DP/4 Articles in Transoniq Hacker

Introduction

This document is a collection of articles about the Ensoniq DP/4 and DP/4+ effect processor. The articles were originally published in Transoniq Hacker - a magazine dedicated to the musical products of the Ensoniq Corporation.

The goal of this document is to collect and organize the information published about the DP/4(+) with proper attribution to the original authors. If you are one of the original authors and feel uncomfortable publishing your work here, please get in touch and we will remove the article in question.

The articles were pulled from the online Transoniq Hacker archive here:
http://www.buchty.net/~buchty/ensoniq/transoniq_hacker/

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DP/4 Announcement

(Transoniq Hacker #76)

RND (♪♫♪)

Ensoniq News

Ensoniq is pleased to announce the development of the DP/4 Parallel Effects Processor. The DP/4 uses 4 effects chips, coupled to 4 inputs and 4 outputs to give new-found versatility to your signal processing needs. This 2-rack space unit contains 400 Presets covering single input, dual stereo inputs, 1 stereo and 2 mono inputs, and 4 mono input configurations.

Algorithms provided include Hall, Room, Plate, Reverse, Gated and Non-Linear Reverbs, Delays, Parametric EQ, Phaser, Flanger, Chorus, Pitch Shifter, Compressor, Limiter, Tremolo, Panner, Vibrato, and Distortion. Chains of up to 16 effects can be set up if desired.

Complete flexibility in signal routing is possible (Serial, Parallel, Feedback, Feedforward) and every parameter is modulatable from any of 8 user chosen modulators. The DP/4 provides for complete signal routing and signal mixing in software, giving you the convenience of 4 effects processors, a patch bay, and a mixing board all in one integrated unit. The DP/4 has a suggested retail price of $1395 and will be available in late October.

Ensoniq DP/4 Parallel Effects Processor Specifications

In Issue #76 Ensoniq announced their new DP/4 Parallel Effects Processor. Here is an up-to-date listing of the specs. Introduction is set for March, 1992. A review will follow shortly thereafter.

Memory

- 400 Presets (200 ROM, 200 RAM)
- 100 each (50 ROM, 50 RAM) of the following:
  - > 1 Unit Presets
  - > 2 Unit Presets
  - > 4 Unit Presets
  - > Config Presets (4 Units plus all the signal routing parameters)
- 256k words (512 kbytes) of delay memory
- Max delay time per unit: 1.6 seconds
- Max single delay time possible (no regeneration): 6.4 seconds

Algorithms

1) Small Room Reverb
2) Large Room Reverb
3) Hall Reverb
4) Small Plate
5) Large Plate
6) Reverse Reverb
7) ReverseReverb2
8) Gated Reverb
9) NonLin Reverb
10) NonLin Reverb 2
11) NonLin Reverb 3
12) MultiTap Delay
13) 3 sec. Delay (2U)
14) Dual Delay
15) EQ-DELAY-LFO
16) Tempo Delay
17) VCFPDistortion
18) Guitar Amp1
19) Guitar Amp 2
20) Guitar Amp 3
21) Speaker Cabinet
22) Tunable Speaker
23) Rotating Speaker
24) EQPChorusPDDL
25) EQPVibratoPDDL
26) EQPPannerPDDL
27) EQPFlangerPDDL
28) EQPTremoloPDDL
29) PhaserPDDL
30) 8 Voice Chorus
31) Flanger
32) Pitch Shifter
33) FastPitchShift
34) Pitch Shift 2U
35) Pitch ShiftPDDL
36) EQPCompressor
37) Inverse Expander
38) Expander
39) KeyedExpander
40) DePEsser
41) Ducker/Gate
42) Parametric EQ
43) Aural Enhancer
44) Vocoder
45) Sine/Noise Gen

Input/Output Specifications

- A/D – D/A conversion: 16 bit linear
- Input Level: –12.5 dBV to +18 dBV
- Output Level: +15 dBV maximum
- Input impedance: 1 MΩ
- Output impedance: 2.6 kΩ
- Frequency Response: 2 Hz – 18 kHz
- Dynamic Range: 96dB
- Signal to Noise ratio: –87dB
- THD+Noise: 0.005% (–86dB)
- Crosstalk between channels: better than –80 dB (1 kHz)
- IM distortion (SMPTE): 0.05%

MIDI

- Responds to 6 MIDI channels at the same time
  - > Each Unit can have its own MIDI channel
  - > A separate MIDI channel can be used for controller information
  - > A separate channel can be used to send Config Preset program changes
- Separate program change mapping tables available per Unit
- Responds to the following MIDI controllers for parameter modulation or effect bypass :
  - > Controllers #000 – #127
  - > Pitch bend wheel
  - > Note number
  - > Velocity
  - > Pressure (channel/mono or key/polyphonic)
- Can save/load the following data types as MIDI Sys. Ex.:
  - > Single Presets
  - > Bank of related Presets (1, 2, 4 Unit or Config)
  - > System global data
  - > All data (all Presets plus System global parameters)

Front Panel Controls

- 4 input, 4 output knobs
- 1 Hi-Z input (overrides Input 1 on back panel)
- Individual Peak and Signal LEDs for each channel
- 32 character back-lit LCD display
- 2 character LED display w/MIDI indicator
- Digital 32 step Data Entry Knob
- 12 buttons (Write/Copy, Cancel/Undo, Left Arrow, Right Arrow, Select, Edit/Compare, System/MIDI, Unit A, Unit B, Unit C, Unit D, Config)
- Active/Bypassed status LED for each Unit
- Power Switch

Rear Panel Connections

- (4) 1/4" Input connectors (can be grouped as 2 mono/stereo pairs)
- (4) 1/4" Output connectors (can be grouped as 2 mono/stereo pairs)
- Pedal/Control Voltage input (allows modulation from external sources, such as the CV pedal)
- Dual/Single foot switch input (for use w/SW-5 or SW-2)
- MIDI In/Out/Thru
- A/C line connector

DP/4 First Look

(Transoniq Hacker #87)

The DP/4 — First Look

Dennie Edwards

Ladies and gentlemen, let me introduce to you to (what I feel) is the coolest black box on the planetface, Ensoniq's DP/4 Digital Effects Processor. If you have not yet played with this piece of musical gear at an Ensoniq clinic or dealer, do so. Following is a brief "get started" tutorial to help you listen to the unit right out of the box or while on display at a dealer. If there isn't one on display, offer to hook it up for the keyboard salesperson.

First get the stuff you need to hook up the DP/4 — an amp or a PA to listen through, two cables to connect a stereo output of the DP/4 into the stereo PA (totally necessary to get full coolness effect), a mono signal such as a guitar (with cable for input to the DP/4), and something to pick your jaw up off the floor with. Second, find an AC outlet with a three prong outlet and hook it up. Then, plug the two shielded cables into the output jacks 1 and 2. This will be our stereo feed to the PA. Next, plug the guitar into the input 1 jack on the front panel. Any signal like a microphone, that has a low-to-high impedance transformer, or a keyboard can be used. The input on the back labelled 1 is okay, too. The front panel input jack disables the rear number 1 input. Set all of the inputs and outputs controls to approximately 12 o'clock and that'll do it for now.

Setting Input Levels

A technique that works most of the time to set the input levels is to set the input at the optimum level without causing distortion, press the config (configuration) button twice to bypass all the units, then play the inputted instrument at its loudest level. Rotate the pot for input one until the peak LED flashes above the input knob. Then, rotate the knob counter- clockwise until the LED doesn't flash anymore. Simply press the config button once more and the unit will be back on-line. Now we can select the different types of configurations.

A config preset is the most powerful of all the presets on the DP/4. It lets you save everything that is in the DP/4 at that time and recalls all algorithms, signal routing among the four units, and all the mixing parameters. It also allows you to select the number of inputs. In short, a config preset shows how the inputs, units, and outputs are routed.

Let's Get Started

I'm gonna assume that you'll use an electric guitar to scope out the coolness of Mr. DP/4. First, select the proper configuration for a "1-Source Mono Input." Press config, then use the data knob to select config #53. Next, press the select button. Now that we have that done, we can listen to some effects.

We are going to listen to the DP/4 as one super unit. This config allows all four units within the DP/4 to act as one. This is easily done since all of the signal routing is done in the digital domain (a digital patchbay!). We are now in the one source config. To select the 4 unit presets, press the config button. Turn the large data entry knob until the screen shows "Select 4 U Presets," (50). Press Select. Rotate the data entry knob to choose among the 100 4-unit presets. Remember that there are 50 RAM and 50 ROM that make up the 100 presets. Find a preset that you want to hear and press the select button. After you have given these presets a listen, get ready to do some knob turning and button pushing.

Okay, so you are impressed with using 4 24-bit processors at once, but wouldn't you like to hear how each processor sounds by itself? 'Course you would. Press the config button once again and select the "1 U Presets," (52). Also remember to press the select button. The yellow light above the A unit should light showing that you have selected the A processor for use. The bypass LED (the red ones) should now light over units B, C, D. With our current input config we can only listen to processor A. One can still edit and select one unit presets for the other processors, but we only have an input going to unit A. Next, turn the data entry knob to display the names of the 1-unit presets. Remember, an algorithm will not load into the processor until you press the select button. Again, have fun and listen to the clarity of 24-bit fidelity.

Speaking about this 24-bit stuff, why does Ensoniq choose to use 24-bit processor instead of 16- or 18-bit processors? The answer is both simple and complex. 16-bit processing gives us the full 20k bandwidth of frequency response that our ears have now come to expect from digital products. If a signal is to be processed to create a reverb, it then must be allowed to sustain and retain its integrity at 20k bandwidth. If more signal is pumped through the processor, it then becomes very difficult for that processor to sustain such long reverbs while maintaining true fidelity. With a 24-bit processor there is a ton of processing power within the unit. So how many bits does it take, then, to double the resolution of a 16-bit machine? 32-bits do you say? WRONG! Believe it or not, it only takes 17-bits to double the resolution of 16-bits. It only takes 18-bits to double the resolution of 17-bits. Which is why the Room/Hall Decay parameter for the Small Room, Large Room, and Hall reverbs can last from 100 seconds, 150 seconds, and 250 seconds respectively. This, friends, is why Ensoniq uses nothing less than 24-bit effects on all of their machines.

Okay, so we're impressed with the factory presets that the guys with the pocket protectors made, but is it easy for us normal people to use? Yup! As one would expect from a straight-forward machine, the first button push is the EDIT button. Next, select the processor A, B, C, or D you wish to edit. We now have selected the unit's algorithm for editing. The active unit's LED should be lit and the two-digit display should read "00." If not, press the left arrow until it does. You can now scroll through the parameters and listen to each one. With each new algorithm, there will also be parameters specifically for that algorithm. To change a parameter all you have to do is use the left and right arrow to select the appropriate parameter and then use the data entry knob the change the value. As soon as you change a parameter, the edit LED flashes, indicating that you have now entered the edit buffer. To compare between the original and your edited version, press the Edit button.

Are you excited yet? This appears to be first processor on the market that will allow the average musician to get the same quality as the BIG STUDIOS get out of their outrageously expensive machines. It will help the everyday musician create better demos and have a better shot at getting in with the big A/R guys who are used to listening to product coming from these mega studios. The bottom line is this, if you want to sound good, I mean really good, a DP/4 can be a major factor in achieving this goal. It is a little complex, but don't worry. There are a series of instructional videos coming from Ensoniq and I am sure we're gonna hear from some third parties to help you master the machine. Also, Ensoniq is making a huge effort in training all of their dealers on the DP/4, so make sure you ask your local Ensoniq dealer for help. One last thing — I understand that a lot a major artists and producers are buying these toys, like The System, Nile Rodgers, Joey DeFrancesco, David Was, George Duke, and Randy Jackson to name a few.

So what it comes down to is that the gap of audio quality that lies between the pros and the rest of us working musicians has just been narrowed considerably by the DP/4. Good deal. ▬

Bio: Dennie Edwards is the Asst. Manager/Keyboard salesman for Vince's Backstage Music in Lafayette, La. Dennie also does MIDI consulting and sound programming for local jingles and other productions. His favorite colors are blue and florescent pink. It is rumored that Dennie is the son of the Shell Answer Man.

