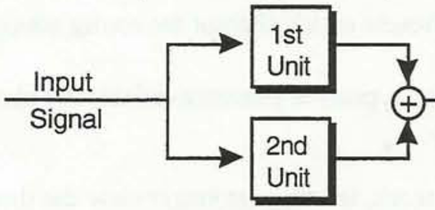


If, for example, the first unit is a chorus, and the second a reverb, you have the signal first going through the chorus, then into the reverb. As a result you would hear the chorused sound with reverb applied to it.

In the DP/4, wherever units (or pairs of units) are connected together in series, the display will show a → symbol between the units.

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

This is *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.

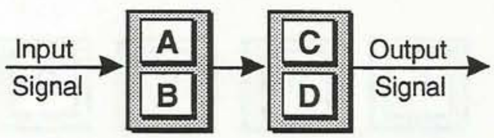
Wherever units (or pairs of units) are connected together in parallel, the display will show a **+** symbol between the units.

Note: *Feedback* routing (shown by a ≑ symbol) is similar to serial routing, but there is a feedback loop returning the output of the second unit to the input of the first. Feedback routing is covered in detail in the Config Parameters section later in this manual.

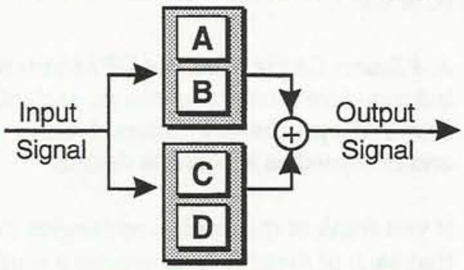
AB to CD Signal Routing

In a *1 Source* configuration (where all four units are used to process a single input signal) the DP/4 gives you the choice of serial or parallel routing between the pairs of units (AB and CD).

Serial routing between AB and CD



Parallel routing between AB and CD



Any combination of serial and parallel routing can be chosen, both *within* the pairs and *between* the pairs of units. This allows a huge variety of different configurations, each of which will sound distinctly different.

Selecting 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:

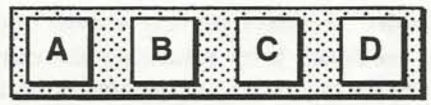
- 100 of the presets are Config Presets which change the config set-up as well as loading new effects algorithms; and
- of the remaining 300 presets, the type(s) of presets available for viewing and selecting depends on the current config.

If you're not clear on how configs work, we suggest you review the discussion earlier in this section.

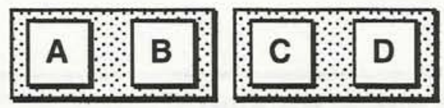
How the Config Affects Selecting Presets

As we discussed earlier, the four different config types effectively turn the DP/4 into 1, 2, 3, or 4 independent effects devices, with a varying number of DP/4 Units per "device":

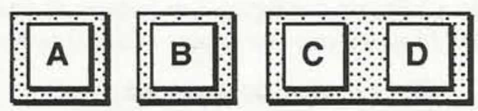
A *1 Source Config* turns the DP/4 into one giant multi-effects processor with 4 effects units all processing the same input signal.



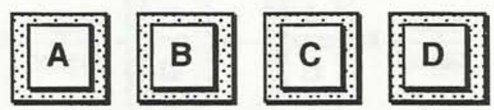
A *2 Source Config* turns the DP/4 into two multi-effects processors, with 2 effects units applied to each input signal.



A *3 Source Config* turns the DP/4 into three effects processors; two with 1 effects unit apiece (A and B), and one with two effects units (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.



If you think of the shaded rectangles in the above illustration as different "devices," you'll notice that each of these configs creates a number of "devices" that are either 1, 2, or 4 Units "wide."

This, in fact 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.

☞ **Important:** There are two ways that the DP/4 tells you which type of Preset you are selecting:

- 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.
- Also, 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:

`Dark Hall` `A:Hall Reverb`	`Vocal Chain 32` `A:CMP → B:rev`	`CleanFlangGuitar` `CMP→fla→ddl+rev`
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.	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).	In a 4-Unit preset, the lower line of the display shows 3-letter abbreviations for the algorithm in all 4 units. By definition a 4-Unit preset will load new algorithms into all four units.

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 **Data Entry Knob**. 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: In its default state, the DP/4 only allows you to select among twelve Config presets (ROM locations 50-61). These have been programmed to give you easy access to the most commonly used configs.

When you want to select among more Config presets, or to save your own, you can "reveal" the remaining 88 Config presets by turning the "Show 100 Config Presets" switch to Yes. This switch (parameter 59 in System•MIDI mode) unveils functions which have been hidden to avoid confusion for the first-time user.

Selecting 4 Unit Presets in a 1 Source Config

First, let's select a 1 Source Config:

1. Press **Select**, then press **Config**.
2. Use the **Data Entry Knob** to choose Config Preset 54:

3. Press **Select** again. This loads the Config Preset, changing the current Config type to 1 Source, and loading all 4 Units with the Dry (No Effect) algorithm. The DP/4 automatically sends you to Unit A.

   Now, press any of the other Unit buttons, **B**, **C**, or **D**. Notice that all four Unit LEDs stay lit no matter which is selected (capitalized). This is because in a 1 Source Config, all four Units function together, processing the same input signal.

   Remember: In a 1 Source Config, only *4 Unit Presets* can be selected. In Select Mode, you can't view or select 1 or 2 Unit presets without changing to a different Config. However, it is possible to change 1 Unit presets in Edit mode (for more about loading a 2 Unit preset while in a 1 Source Config, see *Section 8 — Applications*).

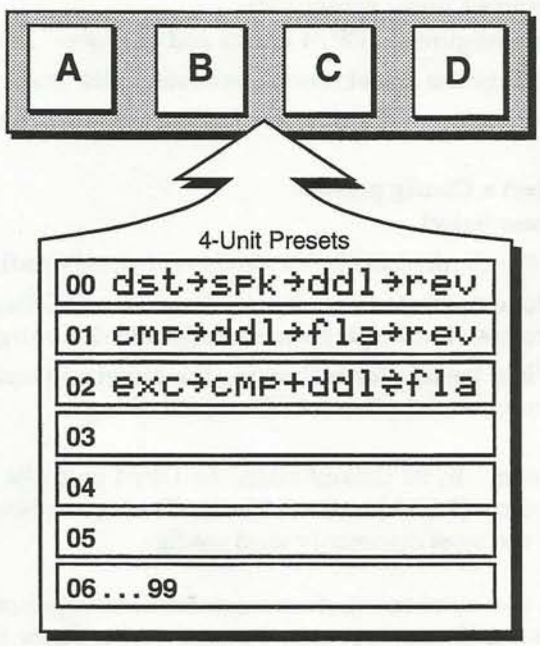
4. Rotate the **Data Entry Knob**. Now you can select among the 100 4-Unit presets in the DP/4 Memory. When you have a preset showing that you want to hear, simply press **Select** to load the preset.

In a 1 Source Config
Units A, B, C, and D function together as one device.

Only 4 Unit Presets can be selected.

Each 4 Unit Preset will load new algorithms into A, B, C, and D, along with routing info between units.

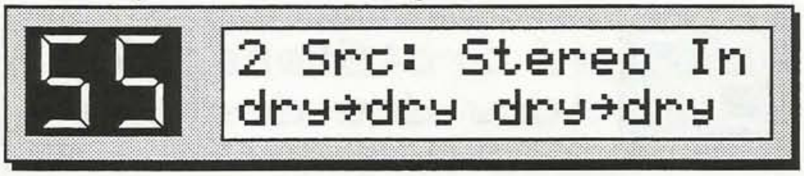
There are 50 ROM and 50 RAM 4 Unit Presets to choose from.



Selecting 2 Unit Presets in a 2 Source Config

Next, we will select a 2 Source Config:

1. Press **Select**, then press **Config**.
2. Use the **Data Entry Knob** to choose Config Preset 55:



3. Press **Select** again. This loads the Config Preset, changing the current Config type to 2 Source, and loading all 4 Units with the Dry (No Effect) algorithm. The DP/4 automatically takes you to Unit A.

   Now, press any of the other Unit buttons, **B**, **C**, or **D**. Notice that the Unit LEDs light only in pairs — A & B light when you press **A** or **B**; C & D light when you press **C** or **D**. This is because in a 2 Source Config, the DP/4 is divided into 2 separate multi-processors, each with 2 Units of processing power.

   Remember! In Select mode, in a 2 Source Config, only ***2 Unit Presets*** can be selected. They can be selected independently for Units A & B or for C & D. The preset will be loaded into whichever pair has its Unit LEDs lit. However, any of the individual units can be changed in Edit mode.

4. Press **A**, and rotate the **Data Entry Knob**. Now you see the 100 2-Unit presets in the DP/4 Memory. Press **Select** to load the preset into Units A & B.
5. Press **C**, and rotate the **Data Entry Knob**. You will see the same 100 2–Unit presets, except now you are selecting them for the pair of C & D. Press **Select** to load the preset into C & D.

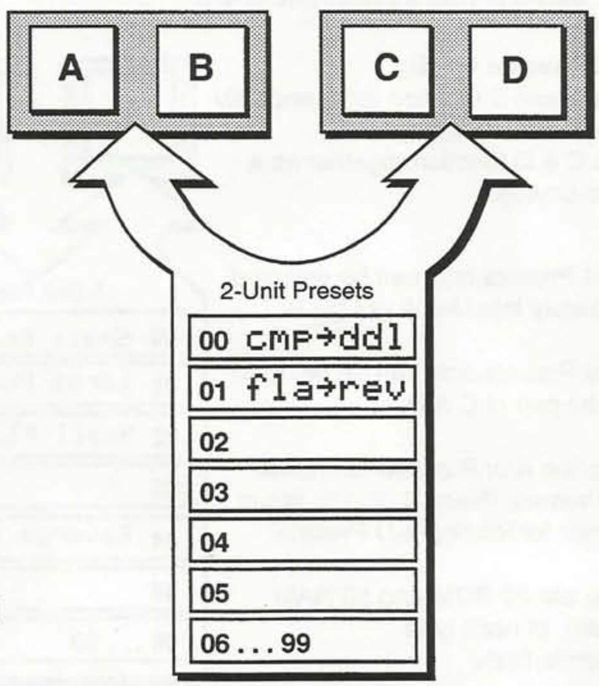
In a 2 Source Config
Units A & B function together as a single device;
Units C & D function together as a single device.

Only 2-Unit Presets can be selected.

Each 2-U Preset will load new algorithms into A & B, or into C & D, depending on which pair is selected.

The same 100 presets are available to load either into A & B or into C & D.

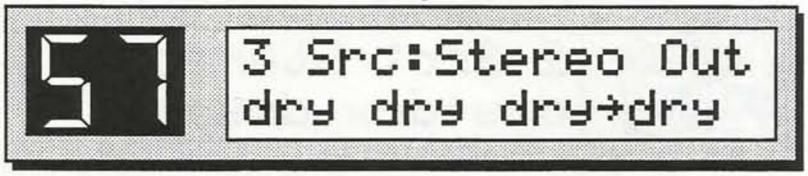
There are 50 ROM and 50 RAM 2-U Presets to choose from.



Selecting 1 and 2 Unit Presets in a 3 Source Config

Next, we will select a 3 Source Config:

1. Press **Select**, then press **Config**.
2. Use the **Data Entry Knob** to choose Config Preset 57:



3. Press **Select** again. This loads the Config Preset, changing the current Config type to 3 Source, and loading all 4 Units with the Dry (No Effect) algorithm. The DP/4 automatically takes you to Unit A.

   Now, press any of the other Unit buttons, **B**, **C**, or **D**. Notice that the A and B LEDs light independently when you press either **A** or **B**; however, C & D light together as a pair when you press **C** or **D**. This is because in a 3 Source Config, the DP/4 is divided into 3 separate processors.

   Remember! In a 3 Source Config, only ***1 Unit Presets*** can be selected in Units A or B; and only ***2 Unit Presets*** can be selected in the paired Units C & D. The preset will be loaded into whichever unit (or pair) has its Unit LEDs lit.

4. Press **A**, and rotate the **Data Entry Knob**. Now you see the 100 1-Unit presets in the DP/4 Memory. Press **Select** to load a preset into A.
5. Press **B**, and rotate the **Data Entry Knob**. You will see the same 100 1-Unit presets. Press **Select** to load a preset into B.
6. Press **C** or **D**, and rotate the **Data Entry Knob**. You will see the 100 2–Unit presets. Press **Select** to load a preset into C & D.

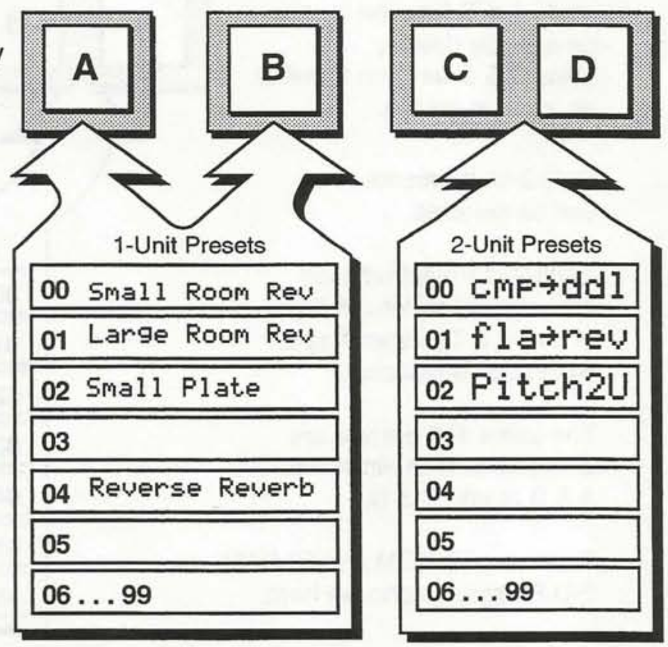
In a 3 Source Config
Units A and B function independently as separate devices;
Units C & D function together as a single device.

1-Unit Presets only can be selected separately into Unit A or Unit B.

2-Unit Presets only can be selected into the pair of C & D.

Press the A or B button to choose 1-U Presets; Press C or D to select that pair for loading 2-U Presets.

There are 50 ROM and 50 RAM Presets of each type to choose from.



Selecting 1 Unit Presets in a 4 Source Config

Now, select a 4 Source Config:

1. Press **Select**, then press **Config**.
2. Use the **Data Entry Knob** to choose Config Preset 59:

3. Press **Select** again. This loads the Config Preset, changing the current Config type to 4 Source, and loading all 4 Units with the Dry (No Effect) algorithm. The DP/4 automatically takes you to Unit A.

   Now, press any of the other Unit buttons, **B, C,** or **D**. Notice that each Unit lights independently, and the display shows its full algorithm name. This is because in a 4 Source Config, the DP/4 is divided into 4 separate 1 Unit devices.

   Remember! In a 4 Source Config, only *1 Unit Presets* can be selected. They can be selected independently into Unit A, B, C, or D. The preset will be loaded into whichever unit has its Unit LED lit when you begin moving the knob.

4. Press **A, B, C,** or **D** and rotate the **Data Entry Knob**. Now you see the 100 1-Unit presets in the DP/4 Memory. Press **Select** to load the preset into the currently selected unit.

Experiment with moving between the four Units and selecting 1 Unit presets for them, thinking of each unit as an independent effects processor.

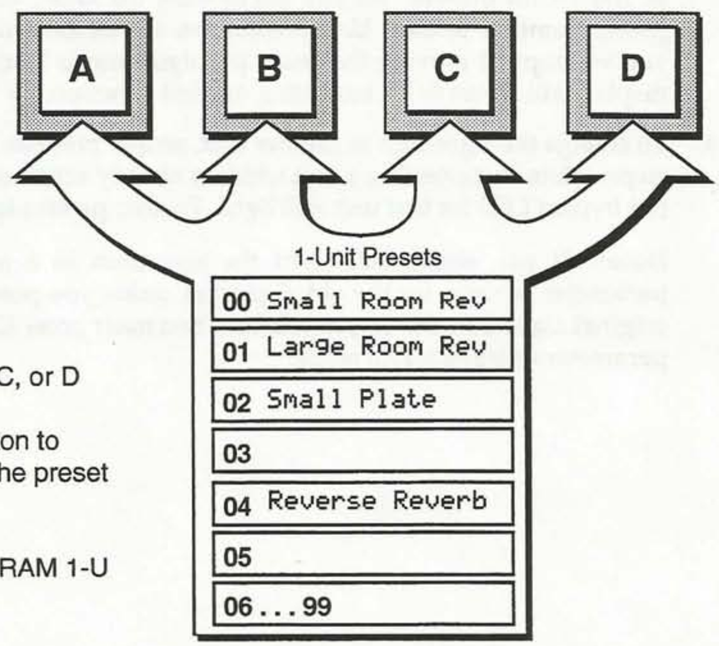
In a 4 Source Config,
Units A, B, C, and D all function independently as separate devices.

Only 1-Unit Presets can be selected.

The same 100 presets are available to load into A, B, C, or D

Press the A, B, C, or D button to choose which Unit to load the preset into.

There are 50 ROM and 50 RAM 1-U Presets to choose from.



Editing DP/4 Parameters

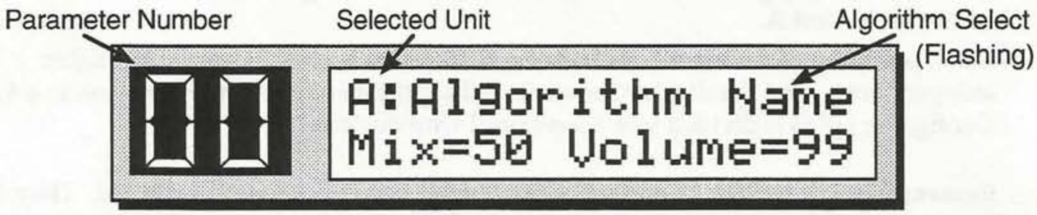
Once you have selected a preset, you can edit (or replace) the algorithm in any of the four units.

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 LED display should indicate parameter 00, which is the *Algorithm Select* parameter, and the algorithm name should be flashing in the upper line of the LCD display. (If not, press the **Left Arrow** button until this is the case.)

3. Move the **Data Entry Knob** to select among the algorithms in memory. When you stop moving the knob, the algorithm that's showing on the display will be loaded into the Unit.

   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 LED display will revert to 00, indicating the first parameter of the algorithm.

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 **Cancel•Undo** to recall the original algorithm and its parameters. You must press **Undo** *before* scrolling to another parameter or leaving Edit mode.

Editing Algorithm Parameters

If you are happy with the algorithm that is in a unit but want to adjust the wet/dry mix, the level of that unit, reverb decay time, or any other parameter within the effect, you can do this in Edit mode.

To modify the parameters of the algorithm in a unit:

1. Press **Edit** (unless you are already in Edit mode).
2. Press Unit **A, B, C,** or **D** to select that unit for editing. The active unit's yellow LED should be lit.
3. Press the **Right Arrow** button once to move to parameter 01, Wet/Dry Mix. The display shows:

The LED display should indicate parameter 01, which is the Wet/Dry Mix, and the mix value should be flashing (if not, press the **Left** or **Right Arrow** button until this is the case).

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

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

   Note: The first three parameters, #00 Algorithm Select, #01 Wet/Dry Mix, and #02 Unit Volume, will be the same for all algorithms. The remaining parameters (those reached by continuing to scroll right past parameter #02) will vary, depending on the algorithm.

7. To perform further edits to the effect, use the **Left** and **Right Arrow** buttons to select parameters and the **Data Entry Knob** to change the value of the active (flashing) parameter. You will find a wide assortment of parameters available for each algorithm, allowing you to customize the DP/4's effects to your exact needs.

Edit Buffer

As soon as you change a parameter's value, you will notice that the Edit LED begins flashing. This means that the 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.

Quick Tips and Shortcuts

Here are a few quick tips to find your way around the DP/4.

Tip: While holding down either the **Left** or **Right Arrow** button, press **Cancel•Undo**. This is a great way to get to the Algorithm Select parameter in Edit mode without having to scroll through lots of parameters.

Tip: While holding down either the **Left** or **Right Arrow** button, press the other arrow button. This is useful when there are several different parameters on the screen at one time, and you want get to the next screen without having to cursor through each individual parameter.

Tip: While holding down either the **Left** or **Right Arrow** 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: Press the **Cancel•Undo** button. This will return the last parameter you edited to its original value, but *only as long as you still have that parameter selected.* If have you scrolled to another parameter, this undo function will no longer work.

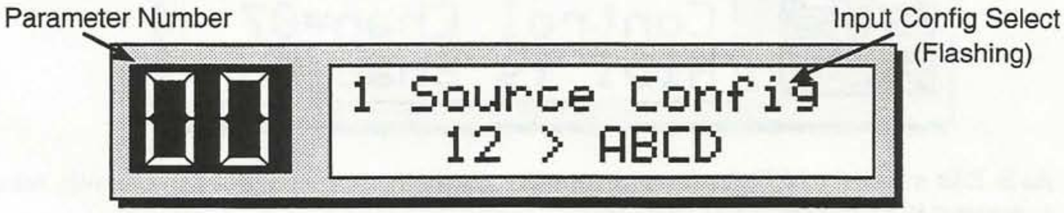
Tip: If you accidentally change the algorithm in a unit, you will lose all of the parameter settings for the old algorithm *unless* you press **Cancel•Undo** to recall the original algorithm and its parameters. You must press **Undo** *before* scrolling to another parameter or leaving Edit mode.

Editing Config Parameters

Editing the parameters of a config is similar to editing the algorithm parameters in a unit — parameter #00 is the *Input Config Select,* and the remaining parameters will vary depending on the config you select. Press the **Left/Right Arrow** buttons to select parameters and move the **Data Entry Knob** to change the value of the active (flashing) parameter.

To edit Config parameters:

1. Press **Edit**.
2. If the yellow Config LED is not already on, press **Config**. The display shows:



   The 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 the **Left Arrow** button until this is the case.)

3. If you move the **Data Entry Knob** now, you will select among the basic config types. When you stop moving the knob, the DP/4 is reconfigured into the config showing on the display.
4. To edit the remaining config parameters, press the **Right Arrow** button to scroll to other parameters and move the **Data Entry Knob** to change their values. You will find a full discussion of all the parameters for each config type later in this manual.

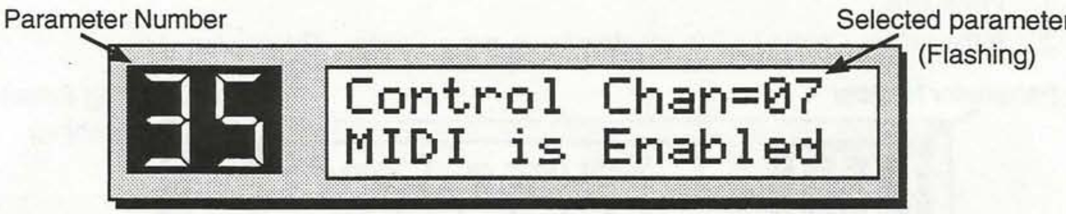
- Note that editing some config parameters may cause a brief interruption in the audio output. This is normal and is required for the system to reconfigure its signal routing.

Editing System•MIDI Parameters

The System•MIDI mode parameters control all the system-wide ("global") functions of the DP/4, including MIDI setup, controller selection, and "user preference" switches that let you set up the DP/4 in a way that best matches your needs.

To edit System parameters:

1. Press **System•MIDI**. The display shows the selected parameter. For example:



As in Edit mode, the LED display indicates the parameter number, and the currently selected parameter is flashing in the LCD display.

2. To edit the System and MIDI parameters, press the arrow buttons to scroll to the one you want to modify, and move the **Data Entry Knob** to change its value.

To make it easier to get through the large number of System parameters, they have been organized into logical groups and the first parameter of each group can be accessed through the following button presses:

Param#	Parameter function	Press:
00-06	Unit A MIDI setup	System, then Unit A button for #00
07-13	Unit B MIDI setup	System, then Unit B button for #07
14-20	Unit C MIDI setup	System, then Unit C button for #14
21-27	Unit D MIDI setup	System, then Unit D button for #21
28-34	Config MIDI setup	System, then Config button for #28
35-36	MIDI chan for controllers	System button repeatedly, until #35 is displayed
37-44	Defining 8 DP/4 controllers	System button again for #37
45-49	Foot switch function & preset chains	System button again for #45
50-51	MIDI Sys-Ex enable and ID number	System button again for #50
52-62	User Preference parameters	System button again for #52
63	Software Version number	System button again for #63

To go to a parameter far up in the list, use the above button presses to get to the parameter nearest the one you want to edit, then scroll to it using the **Left** or **Right Arrow** button.

> **Note:** If you are already in System•MIDI mode (the System•MIDI LED is lit) you don't have to press the **System•MIDI** button again each time to go to the MIDI parameters for Unit A, B, C, etc. Just press **A, B, C, D,** or **Config.**

You will find a full discussion of all of the System•MIDI parameters in the System•MIDI Mode section found later in this manual.

Section 3 — Config Parameters

What is a Config?