DP/4 Review

(Transoniq Hacker #88)

Ensoniq's DP/4 — The Review

Charles R. Fischer

Product: DP/4 Parallel Effects Processor
Price: $1495.
From: Ensoniq Corp., 155 Great Valley Parkway, Malvern, PA 19355, (215) 647-3930 or contact your local Ensoniq dealer.

If you've kept up with the product reviews in the magazines lately, you've seen a number of folks who appear to be in competition to see who could sing the loudest praises about the new Ensoniq effects processor, you guessed it, the DP/4. Well, if you're holding out hope that I'm going to contradict the rest of the reviewers, you're encouraged to skip over the rest of this story — I know what they're excited about.

The DP/4 Parallel Effects Processor is Ensoniq's first attempt at producing a stand-alone digital signal processor, which makes the results even more significant. The DP/4 is noteworthy for a number of reasons, which I'll quickly list here:

* The DP/4 is the first multiple effects processor with 4 sets of audio ins and outs, which may be used with the 4 onboard signal processors in just about any way you might want. Almost all other multi-effects units have their effects hardwired in between their inputs and outputs — there's no way you can use one or two effects on one signal, and use the rest on something else.

• The DP/4 is furnished with a set of useful effects algorithms (45 total), making it a versatile audio toolbox, capable of covering a wide variety of signal processing tasks here. In addition to the garden-variety reverbs and choruses, there're exotic treats like a Van Der Pol filter, tube amp simulators, — even a vocoder.

• And the thing is equipped with an extremely powerful MIDI implementation, which raises its usability by a large leap.

Of course, all the dope features in the world don't mean a thing if the sound doesn't deliver. The audio quality of the DP/4 is remarkably good for the most part. I know, some people still associate Ensoniq with second-rate sound quality because of early products like the Mirage and ESQ-1. There's nothing second-rate about this machine; the sounds are the best I've heard for a signal processor in its price range — $1495 US.

In other words, if this is gonna be another one of those stupid enthusiastic reviews wherein a legendary cynic like myself ends up sounding like a leader of the Ensoniq P.R. staff, you can be certain that this product being covered is exceptional.

The System

It's all too easy to be confused by the architecture of this device, so we'll begin with an excellent explanation borrowed from the owners manual:

"Picture a rack of effects containing the following:

• four state-of-the-art effects processors

* 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 submixer, all with the press of a button (or with a MIDI program change, or a press of a footswitch).

"That's the DP/4."

Of course, there's a little bit more to it than that, but it pretty much describes the power behind this unit. Because of this versatility, there are a few opportunities for confusion as you learn your way around; this can be minimized by learning Ensoniq's insider terms for the features and concepts used in the DP/4.

For example, Ensoniq refers to each effects processor as a "unit." The routing of the audio input to the units is called a "source" (each source can consist of either one or two inputs, depending on how the unit is set up). But the most important of these concepts is what is called the "config," short for configuration.

The config consists of the various parameters that control how the sources get wired up to the units, as well as how the various units are interconnected. You can chain multiple units in parallel or series, or create feedback loops between units. The latter is useful for producing Carlsbad Cavern reverbs, unbelievable Doppler effects, and other def sounds. The DP/4 design offers a global configuration setup, as well as 100 config presets.

The Hardware

The DP/4 is neatly packaged in a 2U rackmount case. The front panel boasts a large rotary knob for editing parameters, a two-digit LED display (indicating the present preset number), a backlit LCD (much like the ones on the SQ synths) and an assortment of buttons and LEDs. There's even an additional audio input jack on the front; this lets guitarists plug in and play without having to mess with the cables on the back panel — a nice touch.

On the back, you'll find a socket for a standard AC line cord (no wall warts to break or lose), MIDI in, thru, and out ports, and ten 1/4-inch phone jacks. These are set up as four sets of audio inputs and outputs, with the last two offering inputs for a footswitch and an optional footpedal or external control voltage.

Of course I removed the DP/4's cover. I'm pleased to report that the unit appears to be very well put together; aside from a few added jumper wires, there was no evidence of last-minute modifications or rework (a fairly frequent occurrence in high-tech products). It appears to be rugged enough to handle touring without any problems.

The Software

Another strength of this unit is the large number of highly usable effects algorithms. If you want variety, you've come to the right place, my friend — you'll find everything from choruses and compressors to wahwah pedals and vocoders. This thing is literally a "greatest hits" of effects in one box — the only things I missed were a state-variable filter and the rarely-used ring modulator.

One thing that must be mentioned is that units and effects are not always the same thing; some algorithms can produce as many as three effects simultaneously, while others make use of multiple units (for example, the vocoder uses all 4 units in concert).

First, you'll find that most of the algorithms were apparently chosen for their musical usefulness; there are lots of bread-and-butter effects here (reverbs, chorusing, flanging, distortion, speaker simulators, and delay lines). While this means that most of the presets are geared to the so-called working musician, technos and other fiends need not worry — this machine can be programmed to produce some truly bizarre sounds.

There's a number of less-often used effects, as well. There's a good rotating speaker simulator with overdrive (so you don't have to waste a unit to get the sound of a Hammond in heat), an envelope follower with distortion (which can also be used as a wahwah pedal), several types of pitch shifters, compressors, expanders, and a few more specialized items. One algorithm generates sine waves and noise, allowing the unit to double as a function generator in a pinch (very clever!).

That's all for now. In the next installment, I'll get into the sound quality, the MIDI capabilities, and some of the possible uses for this unit. Talk to ya soon. ▬

Bio: Charles R. Fischer is a test technician for AKG Acoustics, Inc. In addition, he writes for several magazines, designs custom MIDI controllers, plays keyboards, and finds time to sleep once or twice a week.

Programming the DP/4

Dennie Edwards

One of the most important ingredients in musical production is signal processing. Signal processing takes ordinary sounds and brings them to life. It can bring clarity and separation to the ear. Signal processing creates a picture to the ear and mind similar to the way visual arts bring paintings to the eye and mind. So with this in mind, let's learn how to paint with a new and innovative paint brush...the DP/4.

A major key to unlocking the power of the DP/4 is to understand exactly what it is. It may appear to be as simple as a little box that stands only two rack spaces tall, but we can't judge a book by its cover and neither can we look at the DP/4 and call it just another effects processor. First, we will review just what is inside the DP/4.

The heart of the DP/4 contains four 24-bit Ensoniq Signal Processor (ESP) chips. These ESPs are connected with a digital patchbay and are mixed with a digital mixer. Because of its unique construction, the DP/4 may be considered the most flexible effects product ever invented. With the ability to patch effects in series, parallel, and with feedback loops never before possible in conventional units, the DP/4 delivers audio quality that rivals products costing hundreds of dollars more.

We'll begin with the types of presets and examine the most powerful of all presets, the configuration, or config preset. Depending on the number of inputs, there are many different presets to choose from. There are two hundred ROM and two hundred RAM presets.

The first preset is the one-unit preset. Simply put, it is used when one wants to use one effect unit at a time and several independent sources. A one-unit preset consists of an algorithm and all of its specific parameters. An algorithm defines what type of effect(s) will be loaded into each ESP chip.

The second type of unit preset is called the two-unit preset. These include more complex effects that require two processors or two processors used in series, parallel, or feedback loops. Also, there can be multiple input sources, both stereo and mono.

The third type of preset is that of the four-unit preset. This preset is a snapshot of all four units at a time. This preset is used with a one-source input, whether it is mono or stereo. Using this type of preset can give the user some of the most wonderful sounds available. Imagine the possibilities of using four 24-bit ESPs for one source!

The most powerful of all presets is the config preset which is the main topic of our discussion here. Also, included in the following are some helpful short cuts and tips on general use of the DP/4. Understanding the DP/4 config preset will unlock the power of the DP/4.

What is a config preset?

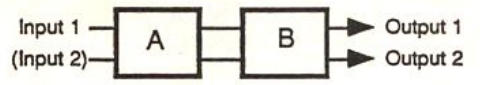
Config presets let you save all input source configurations, effects algorithms, signal routing, and mixing of all four units. There are 100 config presets in the DP/4 (50 ROM and 50 RAM). Presets 50-99 are the ROM and 00-49 are the RAM.

Before we get started, we must prepare the DP/4 for our advanced use. System parameter number 59 is "Show 100 config presets=(yes/no)." Let's throw caution to the wind and set this to Yes. We now can select all 100 config presets! First, press System/MIDI, then using the right arrow, scroll to parameter number 59. Next, turn the data knob, the large silver round thing, and change the parameter to Yes. Now we will select some presets.

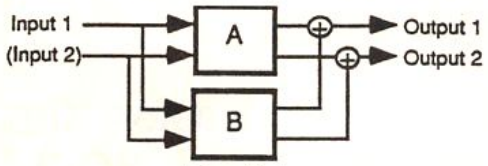
To select a config preset, first press Config, then use the data knob to scroll fast through the 100 presets. Next, press the Select button. We have just selected a config preset. As long as the LED above the config button is lit, we can select config presets. Let's go a bit further. Now press the unit A button. Notice that the display for unit A shows the preset effect and the algorithm that is used to create that preset. The presets that we will have access to by turning the data knob, will be determined by the input configuration loaded by the selected config preset. When a one-source config is selected, four-unit presets can be selected. If the source config is a two-source config, then we can pick two-unit presets, and lastly, if there are four independent sources, a single unit preset can be chosen for each unit.

Before we talk about editing a config preset parameters, let's edit single unit parameters first. Select config preset 00, "MonInVocalSetup." (Press Config, use the data knob, then press Select to confirm.) Next, press the unit A button. Cmp should be capitalized in the display. Each of the four abbreviations for the effects that are found in the bottom row of the display corresponds to each of the four units A, B, C, and D. The currently selected unit will have its effect abbreviation capitalized. Go ahead and try pressing each of the unit buttons and see how it works. With the A unit selected, press the edit button. Parameter 00 should appear. If not, use the left arrow to scroll till the LED display says 00. The display

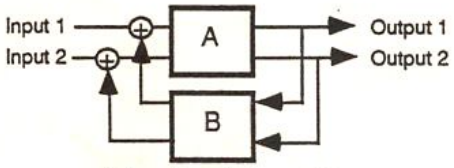
Figure 2.



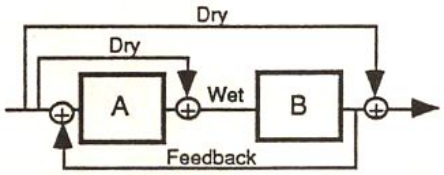
Serial



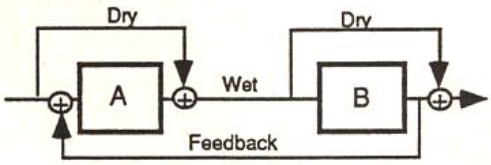
Parallel



Feedback 1 and 2



Feedback 1



Feedback 2

Parameter 06 in the 1 source config is the AB input select. If the display shows (12) this means that the input will be in stereo using inputs 1 and 2. Therefore, the (1) means a mono input using input 1.

The next of the basic config parameters are parameters number 07 (unit A), 08 (unit B), 09 (unit C), and 10 (unit D). These parameters determine what the bypass function (double clicking on the unit button until the red LED is lit) does to the output of each unit. If it is set to (b) bypass, it is like setting the Mix to 00, and if the parameter is set to (k) kill, it is like setting the Volume to 00. Depending on the input source, the config parameters can differ slightly. On page 3-6 of the DP/4 owner's manual are 18 of the possible 36 ABCD Routings. As we can see, the possibilities are almost unimaginable. Go ahead and experiment — remember we can get all of the factory presets back.

The above parameters are the most important ones in setting up the config parameters. Remember that editing a single unit is very similar. Just press the unit button, A, B, C, or D, then press the edit button. Also, a config preset remembers all parameters for the units. In future articles we will discuss more about editing individual units.

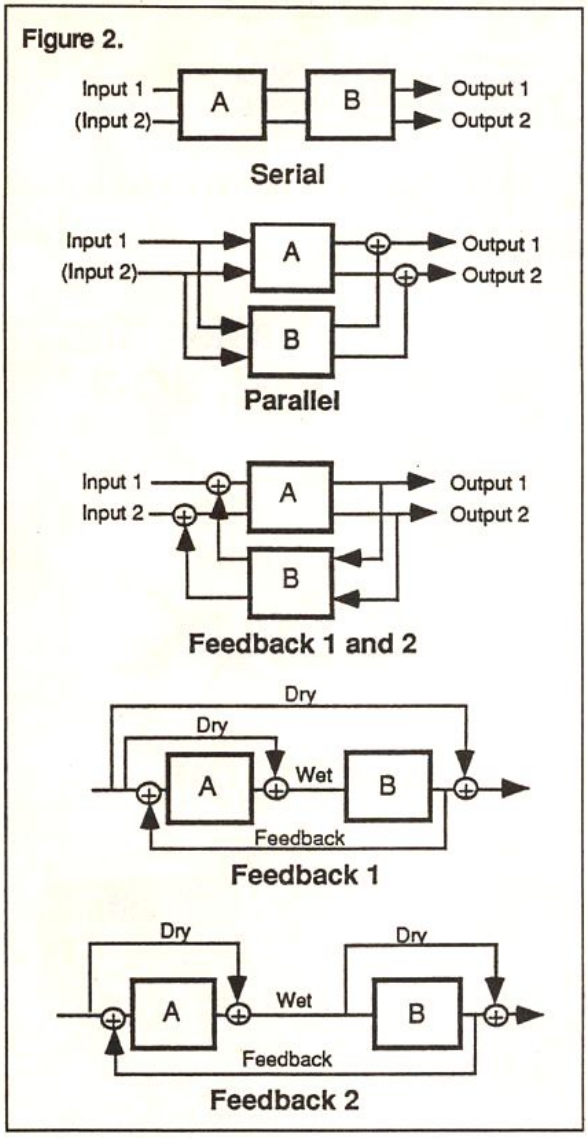
Other important parameters are found in the System/MIDI pages. These parameters control the entire system. First, we will again mention System parameter number 59. The default setting is "no" for software versions before 1.00. To unlock the full potential of the DP/4 it should be set to "yes." Another one is System parameter 56, "Set all 1 Unit Pset mixes to wet= yes/no." This is a great parameter! If you're using the DP/4 with a mixing console for live applications, this parameter is very handy. The mix output of all units will be wet. This is really great when using units in parallel and have a one source input. The same principle can also apply to multiple input configs. We can then use the Aux returns or separate channels on the mixing board to control the sound system effects mix. Bypassing a unit when set to kill will not cause an increase in volume and the mix levels on the board will not have to be adjusted.

The last parameter we will discuss is that of system parameter 63, "Ensoniq * DP/4 OS version x.xx." This may not seem important, but can be useful when referring questions to the local dealer or even in speaking with the Ensoniq Customer Service Department (215-647-3930). Very few people know how to find this information in their computerized musical equipment, whether it's a keyboard, computer software, or effects processors.

The DP/4 is a very flexible product. It is important, however, to not get tangled up in its flexibility. The best advice is to take it slow. Try to learn a little every day or so. The manual is intended as a road map or a guide book. It will not tell us where to stop on the road to a successful music project — that is entirely up to you. The DP/4 will only help you get there sounding much better. ▬

Bio: Dennie Edwards is the Customer Service Representative at Ensoniq for the Midwest Region. While living in the Philadelphia area, he uncovered the secret of the Quakercaster electric guitar.

Figure 2.



Parameter 06 in the 1 source config is the AB input select. If the display shows (12) this means that the input will be in stereo using inputs 1 and 2. Therefore, the (1) means a mono input using input 1.

The next of the basic config parameters are parameters number 07 (unit A), 08 (unit B), 09 (unit C), and 10 (unit D). These parameters determine what the bypass function (double clicking on the unit button until the red LED is lit) does to the output of each unit. If it is set to (b) bypass, it is like setting the Mix to 00, and if the parameter is set to (k) kill, it is like setting the Volume to 00. Depending on the input source, the config parameters can differ slightly. On page 3-6 of the DP/4 owner's manual are 18 of the possible 36 ABCD Routings. As we can see, the possibilities are almost unimaginable. Go ahead and experiment — remember we can get all of the factory presets back.

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Using the DP/4

(Transoniq Hacker #93)

Using the DP/4

Clark Salisbury

Music technology is just getting too cool lately: digital multi-track, inexpensive DAT recorders, tiny audio mixers with great specs — it gets mind-boggling after a while (of course, my wife thinks my mind is permanently boggled anyway, so maybe I'm not the greatest judge of degrees of boggled-ness). Still, one of the coolest things I've had the fortune of adding to my humble studio lately is a DP/4, the (relatively) new Parallel Effects Processor.

Of course, if you've read the reviews, or better yet, if you have a DP/4, you already know it's cool, so I won't bore you with the details. Still, the DP/4 is such a great creative tool that I thought it might be fun to share some stuff. So here goes.

Many effect devices offer some form of dynamic control — allowing you to control the reverb mix from your keyboard's mod wheel, for example, or chorus rate from a MIDI footpedal. Not only does the DP/4 offer dynamic control of this type, you can assign this control to any parameter. This opens up some interesting possibilities, both in the studio and in live performance.

There are 138 possible sources for modulation of any of the DP/4's parameters, including both MIDI controllers and analog inputs for footpedals and such. Up to eight DP/4 parameters can be controlled at any one time — two for each Unit in a Config (for those of you new to the DP/4, think of the Unit as an effects device and a Config as the way in which four of these effects devices (units) are interconnected, along with the programming for the individual Unit's parameters).

The 8 controllers you can use are assigned globally, from the System page. Once assigned, each controller is then available to modulate any parameter in any Unit. Let's take a look at a basic application — using a mod wheel to control DP/4 parameters — to get a sense of how this works.

It will be easiest to hear the effects we'll be controlling if we listen to just a single DP/4 unit, so select "Config 52 1U Psets" before trying these examples. Also, to use a MIDI controller with the DP/4 you must first make sure that "Control Chan" (parameter #35 in the System-MIDI menu) is set to the same MIDI channel that you are sending the controller on — in this case, MIDI channel 1 — and that MIDI is enabled (parameter #36). Otherwise, the MIDI controller will not be recognized by the DP/4.

First, connect the MIDI out from your keyboard to the MIDI in on the DP/4, and set the keyboard to transmit on MIDI channel 1. The default settings for the DP/4 include having the mod wheel assigned as MIDI controller #8, and DP/4 unit A set to

respond to controllers sent on MIDI channel 1. Therefore, assigning DP/4 Controller8 to control one or more DP/4 parameter(s) in Unit A will allow you to vary the parameter(s) from the mod wheel of the keyboard you've connected. Here are a couple of examples:

To control reverb mixing

- First, load unit A with a reverb program; press Select, then Unit A. Let's use #10, Summer Hall, so use the Data Entry Knob to scroll to this program, then hit the Select button to load the algorithm.
- Press Edit and scroll to parameter #23, and set the Mod 1 parameters to the following:

Param #:	Parameter:	Set to:
23	Mod 1 Src=	Cntrl-8
24	Mod 1 Destination	Parameter=002*
25	Mod 1 Param Range Min	00%
26	Mod 1 Param Range Max	99%

* Setting this parameter to 002 will control the volume of the reverb. If you are using the DP/4 with the source signal patched directly in, as you would if you were using it as a guitar processor, you may have better results setting this parameter to 001, effect Mix.

You can use parameters 25 an 26 to adjust how much effect the mod wheel will actually have on the reverb mix; setting the Max range lower and/or the Min range higher will restrict how far in one direction or the other the change will be. Note, too, that once you've set up these basic parameters, you can control any parameter in this effect simply by re-assigning parameter 24, Mod 1 Destination. For example, if you assign this to parameter 03 you can control reverb decay time.

Now let's take a look at another example of using the mod wheel to control a DP/4 parameter — in this case, chorus rate.

To control chorus rate

- Press Select, then Unit A and scroll to effect #30, Lush Keys.
- Press Edit and scroll to parameter #23, and set the Mod 1 parameters to the following:

Param #:	Parameter:	Set to:
18	Mod 1 Src=	Cntrl-8
19	Mod 1 Destination	Parameter=003
25	Mod 1 Param Range Min	00%
25	Mod 1 Param Range Max	70%*

* A value of 99% for this parameter makes the chorus sound pretty wobbly at maximum setting.

Just remember that you can control any parameter — filter cutoff, rotary speaker rate, compressor threshold, delay time, you name it. And, of course, you can use other controllers besides the mod wheel to vary effect parameters — use pressure, the pitch wheel, timbre control, patch selects, and so on.

Of course, you don't have to use MIDI to control effect parameters. As a matter of fact, if you use a footswitch or control pedal with the DP/4, the DP/4 will translate your pedal moves into MIDI data, which can be recorded into a sequencer. Assigning a control voltage pedal or foot switch works much the same as assigning a MIDI controller; the main difference is that you'll need to assign the foot pedal or foot switch as a controller from the System-MIDI menu, parameters #37 through #44.

One overlooked feature of the DP/4 is the ability to create "Songs" — sequences of presets that can be chained together for quick access via footswitch during performance. This feature makes it easy to organize and "cue up" groups of DP/4 presets for quick recall during performance. You could create a song and assign a footswitch (the SW-5/10 foot switch is recommended) to control preset selection. Here's how:

- Press System-MIDI and scroll to parameter #45. Set the following values:

Param #:	Parameter:	Set to:
45	DP4 Footswitch1=	Song Preset Up
46	DP4 Footswitch1=	Increment song

This sets Foot Switch 1 (on the right) to step sequentially through the 5 presets defined in the current song. Pressing Foot Switch 2 (on the left) will advance to the next song.

- Press the Right Arrow button to move to the Song Editor (parameters #47-49).
- Use the three parameters on this screen to define whatever 5-step songs you want. When you are done, use parameter #47 to reset to Song #01.
- Now each time you press Foot Switch 1 the DP/4 will select the next preset in the current Song, and will "wrap around" back to step 01 after step 05. Pressing Foot Switch 2 will advance to the next song.

Finally, here's an idea for you guitar players who use the DP/4. You know how it takes most effect devices a moment to reprogram themselves when you call up a new effect? And you know how sometimes you need to be able to go from, say, a clean rhythm guitar sound to a distorted solo sound rather quickly? Here's a way to do it:

Program one (or more) of the DP/4 footswitches to act as effect bypass buttons. Parameter 06 on the System page allows you to assign a controller to bypass Unit A; parameter 13 assigns a controller to bypass Unit B; parameter 20 assigns a controller to bypass Unit C; and parameter 27 assigns a controller to bypass Unit D. Once you've assigned a footswitch (or whatever) to bypass one of the units, call up a preset that includes a distortion sound. Bypass the Unit that's responsible for the distortion effect (either by pressing the Unit button, or hitting the bypass switch which you just programmed), and go ahead and jam using the clean sound that remains. When it's time to tear into your solo, just hit your bypass switch, and the distortion effect kicks in. And since the DP/4 doesn't have to re-program anything, it's pretty much instantaneous.

So that should give you some ideas for stuff you can do with the DP/4. Of course, we're not even scratching the surface here — I'll have more to say in future columns. Be well. ▬

Getting Started with the DP/4
Configs

Clark Salisbury

Because of its unique architecture — four discrete processors which can be digitally interconnected in any configuration — the DP/4 presents a vast potential for effects design not previously available in stand-alone processors. With the DP/4, for example, you could have a preset that includes a reverb and a delay fed by a single source, such as a guitar, and set up in parallel, mixed to stereo, along with a pair of compressors (each with EQ) set up in mono, each fed by its own source (such as the left and right channels of a stereo mix). You could have a preset with a single source feeding a plate reverb routed into a phaser, then to a DDL, and also feeding an 8-voice chorus, routed into another plate reverb, with some of the output of the plate routed back into the chorus in a feedback loop. This could all be mixed into two pair of stereo outs, with the plate -> phaser -> DDL chain appearing at one set, and the 8 voice chorus <-> plate chain appearing at the other. Why, it can get overwhelming just writing it out, much less creating these monster combinations.