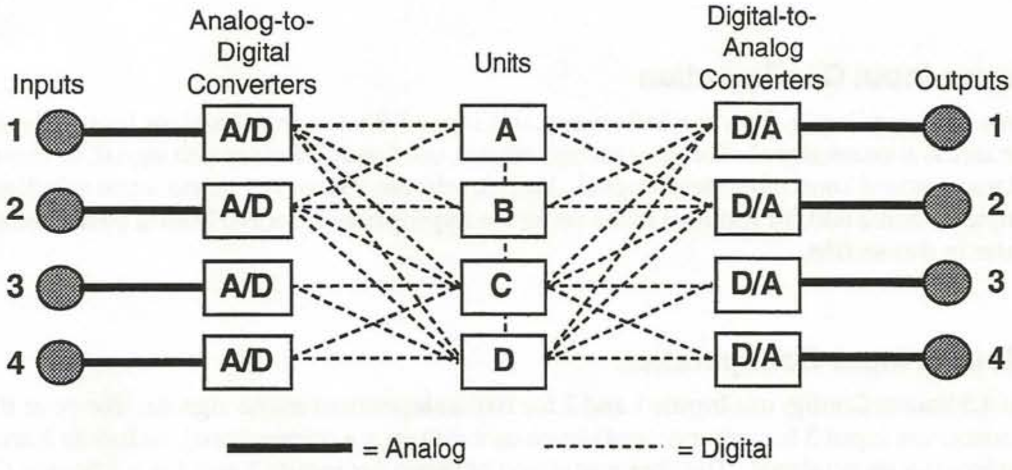
Config (short for CONFIGuration) determines the number of input sources that are to be processed by the DP/4, and how the inputs, units, and outputs are routed to each other. A 1 Source config means that one signal (stereo or mono) is going into the DP/4. Two, three, and four source configs are also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

Digital and Analog Converters

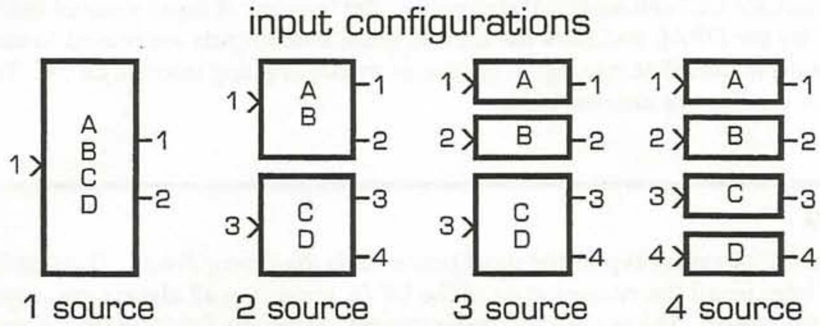
The four audio inputs are analog signals and are fed to four analog-to-digital converters. The four units are audio signal processors which have digital inputs and outputs. Routing between the units is all digital. The output of a unit is converted back to analog audio for the output jack.



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

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, 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, 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, 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 four source config, four separate mono sources are plugged into Inputs 1, 2, 3, and 4.

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 parameters) into each of the four Units.

To select a Config preset:

1. Press **Select.**
2. Press **Config.**
3. Move the **Data Entry Knob.** 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, sends you back to Unit A, and the Select LED stops flashing (solidly lit).

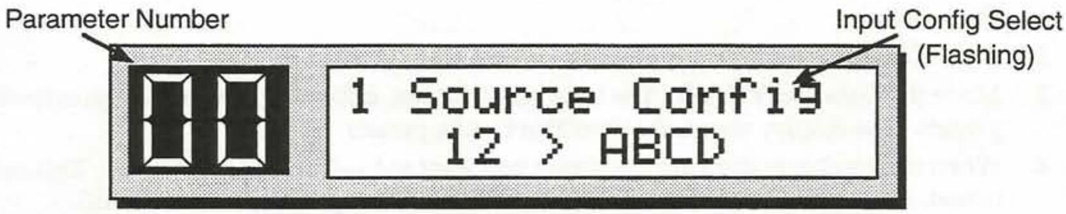
Note: In its default state, the DP/4 only allows you to select among twelve Config presets (ROM locations 50-61). These have been programmed to give you easy access to the most commonly used configs.

When you want to select among more Config presets, or to save your own, you can "reveal" the remaining 88 Config presets by turning the "Show 100 Config Presets" switch to Yes. This switch (parameter 59 in System•MIDI mode) reveals functions which have been hidden to avoid confusion for the first-time user.

Editing a Config Preset

In Edit mode, you can select different input configurations and edit their related parameters (which contain other signal routing information) using the **Left/Right Arrow** buttons and the **Data Entry Knob**. To edit a Config:

1. Press the **Edit** button, then
2. Press **Config.** The yellow Config LED should be lit. The display shows:



The LED Numeric 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 the **Left Arrow** button until this is the case).

3. If you move the **Data Entry Knob** now, you will select among the config types. When you stop moving the knob, the display momentarily shows:

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

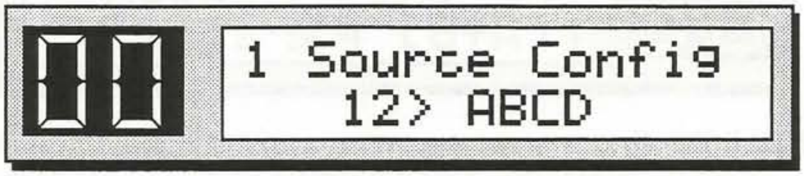
4. To edit the remaining config parameters, press the **Right Arrow** button to scroll to other parameters and move the **Data Entry Knob** to change their values.

There are four different types of configs that can be edited:

- 1 Source Config
- 2 Source Config
- 3 Source Config
- 4 Source Config

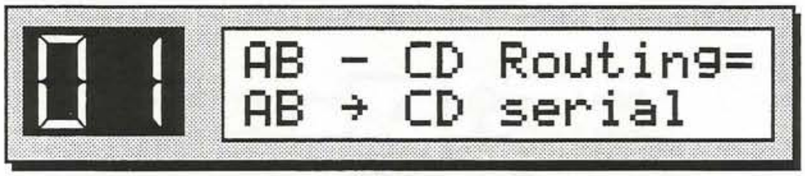
5. Turn the **Data Entry Knob** to select different configs. When you move the knob, the LED display stays at parameter number 00. That's because the Config Select parameter is the first parameter for each source config type.

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 06).

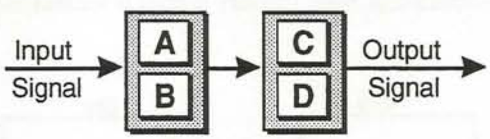


01 — AB - CD Routing

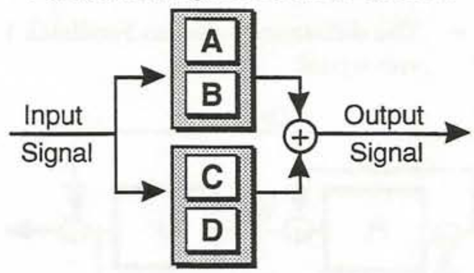
Range: serial or parallel

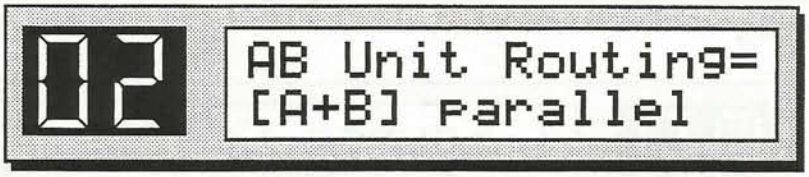
Units A and B can be routed to Units C and D in one of two different ways:

Serial routing between AB and CD



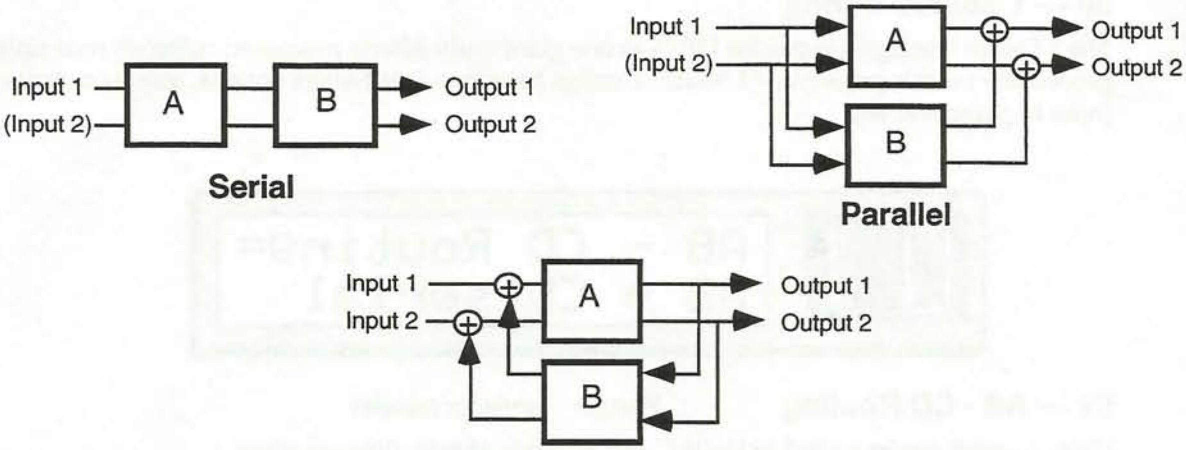
Parallel routing between AB and CD





02 — AB Unit Routing Range: serial, parallel, feedback1 or feedback2

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

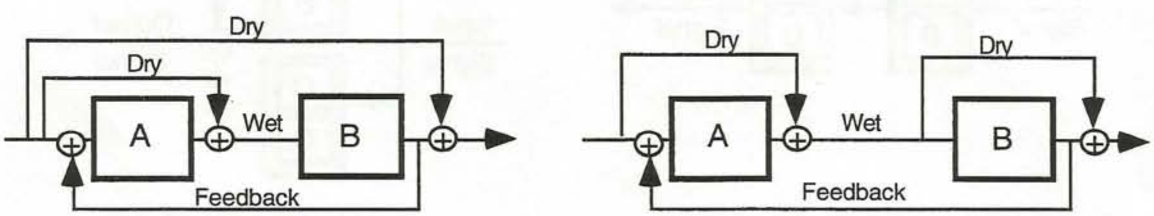


Serial

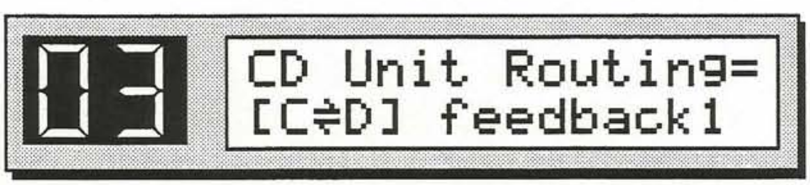
Parallel

Feedback 1 and 2

- The difference between Feedback 1 and Feedback 2 is how the dry signal is mixed into the wet signal:



Feedback 1 **Feedback 2**

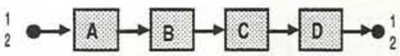


03 — CD Unit Routing Range: serial, parallel, feedback1 or feedback2

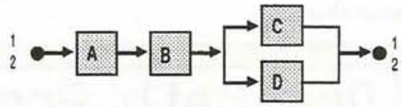
Units C and D can also be routed together in one of four ways. By combining parameters 01, 02, and 03, there are 32 different ABCD routing possibilities.

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

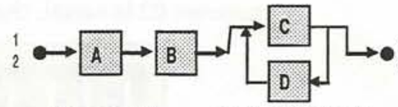
Available ABCD Routings



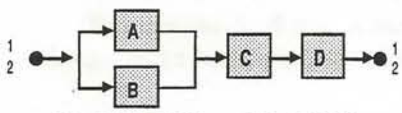
AB SERIAL —> CD SERIAL



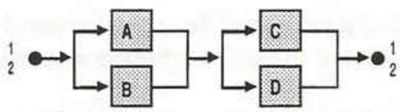
AB SERIAL —> CD PARALLEL



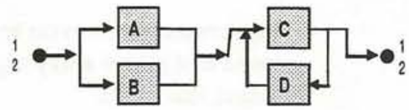
AB SERIAL —> CD FEEDBACK



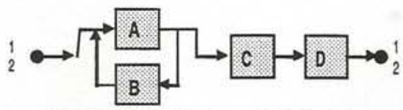
AB PARALLEL —> CD SERIAL



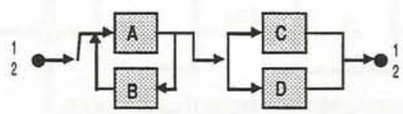
AB PARALLEL —> CD PARALLEL



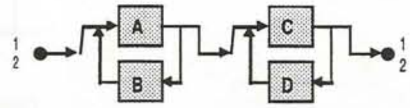
AB PARALLEL —> CD FEEDBACK



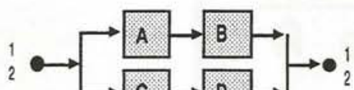
AB FEEDBACK —> CD SERIAL



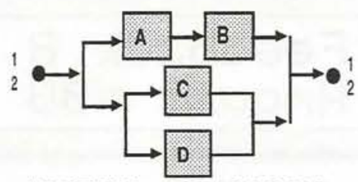
AB FEEDBACK —> CD PARALLEL



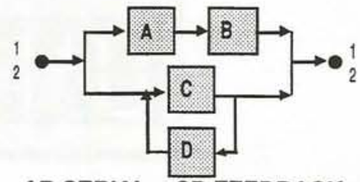
AB FEEDBACK —> CD FEEDBACK



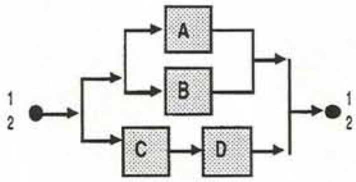
AB SERIAL + CD SERIAL



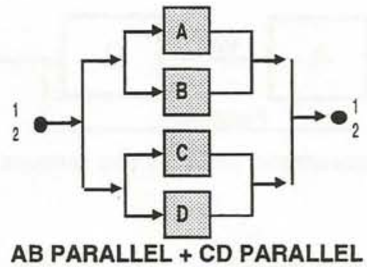
AB SERIAL + CD PARALLEL



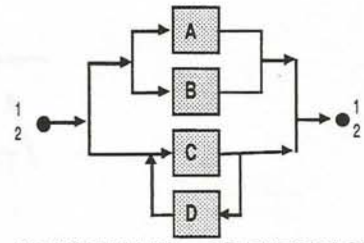
AB SERIAL + CD FEEDBACK



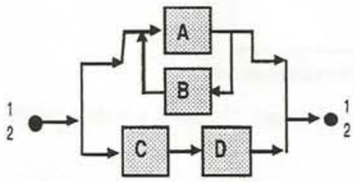
AB PARALLEL + CD SERIAL



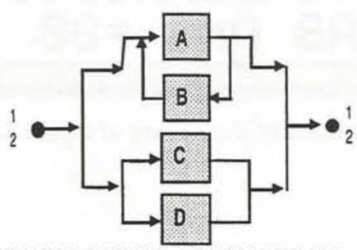
AB PARALLEL + CD PARALLEL



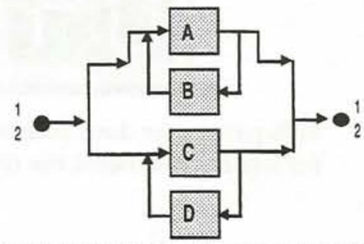
AB PARALLEL + CD FEEDBACK



AB FEEDBACK + CD SERIAL



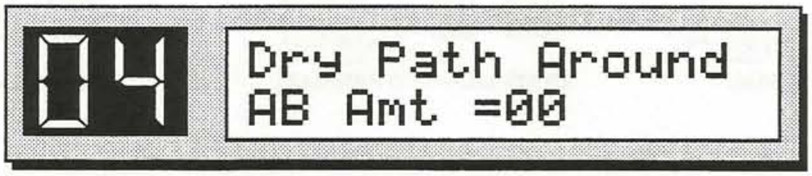
AB FEEDBACK + CD PARALLEL



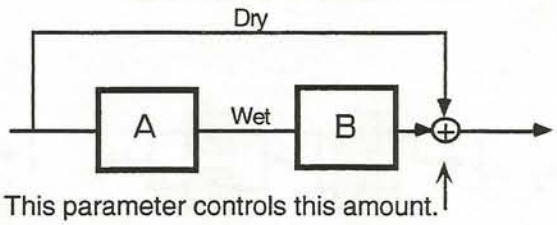
AB FEEDBACK + CD FEEDBACK

04 — (Config Dependent)

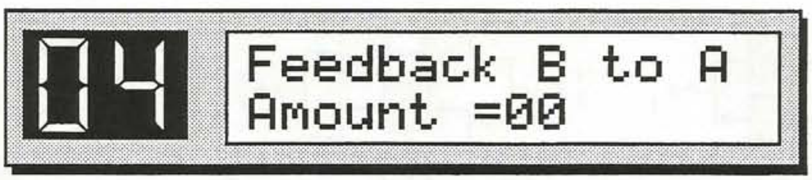
This parameter is dependent on how Units A and B are routed (determined by parameter 02). If parameter 02 is serial, this screen shows:



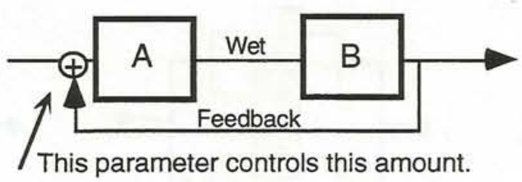
This screen allows you to control an external dry signal around Units A and B. A setting of 00 would not allow a dry signal around the units, whereas a setting of 99 would permit a full signal around the units.



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

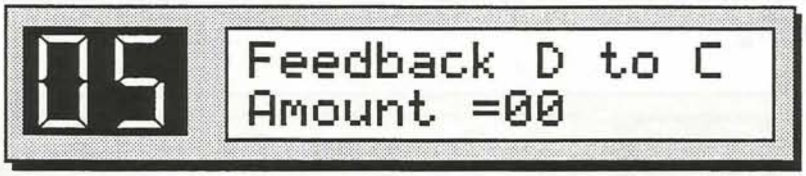


This screen allows you to control a feedback signal from Units B to the front of Unit A. A setting of 00 would not allow the feedback signal back into the units, whereas a setting of 99 would permit a full feedback signal.



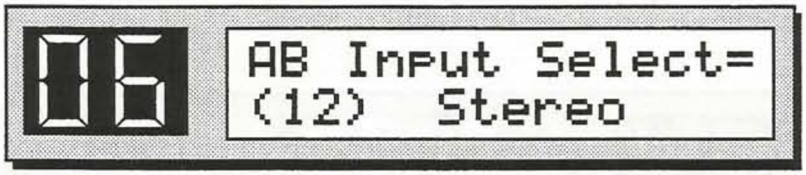
If parameter 02 is parallel, this screen shows:

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



05 — (Config Dependent)

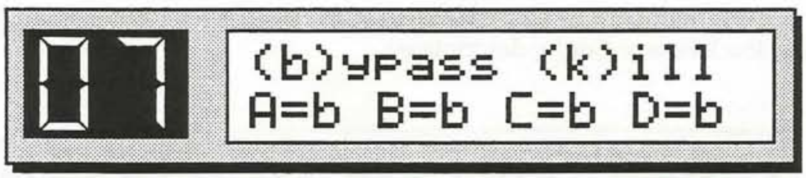
This parameter, identical to parameter 04, is dependent on how Units C and D are routed together (determined by parameter 03).



06 — AB Input Select

Range: (12) Stereo or (1) Mono

This parameter selects either a mono (Input 1) or stereo (Inputs 1 and 2) signal.



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

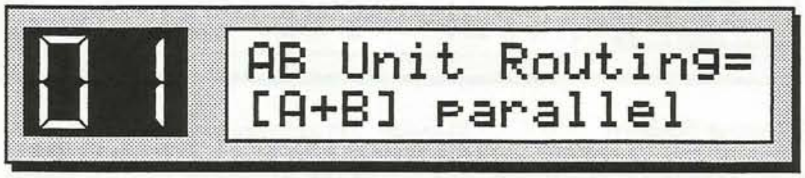
These parameters determine what happens when you bypass a unit (the red LED is lit). When set to bypass (b), only the dry signal passes through the unit. When set to kill (k), no signal passes through the unit.

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.

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 Unit Routing

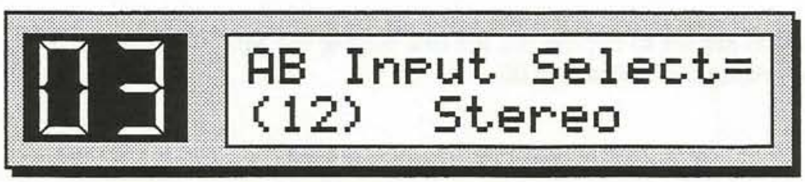
Range: serial, parallel, feedback1, or feedback2

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).

02 — CD Unit Routing

Range: serial, parallel, feedback1, or feedback2

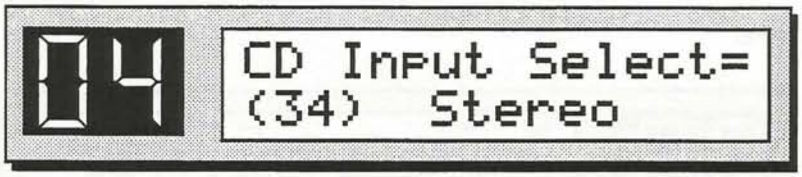
Units A and B can be routed together in either serial, parallel, or two different kinds of feedback (explained in the 1 Source Config description).



03 — AB Input Select

Range: (12) Stereo or (1) Mono

This parameter selects either a mono or stereo input.



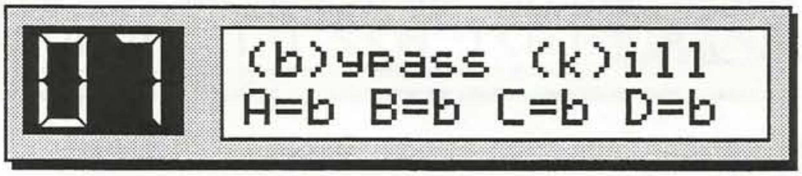
04 — CD Input Select Range: (34) Stereo or (3) Mono

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

05 — AB (Config Dependent)

06 — CD (Config Dependent)

For a complete discussion of config dependent parameters, please refer to 1 Source Config parameters 04 and 05 earlier in this section.



07 — Bypass Kill (Unit) A

08 — Bypass Kill (Unit) B

09 — Bypass Kill (Unit) C

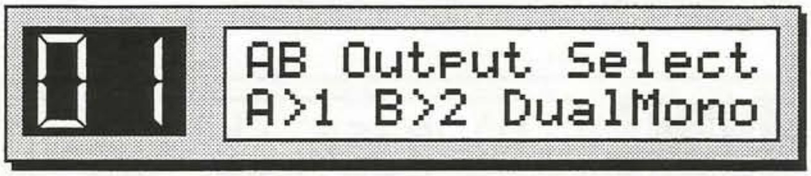
10 — Bypass Kill (Unit) D

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.

3 Source Config

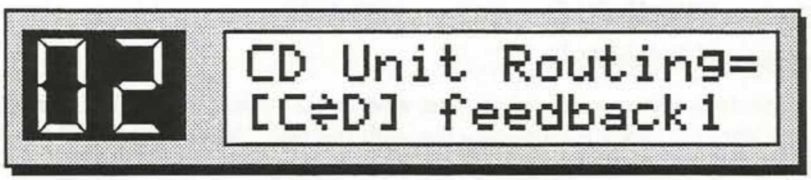
00 — 3 Source Config

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.



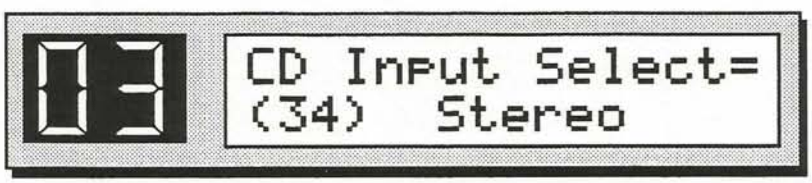
01 — AB Output Select   Range: Dual Mono or Mixed Stereo

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.



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.

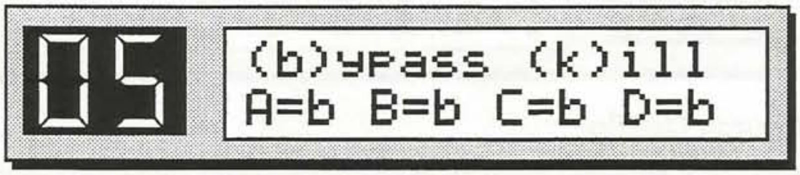


03 — CD Input Select   Range: (34) Stereo or (4) Mono

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

04 — (Config Dependent)

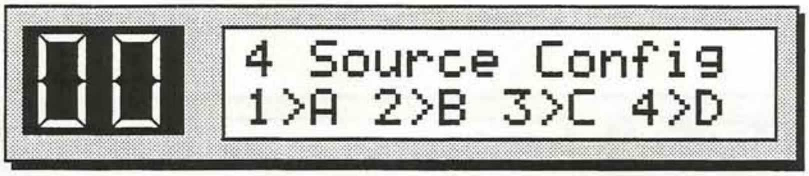
This parameter is dependent on how Units C and D are routed. See the description about config dependent parameters under 1 Source Config (parameter 04) for more information.



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

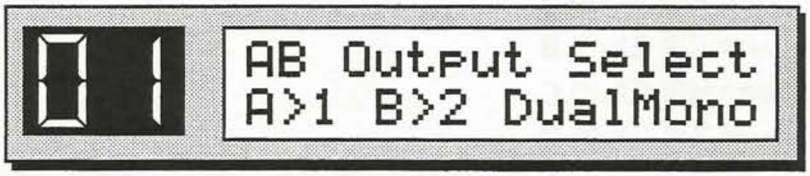
These parameters determine what happens when you bypass a unit (the red LED is lit). When set to bypass (b), only the dry signal passes through the unit. When set to kill (k), no signal passes through the unit.

4 Source Config



00 — 4 Source Config

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

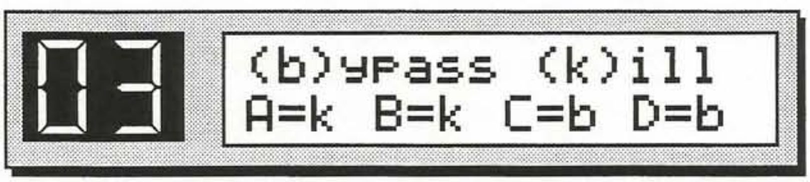


01 — AB Output Select   Range: Dual Mono or Mixed Stereo

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.

Section 4 — Algorithms

Understanding DP/4 Algorithms

The DP/4 is a powerful signal processor which can produce a wide variety of algorithms (effects). The flexible config routing scheme and the extensive parameter control give the DP/4 its dynamic effects capability.

The DP/4 is equipped with an advanced digital signal processing system based on the ENSONIQ Signal Processor (ESP) chip. The ESP 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 very high-quality output signal.

The digital effects processing has been designed to complement any input source, and all of the algorithms can have specific parameters modulated by various MIDI and non-MIDI controllers such as a keyboard's pitch wheel, a CV Pedal, etc.

The algorithms are fully programmable, and may be customized for particular applications. Algorithms are stored as parts of a preset, although each algorithm can be individually selected (in the form of 1 Unit presets). Each of these algorithms are treated a little differently, and will be described individually on the following pages.

About the Parameters

Each algorithm in the DP/4 contains a complete set of parameter values which determine how that algorithm will sound. The algorithm is present even if the signal is not routed through the effect (e.g. when the units are bypassed). Whenever you copy or save an algorithm as a preset, the parameter settings are also saved.

The algorithm parameters are displayed by pressing the **Edit** button then the appropriate **Unit** button, and edited by pressing the **Left/Right Arrow** buttons and turning the **Data Entry Knob**. The parameters that pertain to each algorithm are described later in this section.

Programming Algorithms

The DP/4 algorithms are highly programmable. There are several common parameters for every algorithm as well as algorithm-specific parameters. The first parameter selects which algorithm will be used. When this parameter is changed, a new algorithm is selected which causes several important things to occur.

When a new algorithm is selected:

- a new program is loaded into the ESP chip, causing a brief pause in the audio output,
- the parameter screens are redefined for the particular algorithm selected, and
- the parameters are automatically set to their preset values for the new algorithm.

About Presets

The complete DP/4 setup, including the values of all unit and config parameters, is saved when you save a Config Preset. The DP/4 is smart about switching effects, since all sound must stop for an instant when it changes effects.

When are New Algorithms Loaded into the ESP Chips?

- When you select an algorithm from one of the internal ROM or RAM banks, the algorithm and its parameters will be loaded into the ESP(s), and you will hear that algorithm.
- When you select a 1, 2, 4 or Config preset, the algorithm and its parameters will be loaded into the ESP(s).
- When you bypass the units, the algorithms will *not* be changed.

Whenever a new algorithm is loaded into the ESP(s), the audio output may pause briefly (depending on the config and type of preset being selected), allowing the instructions which create the new algorithm to be loaded into the DP/4. If an algorithm differs only by variation in parameter values, then this pause may not occur.

Mix and Volume Parameters

All of the algorithms in the DP/4 share common Mix and Volume parameters:

01 — Mix Range: 00 to 99

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

02 — Volume Range: 00 to 99

The Volume parameter (always parameter 02) adjusts the output volume of the signal. Setting this parameter to 00 will eliminate the signal, and any algorithms and/or configs that follow will also receive no signal, and will produce no sound.

Algorithm Modulators

All the algorithms allow real-time control of particular parameters and share common modulation control parameters. The exact location of these parameters varies depending on the selected algorithm, but it is always the last eight parameters for all of the algorithms:

Mod1 Source

Mod2 Source

Range: Off/Controller 1 - 8

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

Mod1 Destination Parameter

Mod2 Destination Parameter

Range: Off, 01 to 34 (depending on the algorithm)

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

Mod1 Param Range Min

Mod1 Param Range Max

Mod2 Param Range Min

Mod2 Param Range Max

Range: 00 to 99

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

The eight different Mod Source controllers (two for each unit) are assigned in System•MIDI mode and are explained in more detail in that section.

Algorithm Parameters

Each of the algorithms has a particular set of user programmable parameters associated with the effect. Some of the parameters are common to many effects and some are specific to certain effects.

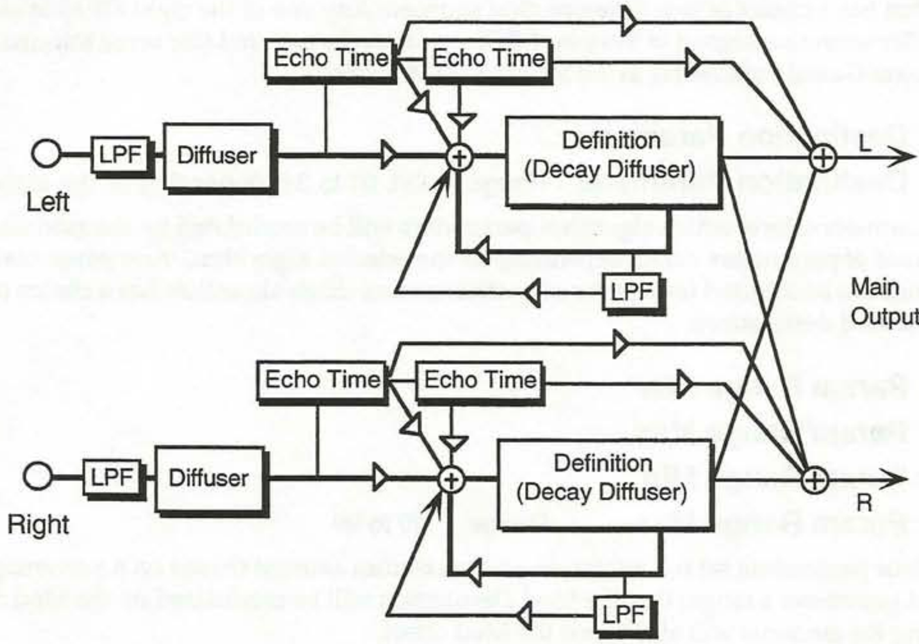
In some cases, several algorithms of the same type will have an identical set of parameters. In these cases, the parameters will only be described once, but the algorithms which share those parameters will be listed at the beginning of the description.

Each algorithm has a mix and a volume control, modulator parameters, plus a set of parameters which is relevant to the algorithm. All of these parameters (except the algorithm name) are programmable, and provide much flexibility for customizing the effects.

SMALL ROOM REV, LARGE ROOM REV, HALL REVERB

The algorithm name is the first parameter for all of the algorithms:
00 — **Small Room Rev** provides ambience.
00 — **Large Room Rev**, larger than Small Room Rev, also provides ambience.
00 — **Hall Reverb** is a large acoustic space, and provides high density reverb.

Small Room Rev, Large Room Rev, and Hall Reverb Signal Routing



These three reverb algorithms share essentially the same signal routing topology. 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).

The parameters available for the small room, large room and hall reverb algorithms are:

01 — Mix
02 — Volume

These parameters are identical for all of the algorithms and are explained in detail under the Mix and Volume Parameters description, found in the beginning of this section. The reverbs sound best with a Mix of wet and dry.

03 — Decay

Small Room Range: 0.20 to 100.0 sec.
Large Room Range: 0.20 to 150.0 sec.
Hall 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. In the room reverbs we don't recommend higher settings, which tend to create an infinite and unnatural sustain. Since most rooms don't naturally have a large decay, set this low for the best sound. However, 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

This parameter acts like a tone control and will boost (positive values) or cut (negative values) the rate at which low frequencies will decay.

06 — HF Damping

Range: 00 to 99

This parameter 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 Room Diffusion1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffuser 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: 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

This parameter 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

This parameter 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

This parameter 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. This parameter should not be set too high, 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/hall. Think of these parameters as three different microphones placing at various distances within the room/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) room/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.

SMALL PLATE, LARGE PLATE

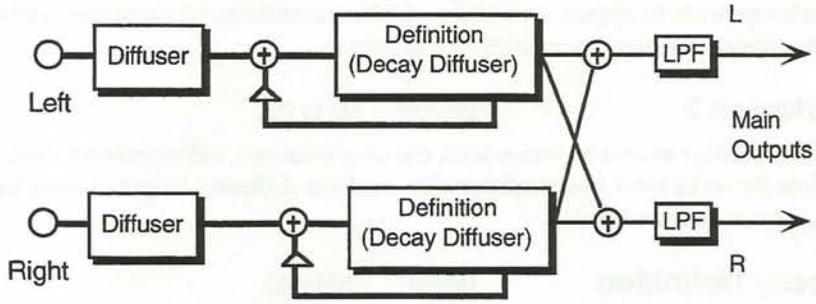
A plate reverb takes the vibrations from a metal plate and uses them to create a metallic sounding reverb.

00 — Small Plate is a tight sounding plate reverb, and

00 — Large Plate simulates a larger plate reverb.

Although they may be named small or large, the parameters still serve the same function in each algorithm. Small plate reverbs are most often used in the studio for drums and percussion, while large plate reverbs are often used to enhance a vocalist's performance.

Small Plate, 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).

The user controllable parameters for the plate reverbs algorithms are:

01 — Mix

02 — Volume

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

03 — Decay

Small Plate Range: 0.20 to 100.0 sec.

Large Plate Range: 0.40 to 140.0 sec.

This parameter 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

Small Plate Range: 0 to 500 ms

Large Plate Range: 0 to 430 ms

This parameter 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

The Diffusion 1 parameter 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. Try to select the highest value that works with your sound source for the best performance.

10 — Early Ref Level 1
11 — Early Ref Level 2
12 — Early Ref Level 3
13 — Early Ref Level 4 Ranges: -99 to +99

These parameters 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 Definition.

14 — Left/Right Balance Range: -99 to +99

This parameter 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.

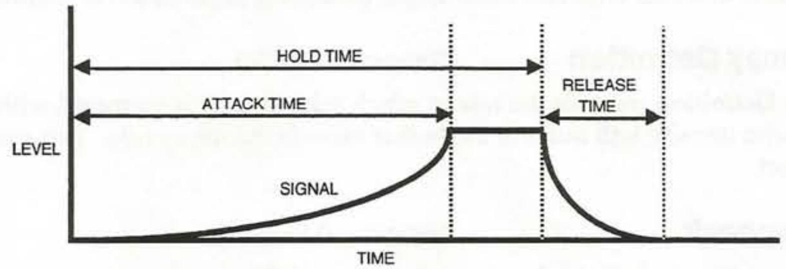
REVERSE REVERB

00 — Reverse Reverb 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. Reverse Reverb is triggered by an input signal level (threshold) determined by the user. Once triggered the reverse envelope will proceed to completion, and will ignore subsequent trigger levels. If you are looking for a reverse effect that will retrigger, try using the Reverse Reverb 2. The topology of the Reverse Reverb is similar to the Plate Reverb.

The parameters available for this algorithm are:

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

This parameter 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

This parameter 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

This parameter 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 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

The Decay Definition 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

This parameter 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 at about the same value as the Envelope Hold Time (parameter 03).

12 — Slapback Level Range: 00 to 99

This parameter 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

00 — 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.

The parameters available for this algorithm are:

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

This parameter 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

This parameter is used to capture transients which occur before the trigger. This parameter is critical to the sound quality. The user determines 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

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.

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

The Decay Definition 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.

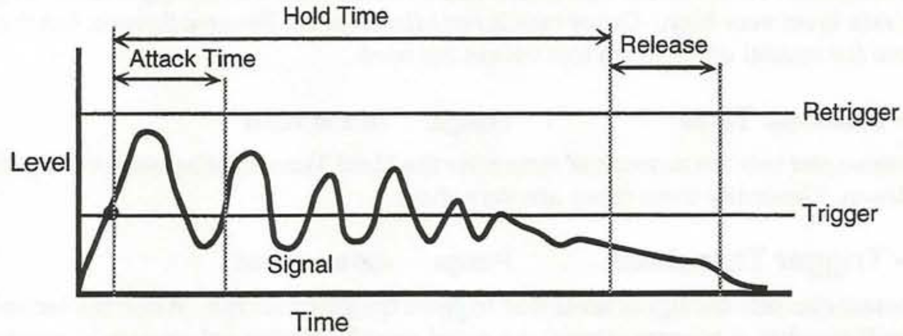
12 — Mod1 Source
13 — Mod1 Destination Parameter
14 — Mod1 Param Range Min
15 — Mod1 Param Range Max
16 — Mod2 Source
17 — Mod2 Destination Parameter
18 — Mod2 Param Range Min
19 — Mod2 Param Range Max

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

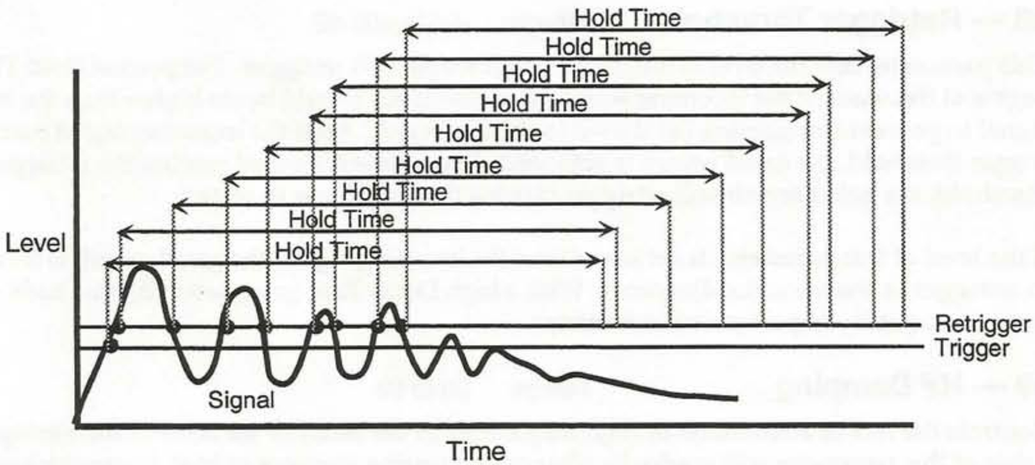
GATED REVERB

00 — Gated Reverb provides a superb gated reverb. When a reverb is muted partway through, 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 input signal. The gate is first opened when the input signal passes the trigger threshold. This trigger threshold is set as low as possible by the user, 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 there are 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



The parameters available for the Gated Reverb algorithm are:

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

This parameter 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.

This parameter 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 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

This parameter 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

This parameter 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 Gated 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. 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: 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

This parameter 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

This parameter 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 Parameter
22 — Mod1 Param Range Min
23 — Mod1 Param Range Max
24 — Mod2 Source
25 — Mod2 Destination Parameter
26 — Mod2 Param Range Min
27 — Mod2 Param Range Max

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

NON LIN REVERB 1, 2, 3

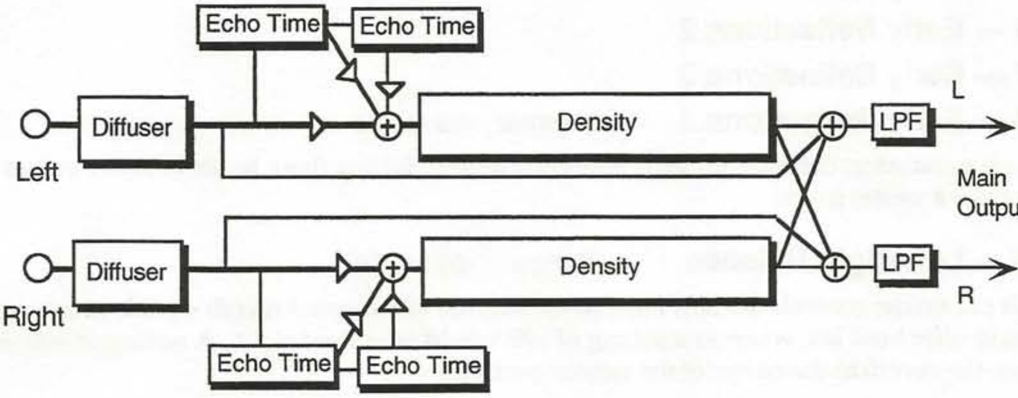
Non Lin can be used to obtain blooming reverb, gated reverb, reverse reverb and early reflections. Non linear reverbs in general do not produce an exponentially decaying reverb. Unlike the Hall, Room and Plate reverbs, Non Lin 1, 2, and 3 passes the input signal through the reverb diffusers only once. 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.

00 — Non Lin 1 is optimized for shorter duration effects (approx. 0.5 sec.).

00 — Non Lin 2 offers approx. a 1.5 sec. duration.

00 — 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



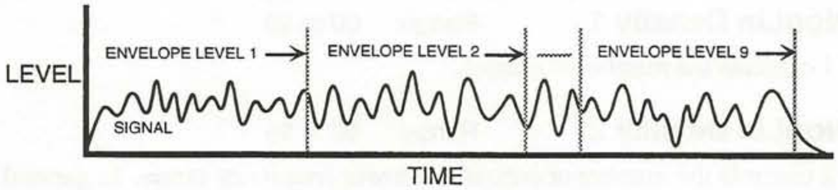
The signal goes directly through a diffuser which smears the signal. The signal is then routed to a decay diffuser, known as Density, and is diffused over a period of time. Within the density the signal goes through a high frequency damper. The signal is then routed to the output. After the density, 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).

The parameters available for the Non Lin Reverbs are:

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 two 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

This parameter 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

This 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

This parameter 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.

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.

MULTI TAP DELAY

00 — **MultiTap Delay** requires only one ESP chip, allowing the other three units to be free for other algorithms. MultiTap Delay produces four independent controllable delays.

The parameters available for this algorithm are:

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

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.

3.3 SEC DDL 2U

00 — 3.3 sec DDL 2U uses two ESP chips to provide a high fidelity delay longer than 3 seconds. This algorithm also allows you to record a signal and play it back as a loop. By using this "Instant Replay" feature, you can play/sing along with a looped passage, creating some interesting effects.

The parameters available for this algorithm are:

01 — Mix

02 — Volume

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

03 — Delay Time

Range: 0 to 3668 ms

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

04 — Delay Regen

Range: 00 to 99

Determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delays. A setting of 99 would offer an infinite delay.

05 — Delay Pan

Range: -99 to +99

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

06 — Delay Regen Damping

Range: 00 to 99

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

07 — Delay Mode

Range: Continuous, Loop/Muted, Loop/Record, Loop/Replay

This mode parameter selects the "instant replay" feature. When set to Continuous, any signal entering the unit will delay. When set to Loop, you can create an "instant replay" loop using any mod source defined in parameter 08.

08 — Delay Set

Range: Off, Controllers 1 - 8

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

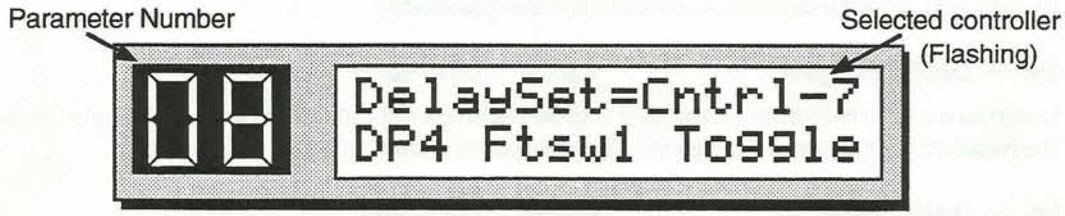
The playback will go into a muted state when a recording is made that is less than 300 ms.

Using the Instant Replay Feature

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

1. Press **System•MIDI** until the display shows Parameter #45:

2. Using the **Data Entry Knob**, set the DP/4 Footswitch 1 to "DP4 Controller." This allows the foot switch to be used as the modulation source.
3. Press **Edit**, and use the **Arrow** buttons to select parameter 08. Turn the **Data Entry Knob** and select "DP4 Ftsw1 Toggle." The display looks like this:



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

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

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

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

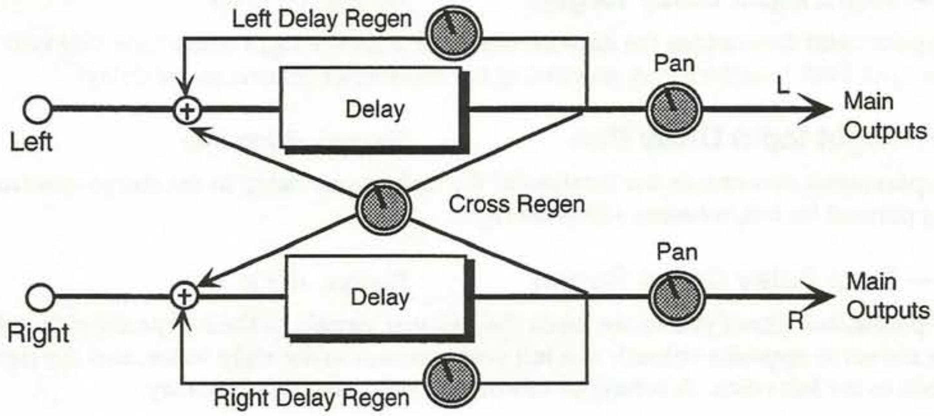
09 — Mod1 Source
10 — Mod1 Destination
11 — Mod1 Param Range Min
12 — Mod1 Param Range Max
13 — Mod2 Source
14 — Mod2 Destination
15 — Mod2 Param Range Min
16 — Mod2 Param Range Max

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

DUAL DELAY

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

Dual Delay Signal Routing



The parameters available for this algorithm are:

01 — Mix

02 — Volume

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

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

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

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

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

05 — Left Input Delay Regen — Range: 00 to 99

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

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

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

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

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

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

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

09 — Right Input Delay Regen
Range: 00 to 99

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

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

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

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

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

12 — Dual Delay Regen Damping
Range: 00 to 99

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

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.

TEMPO DELAY

00 — Tempo Delay features a stereo digital delay (similar to MultiTap) where the tempo is controlled by an assignable modulation source, like a foot switch.

The parameters available for this algorithm are:

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: various

This parameter determines how the tempo will be controlled: Internal clock, MIDI clocks, or FootSwitch 1 Tapping.

In order for Foot Switch 1 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

This parameter 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

This parameter sets the location within the stereo spectrum for the delayed signal.

09 — Tempo Delay Regen Damping Range: 00 to 99

This parameter controls the cut off of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The higher the number, the more the signals are dampened. We recommend lower settings.

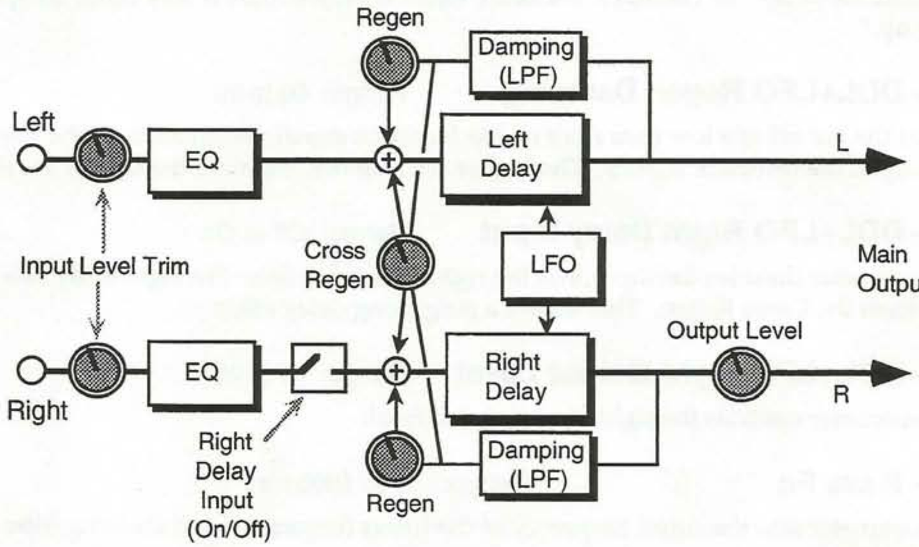
10 — Mod1 Source
11 — Mod1 Destination
12 — Mod1 Param Range Min
13 — Mod1 Param Range Max
14 — Mod2 Source
15 — Mod2 Destination
16 — Mod2 Param Range Min
17 — Mod2 Param Range Max

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

EQ - DDL - WITH LFO

00 — EQ-DDL-withLFO features 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



The parameters available for this algorithm are:

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

This parameter determines the amount of time between the input signal and the left delay output.

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

This parameter determines the amount of time between the input signal and the right delay output. Set this differently from parameter 03 to achieve dotted1/8th note type effects.

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

This parameter 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

This parameter 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 in phase, the left and right choruses will modulate their detunes together. When 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

This parameter controls the amount of regen (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

This parameter 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 delay. Be careful, if the delay regen is set too high, it may cause this parameter to "blow up."

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

This parameter 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

This parameter 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

This parameter allows you to adjust the input volume of the EQs to eliminate the possibility of clipping boosted signals.