Still, it's not as hard as one might think to master the DP/4. The key, in my humble opinion, lies in mastering the Config (short for configuration) — the way in which the four processors can be interconnected.

In the world of multi-effect processors, the DP/4 Config is somewhat unique, and therein lies a potential for confusion. With most processors you plug your source signal into the effect inputs and plug the effect outputs into your mixer or audio system. With the DP/4, though, you have a choice of four inputs and outputs to route signals through. And to further complicate matters, different sets of inputs and outputs might be used depending on your application. To help clear up some potential confusion for the new DP/4 user we're going to spend a little time dealing with the heart of the DP/4, the Config.

Each of the four processors that reside in the DP/4 has an input, called a Source. A single, massive effect that processes one input signal, for example, uses a 1 Source Config — a configuration in which all the DP/4 processors are driven by a single mono or stereo input signal.

Each of the processors in the DP/4 are called Units. In the example above, all four DP/4 Units are driven by the same Source (input signal), so this preset is thought of as a 4 Unit Preset. We can say, then, that an effect which processes a single input signal through all four DP/4 processors is a 4 Unit preset using a 1 Source Config.

Of course, other Configs are available. There are also 2, 3, and 4 Source Configs available. The Config you'll use in a given application will depend on how many different input signals you want to process through how many Units.

There are a couple of different ways to select the type of Config you'll use in a given situation. One is to simply scroll through the DP/4 presets until you find one that more or less matches your needs, and then edit any parameters that you find lacking.

First, hit the SYSTEM/MIDI button, and scroll to parameter 59, Show 100 Config Presets. If this is set to NO, use the data entry knob to change it to YES. This allows you to access the pre-programmed Config Presets that are included with the DP/4 when it's shipped from the factory.

Now press the SELECT, then the CONFIG button. Simply scroll through the Config presets until you find one that matches your needs. For example, if you want to process two mono signals (such as a snare drum and a lead vocal) separately, scroll until you find a likely effect — perhaps preset 06, Drums Vocals, might work in such a case.

Once you've selected the preset, you'll want to check to make sure that it is configured appropriately for your purpose. Press EDIT, and scroll through the parameter listings to check it out.

In the case of Preset 06, Drums Vocals, the first page you come to will tell you that this preset does indeed use a 2 Source Config. Scroll once to the right reveals that Units A and B are routed in parallel. They will each receive input signal directly from the source (these could also be configured in serial, in which Unit A would first receive the source signal, and then pass it along to Unit B, or in either of the two feedback modes, in which one or both Units return their output back to the input of the other Unit).

Scrolling once more to parameter 02 will reveal the routing relationship for Units C and D — in this case, they are connected in series. Scroll again to parameter 03, and you'll find that Units A and B are set up to receive a mono signal from input 1. If you wished, you could use the data entry knob to change this so that Units A and B would receive a stereo signal from inputs one and two. And scrolling once more will allow you to edit the same parameters for Units C and D.

Scrolling again will take you to parameter 05, where you can set the amount of dry, unprocessed signal that will be mixed with the output from Units A and B; scrolling again allows you to set the same parameter for Units C and D. Note that you can also control the wet/dry mix for any effect algorithm

by hitting EDIT, then pressing the appropriate A, B, C, or D button (depending on which effect you wish to edit), and scrolling to parameter 02.

The other way to create a custom Config Preset is to select one of the preprogrammed Config templates, and then assign to it the effects you wish to use. To do this, press the SELECT button, and scroll to Preset 53, 1 Src: Mono In. This is the first of several templates designed to make it easy to put together more complex Config Presets. You'll notice that all effects assigned to this preset (and the other Config templates) are "Dry" — no effects are used.

It's a simple matter, though, to select the Config that most closely matches your needs, and then simply assign the effects you want to use. For example, let's say you'd like to design a distorted guitar effect. The first step is to select the correct Config Preset. In this case, you might want to use all four Units to process the single guitar input. Since you're probably using a mono signal from your guitar, then, it should be pretty obvious that the most appropriate Config template is number 53, 1 Src: Mono In.

Selecting this Config Preset will set up a chain of all four Units configured to process the mono input signal. If you wish, you might press the EDIT button and scroll through all the Config parameters to make sure that they are set up to do what you want. You'll find that this Config is set up to process a mono signal presented to input 1 through all 4 Units, and that the Units are all connected in series. This should be pretty much what you want for this type of application.

Now all you have to do is assign the effects you want to the chain in whatever order you'd like. For example, you may want to place a compressor at the beginning of the chain. To do so, simply press EDIT, then button A (to select Unit A for editing). Make sure that parameter 00 is selected for editing, and simply use the data entry knob to scroll to the appropriate effect — in this case, probably one of the EQ-Compressor effects. Once you've selected the effect, of course, you can scroll through its parameters to make any changes you might wish.

Next, hit button B, (to select Unit B for editing), make sure that parameter 00 is selected, and again scroll through the available effects until you find the one you next like to assign in the chain — maybe a distortion effect or an amp simulator. Once this effect has been selected, you can continue in the same way until you've assigned (and possibly edited) an effect for each of the four Units. Once the entire thing has been tweaked to your satisfaction, you can save it for later recall.

If after doing this you still don't feel completely at ease with Configs and Config Presets, I'd suggest putting on a fresh pot of coffee and spending some quality time getting comfortable with them. If need be, talk to your local dealer, or give Ensoniq a call. The true potential of the DP/4 cannot be realized without a solid understanding of the Configs and Config Presets. But once you nail them, you can do just about anything with this crazy box. ▬

Bio: Clark Salisbury is a partner in the MIDI Connection, a Portland-based consulting firm. He has long been actively involved in the composition, performance, and recording of electronic music and is now producing his own pop-oriented compositions. His favorite color is chrome.

Suggestions, Hints and Tricks on Using the DP/4

Part One

Vance Galloway

The DP/4 is a very powerful tool. Its multitude of parameters, programs, algorithms, units and configurations can be a lot to deal with. What follows are a few general suggestions, hints and ideas that I have come up with to help DP/4 users get more out of this great processor. Since I am a guitarist, I use 1-input configurations most often. While my comments are obviously going to be influenced by this, most of them, nevertheless, hold true no matter what sort of setup you have.

First the Very General

When using more than 1 Unit to affect a single Sound Source remember that changing Effect Routings can radically change the sound of the effect. Routings, for those of you not familiar with the Unit, are the settings which determine which of the DP/4's Units (that is Effects) are working in Series (the output of one going into the input of the other), Parallel (both Units receiving the same signal and affecting that signal separately), or Feedback (output from one being sent to the input of the other as well as being sent back to the input of the first. For information on Routings see sections 3-5 through 3-8 in the DP/4 manual).

Experiment freely with these different Routings. If you have a sound that you like, but wish to make it bigger, try routing some of the Units in parallel instead of series. For instance, if a delay algorithm is loaded into Unit C and a reverb is loaded into Unit D, and CD are in series (as is the case with 4 unit preset #10, "Studio Vocal 1" or CONFIGuration preset #2, "Rock Gtr Setup") try changing the CD Unit Routing. Notice how much wider the effects now sound. If you want to make a sound tighter, try using a Series Routings. Play with some of your favorite DP/4 sounds by simply changing the Unit Routings. And don't forget to experiment with Feedback Routings. On several occasions utilizing the DP/4's Feedback Routings have given me very pleasant surprises, creating entirely new sounds out of otherwise "standard" presets. (See sections 3-6 through 3-12 for information on Routing.)

Remember that changing a particular effect's level as opposed to its mix amount create radically different sounds. This is very important. Changing the mix amount means altering the ratio of processed to dry signal that leaves the Unit whereas the Level (of course) controls the output volume of both processed and dry signals. Experimenting with different settings, especially on a multi-Unit configuration, can reveal very useful changes in the sound. You might use very high mix ratios and very low volumes in one preset and find that you need just the opposite in another. Since there are so many variables here (Unit routing, number of input Sources, which algorithms are being used) I can't give any specific suggestions except to play around with various settings and note the results. In general; higher mix ratios yield wider or bigger sounds while lower ones sound a bit more natural. (See manual section 4-3 for Volume and Mix information.)

There is also a big difference between turning off a Unit by using Bypass as opposed to Kill. Bypass has the effect of simply turning the Unit off while allowing dry signal to pass through. Kill, on the other hand, prevents any signal from passing through. Therefore, if all your Units are in series and you Kill Unit A, none of the other Units will get any signal. However, Kill can be very useful in any routing where any of the Units are in parallel. For example, let's pretend you have Units A and B running in series and have C and D running in parallel and AB and CD running parallel. If you Bypass Unit C, there will then be dry signal running straight from input through the bypassed Unit C, to output. This means you will have dry signal running to the output. Sure, this might be desirable. But if it's not, set Unit C's (B)ypass (K)ill setting [found by pressing Edit then Config then scrolling to (K)ill]. This will keep any signal from passing through Unit C while allowing all of the other Units to work just as they had before. Again, experiment freely and you'll see the profound usefulness of this option.

Now the More Specific

Bypass/Kill can also be very useful in eliminating the glitch which occurs whenever you switch presets. (This happens on virtually all digital effect processors.) Instead of actually switching presets, try simply Bypassing 1 or more Units (using the Bypass/Kill feature). By doing this your sound changes instantly and the change is glitch free.

This ability to turn on and off each Unit individually (Bypass/Kill) is a DP/4 feature I use constantly. As a guitarist, I was used to having multiple footpedal effects on the floor in front of me when I played. With the DP/4 I can once again turn individual effects on or off (like distortion and delay) while leaving other effects (like reverb and chorus) on. This is especially a boon to those of us who want to improvise and use MIDI controllable digital effects. Now we aren't stuck to pre-programing the sounds that we are going to want during the course of an improvisation. Now I simply set up a few presets which each contain effects I am likely to use and turn on and of the unwanted effects (Units) when the need arises.

This individual Unit on/off approach is especially attractive when you consider one very nice feature about the Configuration Presets: Whenever you save one, all of the routing information and the information about which Units are on or off is saved with the preset. So if you save a Configuration Preset (#34 for example) with Units A and C on and B and D off, it will appear that way each time you recall it. By this method I can call up my chorus/reverb preset (#34) and, at the appropriate time in the song, turn on (Un-Bypass) my distortion which is Unit B and at then later turn on my pitch shifter which is Unit D, all without altering my reverb and chorus. All you have to do is define a MIDI (or DP/4 footswitch) message to control the Bypass/Kill for each Unit. (To do this refer to section 6-6 of the manual.) Personally, I use MIDI control messages because that allows me to Bypass each Unit from my MIDI footpedal unit. Depending on your setup you might want to choose something else. Also note that it is easy to turn on or off more than one Unit at a time by sending several MIDI messages to the DP/4 a the same time. On my footpedal unit I have one pedal assigned to Bypass each of the DP/4's four Units and one pedal to Bypass A and B simultaneously and one to Bypass C and D simultaneously. You should use whatever works for your application. *Experiment freely.*

I also recommend heavily utilizing the DP/4's ability to change parameter values in real time. This is probably the best way to keep your effect use sounding fresh and interesting. By using this feature you can change (for example) reverb decay times, delay times, eq settings etc. (See section 4-4 of the manual for information on assigning controllers to specific parameters within any algorithm.) The ability to crossfade between two algorithms is also immensely useful, creating a smooth transition between one timbre and another. (See section 8-5 and 8-6 on how to create these crossfades.) What the manual does not mention is that crossfades can be set up between any parameters that you want. That is, you could have a pedal turn up the preamp gain and turn down the high frequency gain in the Guitar Amp algorithm in Unit 1, while turning down the delay time and up the feedback in the delay algorithm in Unit B, while turning up the stereo spread of the chorus in Unit C, while turning down both the reverb time and mix level of the Large Plate Reverb in Unit D. All of this could be achieved by pushing one pedal. While this example might not be exactly what you (nor I) might actually do, it does express the idea that a huge (or very subtle) smooth (or sudden) timbral transition can easily be performed on the DP/4 by changing some of its parameters in realtime. Again, experiment.