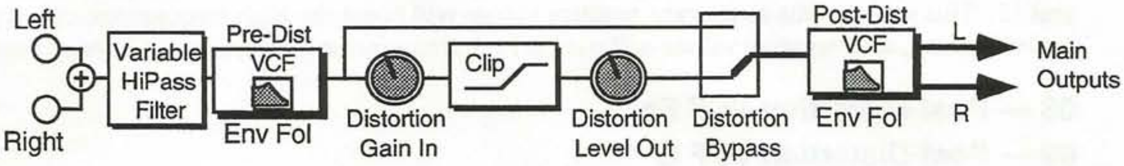
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

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

VCF - DISTORTION

00 — VCF - Distortion combines a voltage control filter and a raspy distortion. 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. It can be set to act like a simple a speaker simulator, or it can be modulated in parallel with the pre-distortion VCF.



The parameters available for this algorithm are:

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

This parameter 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

This parameter 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. At mid positive values, Fc will go high, but then come down to its nominal setting. At negative mid values, 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" sound, and negative values will cut the high frequencies, producing a "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

This parameter 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 previous 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 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

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

GUITAR AMP 1, GUITAR AMP 2

These algorithms recreate the warm sound of a 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.

00 — **Guitar Amp 1** is designed for Hard Rock sounds.
00 — **Guitar Amp 2** is optimized for "bluesy" type sounds.

The parameters available for this algorithm are:

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

This parameter 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 desireable 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 desireable 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

This parameter 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

This parameter 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

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

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

This parameter 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

This parameter 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

This parameter 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 which are provided in the DP/4.

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

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

GUITAR AMP 3

00 — Guitar Amp 3 combines the inverse expander with a bright distortion for amp lead sounds. The inverse expander may be thought of as a compressor which amplifies all signals below the threshold. This algorithm is optimized for heavy metal guitar solos.

The parameters available for this algorithm are:

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

This parameter 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

This parameter 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

This parameter 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

This parameter 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

This parameter 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

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

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

This parameter 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

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 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.

16 — 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.

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

This parameter 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.

19 — 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.

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

This parameter filters out the high frequencies of the speaker. The higher 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.

SPEAKER CABINET

00 — Speaker Cabinet simulates the warm sound of an opened 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 with VCF-Distortion; turn down the volume there and make up for it with the output gain here.

For a brighter speaker emulation, try using Tunable Speaker.

The parameters available for this algorithm are:

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.

TUNABLE SPEAKER

00 — **Tunable Speaker** 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.

The parameters available for this algorithm are:

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

This parameter 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.

ROTATING SPEAKER

00 — Rotating Spkr adds the famous classic rotating speaker sound to any keyboard instrument. This algorithm also sounds great with other instruments as well. A tunable distortion is added to the input signal and is also passed through the rotors.

The parameters available for this algorithm are:

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

This parameter adjusts the slow rate of the rotating speaker. This parameter will only work if the Speaker Speed (parameter 03) is set to Slow. By assigning a modulation controller to this parameter, you can change the speed in real time.

04 — Rotating Speaker Fast Speed   Range: 01 to 55

This parameter adjusts the fast rate of the rotating speaker. This parameter will only work if the Speaker Speed (parameter 03) is set to Fast. By assigning a modulation controller to this parameter, you can change the speed in real time.

05 — Rotating Speaker Speed   Range: Slow or Fast

This parameter determines the rate of the rotating speaker.

06 — Rotating Speaker Inertia   Range: 00 to 99

This parameter controls the rate of change from slow to fast. Adjust this parameter to simulate the effect of the rotary speaker gradually picking up speed.

07 — Distortion Level In   Range: -48 to +48 dB

This parameter 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 how much of the amplifier distortion is added to the input signal (there is an internal clean path in parallel with the distortion). 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 Off, there is no distortion.

10 — Rotating Speaker Stereo Spread Range: 00 to 99

This parameter 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.

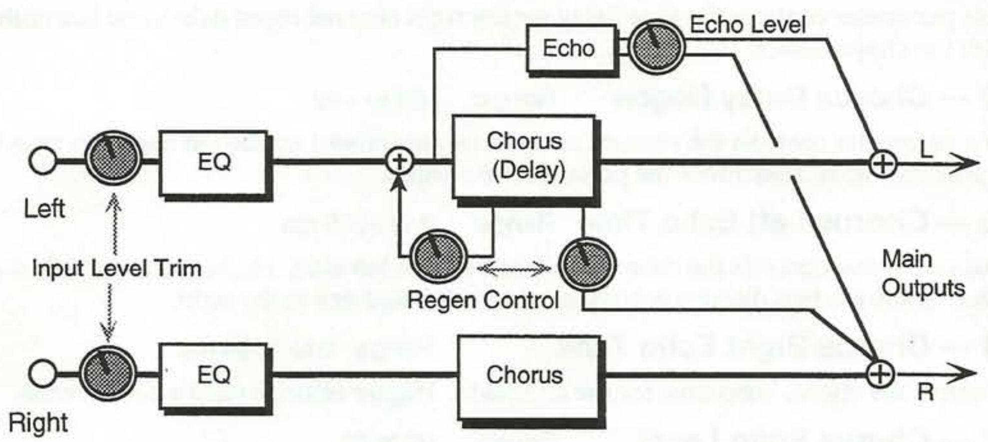
11 — Mod1 Source
12 — Mod1 Destination
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

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

EQ - CHORUS - DDL

00 — EQ-Chorus-DDL combines an EQ with a chorus and a digital delay. This is the industry standard chorus effect, designed with very long delays to provide a modulated detune effect. This algorithm sounds great with a guitar, but try it with any source!

EQ-Chorus-DDL Signal Routing



The signal enters a programmable EQ, which is preceded by an input level trim (parameter 16). The signal is then routed to the chorus which is heard directly at the output. There is also a delayed unchorused signal (sharing the same delay lines) that is routed back into the chorus. There is also a second signal from the delay line that is routed to the right side. There are two discrete echo times tapped out before the chorus delay line. These are unchorused echoes. The signal from the echoes are routed directly to the outputs. There is also a dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

The parameters available for the EQ-Chorus-DDL are:

01 — Mix

02 — Volume

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

03 — Chorus LFO Rate Range: 00 to 99

This parameter controls the rate of pitch modulation which is the chorus. To achieve chorusing, this rate must be very slow.

04 — Chorus LFO Width Range: 00 to 99

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

05 — Chorus Center Range: 00 to 99

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

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

When in phase, the left and right choruses will modulate their detunes together. When out-of-phase, the detune on the left channel will go up while the detune on the right will go down.

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

This parameter controls the time delay for the left channel regen delay, and has nothing to do with the chorus effect.

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

This parameter controls the time delay for the right channel regen delay, and has nothing to do with the chorus effect.

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

This parameter controls the amount of regen (regeneration) applied to the delay time taps. The sign of the value determines the polarity of the regen.

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

This parameter controls the chorus 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 — Chorus Right Echo Time Range: 0 to 1500 ms

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

12 — Chorus Echo Level Range: 00 to 99

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

13 — Bass Fc Range: 0 to 1000 Hz

This parameter 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

This parameter allows you to adjust the input volume of the EQs to eliminate the possibility of clipping boosted signals.

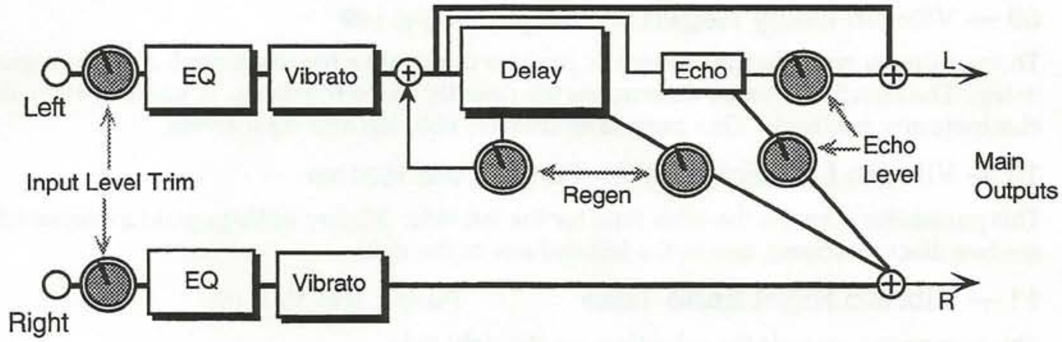
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

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

EQ - VIBRATO - DDL

00 — **EQ-Vibrato-DDL** combines a vibrato effect (a pitch shifter modulating over a very small range) with EQ and 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 which 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. A 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 an external dry signal (not shown) that goes directly from the input to the output and is controlled with the mix parameter (01).

The parameters available for this algorithm are:

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 — Vibrato Rate — Range: 00 to 99

This parameter sets the amount of modulation. Higher values create a faster vibrating rate.

04 — Vibrato Width — Range: 00 to 99

This parameter sets the width (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 degrees from the right. When in phase, both channels will change pitch together.

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

This parameter 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

This parameter controls the time delay on the left regenerated delay.

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

This parameter controls the time delay on the right nonregenerated delay.

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

This parameter 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

This parameter 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

This parameter controls the echo time for the right side.

12 — Vibrato Echo Level   Range: 00 to 99

This parameter 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

This parameter selects the cutoff frequency of the low 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 selects the cutoff of the upper frequency band 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

This parameter allows you to adjust the input volume before the EQs to eliminate the possibility of clipping boosted signals.

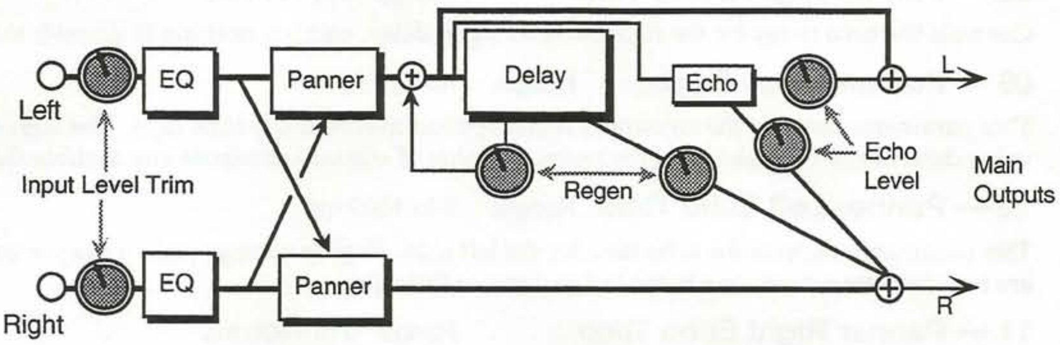
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

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

EQ - PANNER - DDL

00 — 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



In this algorithm the signal enters an input level trim (parameter 17) followed by a programmable EQ, and then is 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. Another signal from the delay is routed to the output on the right side. One regen parameter (09) 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).

The parameters available for this algorithm are:

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. Higher values create a wider separation.

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 a stereo signal becomes mono. Switch between the two settings until it sounds right for your routing config.

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

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. The sample and hold function can be turned off.

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

Controls the time delay for the left channel regen delay, and has nothing to do with the pan effect.

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

Controls the time delay for the right channel regen delay, and has nothing to do with the pan effect.

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

This parameter controls the amount of regen applied to both 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 deeper 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

This parameter 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

This parameter selects 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 selects the cutoff frequency of the upper frequency band 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

This parameter allows you to adjust the input volume of the EQs to eliminate the possibility of clipping boosted signals.

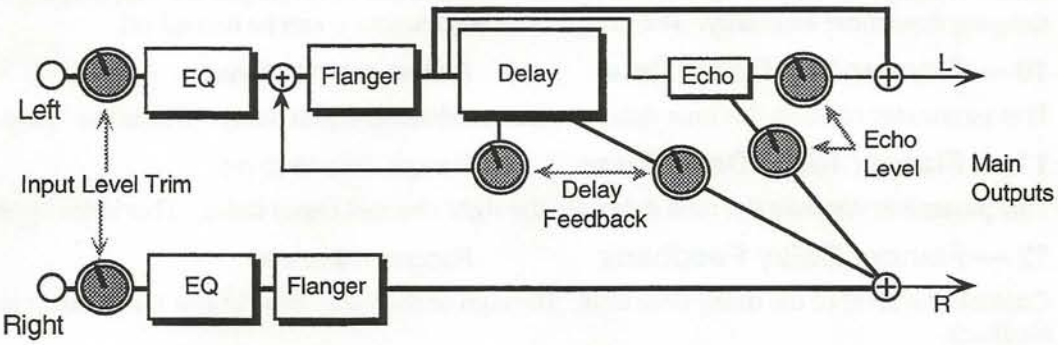
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

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

EQ - FLANGER - DDL

00 — 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).

The parameters available for this algorithm are:

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

This parameter controls the rate of modulation of the flanger effect.

04 — Flanger LFO Width   Range: 00 to 99

This parameter controls the range of the high to low frequency sweep in the flanger effect.

05 — Flanger Center   Range: 00 to 99

This parameter 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

This parameter 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: 001 to 100, or Off

This parameter 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

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

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

This parameter 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

This parameter 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

This parameter sets the cutoff frequency of the lower frequency band shelving filter.

17 — EQ Gain   Range: -48 to +24 dB

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

18 — Treble Fc   Range: 01KHz to 16KHz

This parameter selects the cutoff of the upper frequency band high shelving filter.

19 — EQ Gain   Range: -48 to +24 dB

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

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

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

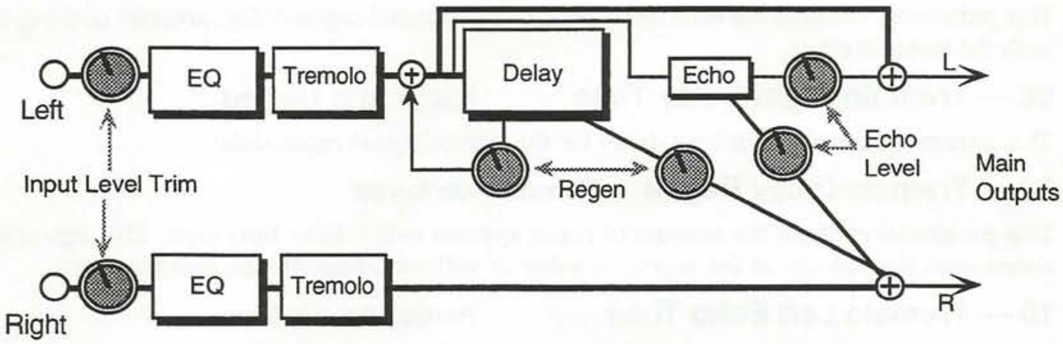
21 — Mod1 Source
22 — Mod1 Destination
23 — Mod1 Param Range Min
24 — Mod1 Param Range Max
25 — Mod2 Source
26 — Mod2 Destination
27 — Mod2 Param Range Min
28 — Mod2 Param Range Max

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

EQ-TREMOLO-DDL

00 — **EQ-Tremolo-DDL** combines a tremolo effect, which is a pulsating change in volume, with an EQ and 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).

The parameters available for this algorithm are:

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

This parameter 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.

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: 001 to 100, or Off

This parameter 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, and has nothing to do with 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

This parameter 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 deeper 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

This parameter controls the volume of the discrete echo for both the left and right sides.

13 — Bass Fc   Range: 0 to 1000 Hz

This parameter selects 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 selects the cutoff frequency of the upper frequency band 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

This parameter allows you to adjust the input volume of the EQs to eliminate the possibility of clipping boosted signals.

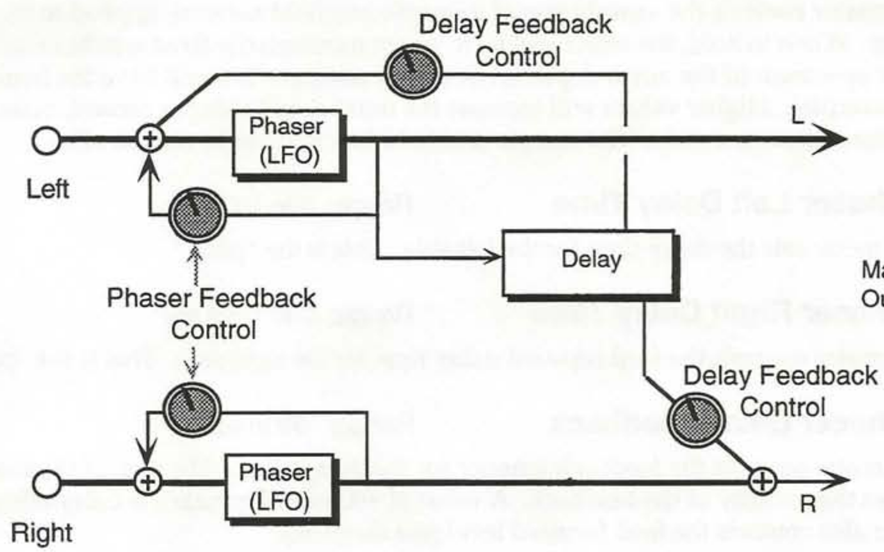
18 — Mod1 Source
19 — Mod1 Destination
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

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

PHASER-DDL

00 — 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; this is what differentiates the phaser from the flanger. The phasing effect is achieved within the Phaser topology, so 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 topology). 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 sec. ping-pong effect, and is very effective as a "poor man's reverb."

Phaser - DDL Signal Routing



The parameters available for this algorithm are:

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

This parameter controls the width of the notch *excursion*. For large excursions set this parameter to 99 which can give a very high "woosh" and a very low "woosh."

05 — Phaser Center   Range: -99 to +99

This parameter 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

This parameter 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

This parameter 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: 001 to 100, or Off

This parameter 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

This parameter controls the feed forward delay time for the right side. This is the "pong."

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

This parameter 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).

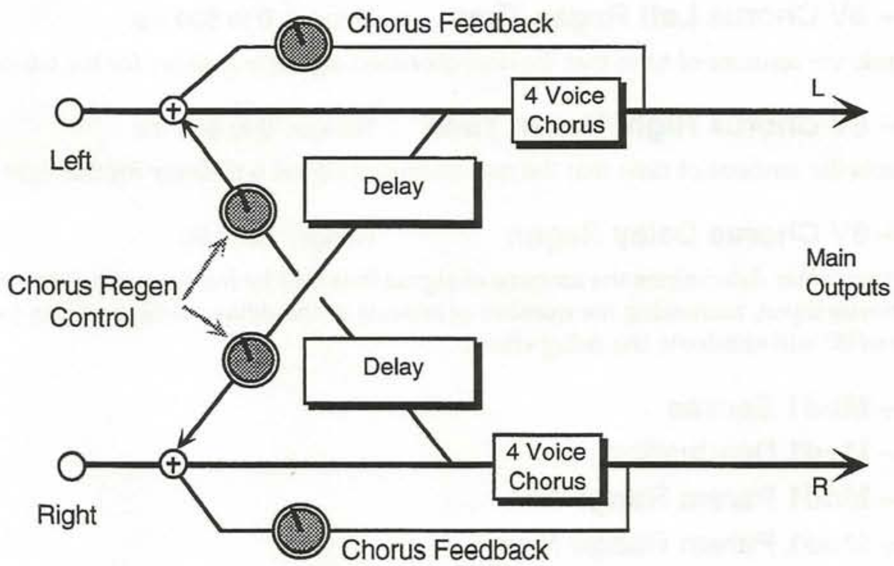
13 — Mod1 Source
14 — Mod1 Destination
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

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

8 VOICE CHORUS

00 — 8 Voice Chorus offers a symphonic chorused sound having eight different voices and using eight separately randomized LFOs. This algorithm also offers a user programmable stereo delay in a cross coupled configuration between the left and right chorused outputs (see diagram). This algorithm is good for creating an ensemble of instruments from single sources (there is no internal filtering applied to any of the chorused voices).

8 Voice Chorus Signal Routing



The parameters available for this algorithm are:

01 — Mix

02 — Volume

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

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

This parameter controls the eight ganged rates of modulation of the respective voices. This modulation produces an effect similar to both vibrato and tremolando occurring at the same time.

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

This parameter controls the excursion of the vibrato of all the individual voices.

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

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

06 — 8V Chorus Regen Range: 00 to 99

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

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

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

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

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

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

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

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

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

FLANGER

00 — 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.

The parameters available for this algorithm are:

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

This parameter 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.

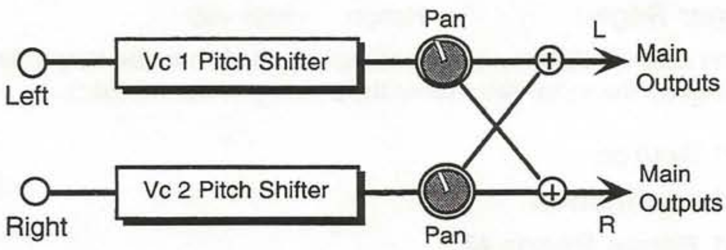
PITCH SHIFTER, PITCH SHIFT 2U

The Pitch Shifters allow you to change the pitch of a signal to any pitch desired within a range of one octave in either direction. The DP/4 offers four different pitch shift algorithms, each designed with a different purpose. There are three different 1 Unit Pitch Shifters, and one 2 Unit Pitch shifter:

00 — **Pitch Shifter** offers a 1 unit splicer type pitch shifter.
00 — **Pitch Shift 2U** is a splicer type incorporating zero crossing detection.
00 — **PitchShift-DDL** combines a pitch shifter with a DDL (described later).
00 — **FastPitchShift** is a 1 unit pitch shifter designed for pitch correction.

Try the different pitch shifters until you find the one that works best with your sound source, and for your application. These first two algorithms listed are "splicer types" of pitch shifters, which means that they drop or add small sections of the original signal to the effect. 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. Pitch Shift (1U) uses only one ESP chip, 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. All pitch shifters except PitchShift-DDL take the left channel input as Voice 1, and the right channel input as voice 2.

Pitch Shifter, Pitch Shift 2U, FastPitchShift Signal Routing



The parameters available for these algorithms are:

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

This parameter 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

This parameter allows you to adjust the volume of Voice 1. A setting of 00 would eliminate any audible pitch shift.

06 — PitchShifter Vc 1 Pan Range: -99 to +99

This parameter 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

This parameter allows you to adjust 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

This parameter allows you to adjust the volume of Voice 2. A setting of 00 would eliminate any audible pitch shift.

10 — PitchShifter Vc 2 Pan Range: -99 to +99

This parameter 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

(Pitch Shifter 1U Only)

This parameter allows you to choose between a long/smooth setting, or a short/coarse 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.

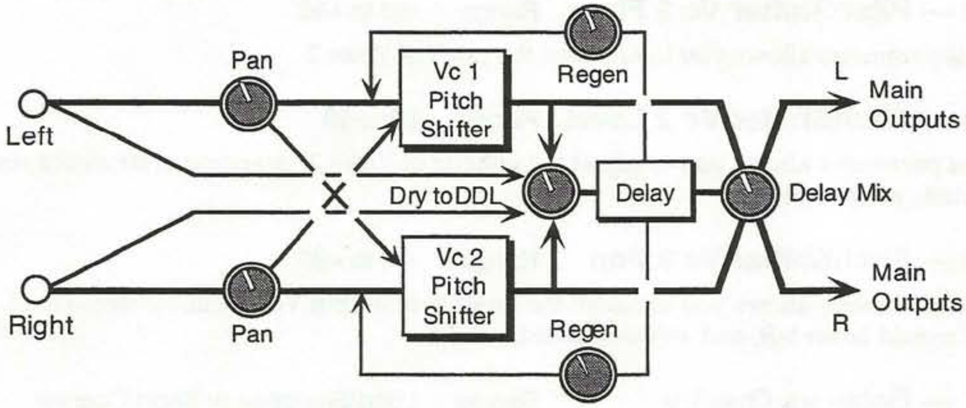
Mod1 Source
Mod1 Destination
Mod1 Param Range Min
Mod1 Param Range Max
Mod2 Source
Mod2 Destination
Mod2 Param Range Min
Mod2 Param Range Max

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

PITCHSHIFT - DDL

00 — 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 which feeds back into the pitch shift. This feature allows spiraling upward or downward pitch shifts.

PitchShift-DDL Signal Routing



The parameters available for this algorithm are:

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 this parameter to fade in or fade out the pitch shifted signal.

03 — PitchShift Vc 1 Semi   Range: -12 to +12

This parameter allows you to adjust 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

This parameter allows you to assign the location of Voice 1 in the stereo field. A value of -99 would be far left, and +99 would be far right.

07 — PitchShift Vc 2 Semi   Range: -12 to +12

This parameter 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

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.

11 — PitchShift Dry Level to DDL Range: 00 to 99

Allows you to 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 (see diagram).

15 — PitchShift Delay Regen Range: -99 to +99

This parameter 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.

FAST PITCH SHIFT

00 — FastPitchShift has a transport delay of only 10 ms 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 should be used for pitch correction (for instance, try hooking up a mod wheel for MIDI).

The parameters available for this algorithm are:

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

This parameter allows you to adjust the volume of Voice 1. A setting of 00 would eliminate any audible pitch shift.

05 — PitchShifter Vc 1 Pan Range: -99 to +99

This parameter 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

This parameter allows you to adjust 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

This parameter 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

This parameter controls the excursion 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

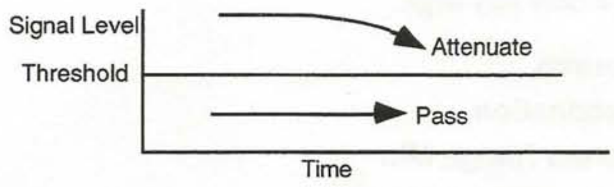
17 — Mod2 Param Range Min

18 — Mod2 Param Range Max

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

EQ - COMPRESSOR

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



The parameters available for this algorithm are:

01 — Mix

02 — Volume

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

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

This parameter boosts the compressed signal level.