Well, these are some of my more general suggestions to get more out of the DP/4. I hope you find them useful jumping off points in your own use and understanding of the DP/4. In the next article I will be getting into some more specific tricks about how to use realtime control on the DP/4.

And Now a Few Requests/Suggestions for the People at Ensoniq

1) Allow the option to use input voltages as controller sources. You use input voltages to alter the VCF in the VCF-Distortion algorithm. Why not make it a universal option selectable in the System/MIDI mode where the other controller sources are?

2) There ought to be a way to switch between a Config preset and a regular preset without having to touch the front panel. In fact, it's very necessary that I be able to do so. I use some sounds which I have saved as Config Presets right after or before ones I have stored as regular one-source presets. I have too many sounds to save them all in the 50 Config Preset spaces and I need to have instant, hands-off access to each of my sounds while I'm playing live.

3)There ought to be a way to have completely different processing on the left and right sides from a single sound source. I want to be able to effect a single source so that the right side has distortion and the left side is clean. Sure, there are a number of ways to try to trick the machine into doing this (e.g., distortion in series with a multitap delay, 100% mix on the delay, pan all taps hard right with 0ms delay time. Then do the same sort of thing to whatever effects you want on the left, running AB and CD in parallel). But none of the ideas I've had are at all satisfactory. Especially when it should be no large feat to put in a parameter that controls the output pan of the effect. It could work like a stereo channel on a mixing board; If set to center, the effect would be in stereo, if set anywhere else the effect would go to the output selected. This would *greatly* increase the pallet of sounds one could coax out of the DP/4. ▬

Bio: Vance Galloway is a Composer/Musician and Sound Engineer who does Mac based digital editing during the day. At night he collaborates with other musicians as well as with choreographers and multimedia artists. His own music, it has been said, has "little or no current commercial potential within the Music Industry as it now exists" which is just fine by him.

Suggestions, Hints and Tricks on Using the DP/4 Part Two

Vance Galloway

Last issue I requested of Ensoniq that they create some way to switch between a "regular" preset and a "configuration" preset without having to touch the front panel. Well, there always was a way via MIDI. All you have to do is send program change information on the MIDI channel that is assigned to control the type of preset you want to recall. For instance, suppose you have the Configuration presets set to respond to MIDI channel 5 and the One Source config presets set to respond to MIDI channel 1. If you are currently on One Source preset #33 and want to get to Config preset #21, simply send program change #21 on MIDI channel 5. When you want to change back, send #33 on channel 1. Easy as that. Be sure to avoid the mistake I made and check to see that you are sending on the right MIDI channel. Also see section 6-3 and 6-4 of the manual for some more information on how the DP/4 uses MIDI channels.

Last time I talked some about general ways in which to get more out of the DP/4. This time I'm gonna explain a couple of tricks I've figured out to help me with realtime control of DP/4 algorithm parameters.

You probably know that you can assign two of many MIDI or analog controllers to alter (or modulate, as Ensoniq puts it) any algorithm parameters. (If you are not familiar with this, see Part One of this article and Section 4-4 of the manual.) Sometimes, however, you might run into limitations with the DP/4's parameter Modulation scheme. What if you need to control more than two parameters per unit? Or what if you only have one source for continuous controller information (like the DP/4's Analog CV pedal, or a synth's mod wheel) but want to send different information at different times to two, three or four (or more) parameters. What if you want the same continuous controller source (again, like the DP/4's Analog CV pedal, or a synth's mod wheel) to have different minimum and maximum values at different points in a song?

In these cases you can assign one Modulation Source to alter which parameter the other Modulation Source controls. This means that you can assign, for example, your footswitch to control which parameter your footpedal is assigned to. Since this is not covered anywhere else that I've seen, I'll give an example.

What you will be doing is assigning the Mod 1 Destination (found on the EDIT page) to the parameter number that corresponds to Mod 2 Destination. (In this example I am assuming that you are using both the DP/4's Footswitch 1 and its Analog CV In as controllers. If not, either do so by following the directions in Section 6-8 of the manual or use ANY controller that works with your setup to achieve the same goal I am about to describe. No matter which controller you choose, the concept remains the same.)

Choose the Large Plate algorithm to be in Unit D by pressing EDIT then pressing the Unit D Unit Button. Then turn the Data Entry Knob until Large Plate appears. Next press the Left or Right arrow buttons until Mod1 Src appears (parameter 15). Using the Data Entry Knob choose Mod1 Src to equal Ftsw1 toggle. Right arrow scroll over to parameter 16 (Mod1 Destination) and set it to equal parameter 20 (Mod 2 Destination). You have just set Mod 1 to control the destination of Mod 2. Now set parameters 17 and 18 (Mod 1 Range Min and Max) to 3% and 9% respectively. Set parameter 19 (Mod 2 Src) to DP/4 Analog CV In. Don't worry about parameter 20 (Mod 2 Destination) since Mod 1 is controlling it. Set parameters 21 and 22 (Mod 2 Range) to 0% and 99%. Now scroll back to parameter 20 and press the DP/4's Footswitch 1. You will see that the Mod 2 Destination changes from parameter 1 (which is Mix) to parameter 3 (which is Decay Time). Now whenever you move the Analog CV In pedal you will be altering the mix or the decay time, depending on the status of the DP/4 Footswitch 1.

Also note that this same concept can be applied to multiple Units at the same time. So, in one part of a song, the Analog CV pedal could control distortion level and reverb balance and then, once you press the DP/4 footswitch 1, the same Analog CV pedal could be controlling distortion tone and reverb time.

You could also set up a footswitch to control the Range of the Analog CV pedal by assigning the footswitch to the parameter which controls the Analog CV pedal's Range. (In our example you would be assigning Mod 1 Destination to equal parameter 21 or 22 which are Mod 2's Min and Max respectively.)

Unfortunately, there does not seem to be a way to assign Mod 1 to control Mod 2's Source. This would be nice because then you could have, for example, reverb time controlled by Analog CV in during the verse of a song, then once you press a Footswitch 1 reverb time could be controlled by MIDI note number. If this is attempted the DP/4 simply ignores it.

The limitation of using a footswitch is that it can send only two values (on and off or Min and Max). Therefore (in our example) we are stuck with switching between only two parameters that will be controllable by the analog CV pedal. We could, however, assign Mod 1 Src to equal a MIDI continuous controller number. Each time we alter the value of that con-

troller, the destination of MOD 2 (which, as you remember, is being controlled by MOD 1) would change. This would have the effect of making Mod 2 control any parameter we want, but only one at a time. But remember that "one at a time" on the DP/4 really means "one at a time per unit." During the verse MOD 2 could be controlling one particular parameter per unit. During the chorus you could send a new MIDI continuous controller value from your sequencer (or whatever MIDI device you are using) to Mod 1. By doing this you will have changed the parameters MOD 2 is controlling. During the second verse you could send yet a third continuous controller value, thereby switching the destination of MOD 2 to a third set of parameters. This can be very tricky to set up however. Since each algorithm has a different number of parameters, it is difficult (and in fact sometimes impossible) to assign one single controller value to select the exact parameters you want to modulate on all four units simultaneously. I have found that this particular trick is best suited for selecting many different parameters of one or maybe two algorithms at a time. It all depends on your particular application and which algorithms you are attempting to use.

Note too that this trick could also be used to select between various MOD MIN or MAX values instead of selecting MOD destinations. To do this simply assign MOD 1 destination to equal MOD 2's MIN or MAX parameter. In the example above, this would mean assigning MOD 1 destination to parameter 21 or 22.

I hope that these little tricks and bits of advice on how to get more out of your DP/4 will be useful to you. They are really meant to serve as jumping-off points for your own experiments. Your own needs and applications are likely to be quite different from mine. Nonetheless the inner workings of the DP/4 remain the same no matter who uses it or what setup it is being used in. With a tool like the DP/4 it is particularly important to really understand how the device is set up. By doing so you are much more likely to get it to do exactly what you want it to do. Then again, the DP/4 is such a flexible device that there are probably things it can do that you (and I) have yet to realize. Therefore I also suggest that you experiment freely with it: Change effect routings, alter mix levels, change every single value of every parameter in your favorite preset, use a variety of input sources, set parameters to random values, substitute one algorithm for another... Play, have fun, and remember to remember exactly what you did to create that new effect that you like. ▬

Bio: Vance Galloway is a Composer/Musician and Sound Engineer who does Mac based digital editing during the day. At night he collaborates with other musicians as well as with choreographers and multimedia artists. His own music, it has been said, has "little or no current commercial potential within the Music Industry as it now exists" which is just fine by him.

So How Many Quadraverbs Does It Take To Match A DP/4?

Michael Harvey

When I heard the incredible rumors some 18 months ago about a new Ensoniq parallel effects processor, I placed an advance order with my dealer: four independent processors; chip designers and programmers recruited from Lexicon; true stereo effects; all from Ensoniq for under $2000. If the box lived up to even half its advance PR, it would be a deal at any price. To date, I haven't had a single cause to regret my impulsive purchase.

The release of the DP/4 immediately raised an innocuous seeming question: just how many Quadraverbs would one need to match the capabilities of a single DP/4. As with all truly important issues, however, reality proves far more intriguing and knotty than one might hope.

The DP/4, as its name suggests, contains four independent processors, each capable of stereo multi-effects. This description suggests that each of the DP/4's four processors equals a single Quadraverb. Unfortunately, the obvious solution is not necessarily the correct one.

The Quadraverb, as its name suggests, can deliver four simultaneous effects in series from a single processor. Recreating a comparable effects chain in the DP/4 requires two of its four processors. The debate still seems easily resolved, then: it must take only two Quadraverbs to match a single DP/4. This view gains support from a simple mathematical fact: two Quadraverbs provide four inputs and outputs, the same number of holes as a DP/4. Once again, however, the easy solution is incorrect.

Let's examine some other important considerations: sound quality, programmability, the variety of algorithms offered, and, not to overlook the obvious, price. A look at the spec sheet for each unit reveals the following:

	DP/4	Quadraverb
Freq. Response	2Hz-18kHz	16Hz-20kHz
Dynamic Range	96dB	85dB
Processing Power	40 MIPS	24 MIPS
DAC/ADC	16 Bit	16 Bit
Max Single Delay Time	6.4 sec	3.3 sec
Preset Memory	400 locations	190 locations

Significant differences exist in total processing power and delay times, with the DP/4 simply having more in each case. This should come as no surprise given the DP/4's multi-processor design. The DP/4 also offers more headroom than the Quadraverb, though in practice, the difference in dynamic range proves mostly insignificant. In truth, we will not find what we're looking for on the spec sheet.

With the help of two colleagues, I staged an old-fashioned A-B listening test. We used the dry sample sounds from Ensoniq's "DP/4 Demo CD" to test each unit's workhorse routines. The DP/4 outshined the Quadraverb hands-down in pitch effects, delivering crisp choruses and flanges, and very usable pitch shifts. The Quadraverb held some surprises for us, however. Widely disparaged as a noisy box, we were astonished to find that when its input and output levels were optimized, it was not appreciably noisier than the DP/4, despite its narrower dynamic range. And while its reverbs had a decidedly metallic edge in several algorithms, the Quadraverb's basic delays were difficult, if not impossible, to distinguish from those of the DP/4.

You may be wondering at this point why this article is appearing in the *Transoniq Hacker* rather than an Alesis newsletter. Let's settle the issue and move on to the areas where the DP/4 is truly unparalleled (even though it is a parallel effects processor).

The DP/4 offers incredible programmability of its effects — complex algorithms require 34 parameter pages! Whereas a basic reverb patch in the Quadraverb is defined in about ten button presses, the DP/4 is just getting going at this point, offering choices for room detuning, multiple early reflection levels, and decay definition. Similarly, a stereo delay on the Quadraverb offers (predictably) control over the length and feedback of the left and right delays. The DP/4 goes much further, allowing control of the panning for each delay along with cross-regeneration in which the output of the left delay can be fed back into the input of the right delay, and vice versa.