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

This parameter sets the amount of compression. The range is based on decibels (dB) above the threshold. If this is set to 4:1 for example, it will compress changes in signals above the threshold by one quarter. When this is set to infinity, it acts as a limiter.

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

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

06 — Gain Change   Range: N/A

This read only parameter displays a gain reduction meter.

07 — Comp Attack   Range: 50µs to 100ms

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

08 — Comp Release   Range: 1ms to 10.0s

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

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

This parameter 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

This parameter 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

This parameter 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

This parameter sets the cutoff frequency of the lower frequency band shelving filter.

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

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

14 — Treble Fc Range: 01KHz to 16KHz

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

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

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

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

This parameter allows you to adjust the input volume of the EQs, to eliminate the possibility of clipping boosted signals.

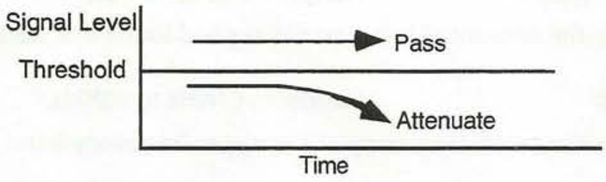
17 — Mod1 Source
18 — Mod1 Destination
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

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

EXPANDER

00 — Expander performs downward expansion of input signals. For high expansion ratios this algorithm functions as a gate. This algorithm operates by expanding (attenuating) signals below the threshold and passing the signals above the threshold. The Threshold is a parameter defined by the user. 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 posesses two features which are new:

1. The ADSR 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 amout of gain reduction in real time.

06 — Exp Attack   Range: 50µs to 100ms

This parameter 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

This parameter 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 parameter is the detection sustain time in the ADSR which constitutes attack, sustain, and release.

09 — Sidechain EQ Gain   Range: -48 to +48 dB

This parameter 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

This parameter 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 — 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 — 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 cut (negative values) or boost (positive values) applied to the expander on the output volume. We recommend a starting application of +00 dB.

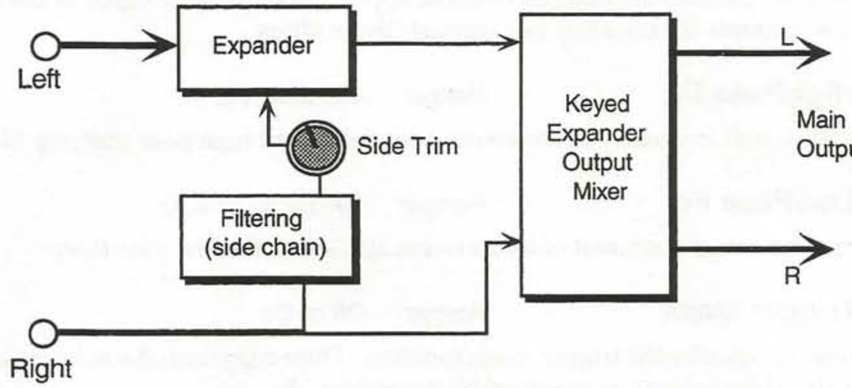
16 — Mod1 Source
17 — Mod1 Destination Parameter
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination Parameter
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

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

KEYED EXPANDER

00 — 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).

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

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 amout of gain reduction in real time.

06 — Exp Attack Range: 50µs to 100ms

This parameter 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

This parameter 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 parameter is the detection sustain time in the ADSR which constitutes attack, sustain, and release.

09 — Sidechain EQ Gain Range: -48 to +48 dB

This parameter 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 parameter sets the cutoff frequency of the lower frequency band high pass shelving filter.

11 — LowPass Fc Range: 100 Hz to 16 KHz

This parameter 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 — 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

This parameter 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

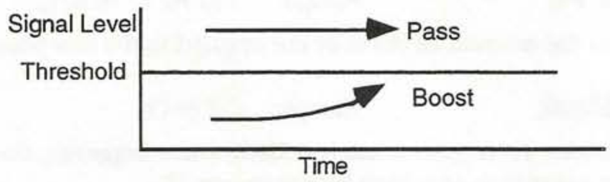
This parameter sets the amount of cut (negative values) or boost (positive values) applied to the expander on the output volume. We recommend a starting application of +00 dB.

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.

INVERSE EXPANDER

00 — 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.



The parameters available for this algorithm are:

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 by three times. 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

This parameter determines the time after the initial signal amplitude has been detected for the expansion to take affect.

07 — Exp Release Range: 1ms to 10.0s

This parameter 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

This parameter 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

This parameter sets the cutoff frequency of the lower frequency band shelving filter.

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

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

12 — Treble Fc Range: 01KHz to 16KHz

This parameter 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

This parameter allows you to adjust the level of the signal entering the EQ, to eliminate the possibility of clipping boosted signals.

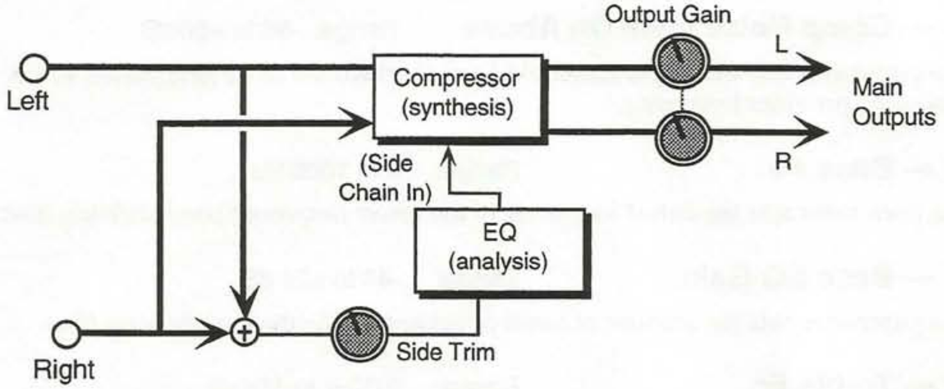
15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

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

DE-ESSER

00 — De-esser is a stereo algorithm that compresses sibilant frequencies (like the "ess" sound) as they become louder. This was designed for vocalists, but it can also be used to control the boomy sound of a guitar or the ringing sound of drums by adjusting the side-chain appropriately.

De-esser Signal Routing



01 — Mix

02 — Volume

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

03 — Output Gain Range: -48 to +48 dB

This parameter sets the amount of cut (negative values) or boost (positive values) applied to the de-esser on the output volume. We recommend a starting application of 00 dB.

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

This parameter sets the amount of compression. The range is based on decibels (dB) above the threshold. If this is set to 4:1 for example, it will compress changes in signals above the threshold by one quarter. When this is set to infinity, it acts as a limiter.

05 — Threshold Range: -96 to +00 dB

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

06 — Gain Change Range: N/A

This read only parameter displays a gain reduction meter.

07 — Comp Attack Range: 50µs to 100ms

This parameter determines the attack rate after the initial signal has been detected and before the compression takes affect.

08 — Comp Release Range: 1ms to 10.0s

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

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

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

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

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

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

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

12 — Bass Fc Range: 0 to 1000 Hz

This parameter sets the cutoff frequency of the lower frequency band shelving filter.

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

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

14 — Mid1 Fc Range: 100 to 9999 Hz

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

15 — Mid1 Gain Range: -48 to +24 dB

This parameter sets the amount of boost or cut applied to this frequency parametric.

16 — 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 band you can produce a narrower bandwidth.

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

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

20 — Treble Fc Range: 01KHz to 16KHz

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

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

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

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

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

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

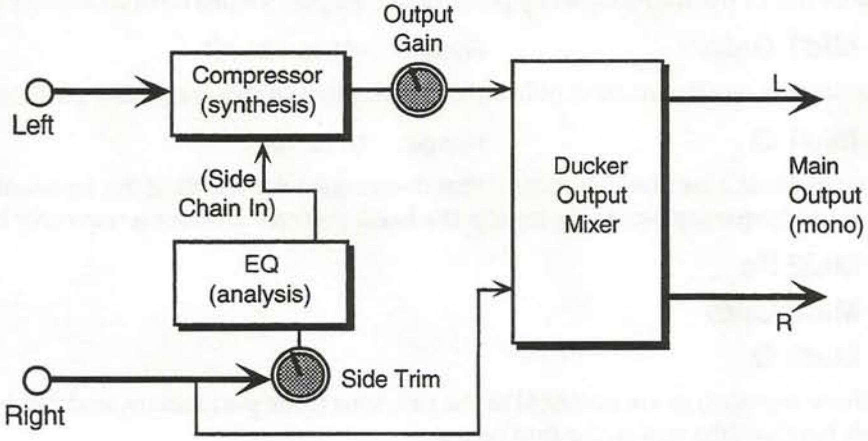
See the descriptions under the Algorithm Modulators earlier in this section.

DUCKER / GATE

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

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

Ducker / Gate Signal Routing



01 — Mix

02 — Volume

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

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

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

04 — Ducker Output Mix   Range: 00 to 99

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

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

This parameter sets the amount of compression. The range is based on decibels (dB) above the threshold. If this is set to 4:1 for example, it will compress changes in signals above the threshold by one quarter. When this is set to infinity, it acts as a limiter.

06 — Threshold   Range: -96 to +00 dB

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

07 — Gain Change   Range: N/A

This read only parameter displays a gain reduction meter.

08 — Comp Attack Range: 50µs to 100ms

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

09 — Comp Release Range: 1ms to 10.0s

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

10 — 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.

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

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

12 — Bass Fc Range: 0 to 1000Hz

This parameter sets the cutoff frequency of the lower frequency band shelving filter.

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

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

14 — Mid1 Fc Range: 100 to 9999 Hz

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

15 — Mid1 Gain Range: -48 to +24 dB

This parameter sets the amount of boost or cut applied to this frequency parametric.

16 — Mid1 Q Range: 01 to 18

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

17 — Mid2 Fc

18 — Mid2 Gain

19 — Mid2 Q

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

20 — Treble Fc Range: 01KHz to 16KHz

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

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

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

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

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

23 — Mod1 Source

24 — Mod1 Destination

25 — Mod1 Param Range Min

26 — Mod1 Param Range Max

27 — Mod2 Source

28 — Mod2 Destination

29 — Mod2 Param Range Min

30 — Mod2 Param Range Max

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

RUMBLE FILTER

00 — **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.

The parameters available for the Rumble Filter are:

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 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

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

PARAMETRIC EQ

00 — Parametric EQ offers a minimum phase four band parametric EQ.

The parameters available for this algorithm are:

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

This parameter sets the center of the low frequency parametric.

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

This parameter sets the amount of boost or cut applied to this low frequency parametric.

05 — Mid1 Fc Range: 100 to 9999 Hz

This parameter 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. 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 (05, 06 and 07), and are used to control different bandwidths within the mid range.

11 — Treble Fc Range: 01 to 16 KHz

This parameter 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

This parameter allows you to adjust 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.

VAN DER POL FILTER

00 — VandrPol Filter adds synthetic high harmonics to the input signal, brightening the overall sound. This newly designed 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.

The parameters available for the VandrPol Filter are:

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 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

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

SINE/NOISE GEN

00 — Sine/Noise Gen is a utility algorithm, but when used with a real time modulator/controller, might provide some interesting musical effects. Filters are provided for the noise, but no filters are provided for the sinusoid.

The parameters available for this algorithm are:

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

This parameter 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

This parameter cuts out the high frequencies and can be used to create pink noise.

06 — Bass Fc   Range: 0 to 1000 Hz

This parameter 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

This parameter allows you to adjust 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.

NO EFFECT (BYPASS EFFECT)

00 — **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*.

The parameters available for this algorithm are:

01 — Mix Range: 00 to 99

Controls the mix of a dry signal with silence. 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. The Mix parameter for this algorithm 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.

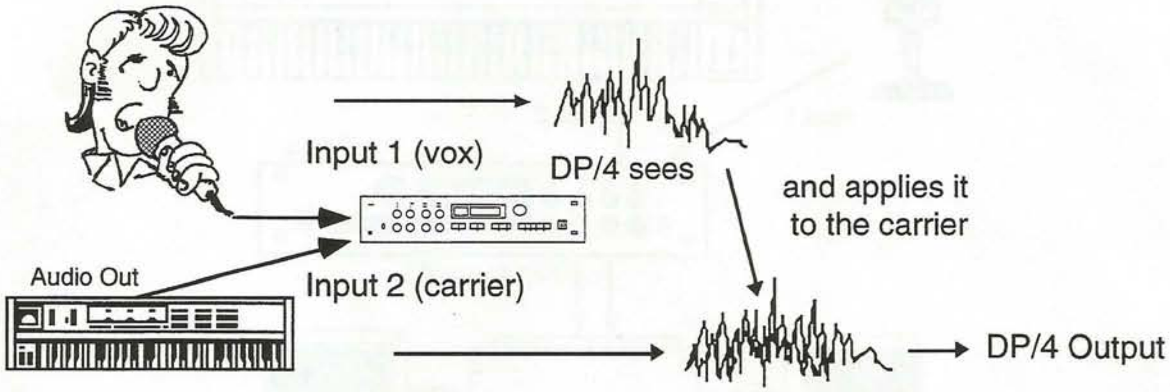
03 — Mod1 Source
04 — Mod1 Destination
05 — Mod1 Param Range Min
06 — Mod1 Param Range Max
07 — Mod2 Source
08 — Mod2 Destination
09 — Mod2 Param Range Min
10 — Mod2 Param Range Max

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

Section 5 — The Vocoder

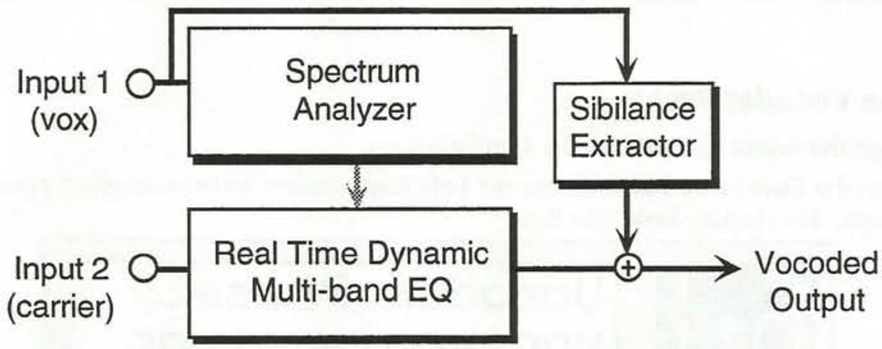
About the Vocoder

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 or sampler).



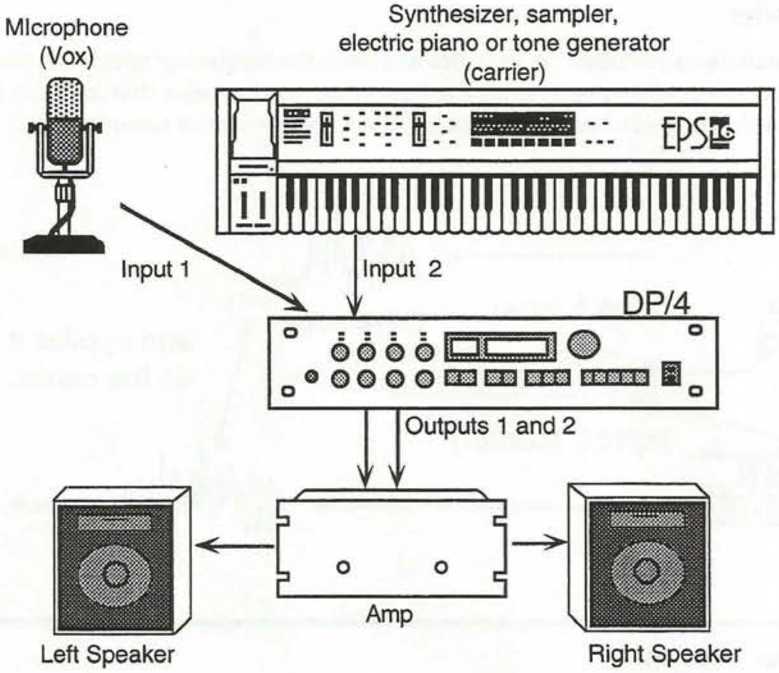
How the Vocoder Works

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

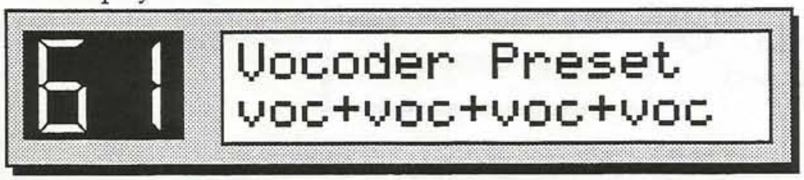


Making the Right Connections

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 the **Select** button, then the **Config** button.
2. Turn the **Data Entry Knob**, or use the **Left/Right Arrow** buttons to select preset #61 Vocoder Preset. The display looks like this:



3. Press the **Select** button to engage the Vocoder preset. The DP/4 automatically takes you to Unit A.

Using the Vocoder

As you speak into the mike, 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 mike while playing a single note, or to create choir sounds by saying "aah" or "ooh" into the mike 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 Parameters

The vocoder algorithms have a particular set of user programmable parameters associated with the effect. The first parameter is the algorithm name:

Vocoder Low

Vocoder Mid 1

Vocoder Mid 2

Vocoder High

The parameters available for the vocoder are:

Mix Range: 00 to 99

This parameter 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

This parameter adjusts the output volume. Setting this parameter to 00 will eliminate the signal, and will produce no sound.

Speech Gain Ranges: -48 to +48 dB

This parameter 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 Ranges: 00 to 99

This parameter 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

This parameter 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.

Vocoder Modulators

The vocoder, like the other algorithms, allows real time control of particular parameters and shares common modulation control parameters:

Mod1 Source

Mod2 Source
Range: Off/Controller 1 - 8

These parameters select the modulation sources used to modulate the parameter destinations. Each algorithm has a choice of two different mod sources. Any one of the controller sources assigned in the System•MIDI mode can be selected (for more information, refer to the System Global Parameters in the System•MIDI Section).

Mod1 Destination

Mod2 Destination
Range: 00 to 34

These parameters select which algorithm parameters will be modulated by the selected modulation sources. Any parameter within an algorithm can be selected (except the algorithm name). Each algorithm within the vocoder has a choice of two different mod destinations.

Mod1 Param Range Min

Mod1 Param Range Max

Mod2 Param Range Min

Mod2 Param Range Max
Range: 00 to 99

These four parameters set the minimum and maximum amount (based on a percentage of the selected parameters range) that the Mod Destination will be modulated by the Mod Source.

The eight different Mod Source controllers (two for each unit) are assigned in System•MIDI mode and are explained in more detail in *Section 6 — System•MIDI Mode.*

Section 6 — System•MIDI Mode

About System•MIDI mode

Press the **System•MIDI** button to enter System•MIDI mode. The System•MIDI LED will light to indicate this mode.

In this mode you will find the system and MIDI parameters which control how the DP/4 responds to MIDI and the foot switch and CV Pedal control inputs. You will also find user preference parameters, such as parameter wrapping and preset Auto-Load which allow you to tailor the user interface to your liking. System parameters are not affected by preset changes.

Furthermore, you will find a flexible system exclusive facility which allows presets and system parameters to be dumped to and loaded from an external MIDI System Exclusive recorder.

All parameters in System mode belong to one of two categories:

- Parameters specific to units (MIDI channels, program change maps, etc.)
- Parameters that affect the operation of the system "globally", such as user preference switches and the DP/4 system controllers used as algorithm parameter modulation sources ("global" means they function on a system-wide basis).

In addition to selecting parameters with the **Left** and **Right Arrow** buttons, you are provided with the following shortcuts to help you get quickly to the parameters you wish to modify:

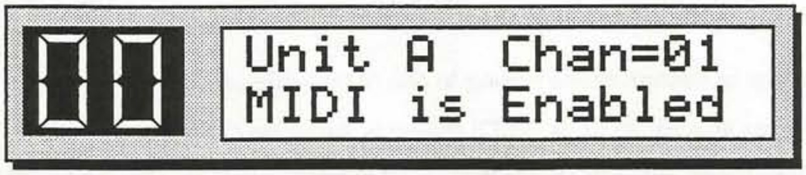
- Press a Unit button (**A**, **B**, **C** or **D**) or **Config** to get to the beginning of its MIDI setup parameter list.
- Press the **System•MIDI** button repeatedly to cycle through groups of system global parameters.

Param#	Parameter function	Press:
00-06	Unit A MIDI setup	Unit A button for #00
07-13	Unit B MIDI setup	Unit B button for #07
14-20	Unit C MIDI setup	Unit C button for #14
21-27	Unit D MIDI setup	Unit D button for #21
28-34	Config MIDI setup	Config button for #28
35-36	MIDI chan for controllers	System button repeatedly, until #35 is displayed
37-44	Defining 8 DP/4 controllers	System button again for #37
45-49	Footswitch function & preset chains	System button again for #45
50-51	MIDI Sys-Ex enable and ID number	System button again for #50
52-62	User Preference parameters	System button again for #52
63	Software Version number	System button again for #63

Unit Specific Parameters

The first 35 System•MIDI parameters are the unit specific MIDI setup parameters. Each unit (A, B, C, D) and Config has seven MIDI setup parameters available. In this context only, the Config may be considered as a "virtual unit" because it has its own set of MIDI parameters which allow program changes to select config presets and bypass controllers to bypass all of the units together as is the case when the **Config** button is toggled.

The following screen displays are all shown for unit A. The other units' screens are similar and can be reached by pushing the respective unit buttons and then scrolling with the **Right Arrow** button. The parameter numbers for the other units are listed following each of the descriptions.



00 — MIDI Channel

Range: 01 to 16

This parameter allows you to set the MIDI channel to which the unit responds, if MIDI reception is enabled for the unit.

This is:	For:
parameter 00	Unit A
parameter 07	Unit B
parameter 14	Unit C
parameter 21	Unit D
parameter 28	Config

01 — MIDI Enable

Range: Disabled or Enabled

This parameter allows the selected unit's MIDI reception to be enabled or disabled.

This is:	For:
parameter 01	Unit A
parameter 08	Unit B
parameter 15	Unit C
parameter 22	Unit D
parameter 29	Config

How the DP/4 Uses MIDI Channels

The DP/4 can respond to a maximum of six MIDI channels at once. Each Unit (A, B, C, D and Config) can have its own MIDI channel on which to receive program changes and MIDI Controller 7 (volume) information. Additionally, there is a separate controller channel used to receive controllers, pitch bends, channel aftertouch, key pressure, note events and velocity.

Each of these channels can be enabled individually. The only restriction is that the Config channel has to be different than the Unit channels. The Unit channels may all be the same, and the controller channel can be the same as any Unit or Config channel.

The current config controls which channels are actively used. When units are ganged, the lower numbered (further left) units channel will be used:

- In a 1 Source Config, Unit A's channel is used.
- In a 2 Source Config, Unit A's and Unit C's channels are used.
- In a 3 Source Config, A, B and C's channels are used.
- In a 4 Source Config, all Unit's channels are used.

Active MIDI Channels

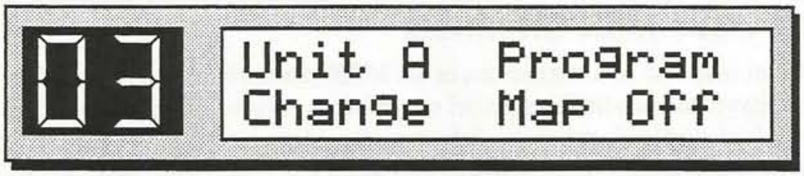
	A	B	C	D	Config
1 Source	●				●
2 Source	●		●		●
3 Source	●	●	●		●
4 Source	●	●	●	●	●

The Config MIDI channel is always active

02 — Program Change Range: Ignored or Received

This parameter determines whether you want to receive or ignore program changes with this particular MIDI setup. The DP/4 can receive MIDI program change messages to select presets. Program changes received on the Config channel select config presets. Program changes received on the Unit channels select 1, 2, or 4 unit presets, depending on the current config and the rules defined in the previous discussion. Program change reception can be enabled separately for each unit. There is also a MIDI program change master switch which has to be enabled in order to receive *any* program changes for individual units (see description of parameter 53).

This is:	For:
parameter 02	Unit A
parameter 09	Unit B
parameter 16	Unit C
parameter 23	Unit D
parameter 30	Config



03 — Program Change Map Range: Off or On

Each Unit and Config setup has a user programmable program change-to-preset map. This allows you to define which DP/4 preset is selected by each MIDI program change number received. The map may be programmed to ignore specific program change numbers or to use them to control the bypass status of the unit(s). The user programmable maps can be enabled or disabled separately for each unit.

If the parameter is set to "Off," the system uses a default preset map which maps program changes as follows:

- MIDI program changes 001 to 100 select presets 00 through 99.
- Program changes 101 to 128 are ignored.

If this parameter is set to "On" MIDI program changes received by the unit are translated into DP/4 presets using the unit's programmable map. Whenever the system is reinitialized, the user-programmable maps are reset to their default Range using the same mapping listed above, but with these exceptions:

- Program change 101 bypasses, 102 kills and 103 activates (un-bypasses) the affected unit(s).