Both products offer excellent, real-time MIDI control of their effect parameters. Each supports up to eight modulation sources (pitch bend, pressure, continuous controllers, etc.) and provides (slightly different) schemes to route any modulation source to any parameter. The DP/4 also offers a few additional niceties like delays synchronized to tempos and MIDI clocks.

Turning to the effects themselves, the DP/4 offers a slew of algorithms that aren't available from a Quadraverb. These effects include Exciters, Duckers, De-essers, Noise Gates, a Vocoder, Speaker and Amplifier simulations, and Digital Distortion. As with its other effects, these DP/4 algorithms sound great. Some of the amplifier and distortion routines will take virtually any input signal and turn it into a wailing guitar sound of which Jimi (and Jimmy) would be proud. Imagine scat singing the solo to *Purple Haze* and sounding like the God of Grind!

"Real" guitar players will be similarly impressed by the DP/4. The unit provides an extra input jack on the front panel for a guitar cord. A dual footswitch attached to the rear panel provides a modulation source, an effect bypass switch, and preset selector. Effect chains can be programmed for performance situations letting a guitarist cycle through presets by using the footswitch as a stomp-box. These functions can, of course, be controlled via MIDI.

Unlike a Quadraverb, which sums and averages its input signals,

the DP/4 maintains discrete signal paths for the left and right channels, allowing it to perform individual processing on each, or combine the signal pair in unique and complex ways. One highly beneficial result is that DP/4 effects retain the stereo character of their source, resulting in spacious, crystal clear mixes.

The DP/4 can also combine the processing power of each its four units. (The Vocoder is an excellent and entertaining example.) Furthermore, routings between processors are programmable, allowing for any combination of serial and parallel effect chains. No matter how many Quadraverbs you own, you can never share delay memory, or set up complex internal mixing and feedback loops between the units. Even if you could, the signal would have to pass through multiple A-D and D-A stages between each unit. The DP/4, in contrast, handles all routing between its processors in the digital domain.

As a result of this flexible architecture, the DP/4 can be used as a single monster effect box, two stereo processors, or four individual effects processors. Regardless of the number of source signals, all outputs can be digitally mixed to a single stereo pair! It's like having a programmable digital patch bay, mixer, and four effects boxes in a single unit. For anyone working with a small mixing board, or driving toward fully automated mixing, this flexibility is an enormous boon and a tremendous value.

The above discussion might suggest that no number of Quadraverbs can match a DP/4. Before gloating too much, we should remind ourselves that the DP/4 also costs considerably more. The list price for a new DP/4 is $1,495. By Ensoniq's reckoning, this must mean a DP/4 is worth about three Quadraverbs. Is that the answer to our question? Based solely on price, I would argue that the two products feel more or less fairly valued. Nevertheless, the DP/4 simply does things the Quadraverb cannot, making a simple division of one into the other impossible. Each musician must personally assign a value to the revolutionary capabilities and design of the DP/4.

Let me close with an interesting conundrum. Due to the DP/4's capabilities, it becomes unbearably tempting to use its full power for every processing chore. I cannot overstate the anguish of having to use "only" two processors on a lead vocal so that the other two can be used for house reverb. Perhaps I shouldn't be asking how many Quadraverbs match a DP/4. Perhaps I should be asking how many DP/4's I need to live up to the incredible potential this machine opens up to regular Joes like you and me. ▬

Bio: Michael Harvey earns his living as a product manager in Microsoft's Consumer Products Division. He spends his living as the owner/operator of Moose Leap Corner Studios, a Seattle project studio specializing in song demos, audio for video, and multimedia software.

DP/4 "Hackerpatches"

Dave Kelly

Patch: Pebbles

This is a 1-source, 4-unit patch I designed to be used as a vocal patch and on snares. The chorus and flanger provide a sense of smoothness to the patch, especially on snare. If you want to use this with guitars, I suggest lowering the values of Parameters 3 and 4 on the chorus (LFO rate and width). The LFO rate and width on the flanger add a slight tone change when used on snares. If you don't like this effect, lower the value of these two parameters. On the two hall reverbs, if you want a more metallic sound, lower the values of Parameters 11 and 12 (room detune, rate and width). Increasing the values of these two parameters will appear to give a smoother, more pleasing decay when used with snares, but increasing them too high could make vocals sound out of tune. Experiment with Parameters 8 and 9 on the halls. Lowering the values of the diffusers will create a coarser decay which can still be very pleasing. Increasing the pre delay times can create an echo type effect which can sometimes be useful. Altering the times of the early reflections can drastically change the tone of the sound. Experiment. Changing Config Parameter 01 to serial will make the patch sound more distant.

Patch: Owl's Eyes

This is another 1-source, 4-unit patch. I designed it to be primarily used as an ethereal guitar patch but have also used it on vocal phrases to create some interesting effects. I recommend altering the delay times of the phasers to correspond with the tempo of your music. The pitch shift and chorus patches are pretty standard but they add a sense of smoothness. If you prefer a more natural tone, lower the values of Parameters 3 and 4 on the chorus and Parameters 4 and 8 on the pitch shifter. This will have a subtle, but still noticeable, effect on the sound. With the phasers you can get as weird as you want. If you want an underwater sound, increase the values of Parameters 3 and 4. Altering the value of Parameter 6 in the phasers can produce some strange feedback effects. Changing the value of Parameter 12 on the phasers will drastically change the way the delays are perceived. The tones you can create from this type of patch are endless. Enjoy.

Name: Pebbles

Use: Vocals / Snare / Ambience

Alg	8 Voice Ch	Alg	Hall Rv	Alg	Flanger	Alg	Hall Rv
01	85	01	58	01	46	01	68
02	99	02	99	02	99	02	99
03	19	03	2.86	03	13	03	3.08
04	47	04	3	04	06	04	4
05	99	05	+07	05	01	05	+06
06	00	06	37	06	00	06	38
07	0 ms	07	49	07		07	51
08	0 ms	08	93	08		08	74
09	00	09	81	09		09	99
10		10	35	10		10	33
11		11	44	11		11	14
12		12	48	12		12	20
13		13	+40	13		13	+37
14		14	12 ms	14		14	14 ms
15		15	14	15		15	13
16		16	18	16		16	12
17		17	29 ms	17		17	47 ms
18		18	30	18		18	17
19		19	31	19		19	29
20		20	+94	20		20	+78
21		21	+49	21		21	+62
22		22	+29	22		22	+34
23		23		23		23	

Notes: Configs: (00) 1 (01) Par (02) Ser (03) Ser (04) 00 (05) 00 (06) Mono (07) b (08) b (09) b (10) b

Name: Owl's Eyes

Use: Ethereal Guitar / Strange Vocal Ambiences

Alg	8 Voice Ch	Alg	Phaser/DDL	Alg	Pitch Shft	Alg	Phaser/DDL
01	50	01	99	01	89	01	99
02	99	02	85	02	99	02	99
03	09	03	12	03	00	03	26
04	18	04	23	04	-10	04	41
05	99	05	+09	05	99	05	+03
06	00	06	+07	06	-99	06	+56
07	30 ms	07	+99	07	00	07	+93
08	65 ms	08	Out	08	+10	08	In
09	02	09	Off	09	99	09	Off
10		10	250 ms	10	+99	10	250 ms
11		11	500 ms	11	Long	11	500 ms
12		12	00	12	02	12	-72
13		13		13	02	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	

Notes: Configs: (00) 1 (01) Ser (02) Ser (03) Ser (04) 00 (05) 00 (06) Stereo (07) b (08) b (09) b (10) b

Adjust delay times on phaser to match song tempo.

Using the DP/4 with a MIDI Controller Keyboard — Part 1

Steve Byhurst

The DP/4 is ideal for musicians like myself who love putting sounds through weird and wonderful combinations of effects. Even those of us who have synths with on-board processors yearn for more power and flexibility, and here the DP/4 certainly delivers. The combination of a keyboard like the VFX-sd working with a DP/4 broadens your musical horizons.

Amongst the sort of things you can do with this setup are: split separate outputs from the synth and add effects to dry or processed sounds; send any one or a combination of synth outputs through one or more of the DP/4's units; control effects parameters in real-time from the keyboard; choose effects from the keyboard's control panel; and automate selection of effects from the sequencer.

This article concentrates on the sort of uses I make of this combination of equipment with the aim of making the DP/4 an extension of my controller keyboard. I use a VFX-sd and therefore the synth details are based around using an SD, although many of the comments will also apply to using other keyboards.

Basic Connections

Before we consider how to configure the SD and the DP/4 to work with one another, let's begin with the basics — MIDI and signal connections. MIDI-wise, as the synth will be doing all the controlling, we simply need to connect the MIDI Out of the keyboard to the MIDI In of the DP/4, perhaps via a MIDI patchbay or MIDI thru unit.

Signal routings can be configured in many different ways according to your own particular setup, but I shall describe my own as an example. I send all four of the main and aux synth outputs to four separate channels of a mixer, and then route four mixer auxiliary sends to the four DP/4 inputs. This gives me the flexibility to send any of the four SD outputs (and any other mixer inputs) to any of the four DP/4 inputs. Signal levels should be set using the guidelines in your manuals.

SD MIDI Configuration

There are a few SD parameters which need to be checked, and these are found on the System MIDI Control page. The first sub-page features the SEND-CHAN parameter, and this should be set to TRACK. On the second sub-page is CNTRLS, which should be turned ON, and PROG-CHG, which should be ON or NEW. This will make sure that the SD can send controller and program change data to the DP/4 from each track of a preset or sequence. Other controller keyboards should offer similar MIDI parameters.

DP/4 MIDI Configuration

The DP/4 has a comprehensive MIDI implementation which means it can be difficult to decide how to set it up. Here I'll describe a basic configuration to use as a good starting point, but remember that there are usually several ways of accomplishing the same result with the DP/4 due to its inherent flexibility.

There are a number of parameters specific to each effects unit and we shall look at those soon, but first of all let's check out the global parameters. Press the System MIDI button and go to parameter 35. This specifies the channel number of the Control Channel which is the only one able to receive modulation controllers. Give it a number and use parameter 36 to enable it.

Parameters 37-44 specify the eight system controllers you can use as modulation sources. Each effect algorithm allows you to use two of these possible eight to control modulation of any two effect parameters. The controller data is received by the channel defined in parameter 35. Choose the eight controllers you think you would use most when modulating effects from your keyboard. Remember that you may want to keep those you regularly use for synth performance, like velocity and pitch bend, separate from effects controllers. In some cases however, using the same one for both purposes can give useful results (velocity controlling reverb depth on a piano sound for example).

Finally, go to parameter 53. This acts as a master switch for program change reception and we need to turn this ON.

Now the unit specific commands. The System MIDI button accesses five sets of identical parameters, one for each of the four effects units A, B, C, and D, and one for config presets. You can quickly get to the first parameter of each these by pressing the relevant unit or config button. The parameters we are most interested in are the first three in each set. MIDI Channel is the first. If you have them free give each unit a different channel, otherwise give them all the same one. Allocate configs the same number as the control channel, but make sure that this number is different from the number(s) you have given to the four units. Set the second parameter, MIDI Enable, to ENABLED, and the third, Program Changes, to RECEIVED. We will consider the remaining unit parameters, and how to use MIDI control of the individual units, in part two.

Configs

Configs provide the key to getting the most out of the DP/4. Each one stores everything the DP/4 needs to know about which inputs to use and whether they are mono or stereo, how those inputs are connected to each of its four effects units, which effect algorithm is used in each unit, how the units are connected to each other, and which outputs are used. Phew! As one simple program change command from your keyboard can trigger an entirely different DP/4 setup, you can see how powerful a feature this is.