This is:	For:
parameter 03	Unit A
parameter 10	Unit B
parameter 17	Unit C
parameter 24	Unit D
parameter 31	Config

Note: The DP/4 *displays* MIDI program changes as being numbered 001 to 128. The actual MIDI program changes *received* are numbered 000 to 127 in accordance with the MIDI Specification standard format. Some systems may display program changes using the MIDI standard format, so be alert for "off by one" problems.

If it does not seem to be working:

- Verify that your synthesizer is really sending MIDI program change events and the DP/4 is receiving those messages. The DP/4's MIDI message indicator (located at the bottom right corner of the LED Numeric display) should light up when the synth is sending program changes (e.g. when selecting patches).

- Make sure that the MIDI transmit channel (of the synthesizer) and the MIDI receive channel of the current unit (in the DP/4) match, and that program changes are enabled on the DP/4 in both the unit MIDI setup and the global switch (system parameter 53).

Program Change-to-Preset Map Editor

This two parameter screen is where you edit the user programmable program change-to-preset map.

04 — Program Change

Range: 001 to 128

The first parameter on this page selects the MIDI program change numbers used in the map.

This is:	For:
parameter 04	Unit A
parameter 11	Unit B
parameter 18	Unit C
parameter 25	Unit D
parameter 32	Config

05 — Preset Select

Range: 00 to 99, bypass, un-bypass, kill or ignore

The second parameter defines which DP/4 preset the displayed MIDI program change number will select.

This is:	For:
parameter 05	Unit A
parameter 12	Unit B
parameter 19	Unit C
parameter 26	Unit D
parameter 33	Config

You can now go back and forth between these two parameters and define which DP/4 preset is selected by each MIDI program change number. This process is called mapping. Multiple program change numbers can map onto the same preset number. A common application is to map your synthesizer's presets onto effects presets so each sound on the synth has an associated external effect.

> **Tip:** There is a quick way to edit the program change map if you have a keyboard or other device which can send program changes connected to the DP/4 MIDI Input:

- Select the second parameter (preset number).
- Send a program change (select a patch on your synthesizer) and notice the first parameter change. It should show the number of the program change received.
- Turn the **Data Entry Knob** to select the preset to be assigned to this program change.

You've just defined one entry in the program change-to-preset map. Send other program changes and use this process to map all 128 locations without ever changing parameters.

06 — Unit Bypass

Range: various

This parameter allows you to choose the controller source which will function as a bypass switch for the unit.

The available bypass controller sources are:

- MIDI Pitch Bend
- MIDI Note Number
- MIDI Note Velocity
- MIDI Aftertouch
- DP/4 Analog CV In
- DP/4 Footswitch 1
- DP/4 Ftsw1 Toggle
- DP/4 Footswitch 2
- DP/4 Ftsw2 Toggle
- MIDI Control #000 to #127
- Turned Off

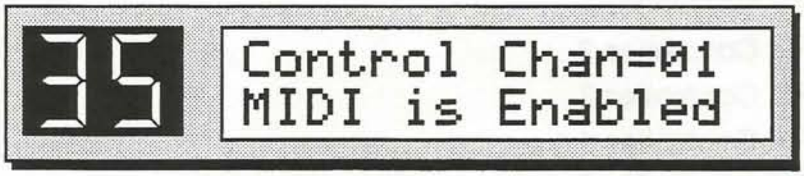
The same controller source can be used to modulate *and* bypass an effect. However, to avoid unexpected side effects, the modulation and bypass functions should usually be assigned to different controllers.

Note that MIDI Controllers used for bypass are only received on the DP/4's Controller channel.

This is:	For:
parameter 06	Unit A
parameter 13	Unit B
parameter 20	Unit C
parameter 27	Unit D
parameter 34	Config

System Global Parameters

There are some system global parameters (beginning with parameter 35) that can be reached by pressing the **System•MIDI** button repeatedly. Parameter 35 is the first page of a sub-group. The following system global parameters are available:



35 — Control Chan Range: 01 to 16

The first parameter on this page allows you to select the MIDI channel (01 through 16) on which MIDI controller messages (modulation sources) are received by the DP/4. This is the *only* channel on which the DP/4 can receive modulation and bypass controllers.

36 — Control Channel Reception Range: Disabled or Enabled

The next parameter (36) allows you to enable or disable the receiving of MIDI controllers for the entire system.

37 DP4 Controller1=
DP4 Analog CV In

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 Ranges: (see Sources list which follows)

Parameters 37 to 44 allow you to define eight system controllers to be used as modulation sources. Every algorithm has parameters which allow you to use two of these DP/4 controllers to modulate any two parameters in the algorithm (except parameter 00 –the algorithm name– which cannot be modulated).

Source	Notes and Examples of modulation application
MIDI Controller Sources	
Pitch Bend	Can be used to control such things as left/right panning and rotor speed.
Note Number	Can be used to enable higher MIDI note numbers from a piano to shorten the decay on the reverb.
Note Velocity	Can raise or lower the effect mix. Higher velocities from drums can increase detune amount.
Aftertouch	Both channel (mono) and key (poly) pressure are recognized and are combined into a single mono source.
Channel Controllers	The full range of controllers (0 to 127) is supported. Some of the most commonly used ones are displayed using their conventional names: Modulation Wheel (1), Breath Controller (2), Volume (7), and Pan (10).
Additional (non-MIDI) Controller Sources	
Control Voltage	Usually a CV Pedal such as ENSONIQ's CVP-1, but could be generated by anything that produces a 0 - 5 volt control voltage, such as an analog synthesizer's CV out.
Foot Switch 1 and 2	The down position generates maximum modulation and the up position sets modulation to minimum. The Foot Switch jack can accommodate both the ENSONIQ SW–2 (single) or the SW–10 (dual) foot switches. "Footswitch 2" is only usable when dual foot switches are connected by a stereo plug (e.g. the SW–10).

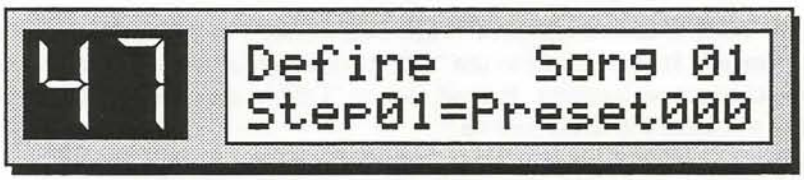
45 — DP/4 Foot Switch 1

46 — DP/4 Foot Switch 2 Ranges: (see the following list of Range)

Parameters 45 and 46 define how foot switch 1 and 2 will be used. These are the possible ranges:

DP/4 Controller	Allows the foot switch to be used as a modulation source. This is the only setting which allows the foot switch to be used for the tap-tempo feature in the Tempo Delay algorithm.
Increment Preset	Selects next higher preset of the current preset type on the current unit(s).
Decrement Preset	Selects next lower preset of the current preset type on the current unit(s).
Increment Song	Selects next higher song.
Decrement Song	Selects next lower song.
Song Preset Up	Selects next higher song step.
Song Preset Down	Selects next lower song step.
Not Used	Ignores foot switch events.

Song Editor



47 — Define Song Range: 01 to 20

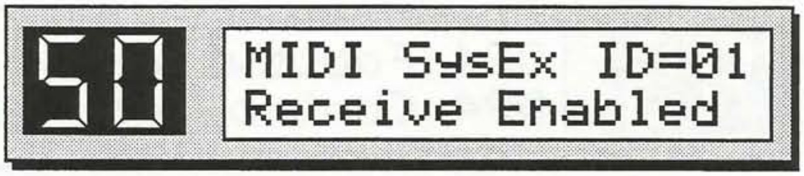
48 — Step Range: 01 to 05

49 — Preset Range: 000 to 099, Goto Step 1

Parameters 47 (Song), 48 (Step) and 49 (Preset) allow you to set up lists of presets that differ from the order they are stored in memory. This list is meant primarily for live performance and can only be accessed by the foot switches.

There are 20 songs available, each containing 5 steps that you can assign any preset (0-99) to. Assign your foot switch(es) to increment/decrement through songs/steps to make use of this feature.

For more information about using the Song Editor parameters, refer to *Section 8 — Applications.*



50 — MIDI SysEx ID Range: 01 to 16

This parameter sets the system exclusive ID. This number is not a MIDI channel. It is simply an ID number imbedded within the system exclusive message. This allows usage of multiple DP/4's with a universal editor librarian etc. All outgoing dumps will contain this ID number and incoming dumps will only be received if the ID in the message matches this parameter value.

51 — SysEx Reception Range: Disabled or Enabled

This parameter (second line) defines whether system exclusive messages can be *received* by the DP/4. Outgoing dumps can always be initiated by pressing the Write•Copy button while in System•MIDI Mode.

52 — Preset Memory Protect Range: Off or On

If set to "On" this parameter will protect any RAM (user programmable) preset from being changed or erased. It must be set to the "Off" position in order to write an edited preset or copy a selected preset to a new location. It must also be "Off" in order to use the system utility command to initialize the RAM presets.

Use the MIDI System Exclusive Receive Enable switch (parameter 51) to protect preset memory against being unintentionally changed by incoming dumps.

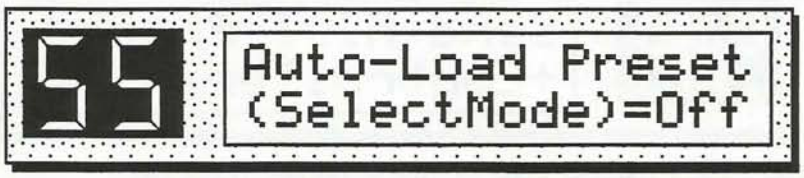
53 — MIDI Prog Change MasterSwitch Range: Off or On

This is the main program change receive switch. If it is set to "Off," all program change messages are ignored regardless of how the individual units are set up. If it is "On," the Range in the individual unit MIDI setups will determine whether or not program changes are recognized.

54 — Parameter Wrap Feature   Range: Off or On

In the "Off" position, when the **Left Arrow** is pressed, the display will stop at the lowest parameter. Likewise, when the **Right Arrow** is pressed, it will stop at the highest parameter.

When the switch is in the "On" position and you are on the highest numbered parameter, pressing the **Right Arrow** button will wrap you around to parameter 00. Likewise, to get to the highest number parameter from parameter 00, press the **Left Arrow** button.



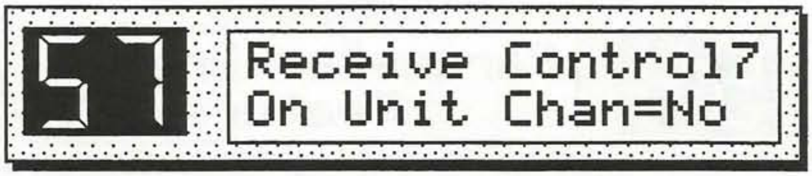
55 — Auto-Load Preset   Range: Off or On

In Select mode, when the Auto-load Preset parameter is set to "On," presets are automatically loaded on second after being selected with the **Data Entry Knob**, and without having to press the **Select** button. This allows you to select effect presets quickly.

When this switch is set to "Off," you must press the **Select** button to actually load the displayed preset. For example, in a live context, a sound engineer may pre-dial an effect preset without activating it. It is now at the engineer's (or your) fingertip to bring in the effect at exactly the correct time simply by pressing the Select button.

56 — Set All 1uPreset Mixes To Wet   Range: No or Yes

When the DP/4 is used in conjunction with a mixing board's auxiliary paths or in any other application where it's desirable to have only the effected (wet) signal appear in the DP/4's audio outputs, set this switch to "Yes." It will automatically set all 1 Unit Preset mix levels to 99 (all wet) when they are selected—or installed in Edit mode—without altering the actual values saved in your presets. When you alternate between using the DP/4 for mixing and as a dedicated instrument effects box, this mix override switch saves you from laborious reprogramming of all of your preset mix Range.



57 — Receive Control7 On Unit Chan Range: No or Yes

When this switch is set to "Yes," the DP/4 will listen to MIDI Controller 7 (volume) messages on unit MIDI channels. This feature allows you to have MIDI controlled effects mixing. This is a "smart" parameter and will only control the volumes of units which are not feeding other units. You may need to adjust the Modulation Response Rate parameter (system parameter 61) for best results.

58 — Send MIDI PrgChg +Controllers Range: No or Yes

When this switch is set to "Yes," the DP/4 will act as a MIDI controller source by generating MIDI messages. These messages are sent out via the DP/4's MIDI Out jack. The DP/4 can send program changes as well as certain controller events, depending on the context.

Every time you select a new preset the DP/4 will send out a program change message corresponding to the preset number. The program change will be sent on the preset's primary unit channel, following the same conventions for receiving events (Refer to the Active MIDI Channel chart in the How the DP/4 Uses MIDI Channels earlier in this section). Program changes are not sent out in response to inbound MIDI program changes.

MIDI Channel Controller messages sent out by the DP/4:

Source	Controller number	Standard Function
CV Pedal	04	Foot control
Footswitch 1	64	Sustain pedal
Footswitch 2	66	Sostenuto pedal

These assignments can not be changed. All controller messages will be send on the DP/4 controller channel. A foot switch event will be transmitted as a controller only when the Footswitch Function (system parameters 45 or 46) is set to "DP/4 Controller."

59 — Show 100 Config Presets   Range: No or Yes

Since the DP/4 is a very complex unit, it is potentially confusing to first time users. Selecting a config preset can radically reconfigure the system, which may not be what the novice user expected while exploring the available presets, so the DP/4 is shipped with most of the config presets hidden away to make it easier for you to get started. Only a few basic config presets are available, and these represent templates which set the unit up for the most common applications.

When this parameter is set to "No" (which is the default) in Select mode, the DP/4 allows you to select the basic config templates (in preset locations 50-61). Once a config preset is selected (by pressing the Select button), the DP/4 automatically takes you to Unit A for selecting presets of that type of configuration.

In Select mode, if you want to bypass all units and not change the config preset, press the **Config** button until the bypass LEDs are lit. If you do not turn the **Data Entry Knob** (which would select a new config), after a brief interval, the DP/4 will automatically put you on Unit A, taking you out of the Select Config mode.

> **Note:** When "Show 100 Config Presets" is set to No, the Auto-Load Preset parameter (#55) is inoperative when selecting a Config Preset. This prevents you from accidentally selecting unwanted configs.

Once you feel comfortable with the DP/4, and you've taken the time to understand the architecture by reading the manual, you are ready to reveal the rest of the config presets. Set the "Show 100 Config Presets" parameter to "Yes." Now, in Select mode, you may still select basic configs (in preset locations 50-61), but you can also select ROM config presets 62-99 and user programmable RAM config presets 00-49.

> **Note:** When this parameter is set to "Yes," the DP/4 will not automatically send you to Unit A when you've selected a config. You must press a **Unit** button before you can select that type of preset.

60 — Data Entry Knob Response   Range: Fast, Normal, Slow

This parameter controls the acceleration rate of the **Data Entry Knob** when spinning quickly through parameter values. The default setting is "Normal." The "Slow" setting will reduce the size of the jumps taken by the values as the knob is rotated quickly, whereas the "Fast" setting will make the jumps larger.

61 — Modulation Response Rate Range: 01 to 30

This parameter controls the rate at which any modulation is applied to the modulation destinations in the DP/4. As parameters are modulated, their values are changed. This parameter controls how quickly the parameters are changed by setting the size of the change. A setting of 01 would have the slowest response with the finest resolution (small changes), whereas a setting of 30 is fast (larger changes), but the modulation is not as smooth.

62 — Use Alternate ROM Presets Range: No or Yes

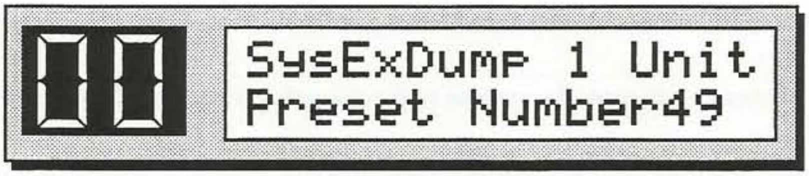
Allows you to replace the original ROM presets (50 through 99) with an alternate ROM preset bank. The alternate ROM bank is loaded into RAM when the DP/4 is reinitialized. This parameter defaults to "No" at reset.

63 — Operating System Version

This read-only page shows the version number of the operating system EPROMs installed in the unit.

System Exclusive Dump

Push the **Write•Copy** button at any time while in System mode to engage the system exclusive dump facility.



This two parameter screen allows you to send various kinds of MIDI System Exclusive dump messages from the DP/4.

You can send dumps containing single presets or preset banks of all four types (1, 2, 4 Unit and Config). You can also save system and MIDI parameters (such as preset maps and user preference switch Range).

The first parameter allows you to select which type of preset you want to dump (1, 2, 4 Unit and Config), as well as allowing you to dump all preset banks, all system global parameters and even all user data contained within the DP/4 (all preset banks and system global parameters).

The second parameter is only available when the first parameter is 1, 2, 4 Unit or Config. It allows you to select an individual preset number (00 through 49), or when you scroll past 49, you can select "Bank" which will dump the entire bank of the selected preset type.

Once the screen shows you what you want, press **Write•Copy** once more to start MIDI transmission of data. You can also press **Cancel•Undo** to exit this screen without sending any data.

Notes

1. At entry, the dump type defaults to the single preset belonging to the currently active unit, whose preset type and number are displayed.

2. ROM presets and ROM preset banks may not be dumped from the front panel. System Exclusive Dump Request commands are provided for external devices which need to extract those banks. Refer to the System Exclusive documentation for more details.

3. System Exclusive message reception is "automatic" and does not have to be enabled by any other action than making sure that System Exclusive reception is enabled and that the ID number setting matches the ID embedded in the dump to be received (see System•MIDI parameters 50 and 51). A confirmation message is displayed to indicate what type of dump has been received when the dump reception is complete, or an error message if there is a problem with the incoming data.

Refer to *Section 7 — Storage* for more information on using System Exclusive messages with the DP/4.

System Utility Functions

The DP/4 has some useful utility functions that are initiated using special button combinations. These include:

Soft Reset

The DP/4 allows you to reset the system without erasing the internal memory. To accomplish this:

1. While holding down the **System•MIDI** button,
2. Press the **Unit A** button.

The system will automatically reset itself as if the power switch had been turned off and back on, with the advantage of not unnecessarily stressing the internal components. The data in the units should not be affected by this procedure, and the system should come back on with the same effects loaded. After the DP/4 has been reset, it defaults to Select mode.

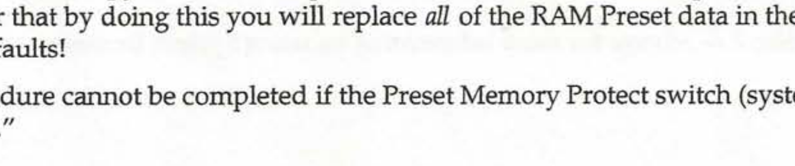
Initializing the RAM Presets

If you want to restore the factory default presets into *all* of the 200 internal memory (RAM) preset locations, there is a convenient command that does so without affecting the System parameter Range.

> WARNING! THIS PROCESS WILL ERASE ALL RAM PRESETS! The 200 User Presets in the internal memory (RAM) are automatically loaded with the factory defaults by this procedure. Make sure that you have saved any RAM presets that you wish to keep by using the SysEx Dump commands.

To initialize the RAM presets:

1. While holding down the **System•MIDI** button,
2. Press the **Unit B** button. The following screen will appear:



Press the **Cancel•Undo** button to quit *without* initializing the presets.

Press the **Write•Copy** button at this point to initialize all of the RAM presets in the DP/4. Remember that by doing this you will replace *all* of the RAM Preset data in the DP/4 with the factory defaults!

This procedure cannot be completed if the Preset Memory Protect switch (system parameter 52) is set to "On."

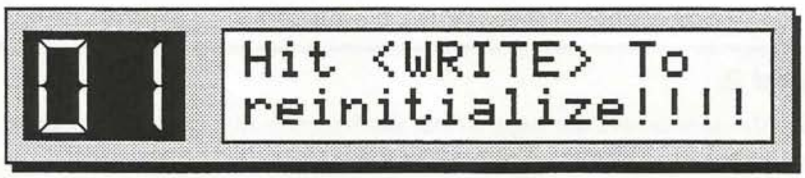
Reinitializing the DP/4

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 **Unit B** button.
3. Then use the **Right Arrow** button to scroll to this page:



Press the **Cancel•Undo** button to quit *without* reinitializing the system.

Press the **Write•Copy** button at this point 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 problems you are encountering, then contact an Authorized ENSONIQ Repair Station.

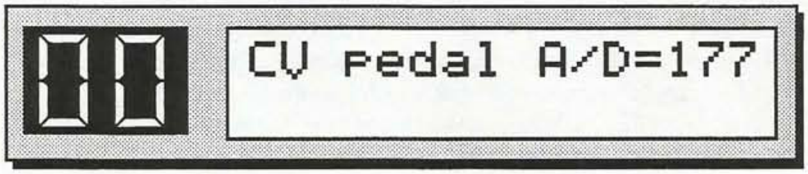
Note: In the unlikely event of a system malfunction, you can save your entire set-up (all Presdt Banks and System parameters) with a System Exclusive dump by pressing the Write button. This will help you restore all of the user-defined parameters. For more information about System Exclusive dumps, see *Section 6 — System•MIDI Mode*.

System Diagnostic Parameters

The DP/4 has a number of diagnostic parameters. To access them:

1. While holding down the **System•MIDI** button,
2. Press the **Unit C** button.
3. Use the **Left** and **Right Arrow** buttons to scroll through the following parameters:

CV Pedal

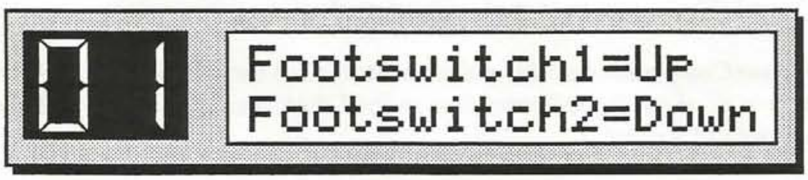


This read-only parameter displays the current control voltage reading of the CV Pedal input's 8 bit analog-to-digital converter. With a CVP-1 pedal connected, the typical range is 000 to 177, with the fully up position showing 000 and fully down showing approximately 177.

If nothing is plugged into the CV Pedal jack, the displayed value should be 255.

Foot Switch 1 and 2

These read-only parameters are used to test the functionality of the foot switch circuitry of the DP/4. They show the current position of Foot Switches 1 and 2.



- If a single foot switch with a mono plug (such as the SW-2) is connected, the display will show "Footswitch1=Up" when the footswitch is *not* depressed, and "Footswitch1=Down" when the foot switch is held down. Foot switch 2 is disabled and should always indicate the "Down" position (as shown above).
- If a dual foot switch with a stereo plug (such as the SW-10) is connected, the display will show "Footswitch1=Up/Down" for the left foot switch controller, and "Footswitch2=Up/Down" for the right foot switch controller.
- If nothing is plugged into the Foot Switch jack, the display should always show "Footswitch1=Up" and "Footswitch2=Up."

> **Important!** There are a number of additional service and diagnostic parameters in the DP/4 that SHOULD NOT BE ALTERED. If altered, they could cause the DP/4 to reinitialize (erasing the RAM presets), sever volume changes could result that may damage your equipment/hearing, or they could affect the internal circuitry making the DP/4 inoperative.

Section 7 — Storage

The storage functions on the DP/4 in Edit mode 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.

In addition you can manually transcribe preset information using the provided Preset Parameter Worksheet.

Internal Storage

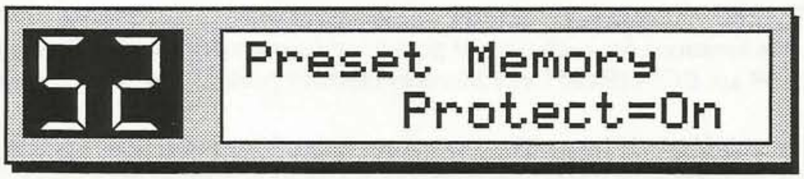
The Preset Memory Protect Switch

Before you can copy or write presets the Preset Memory Protect switch needs to 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." The DP/4 defaults to this setting so that you don't accidentally erase any previously saved presets.

To set the Preset Memory Protect Switch to the "Off" position:

1. Press the **System•MIDI** button, then
2. using the **Left** and **Right Arrow** buttons, scroll until the display shows:



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 the Edit mode. Your newly edited preset should still be intact and ready to save.

Tip: There is a quick way to get to this page. The System•MIDI parameters are divided into sub-groups. By pressing the **System•MIDI** button several times, you can quickly scroll through these sub-groups. Parameter 52 is the first page of one of these sub-groups.

Saving Presets

Presets are written (saved) using a two-step procedure:

1. Select the location where you wish to write the preset ; and
2. Edit the preset name.

After you've created a preset, make sure that the DP/4 is in Edit mode (the Edit LED should be on). If it isn't, press the Edit button. Press the **Write•Copy** 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 particular 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•Copy.** The rules which govern this are the same as in Select mode. You can write what you can select in the current config.

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 selected.

- Press **Cancel•Undo** if you wish 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 wish to save a config preset.

Advanced Feature

Before selecting the destination location for your preset, you may press one of the unit buttons (**A, B, C, 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•Copy** 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 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 cannot be saved this way.

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

Once you have selected the internal location into which the preset will be written (or saved), you are given the opportunity to edit the name of the new preset.