Which config we choose is crucial to what will happen to our sounds once they are received by the DP/4. Let's review the dif-

ferent types and see what they can do for us: —

1 Source: Useful if we want to give one sound a really complex effect as all four units will be used. We would have to use this type if we wanted to use the Vocoder as it is a ganged 4U effect. Any other sounds would have to be dry, use on-board effects, or use other effects processors. We could also set up alternative algorithms in each unit and use the bypass commands to select different effects and combinations of effects. Input 1 is used for mono signals, Inputs 1 and 2 for stereo.

2 Source: This config could be used to put the SD stereo main outs through units A and B whilst sending the stereo aux outs through C and D, giving us two-unit processing for each pair. Useful if both pairs have stereo sounds which need to be processed separately. Also good if 2U effects like the 3.3 second delay are required. All four inputs are used if configured for stereo, Inputs 1 and 3 for two mono sources.

3 Source: With this one we could send separate dry mono signals from Aux Left and Aux Right outputs to be processed by A and B respectively, and send the stereo signals from the main outs to C and D for stereo processing. Inputs 1 and 2 are mono, 3 and 4 are the stereo pair inputs.

4 Source: This is really suited to giving each of four mono signals different effects and can be a really powerful setup. Remember that although the four inputs are mono, the outputs can be a combination of four stereo signals. Also, this config could be used in a similar way to a 3 Source by sending a stereo signal to any two units sharing the same effect algorithm.

The DP/4 comes with config templates (presets 53-60) featuring all the various alternative uses of sources, inputs and outputs already programmed, and these can be used to create configs which particularly suit your own way of working. It is a good idea to program a few configs of your own from scratch, rather than just editing presets, as you then gain a better idea of how they work.

Output Routing

Now that we have a basic configuration, let's start to look at the sort of things we can do, from first of all just processing sounds played from the keyboard, to remote control of the DP/4 from the SD, and then on to automation of that control from the sequencer.

First of all, check where voice outputs are being sent. SD programs usually send all six voices to the main outs via the on-board processor. If we want dry sounds, and/or need to use the aux outputs, we have to edit the program. The SD has four outputs configured as two stereo pairs. Main Out L&R can send dry and/or effected sounds, but Aux L&R can only send dry signals. Each of the six voices in a program can be directed to any of the four outputs using commands on the Output page within the programming section.

The second Output sub-page has the DESTINATION BUS parameter which chooses where stereo bus signals go and whether they are dry or not. Select FX1 or FX2 and a wet signal will use the main outs for that voice. DRY will route a non-effected signal to the main outs, or select AUX to send a dry signal to the aux outs. We can use the PAN parameter on the same sub-page to send signals for mono processing. A value of 00 will send the voice to the left, 99 to the right. By using these commands we can choose the effect content and routing of sounds before directing them through the mixer to the DP/4.

Preset Control

We can now examine some different methods of controlling DP/4 parameters from the keyboard via MIDI. Let's start by looking at SD presets.

The advantage of using presets rather than individual programs is that there is no need to edit originals for separate output use, and therefore you can keep your favourites intact. Also, we can make full use of the performance parameters which presets supply. Select a preset and press the EFFECTS button in the performance section.

The first EFFECTS sub-page gives us the FX BUS OVERRIDE parameter for each track. CNTRL is the default which leaves all output routings as programmed in the Output page, but you can force all voices to DRY, AUX or FX1/FX2 if you wish. Note that if you do this you lose separate processing for each voice. Subsequent sub-pages allow you to change the algorithm and its parameters. The PAN page lets you keep the original routing by programming VOI, or you can change it using the same settings the Output page provides.

Make one of the tracks dry and press the performance MIDI button. This provides three parameters which we can use to control the DP/4. Set STAT to BOTH and CHAN to your chosen config/control channel number. If the mixer is set up correctly, we can now play that sound and hear it processed by the DP/4. The third parameter, PROG, can be used to select different configs until we find something suitable. SD users should note that program numbers 001-100 access DP/4 preset numbers 00-99, so the two displays will always be out by one.

Apart from using MIDI to change configs, we can use controllers like the modulation wheel to alter effect parameters as we play. This is possible because the config and control channels share the same number, and so both types of MIDI data can be received. We can choose what effect the controller information has by using each unit's modulation parameters. Once the SD preset is saved, the chosen config will be loaded every time it is selected.

Presets give us the ability to remotely control the DP/4, but there are many more possibilities to explore if we use sequencer control, and that is exactly what we will do in Part Deux. See you then! ▬

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Using the DP/4 with a MIDI Controller Keyboard — Part II

Steve Byhurst

Last time, we set up our SD-DP/4 system with a basic configuration and then looked at how we could use presets to control the DP/4 from the SD. Now let's see how we can use the sequencer to automate control and try out various other applications that this combination allows.

Sequencer Control

Suppose we've recorded an eleven-track masterpiece with the sequencer and the tracks are a mixture of internal programs (some using internal effects, some not) and external sounds. We now have to decide the processing requirements for each track.

If no processing by the SD or DP/4 is needed, we can route dry tracks straight into the mixer where they can be left as they are or sent to an alternative effects unit if required. Internal effected sounds can be directed via the main outputs to the mixer and onward to the DP/4 for radical treatment, or left at the mixer if on-board processing is enough. This leaves unprocessed sounds from the SD or other sources which require DP/4 effects. These dry outputs are simply routed to the DP/4 by the mixer.

Cast your mind back to part one and you might remember that we used performance parameters to control what happened to the outputs of the three preset tracks. We can use the parameters of the twelve sequencer tracks in exactly the same way. The EFFECTS and PAN buttons give us the same choices as before to control which output a program's voices will be sent to and whether they go through an internal effect or not. The MIDI button also gives us the same choices as presets regarding status, channels, and programs, but the extra number of tracks allows us to specify a spare non-recorded track (number twelve in our example) as a DP/4 control track. Give this MIDI status and allocate it the same channel number as the config/control channel of the DP/4. You could specify a control track like this in any SD templates you may already have.

The next thing to do is to make sure that each recorded track is set up correctly. Any tracks using internal sounds should be programmed as STATUS=LOCAL, and tracks using external sounds should be set to STATUS=MIDI with the relevant channel and program numbers. Also check that the mixer is configured to send outputs to all the right inputs and that you have some basic config presets ready that you can tweak later.

Similar to the way we used presets in part one, we can now select a config for the sequence by using the PROG parameter, and, once the sequence is saved, that config will be recalled whenever the sequence is selected. It is also possible to change the config as the sequence progresses by recording a program change event at any chosen point. Bear in mind that there will be a delay and mute of output if the config you change to is significantly different.

MIDI Controllers

Having chosen our config(s) for the sequence, we might now want to use MIDI controllers like pitch bend or aftertouch to alter some of the algorithm parameters. First of all, make sure that the presets used by configs are programmed to correctly receive the type of data you want to send. Use the Mod1/Mod2 preset parameters to do this.

The DP/4 only receives controller information on the single channel we have specified, so we need to record this data on track twelve. Using record-mode ADD, we can adjust controllers whilst the recorded tracks play back and have those adjustments recorded on the track. An example would be using the modulation wheel to adjust effect volume at appropriate moments during the sequence. These changes will be then be sent whenever the sequence plays. By keeping all of this control data on one track we save it from getting confused with any other data.

An alternative to this method would be to record combined note and controller information on the same track. To use internal programs we need to set the track's MIDI status to BOTH and make its channel number match the config/control number. A config can be chosen using PROG, notes can be recorded, and controller events can either be included as you play or added afterwards. However, there are drawbacks with this method because any changes made to the track will affect both the internal program and the config. This means that you cannot change a config without also selecting a different internal program, and, any controllers used will affect both the program and the config if they are both set up to use them. These problems can normally be overcome by some re-programming of either the SD or the DP/4, depending on what is needed.

It is not possible to use tracks which access external sound modules to control the DP/4 as well. These are normally set to the module's exclusive channel number and will send commands for its use only.

Bypass Commands

A very useful DP/4 feature is the ability to bypass individual units. A bypass command can either send the input signal around a unit and onto the next one it feeds, or stop the input dead at a unit allowing it to go no further. The DP/4 calls the

first option Bypass and the second Kill. We can select the type of bypass mode for each unit when we program a config.

One way that these commands can be controlled from the keyboard is by employing the seventh unit specific MIDI parameter, Unit Bypass. This selects one controller to function as a bypass switch for that unit. There are many to choose from but some are more useful than others. Switches are better than continuous controllers, and avoid volume, pitch bend, note velocity and aftertouch. These are difficult to use and can cause unexpected results. Remember to choose ones that can be easily turned on/off and that do not clash with any used for sound modulation purposes. For SD users, the goodies are mod pedal (#4), sustain pedal (#64), sostenuto pedal (#66) and patch select (#70). We can use track twelve again to record the bypass controllers as although different controllers can be programmed for each unit, all data is received on the control channel and not on unit channels. By using record-mode ADD, we can playback the sequence and record the relevant controller information to bypass/un-bypass units as required. This method can be utilized if there are problems with using config changes due to an unacceptable sound delay/mute, because bypass commands have an almost immediate effect.

Individual Unit Control

So far we have only used one track/channel for control of the DP/4 from the sequencer. This gives us the advantage of having preset change, modulation, and bypass functions, all contained in one place using only one track. For general use this setup is fine and it gives us a convenient, easy way to use the two pieces of equipment. However, if additional control is required we can use extra tracks together with some more DP/4 MIDI functions.

The DP/4 can actually receive up to six different MIDI channels, one for each of the four units, one for the config presets, and one for controllers. If six tracks are available, each one could control each of the six channels, but it is unlikely that all of these would be necessary. Much can be achieved with three or four tracks, as the config and control channels can almost always share a track with no loss of flexibility.

Certain benefits are obtained by giving each unit a different MIDI channel. As well as choosing a config for a sequence, we could also select the preset(s) it uses with program change commands. The type of config decides which unit a command should be sent to and what type of preset will be chosen, as presented below:

1-Source Config: A command sent to any unit will select a 4U preset. **2-Source Config:** Commands to A or B will select the first 2U preset, to C or D the second 2U preset. **3-Source Config:** Commands to A will select a 1U preset, to B a 1U preset, and to C or D, a 2U preset. **4-Source Config:** Commands to each unit will select individual 1U presets.

This shows that it is only necessary to use a track per unit when unit control of a 4 source config is required. Make sure that a config is chosen before presets are loaded by specifying the config control track before the preset control tracks. This is important because commands are sent in track number order by the SD.

We can also use program changes to control bypass commands. If the fourth unit specific MIDI parameter, Program Change Map, is set to ON, a map editor can be accessed using the next two parameters. This map translates program changes to preset numbers and bypass commands. The default makes program changes 001-100 select presets 00-99, 101 bypasses, 102 kills and 103 un-bypasses, but any number (or even multiple numbers) can be chosen to trigger bypass commands.

When using the one track per unit method, modulation controllers for unit algorithms would still need to be sent on the control channel, but if DP/4 global MIDI parameter #57 is set to ON, each unit's volume can be modified by sending controller #7 data from the sequencer. Instant MIDI-controlled effects mixing!

Another possibility would be to give all four units the same channel and send commands from one track set to that channel. By using the program mapping feature, a program number could be translated to select different presets, and/or bypass commands, for each unit.

You may find that when trying out different ways of doing things with the DP/4, flexible as it is, there are trade-offs between the different methods available, and I have tried to mention some of these problems as examples. Usually there are ways around these, but the solutions may then affect other parts of the setup. Ultimately, once the available options are known, the user must decide which method is best for any given situation.

Saving To Disk

Although the 400 presets available in the DP/4 offer the user a fair degree of variation, once you start creating your own presets you may soon find yourself running out of space. I find that the 50 RAM config presets in particular get used up very quickly, as I tend to save each time I make a significant change. Thankfully, with a keyboard like the SD available, we can dump DP/4 data down to disk via the wonders of MIDI System Exclusive.

Connect the DP/4's MIDI Out to the keyboard's MIDI In (just swap the leads around if you don't have extra leads or a patchbay). Press the DP/4 System MIDI button, followed by the Write Copy button, and then choose the type of dump required. We can choose from individual presets, banks of 50 RAM presets, all 200 RAM presets, system parameters, or all RAM presets with system parameters. Make sure that the controller keyboard is ready to receive and then press the DP/4's Write Copy button again to transmit.