- Press the **Write•Copy** 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.

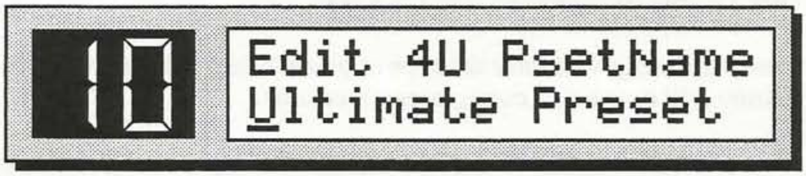
```
Edit Config Name
```

This shows that you are naming a Config preset.

Naming Presets

The name that appears is usually the name of the last selected preset which loaded an algorithm into the active unit. If the name does not appear to match the type of preset that you are saving, you may edit the name to better describe the preset that you are saving.

At this point you can name the preset that you are intending to save. The bottom line of the display offers you 16 spaces to create your own preset name. The display looks like this:



The **Left** and **Right Arrow** buttons will move the cursor left and right, and the **Data Entry Knob** will change the alpha-numeric characters at the current cursor position.

Tip: There is a quick way to select and scroll through the alphanumeric characters. While on this page, the **Unit** buttons will act as shortcuts, as follows:

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

Once you have named your preset, you must confirm that the name and the destination you have chosen are correct and that you wish to complete the saving process.

- Press the **Cancel•Undo** if you wish to return to the Edit Preset Name page or quit from the writing procedure.

- Press the **Write•Copy** button 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 52) back to the "On" position if you wish to eliminate any risk of accidentally erasing or changing your new preset.

Copying Presets

The DP/4 can also copy presets to other RAM locations. To do this:

1. If the Select LED is not on, press the **Select** button.
2. Press the **Write•Copy** button. The top line of the display shows one of four possible screens:

Copy to 1U Pset	This shows that you are copying a 1 Unit preset.
Copy to 4U Pset	This shows that you are copying a 4 Unit preset.
Copy to 2U Pset	This shows that you are copying a 2 Unit preset.
Copy to Config	This shows that you are copying a Config preset.

The screen that appears and the type of preset that will be copied is determined by the current config and the unit that is active when you press **Write•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 the **Cancel•Undo** 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 **Write•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 above).
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).

Sending DP/4 Data Out via MIDI System Exclusive Dump

Push the **Write•Copy** button at any time while in System mode to engage the system exclusive dump utility.

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.

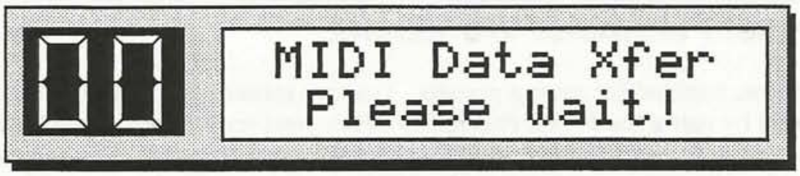
Dump Types:	What's included:
Single Preset	1 RAM preset
Preset Bank	50 RAM presets
System	All system parameters
All Preset Banks	200 RAM presets
All Preset Banks with System Parameters	200 RAM presets and all system params

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•Mode (such as program change-to-preset maps and user preference switch settings). It is best to use the smallest single dump type which contains all of the data you wish to reload at one time.

The 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.

You can press **Cancel•Undo** 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•Copy** 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.



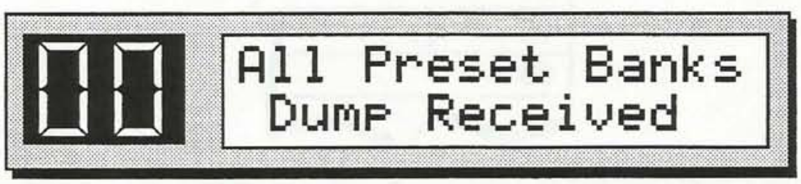
When the dump is complete, the following message will appear for a moment to indicate that the transmission occurred without errors.

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! Remember that the System Exclusive ID number (system parameter 50) 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 50 and 51). The MIDI LED 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.



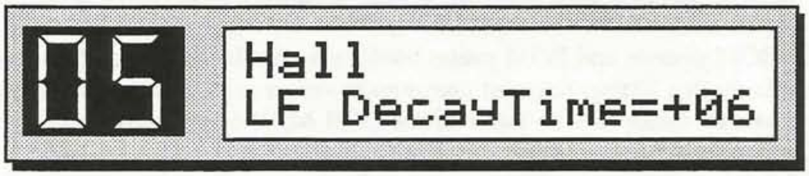
The top line of this message will describe the type of dump received. The preset type and number are shown for single preset dumps. Only the type is shown for preset bank dumps. Dumps containing system parameters will have an additional message which follows the confirmation message to indicate that 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. For more information about error messages , refer to the *DP/4 MIDI System Exclusive Specification.*

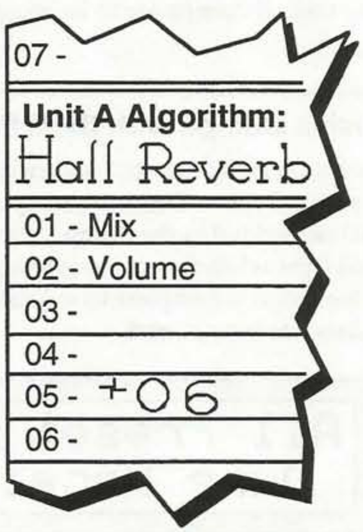
Using the Preset Parameter Worksheet

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.

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 **Left/Right Arrow** 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 **Left/Right Arrow** 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 8 — Applications

This section gives you step by step procedures for a number of the most commonly used DP/4 applications. Working through these applications will help you to get better acquainted with the DP/4 and its features.

Bear in mind that in most of these examples we have illustrated a concept using very specific examples. To get the most out of this section, you should take these applications as starting points and experiment with applying the same concepts to different algorithms, configs, modulators, etc.

Loading a 2 Unit Preset While in a 1 Source Config

Objective: You are using the DP/4 in a 1 Source Config (where you can only select 4 Unit Presets) but you want to load the effects combination from a 2 Unit preset (or a 2 Unit algorithm) into A & B or C & D.

Answer: In Edit mode you can easily select 2 Unit presets (or algorithms) by following these steps:

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 (or 2 Unit algorithm) you desire, wait a moment and it will load automatically.

Saving a 2 Unit Preset While in a 1 Source Config

Objective: You are working in a 1 Source config, and have edited the algorithm parameters in two of the four units to create a sound you really like. Now you want to save that as a 2 Unit preset. But in a 1 Source config, only 4 or 1 Unit presets can be saved.

Answer: 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 **Left Arrow** 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."
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 in *Section 7 — Storage.*
6. 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.

Using a 1 Source Config for 4 Selectable Effects

Objective: You want to "try out" a number of different effects on the same input signal for comparison purposes.

Answer: You can "cue up" four different 1 Unit effects (for example, four different reverbs) and select among them by putting A, B, C, and D all in parallel in a 1 Source config, and bypassing all but the one you want to hear.

1. Set System•MIDI parameter #59 "Show 100 Config Presets," to "No."
2. Press **Select**, then **Config**, and select Config Preset #54, "1 Src: Stereo In."
3. Press **Select**, then **Edit**. Select an effects algorithm for unit A.
4. Press **Unit B**. Select a different effects algorithm for unit B.
5. Press **Unit C** and choose an effects algorithm for unit C.
6. Press **Unit D** and choose an effects algorithm for unit D.
7. Press **Config**, and use the **Left/Right Arrows** and the **Data Entry Knob** to set the config parameters to the following settings:

Param #:	Parameter:	Set to:
01	AB – CD Routing=	AB + CD parallel
02	AB Unit Routing	[A + B] parallel
03	CD Unit Routing	[C + D] parallel
07-10	(b)ypass (k)ill	A=k B=k C=k D=k

8. Now press **Config** to bypass all units.

   This will kill all signal through the DP/4 since we set all four units to (k)ill.
9. Press one of the unit buttons twice to un-bypass it. You will now hear that unit's effect algorithm on the input signal. Bypass that unit and un-bypass another to hear its effect, and so on. This way you can shop around among different algorithms, instantly comparing them. Or you could un-bypass more than one unit at a time to layer algorithms (effects).

Using the DP/4 Song Feature to Chain Presets

Objective: You want to define a series of presets to be selected in a specified order by pressing a Foot Switch.

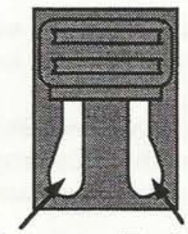
Answer: Using the DP/4 Song feature, you can specify up to 20 *Songs*, groups of 5 presets that can be selected sequentially with the optional SW-10 (or SW-2) foot switch. We recommend the SW-10 Dual Foot Switch for this purpose, as its two pedals let you change from song to song using one pedal and select steps within a song using the other.

- Press **System•MIDI**, until parameter #45 is showing on the display. Set the Foot Switch 1 and 2 parameters to the following values:

Param #:	Parameter:	Set to:
45	DP4 Footswitch1=	Song Preset Up
46	DP4 Footswitch2=	Increment song

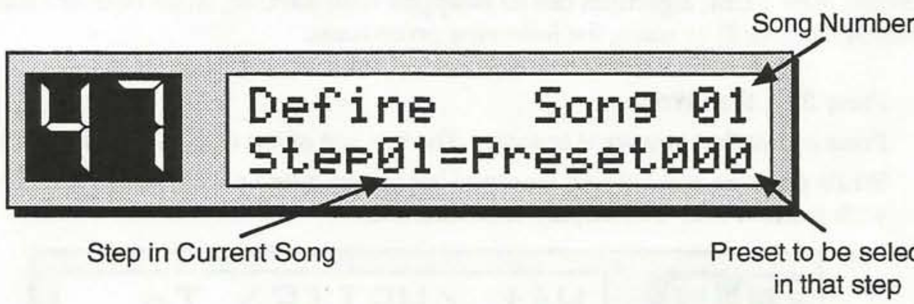
This sets Foot Sitch 1 (on the right) to step sequentially between the five presets defined in the current song.

Pressing Foot Switch 2 (on the left) will advance to the next song.



Note: If you are using a foot switch manufactured by another company, there is a possibility that the wires inside the foot switch may be reversed. This would make the DP/4 recognize Foot Switch 1 as left, and Foot Switch 2 as right.

- Press the **Right Arrow** button to get to the Song Editor (params #47-49). The display shows:



- Use the three parameters on the Song Editor screen to define as many 5-step songs as you will need (see the discussion in Section 6 for more details on the Song Editor). When you are done editing, go back to parameter #47 and reset it to Song #01.

- Now, press Foot Switch 1. Each time you press it, the DP/4 will select the preset that is the next step in the song, returning to step 01 after step 05. Pressing Foot Switch 2 will advance to the next song.

Using a Foot Switch to Alternate Between Two Presets

Objective: You want to dedicate a foot switch to simply alternate between two presets. For, example, a guitarist might want to switch between a lead tone and a rhythm tone by pressing a single foot switch.

Answer: This is another use for the DP/4 Song feature. In this case we will need just one song, and a feature that lets you limit a song to less than five steps.

1. Press **System•MIDI**, until parameter #45 is showing on the display. Set the Foot Switch 1 parameter to "DP4 Footswitch1= Song Preset Up."
2. Press the **Right Arrow** button twice to get to the Song Editor (params #47-49). Select Song 01.
3. Let's suppose you want to alternate between Preset 25 and Preset 42. Set parameter #48 to Step 01, and set the preset (param #49) to "Step01=Preset 025."
4. Press the **Left Arrow** button to go back to the Step, select Step 02, and set the preset to "Step02=Preset 042."
5. Scroll back to the Step, select Step 03, and set it to "Step03=GotoStep1." You will find this choice at the top of the list, after Preset 99.

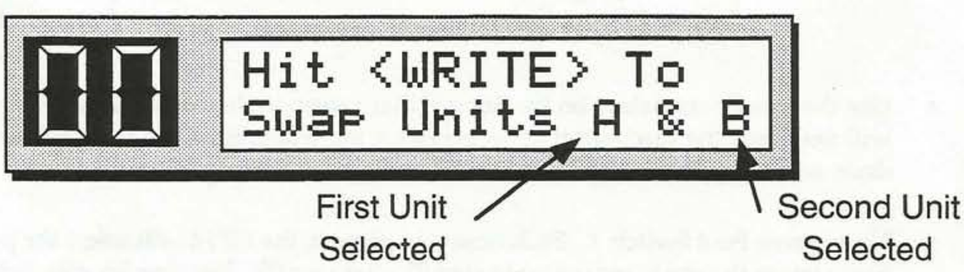
Now, each time you press Foot Switch 1, it will toggle between the two presets you defined. The other Foot Switch can be programmed for some other purpose, if you wish. Bear in mind that the "GotoStep1" option could also be entered as Step 04 or Step 05 to create songs of 3 or 4 steps instead of 2.

Swapping 1 Unit Presets

Objective: You have edited the algorithm parameters and created a sound you really like, but you want to swap the algorithms around between the units.

Answer: Any 1 Unit algorithm can be swapped with another, when both are loaded into either Units A, B, C, or D by using the following procedures:

1. Press **Edit**, then **Write**.
2. Press a Unit 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!"

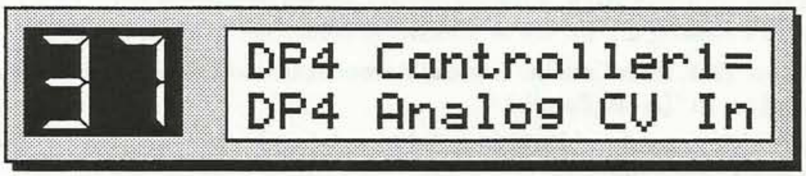
You can use this procedure any time you want to swap a 1 Unit Preset with another.

Modulating Effects Parameters with the CV Pedal

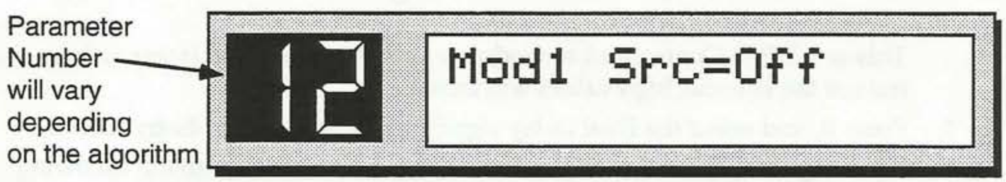
Objective: You want to use the optional CV Pedal to alter some aspect of an effect algorithm in real time.

Answer: Almost any parameter of any algorithm can be modulated using one of the eight programmable controllers which you select in System•MIDI mode. Let's suppose you want to use the CV Pedal to control the wet/dry mix of the effect in one of the units.

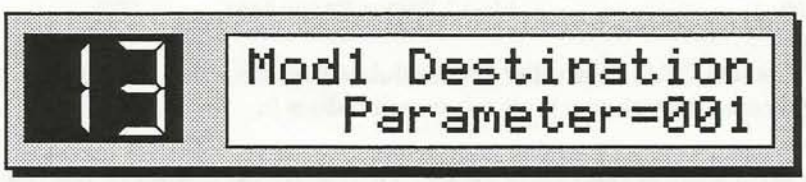
1. Press **System•MIDI** until the display shows parameter #37:



2. Set the "DP4 Controller1" parameter to "DP4 Analog CV In" as shown above (you will use this and the next seven parameters to define the eight controllers the DP/4 will recognize as modulators).
3. Press **Edit**, and select the unit (A, B, C, or D) whose mix you want to modulate using the pedal.
4. Press the **Right Arrow** button until the "Mod 1 Src" parameter is showing on the display:



5. Set the parameter to "Mod 1 Src=Cntrl 1." The lower line of the display shows the name of the controller that has been selected as "Cntrl 1" in System•MIDI mode. It should read "DP4 Analog CV In."
6. Press the **Right Arrow** button one time. The display shows:



7. Here you select which parameter in the algorithm will be modulated by the selected controller. Since Wet/Dry Mix is parameter 01 in *every* algorithm, set it to "Mod1 Destination Parameter=001."
8. The Pedal will now control the mix of the algorithm. If you scroll left to select parameter 01, you can actually see it change as you move the pedal. Remember to write (save) this as a 1 Unit preset if you want to keep it.

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

Crossfading Effects

Objective: You want to crossfade two different algorithms using some controller, such as the optional CV Pedal, MIDI mod wheel, etc.

Answer: You can set up the modulation controllers in a 2 Unit or 4 Unit preset allowing you to crossfade between two different algorithms. Crossfading means that one algorithm is increased as the other is decreased by a controller. In the example below, we will crossfade between a Hall Reverb (Unit A) and a Dual Delay (Unit B) in a 2 Source Config.

1. Press **System•MIDI** until the display shows parameter #37, "DP4 Controller1," and set to "DP4 Analog CV In" (or to whichever controller you want to use).
2. Press **Edit**, then **Config**. Use the **Arrow** buttons to scroll to parameter #01, AB Unit Routing, and set to "[A+B] Parallel."
3. Press **A**, (you should still be in Edit mode) and select the Hall Reverb algorithm using the **Data Entry Knob**.
4. Scroll right to parameter #23, and set the Mod 1 parameters to the following:

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

This sets DP/4 Controller 1 to modulate Unit A's volume in linear fashion — low values will reduce the volume; high values will increase it.

5. Press **B**, and select the Dual Delay algorithm using the **Data Entry Knob**.
6. Scroll right to parameter #13, and set the Mod 1 parameters to the following:

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

This sets DP/4 Controller 1 to modulate Unit B's volume in reverse fashion — low values will increase the volume; high values will reduce it.

Now send a signal (play) into the DP/4 and move the controller you selected (be it a CV Pedal or an external MIDI Controller) and you should hear the two effects algorithms crossfade as you move the controller.

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

Note: If you are using a MIDI controller for this or any other effects modulation, make sure that "Control Chan" (parameter #35 in System•MIDI mode) is set to the same MIDI channel that you are sending the controller on, and that MIDI is Enabled (parameter #36). Otherwise, the MIDI controller will not be recognized by the DP/4.

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		O Controller 4 (CV Pedal)	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**

00 - Gated Reverb

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Gate Attack
04 — Hold Time
05 — Gate Decay
06 — Release Time
07 — Gate Trigger Threshold
08 — Gated Retrigger Threshold
09 — Gated HF Damping
10 — Gated Diffusion 1
11 — Gated Diffusion 2
12 — Gated Decay Definition
13 — Gated Slapback
14 — Gated Slapback Level
15 — Early Refs (1)
16 — Early Refs (2)
17 — Early Refs (3)
18 — Early Refs (4)
19 — Left/Right Balance
20 — Mod1 Source
21 — Mod1 Destination Parameter
22 — Mod1 Param Range Min
23 — Mod1 Param Range Max
24 — Mod2 Source
25 — Mod2 Destination Parameter
26 — Mod2 Param Range Min
27 — Mod2 Param Range Max

00 - Non Lin Reverb 1
00 - Non Lin Reverb 2
00 - Non Lin Reverb 3

The parameters available for these algorithms are:

01 — Mix
02 — Volume
03 — Envelope Level 1
04 — Envelope Level 2
05 — Envelope Level 3
09 — Envelope Level 4
07 — Envelope Level 5
08 — Envelope Level 6
09 — Envelope Level 7
10 — Envelope Level 8
11 — Envelope Level 9
12 — NonLin HF Damping
13 — NonLin HF Bandwidth
14 — NonLin Diffusion1
15 — NonLin Diffusion2
16 — NonLin Density 1
17 — NonLin Density 2
18 — NonLin Primary Send
19 — Reflection 1 Time
20 — Reflection 1 Send
21 — Reflection 2 Time
22 — Reflection 2 Send
23 — Left/Right Balance
24 — Mod1 Source
25 — Mod1 Destination Parameter
26 — Mod1 Param Range Min
27 — Mod1 Param Range Max
28 — Mod2 Source
29 — Mod2 Destination Parameter
30 — Mod2 Param Range Min
31 — Mod2 Param Range Max

00 - MultiTap Delay

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — MultiTap 1 Time
04 — MultiTap 1 Level
05 — MultiTap 1 Regen
06 — MultiTap 1 Pan
07 — MultiTap 2 Time
08 — MultiTap 2 Level
09 — MultiTap 2 Regen
10 — MultiTap 2 Pan
11 — MultiTap 3 Time
12 — MultiTap 3 Level
13 — MultiTap 3 Regen
14 — MultiTap 3 Pan
15 — MultiTap 4 Time
16 — MultiTap 4 Level
17 — MultiTap 4 Regen
18 — MultiTap 4 Pan
19 — Regen Damping
20 — Mod1 Source
21 — Mod1 Destination Parameter
22 — Mod1 Param Range Min
23 — Mod1 Param Range Max
24 — Mod2 Source
25 — Mod2 Destination Parameter
26 — Mod2 Param Range Min
27 — Mod2 Param Range Max

00 - 3.3 sec DDL 2U

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — 3.3 Sec Delay Time
04 — 3.3 Sec Delay Regen
05 — 3.3 Sec Delay Pan
06 — 3.3 Sec Delay Regen Damping
07 — 3.3 sec Delay Mode
08 — DelaySet
09 — Mod1 Source
10 — Mod1 Destination Parameter
11 — Mod1 Param Range Min
12 — Mod1 Param Range Max
13 — Mod2 Source
14 — Mod2 Destination Parameter
15 — Mod2 Param Range Min
16 — Mod2 Param Range Max

00 - Dual Delay

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Left Input Delay Time
04 — Left Input Delay Time (fine)
05 — Left Input Delay Regen
06 — Left Input Delay Pan
07 — Right Input Delay Time
08 — Right Input Delay Time (fine)
09 — Right Input Delay Regen
10 — Right Input Delay Pan
11 — Dual Delay CrossRegen
12 — Dual Delay Regen Damping
13 — Mod1 Source
14 — Mod1 Destination Parameter
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination Parameter
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

00 - Tempo Delay

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Tempo Delay Time
04 — Internal Clock Tempo
05 — TempoDelay Fine Tune
06 — Tempo Control
07 — Tempo Delay Regen
08 — Tempo Delay Pan
09 — Tempo Delay Regen Damping
10 — Mod1 Source
11 — Mod1 Destination Parameter
12 — Mod1 Param Range Min
13 — Mod1 Param Range Max
14 — Mod2 Source
15 — Mod2 Destination Parameter
16 — Mod2 Param Range Min
17 — Mod2 Param Range Max

00 - EQ-DDL-withLFO

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — DDL+LFO Left Delay Time
04 — DDL+LFO Right Delay Time
05 — DDL+LFO LFO Rate
06 — DDL+LFO LFO Width
07 — Left/Right LFO
08 — DDL+LFO Delay Regen
09 — DDL+LFO Delay Cross Regen
10 — DDL+LFO Regen Damping
11 — DDL+LFO Right Delay Input
12 — DDL+LFO Right Output Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

00 - VCF-Distortion

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

00 - Guitar Amp 1
00 - Guitar Amp 2

The parameters available for these algorithms are:

01 — Mix
02 — Volume
03 — Amp Preamp Gain
04 — Amp Output Level
05 — Amp Tube Bias
06 — Pre-EQ Input Level Trim
07 — Pre-EQ High Pass Cutoff
08 — Pre-EQ Fc
09 — Pre-EQ Gain
10 — Pre-EQ Q
11 — Noise Gate Off Below
12 — Noise Gate On Above
13 — Gate Release Time
14 — Speaker High Pass Cutoff
15 — OutEQ1 Fc
16 — OutEQ1 Gain
17 — OutEQ1 Q
18 — OutEQ2 Fc
19 — OutEQ2 Gain
20 — OutEQ2 Q
21 — Speaker Low Pass Cutoff
22 — Mod1 Source
23 — Mod1 Destination Parameter
24 — Mod1 Param Range Min
25 — Mod1 Param Range Max
26 — Mod2 Source
27 — Mod2 Destination Parameter
28 — Mod2 Param Range Min
29 — Mod2 Param Range Max

00 - Guitar Amp 3

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — AmpPreamp Gain
04 — Amp Output Level
05 — PreEQ Input Level Trim
06 — Pre-EQ Fc
07 — Pre-EQ Gain
08 — Pre-EQ Q
09 — ExpndRatio
10 — Threshold
11 — Gain Change
12 — Noise Gate Off Below
13 — Noise Gate On Above
14 — Gate Release Time
15 — Speaker High Pass Cutoff
16 — OutEQ1 Fc
17 — OutEQ1 Gain
18 — OutEQ1 Q
19 — OutEQ2 Fc
20 — OutEQ2 Gain
21 — OutEQ2 Q
22 — Speaker Low Pass Cutoff
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

00 - Speaker Cabinet

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Speaker Output Gain
04 — Mod1 Source
05 — Mod1 Destination Parameter
06 — Mod1 Param Range Min
07 — Mod1 Param Range Max
08 — Mod2 Source
09 — Mod2 Destination Parameter
10 — Mod2 Param Range Min
11 — Mod2 Param Range Max

00 - Tunable Speaker

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Mid1 Fc
04 — Mid1 Gain
05 — Mid1 Q
06 — Mid2 Fc
07 — Mid2 Gain
08 — Mid2 Q
09 — Mid3 Fc
10 — Mid3 Gain
11 — Mid3 Q
12 — Speaker Input Attenuation
13 — Speaker Output Gain
14 — Mod1 Source
15 — Mod1 Destination Parameter
16 — Mod1 Param Range Min
17 — Mod1 Param Range Max
18 — Mod2 Source
19 — Mod2 Destination Parameter
20 — Mod2 Param Range Min
21 — Mod2 Param Range Max

00 - Rotating Speaker

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Rotating Speaker Slow Speed
04 — Rotating Speaker Fast Speed
05 — Rotating Speaker Speed
06 — Rotating Speaker Inertia
07 — Distortion Level In
08 — Distortion Level Out
09 — Rotating Speaker Distortion Tone
10 — Rotating Speaker Stereo Spread
11 — Mod1 Source
12 — Mod1 Destination Parameter
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination Parameter
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

00 - EQ-Chorus-DDL

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Chorus LFO Rate
04 — Chorus LFO Width
05 — Chorus Center
06 — Left/Right LFO
07 — Chorus Left Delay Time
08 — Chorus Right Delay Time
09 — Chorus Delay Regen
10 — Chorus Left Echo Time
11 — Chorus Right Echo Time
12 — Chorus Echo Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

00 - Dual Delay

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Left Input Delay Time
04 — Left Input Delay Time (fine)
05 — Left Input Delay Regen
06 — Left Input Delay Pan
07 — Right Input Delay Time
08 — Right Input Delay Time (fine)
09 — Right Input Delay Regen
10 — Right Input Delay Pan
11 — Dual Delay CrossRegen
12 — Dual Delay Regen Damping
13 — Mod1 Source
14 — Mod1 Destination Parameter
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination Parameter
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

00 - Tempo Delay

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Tempo Delay Time
04 — Internal Clock Tempo
05 — TempoDelay Fine Tune
06 — Tempo Control
07 — Tempo Delay Regen
08 — Tempo Delay Pan
09 — Tempo Delay Regen Damping
10 — Mod1 Source
11 — Mod1 Destination Parameter
12 — Mod1 Param Range Min
13 — Mod1 Param Range Max
14 — Mod2 Source
15 — Mod2 Destination Parameter
16 — Mod2 Param Range Min
17 — Mod2 Param Range Max

00 - EQ-DDL-withLFO

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — DDL+LFO Left Delay Time
04 — DDL+LFO Right Delay Time
05 — DDL+LFO LFO Rate
06 — DDL+LFO LFO Width
07 — Left/Right LFO
08 — DDL+LFO Delay Regen
09 — DDL+LFO Delay Cross Regen
10 — DDL+LFO Regen Damping
11 — DDL+LFO Right Delay Input
12 — DDL+LFO Right Output Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

00 - VCF-Distortion

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

00 - Guitar Amp 1
00 - Guitar Amp 2

The parameters available for these algorithms are:

01 — Mix
02 — Volume
03 — Amp Preamp Gain
04 — Amp Output Level
05 — Amp Tube Bias
06 — Pre-EQ Input Level Trim
07 — Pre-EQ High Pass Cutoff
08 — Pre-EQ Fc
09 — Pre-EQ Gain
10 — Pre-EQ Q
11 — Noise Gate Off Below
12 — Noise Gate On Above
13 — Gate Release Time
14 — Speaker High Pass Cutoff
15 — OutEQ1 Fc
16 — OutEQ1 Gain
17 — OutEQ1 Q
18 — OutEQ2 Fc
19 — OutEQ2 Gain
20 — OutEQ2 Q
21 — Speaker Low Pass Cutoff
22 — Mod1 Source
23 — Mod1 Destination Parameter
24 — Mod1 Param Range Min
25 — Mod1 Param Range Max
26 — Mod2 Source
27 — Mod2 Destination Parameter
28 — Mod2 Param Range Min
29 — Mod2 Param Range Max

00 - EQ-Vibrato-DDL

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Vibrato Rate
04 — Vibrato Width
05 — Left/Right LFO
06 — Vibrato Sample & Hold Rate
07 — Vibrato Left Delay Time
08 — Vibrato Right Delay Time
09 — Vibrato Delay Regen
10 — Vibrato Left Echo Time
11 — Vibrato Right Echo Time
12 — Vibrato Echo Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

00 - EQ-Panner-DDL

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Panner Rate
04 — Panner Width
05 — Left/Right LFO
06 — Panner Sample & Hold Rate
07 — Panner Left Delay Time
08 — Panner Right Delay Time
09 — Panner Delay Regen
10 — Panner Left Echo Time
11 — Panner Right Echo Time
12 — Panner Echo Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

00 - EQ-Flanger-DDL

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Flanger LFO Rate
04 — Flanger LFO Width
05 — Flanger Center
06 — Flanger Feedback
07 — Flanger Notch Depth
08 — Left/Right LFO
09 — Flanger Sample & Hold Rate
10 — Flanger Left Delay Time
11 — Flanger Right Delay Time
12 — Flanger Delay Feedback
13 — Flanger Left Echo Time
14 — Flanger Right Echo Time
15 — Flanger Echo Level
16 — Bass Fc
17 — EQ Gain
18 — Treble Fc
19 — EQ Gain
20 — EQ Input Level Trim
21 — Mod1 Source
22 — Mod1 Destination Parameter
23 — Mod1 Param Range Min
24 — Mod1 Param Range Max
25 — Mod2 Source
26 — Mod2 Destination Parameter
27 — Mod2 Param Range Min
28 — Mod2 Param Range Max

00 - EQ-Tremolo-DDL

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Tremolo Rate
04 — Tremolo Depth
05 — Left/Right LFO
06 — Tremolo Sample & Hold Rate
07 — Tremolo Left Delay Time
08 — Tremolo Right Delay Time
09 — Tremolo Delay Regen
10 — Tremolo Left Echo Time
11 — Tremolo Right Echo Time
12 — Tremolo Echo Level
13 — Bass Fc
14 — Bass EQ Gain
15 — Treble Fc
16 — Treble EQ Gain
17 — EQ Input Level Trim
18 — Mod1 Source
19 — Mod1 Destination Parameter
20 — Mod1 Param Range Min
21 — Mod1 Param Range Max
22 — Mod2 Source
23 — Mod2 Destination Parameter
24 — Mod2 Param Range Min
25 — Mod2 Param Range Max

00 - Phaser-DDL

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Phaser LFO Rate
04 — Phaser LFO Width
05 — Phaser Center
06 — Phaser Feedback
07 — Phaser Notch Depth
08 — Left/Right LFO
09 — Phaser Sample & Hold Rate
10 — Phaser Left Delay Time
11 — Phaser Right Delay Time
12 — Phaser Delay Feedback
13 — Mod1 Source
14 — Mod1 Destination Parameter
15 — Mod1 Param Range Min
16 — Mod1 Param Range Max
17 — Mod2 Source
18 — Mod2 Destination Parameter
19 — Mod2 Param Range Min
20 — Mod2 Param Range Max

00 - 8 Voice Chorus

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — 8V Chorus LFO Rate
04 — 8V Chorus LFO Width
05 — 8V Chorus Stereo Spread
06 — 8V Chorus Regen
07 — 8V Chorus Left Regen Time
08 — 8V Chorus Right Regen Time
09 — 8V Chorus Delay Regen
10 — Mod1 Source
11 — Mod1 Destination Parameter
12 — Mod1 Param Range Min
13 — Mod1 Param Range Max
14 — Mod2 Source
15 — Mod2 Destination Parameter
16 — Mod2 Param Range Min
17 — Mod2 Param Range Max

00 - Flanger

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Flanger LFO Rate
04 — Flanger LFO Width
05 — Flanger Center
06 — Flanger Regen
07 — Mod1 Source
08 — Mod1 Destination Parameter
09 — Mod1 Param Range Min
10 — Mod1 Param Range Max
11 — Mod2 Source
12 — Mod2 Destination Parameter
13 — Mod2 Param Range Min
14 — Mod2 Param Range Max

00 - Pitch Shifter
00 - Pitch Shift 2U

The parameters available for these algorithms are:

01 — Mix
02 — Volume
03 — PitchShifter Vc 1 Semi
04 — PitchShifter Vc 1 Fine
05 — PitchShifter Vc 1 Level
06 — PitchShifter Vc 1 Pan
07 — PitchShifter Vc 2 Semi
08 — PitchShifter Vc 2 Fine
09 — PitchShifter Vc 2 Level
10 — PitchShifter Vc 2 Pan
11 — Delay vs Quality (Pitch Shifter 1U Only)
— PitchShifter LFO Rate
— PitchShifter LFO Width
— Mod1 Source
— Mod1 Destination Parameter
— Mod1 Param Range Min
— Mod1 Param Range Max
— Mod2 Source
— Mod2 Destination Parameter
— Mod2 Param Range Min
— Mod2 Param Range Max

00 - PitchShift-DDL

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — PitchShift Vc 1 Semi
04 — PitchShift Vc 1 Fine
05 — PitchShift Vc 1 Level
06 — PitchShifter Vc 1 Pan
07 — PitchShift Vc 2 Semi
08 — PitchShift Vc 2 Fine
09 — PitchShift Vc 2 Level
10 — PitchShifter Vc 2 Pan
11 — PitchShift Dry Level to DDL
12 — PitchShift Left Delay Time
13 — PitchShift Right Delay Time
14 — PitchShift Delay Mix
15 — PitchShift Delay Regen
16 — Mod1 Source
17 — Mod1 Destination Parameter
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination Parameter
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

00 - FastPitchShift

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — PitchShifter Vc 1 Fine
04 — PitchShifter Vc 1 Level
05 — PitchShifter Vc 1 Pan
06 — PitchShifter Vc 2 Fine
07 — PitchShifter Vc 2 Level
08 — PitchShifter Vc 2 Pan
09 — PitchShifter LFO Rate
10 — PitchShifter LFO Width
11 — Mod1 Source
12 — Mod1 Destination Parameter
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination Parameter
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

00 - EQ-Compressor

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Compressor Gain
04 — Compressor Ratio
05 — Compressor Threshold
06 — GainChange
07 — Comp Attack
08 — Comp Release
09 — Comp Noise Gate Off Below
10 — Comp Noise Gate On Above
11 — Gate Release Time
12 — Bass Fc
13 — Bass EQ Gain
14 — Treble Fc
15 — Treble EQ Gain
16 — EQ Input Level Trim
17 — Mod1 Source
18 — Mod1 Destination Parameter
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination Parameter
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

00 - Expander

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Exp Ratio
04 — Threshold
05 — Gain Change
06 — Exp Attack
07 — Exp Release
08 — Exp Gate Hold Time
09 — Sidechain EQ Gain
10 — HighPass Fc
11 — LowPass Fc
12 — Trigger Mask
13 — TriggeTime
14 — Trig Mask Lower Threshold
15 — Expander Output Gain
16 — Mod1 Source
17 — Mod1 Destination Parameter
18 — Mod1 Param Range Min
19 — Mod1 Param Range Max
20 — Mod2 Source
21 — Mod2 Destination Parameter
22 — Mod2 Param Range Min
23 — Mod2 Param Range Max

00 - Keyed Expander

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Exp Ratio
04 — Threshold
05 — Gain Change
06 — Exp Attack
07 — Exp Release
08 — Exp Gate Hold Time
09 — Sidechain EQ Gain
10 — HighPass Fc
11 — LowPass Fc
12 — Trigger Mask
13 — TriggeTime
14 — Trigger Mask Threshold
15 — Expander Output Mix
16 — Expander Output Gain
17 — Mod1 Source
18 — Mod1 Destination Parameter
19 — Mod1 Param Range Min
20 — Mod1 Param Range Max
21 — Mod2 Source
22 — Mod2 Destination Parameter
23 — Mod2 Param Range Min
24 — Mod2 Param Range Max

00 - InversExpander

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Expnd Ratio
04 — Threshold
05 — Gain Change
06 — Exp Attack
07 — Exp Release
08 — Exp Noise Gate Off Below
09 — Comp Noise Gate On Above
10 — Bass Fc
11 — Bass EQ Gain
12 — Treble Fc
13 — Treble EQ Gain
14 — EQ Input Level Trim
15 — Mod1 Source
16 — Mod1 Destination
17 — Mod1 Param Range Min
18 — Mod1 Param Range Max
19 — Mod2 Source
20 — Mod2 Destination
21 — Mod2 Param Range Min
22 — Mod2 Param Range Max

00 - De-esser

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — De-esser Output Gain
04 — Comp Ratio
05 — Threshold
06 — Gain Change
07 — Comp Attack
08 — Comp Release
09 — Noise Gate Off Below
10 — Noise Gate On Above
11 — Sidechain EQ HighPass Fc
12 — Bass Fc
13 — Bass Gain (loShv)
14 — Mid1 Fc
15 — Mid1 Gain
16 — Mid1 Q
17 — Mid2 Fc
18 — Mid2 Gain
19 — Mid2 Q
20 — Treble Fc
21 — Treble Gain (HiShv)
22 — Sidechain EQ Input Trim
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

00 - Ducker / Gate

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Ducker Output Gain
04 — Ducker Output Mix
05 — Comp Ratio
06 — Threshold
07 — Gain Change
08 — Comp Attack
09 — Comp Release
10 — Noise Gate Off Below
11 — Noise Gate On Above
12 — Bass Fc
13 — Bass Gain (loShv)
14 — Mid1 Fc
15 — Mid1 Gain
16 — Mid1 Q
17 — Mid2 Fc
18 — Mid2 Gain
19 — Mid2 Q
20 — Treble Fc
21 — Treble Gain (HiShv)
22 — Side Chain EQ Input Trim
23 — Mod1 Source
24 — Mod1 Destination Parameter
25 — Mod1 Param Range Min
26 — Mod1 Param Range Max
27 — Mod2 Source
28 — Mod2 Destination Parameter
29 — Mod2 Param Range Min
30 — Mod2 Param Range Max

00 - Rumble Filter

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — HighPass Fc
04 — LowPass Fc
05 — Filter Gain
06 — Mod1 Source
07 — Mod1 Destination Parameter
08 — Mod1 Param Range Min
09 — Mod1 Param Range Max
10 — Mod2 Source
11 — Mod2 Destination Parameter
12 — Mod2 Param Range Min
13 — Mod2 Param Range Max

00 - Parametric EQ

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Bass Fc
04 — Bass Gain (loShv)
05 — Mid1 Fc
06 — Mid1 Gain
07 — Mid1 Q
08 — Mid2 Fc
09 — Mid2 Gain
10 — Mid2 Q
11 — Treble Fc
12 — Treble Gain (HiShv)
13 — EQ Input Level Attenuation
14 — Mod1 Source
15 — Mod1 Destination Parameter
16 — Mod1 Param Range Min
17 — Mod1 Param Range Max
18 — Mod2 Source
19 — Mod2 Destination Parameter
20 — Mod2 Param Range Min
21 — Mod2 Param Range Max

00 - 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

00 - Sine/Noise Gen

The parameters available for this algorithm are:

01 — Mix
02 — Volume
03 — Sine/Noise Gen Sine Freq
04 — Sine/Noise Gen Balance
05 — Noise Filters - Low Pass Fc
06 — Bass Fc
07 — EQ Gain
08 — Treble Fc
09 — EQ Gain
10 — EQ Input Level Trim
11 — Mod1 Source
12 — Mod1 Destination Parameter
13 — Mod1 Param Range Min
14 — Mod1 Param Range Max
15 — Mod2 Source
16 — Mod2 Destination Parameter
17 — Mod2 Param Range Min
18 — Mod2 Param Range Max

00 - Vocoder Low
00 - Vocoder Mid 1
00 - Vocoder Mid 2
00 - Vocoder High

The parameters available for these algorithms 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

00 - No Effect

The parameters available for this are:

01 — Mix
02 — Volume
03 — Mod1 Source
04 — Mod1 Destination Parameter
05 — Mod1 Param Range Min
06 — Mod1 Param Range Max
07 — Mod2 Source
08 — Mod2 Destination Parameter
09 — Mod2 Param Range Min
10 — Mod2 Param Range Max

Edit Config Parameters

1 Source Config

00 — 1 Source Config
01 — AB - CD Routing
02 — AB Unit Routing
03 — CD Unit Routing
04 — AB (Config Dependent)
05 — CD (Config Dependent)
06 — AB Input Select
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 Unit Routing
02 — CD Unit Routing
03 — AB Input Select
04 — CD Input Select
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 — AB Output Select
02 — CD Unit Routing
03 — CD Input Select
04 — CD (Config Dependent)
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
46 — DP/4 Foot Switch 2
47 — Define Song
48 — Define Step
49 — Define Preset

50 — MIDI System Exclusive ID
51 — MIDI Sys Ex Reception

52 — Preset Memory Protect
53 — MIDI Prog Change Master Switch
54 — Parameter Wrap Feature
55 — Auto-load Preset (Select Mode)
56 — Set All 1U Preset Mixes To Wet
57 — Receive Control 7 On Unit Chan
58 — Send MIDI PrgChg & Controllers
59 — Show 100 Config Presets
60 — Data Entry Knob Response
61 — Modulation Response Rate
62 — Use Alternate ROM Presets

63 — Operating System Version

System Exclusive Parameters

Soft Reset (without erasing the internal memory)
- While holding down the **System•MIDI** button,
- press the **Unit A** button.

Initializing the RAM Presets
- While holding down the **System•MIDI** button,
- press the **Unit B** button.
- Press the **Write•Copy** button to initialize all of the RAM presets.

Reinitializing the DP/4
- While holding down the **System•MIDI** button,
- press the **Unit B** button.
- Press the **Right Arrow** button once.
- Press the **Write•Copy** button to reinitialize the DP/4.

List of Algorithms

These 1 Unit algorithms are included in the DP/4:

- Small Room Rev
- Large Room Rev
- Hall Reverb
- Small Plate
- Large Plate
- Reverse Reverb
- ReverseReverb 2
- Gated Reverb
- Non Lin Reverb
- NonLin Reverb2
- NonLin Reverb3
- MultiTap Delay
- Dual Delay
- Tempo Delay
- EQ-DDL-withLFO
- VCF-Distortion
- Guitar Amp 1
- Guitar Amp 2
- Guitar Amp 3
- Speaker Cabinet
- TunableSpeaker
- Rotating Spkr
- EQ-Chorus-DDL
- EQ-Vibrato-DDL
- EQ-Panner-DDL
- EQ-Flanger-DDL
- EQ-Tremolo-DDL
- Phaser - DDL
- 8 Voice Chorus
- Flanger
- Pitch Shifter
- Pitch Shift-DDL
- FastPitchShift
- EQ-Compressor
- Expander
- Keyed Expander
- InversExpander
- De-esser
- Ducker / Gate
- Rumble Filter
- Parametric EQ
- VandrPolFilter
- Sine/Noise Gen
- No Effect (Bypass Preset)

These 2 Unit algorithms are included in the DP/4:

- Pitch Shift 2U
- 3.3 sec DDL 2U

This 4 unit algorithm is included in the DP/4:

- Vocoder

Specs

Frequency response (wet and dry) = 2 Hz–18 KHz
Signal-to-noise = -87dB
THD+Noise = .005% (-86dB)
Dynamic range = 96dB
IM distortion (SMPTE) = 0.05%
Crosstalk between channels better than -80 dB (1 KHz)
Input impedance = 1Meg Ω
Output impedance = 2.6K Ω

(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)

Physical

4 audio inputs, 4 audio outputs (phone jacks)
separate input and output level controls for 4 channels; accommodate -10 to +4dB
two level indicator LEDs per channel
32 character back-lit LCD display
digital 32 step parameter knob
MIDI in/out and thru
analog control voltage pedal input
dual foot switch input
internal power supply, detachable power cord, internal fuse

Dimensions

19" (48.26cm) wide x 3 1/2" (8.87cm) high x 15 5/8" (39.68cm) deep
19" rack mount standard, 2U high
12 lbs. (5.44 kilograms)

Glossary

There are a few terms that need to be understood before you can unlock the DP/4's full potential as a programmable effects processor.

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

By-pass Units The last parameter 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 is 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 combined with time varying delayed versions of itself. These multiple delay lines 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 the signals that go above threshold are attenuated by 25 per cent. 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. The threshold in this effect is the amplitude level below which a signal is passed without change.

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 section. Read these definitions again after reading the rest of the section.

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 — The DP/4 preset type which contains algorithms and parameter settings for all of the units as well as all of the Config parameter settings.

☞ **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 read the rest of this section and then refer to *Section 3 — Config Parameters* for more details on this essential concept.

Damping A parameter in the DP/4 that allows control of frequency information found in reverb algorithms. You can use damping to customize the perceived size and ambience of an environment (making it wetter/drier or brighter/darker).

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 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 create the sonic information that determines how we localize and perceive size in 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. This is more accurately termed downward expansion.

Feedback A signal routing in which some of the output 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 (see Section 3 of the Musician's Manual for a signal-flow diagram). The A and B units are in series; the output of the B unit is mixed back into the input of the A unit. A 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 (see Section 3 of the Musician's Manual for a signal-flow diagram). The A and B units are in series; the output of the B unit is mixed back into the input of the A unit. A 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.

Flanger A processor that originally simulated the effect of two synchronized tape machines in playback of the same signal, where one machine was manually slowed. The small delay causes a phasing cancellation that produces a comb filter which is 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 from recordings, as well as controlling effected signals.

Hysteresis The property of a system whose event threshold 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 an 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 recorded signal, while reducing the variability in the noise floor. 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. 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).

Limiter A device that will prevent a source signal from going above a pre-set level/threshold. A compressor with a high 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.

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).

Generally, in either of these two modes, you use the **Left** and **Right Arrow** buttons to scroll to the parameter you want to modify, then use the **Data Entry Knob** to change the parameter's value.

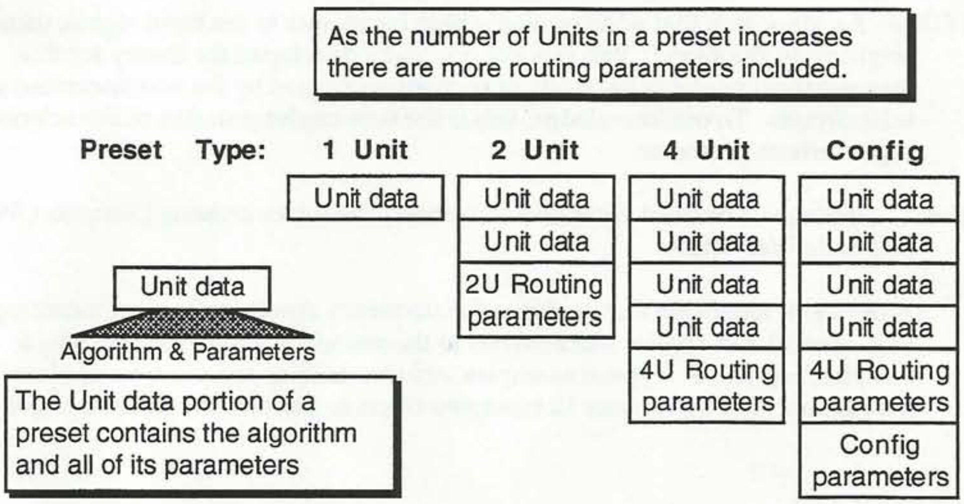
Parametric EQ An algorithm that raises or lowers specified frequency regions in program material. A parametric EQ has variable center frequency, gain, and "Q" - the ratio of center frequency to bandwidth.

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.

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. These groups of 50 presets are referred to as *Preset Banks*.

Preset Number	1 Unit	2 Unit	4 Unit	Config
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.

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 used 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. The new signal can be modulated and processed to create interesting sonic effects.

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. These two signals are electrically unbalanced.

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. To our knowledge, this is the first implementation of this science in a digital effects processor.

VCF-Distortion Voltage Controlled Filter and distortion. Useful for creating Distortion, Wah Wah, and 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: talking orchestra, vocal electronic percussion. The DP/4 uses 12 bandpass filters to perform the analysis (3 per unit).

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"INSTRUCTIONS PERTAINING TO A RISK OF FIRE, ELECTRIC SHOCK, OR INJURY TO PERSONS"

IMPORTANT SAFETY INSTRUCTIONS

WARNING—When using electric products, basic precautions should always be followed, including the following:

1. Read all the instructions before using the product.

2. To reduce the risk of injury, close supervision is necessary when a product is used near children.

3. Do not use this product near water - for example, near a bathtub, washbowl, kitchen sink, in a wet basement, or near a swimming pool, or the like.

4. This product should be used only with a cart or stand that is recommended by the manufacturer.

5. This product, either alone or in combination with an amplifier and headphones or speakers, may be capable of producing sound levels that could cause permanent hearing loss. Do not operate for a long period of time at a high volume level or at a level that is uncomfortable. If you experience any hearing loss or ringing in the ears, you should consult an audiologist.

6. The product should be located so that its location or position does not interfere with its proper ventilation.

7. The product should be located away from heat sources such as radiators, heat registers, or other products that produce heat.

8. The product should be connected to a power supply only of the type described in the operating instructions or as marked on the product.

9. This product may be equipped with a polarized line plug (one blade wider than the other). This is a safety feature. If you are unable to insert the plug into the outlet, contact an electrician to replace your obsolete outlet. Do not defeat the safety purpose of the plug.

10. The power supply cord of the product should be unplugged from the outlet when left unused for a long period of time.

11. Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.

12. The product should be serviced by qualified service personnel when:

    a. The power supply cord or the plug has been damaged; or
    b. Objects have fallen, or liquid has been spilled into the product; or
    c. The product has been exposed to rain; or
    d. The product does not appear to operate normally or exhibits a marked change in performance; or
    e. The product has been dropped, or the enclosure damaged.

13. Do not attempt to service the product beyond that described in the user-maintenance instructions. All other servicing should be referred to qualified service personnel.

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