I like to save all information for a particular piece on one disk. This could include sequencer data, DP/4 preset and system parameter data, and data for other MIDI modules used. All this can then be re-transmitted later and the whole setup will be ready for playback!

Before sending Sys-Ex data back to the DP/4, make sure that it can receive it by checking that system global parameter #51 is set to Enabled. Reception of data is automatic, but when the transmission is completed the display will say what has been received and whether there was an error or not.

That's about it! The procedures and uses I have set out here are just some of the ways I have discovered of using a controller keyboard with the DP/4. The amount of flexibility involved means that this can only be the start, so why not experiment further? I would be pleased to hear from anyone who would like to share their own way of doing things. Until next time... ▬

Bio: Steve Byhurst is a British composer of electronic-based instrumental music. He is an aspiring soundtrack writer who would love to achieve the seemingly impossible — to make a living from the results of using his (mainly USA-made!) gear. You can write to him at 1 Oaklands, Oakhill Road, Horsham, West Sussex, RH13 5LG, U.K.

A DP/4 Application

Ray Legnini

As many of you know, the DP/4 is an extremely versatile sound processing device. With the ability to be configured in a variety of ways, it can be used to simulate screaming guitar amps, create a variety of ambiences ranging from a train station to a concert hall, add silky smooth chorusing to a recorded keyboard track, or tame the dynamics of a vocalist, just to name a few.

One effect that seems to go unnoticed in the DP/4 is the 4-unit Vocoder preset. In fact there are two of them set up for you as it comes from the factory. They are Config Presets #61 and #62. The vocoder is the effects patch you hear creating the popular "robot speech" or "talking keyboard pad" effects. Basically, the frequency spectrum of the input to the vocoder (plugged into DP/4 Input #1) is analyzed and that signal is then used to alter the sound coming from a second source (most often a sustaining keyboard sound patched to DP/4 Input #2). The new sound is then sent to the stereo outputs of the DP/4. The signal from the second source, patched into Input #2, is called the "carrier." The carrier's sound will only be heard when the trigger input signal is present at Input #1. This means that although you may be holding down a sustained chord on your keyboard being used as the carrier, if your input is of a rhythmic nature, the output signal will be rhythmic as well. For more theory on how and why the vocoder does what it does, and the editable parameters available, consult the *DP/4 Musician's Manual*, Section 5.

We'll presume you have a working knowledge of selecting the DP/4 effects algorithms, its Config. structure and how to hook its stereo outputs #1 and #2 up to your audio system. What you'll need for this little tutorial is a DP/4 hooked up to your audio system, a CD player, a mic (either high or low impedance), and a keyboard. Optional items could include an electric guitar, a drum machine, sequencer, turntable, cassette, or a DAT machine.

While there are many ways to configure your audio system, we'll presume also that you can route the signal coming from your microphone or other sound source we'll need to use as a trigger directly into Input #1 of the DP/4. Without going into too much detail, one possibility would be routing the signals through your mixer's submix bus, effects send or aux. send, and patching that directly to Input #1 on the DP/4, so that the original signal is not heard through the mixing console. You can use either of the DP/4's Input #1 jacks, front or rear; they function the same.

Patch your keyboard's output directly to Input #2 of the DP/4, instead of into your mixer. It will become our carrier. For starters, set the input volumes on the DP/4 to 12 o'clock for Inputs #1 and #2. Set the output levels of the DP/4 to maximum. Once you have everything hooked up, it's time to get down to business. Select a sustaining patch on your synth. String pads or vocals are a good starting point. While holding down a chord on the synth, speak into the mic. If you have everything working properly, you should hear the synth play in the same rhythm as your speech. Experiment for a while with different speech patterns, singing "aah," "ooo," or "ooh," etc. to get used to the possibilities. Okay, so now you've heard everything that's ever been done with a vocoder. But wait! There's more! Now for the really fun stuff.

Since the input source is not heard at the outputs of the vocoder, almost any sound source can be used to create really interesting sonic textures. Let's explore a few of the possibilities here. Grab your CD player and plug its output directly into the DP/4's Input #1, instead of the mic. Get a favorite CD spinning and play some chords on the carrier keyboard. Based on the harmonic and rhythmic content of the CD you've chosen you'll get some way cool sounds happening. I experimented with a bit of Aerosmith, some acoustic guitar music, a dance track, a drum loop sampling CD and some sound effects. If your CD player can repeat a small section over and over, you'll get very predictable results and some really cool rhythmic effects. Now put your multi-track in record and add some spice to your next hit song. This same technique can be used if your input source is a cassette, a DAT, a track already recorded on a multi-track master, or a turntable, as well. Guitar players should experiment with plugging a guitar, with or without effects, into Input #1 and using it as the trigger source. Instant James Brown strummed chords meets violin section....hmmmm. What!, no free hands left to play a chord on your keyboard? Use some masking tape to hold down a chord or record a part into the onboard sequencer first.

Of course, you should also experiment with different types of patches on your synth. Although sustaining patches have been the staple food of vocoders over the years for people like Brian Eno and Herbie Hancock, I've had good results with pizzicato violins, nylon guitar, piano and synth bass.

Now, for a DP/4 tweak to go along with the CD as the input source setup: The DP/4 vocoder algorithm has a setting

called "Sibilance Level" (parameter #04). This is used to allow some elements of speech through the vocoder directly to the outputs to help improve intelligibility when using a mic as the input source. But, when using a sound source from a CD like a rock or dance track, you may not want bits of vocals and percussion sneaking into the mix, especially if they're in a different key. They can be turned down or tuned out completely by lowering the value of the Sibilance Level parameter. Remember, the vocoder is made up of four units tuned to various portions of the audio spectrum, so you may want to experiment with this particular parameter in each of the four units to achieve the effect that fits your application.

But wait, we're not done yet! You say that you like the effect of playing a drum loop or track through the input but wish it were more predictable, even sync-able to the song you're working on? Hook up your keyboard workstation's sequencer to be the input source and record your own rhythmic vocoder food. This way you can sync the vocoded synth parts up to a tape track with SMPTE or MIDI clocks. How? For my example I'm using a TS-10, but any workstation synth that is capable of routing separate signals to its outputs will work. Start by recording a drum or percussion pattern on one track of your sequencer. Assign this track to come out of the auxiliary output on your synth and patch this directly into the DP/4's Input #1. We'll use this to trigger the carrier signal coming from the same synth! Assign a synth pad to another track on the sequencer, but you don't need to record anything (of course you could record some chords and leave your hands free to do tweaks or play another instrument). Have this synth pad track come out of the main output of the synth and patch it into the DP/4 Input #2 (if you've been following this tutorial all along, the synth should already be plugged in there).

Start the sequencer playing and then play some chords on the carrier synth. Sounds cool, huh? Want more? Try changing the effects on the percussion track. Try guitar amp simulators, phaser, delays, etc. With a little bit of experimenting you'll open up a whole new world of sounds. I hope you enjoy your new-found sonic toolbox. ▬

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Putting the DP/4 to Work

Setting Up

Michael Harvey

In an industry addicted to new products, the *Transoniq Hacker* provides a unique forum in which to discuss established pieces of gear. In that spirit, this article kicks off covering day-to-day use of the DP/4 Parallel Effects Processor in home and project studios. We'll start basic but accelerate rapidly, examining each step in a typical recording project — tracking, mixing, mastering — and how the DP/4 delivers unprecedented ease-of-use and professional sound to engineers on a budget. Due to the power and flexibility of the DP/4, even small studios can be used to obtain completely professional results. Let's begin.

For discussion purposes, I'll assume a basic multitrack studio set-up with MIDI capabilities... and, of course, a DP/4. The goal of our hypothetical project is to record a high quality, pop/rock song demo suitable for submission to an A&R rep or inclusion on a self-published CD. The project could as easily be audio for video, a jingle, or a multimedia production.

We will use the DP/4 for two basic tasks during our recording project: as an effects processor and as a signal processor. Until now, at least two distinct pieces of gear would be required. The DP/4's flexible design, however, lets us actually use a single device for both chores simultaneously. To do so, we must first configure the DP/4 appropriately. Config presets on the DP/4 are very powerful: each defines a routing scheme, effect algorithms, and mixing setup for the entire DP/4. By changing Config presets, the DP/4 can morph — at the touch of a button — from a four-unit vocoder into four separate stereo reverbs. Since they are the key to tapping the full power of the DP/4, it is a good idea to get used to working directly with Config presets from the very start.

Your DP/4 ships from the factory with most of its Config presets disabled, a situation you will quickly want to rectify. With the DP/4 powered up, press the SYSTEM /MIDI button. Use the left/right cursor buttons to scroll to parameter #59. (Hint: The DP/4 provides many shortcuts to reduce needless scrolling through parameter pages. In this case, pressing the SYSTEM/MIDI button repeatedly cycles through groups of system parameters.) If you haven't reset this parameter yet, it will read: "Show 100 Config Presets=No." Use the data entry knob to change the parameter to "Yes." Press the SELECT button, then press the CONFIG button.

The jog wheel will now scroll through all 100 Config presets. Let's pick one that meets our requirements. Since we want to operate the DP/4 as two virtual devices (as both an effects processor and a signal processor), we will essentially split the DP/4 in two. Scroll to preset #52, "2 Src: Stereo In," and press the SELECT button. This Config is now loaded. In the LCD display you will see that there are no effect algorithms selected; each unit is "dry." Furthermore, since we selected a 2-source Config, the four units of the DP/4 are divided into two distinct groups: units A and B form one group, while units C and D form a second group. The processors within each group are connected in series, as indicated by the small arrows in the LCD display. The groups themselves, however, form two completely distinct signal paths. Finally, since we specified stereo sources when we selected our starting Config, input jacks 1 and 2 (on the rear panel of the DP/4) will feed the left and right inputs of group AB, while input jacks 3 and 4 will feed the inputs of group CD.

We've achieved our goal of splitting the DP/4 in two. We don't have to give up the advantages of owning a single box, however. The output path of the DP/4 will be determined by how we connect it to our system. The DP/4 actually senses

how many cords are patched into its output jacks. If we connect cables only to output jacks 1 and 2, then both pairs of stereo outputs (one from group AB, another from group CD) will be automatically and digitally mixed within the DP/4 to appear as a single stereo pair at output jacks 1 and 2. (This feature is very useful if you have limited mixer inputs or effect returns, for example.) If instead we connect cables to all four output jacks, the output of group AB will appear on output jacks 1 and 2 as a stereo pair, while the output of group CD will appear on output jacks 3 and 4 as a second stereo pair.

We will be tweaking this preset shortly; giving it a new name will help us keep track of our work. Since we will be giving the DP/4 a dual personality, let's call the patch "Schizoid." (The DP/4 owner's manual provides a straightforward explanation of how to name and store presets as well as retrieve factory patches. Don't overlook the handy shortcuts for avoiding unnecessary scrolling.) When you save the renamed preset, you will see the message "***Write OK***." "Schizoid" is now loaded as the active preset.

To use group AB as an effects processor and group CD as a signal processor, we must patch input/output jacks 1 and 2 on the DP/4 into our mixer's effects loop, and input/output jacks 3 and 4 into one of our mixer's insert points. (If your mixer does not provide insert points on its main buss or subgroups, simply patch a pair of your mixer's subgroup outputs (or even its main buss outputs) directly into input jacks 3 and 4 of the DP/4 and then patch output jacks 3 and 4 from the DP/4 into your multitrack. You can then monitor the processed signal via the tape returns from your multitrack. You must be very careful not to inadvertently create a feedback loop if you take this latter approach. Keep monitoring levels low until you're sure the patch is correct. However you end up completing this patch, the goal is the same: to put processing group CD in-line between your mixer and multitrack so that the DP/4 processes the entire signal before it reaches tape.)

There is one final housekeeping step to perform before recording: calibrating input and output levels. To obtain the best signal-to-noise ratio for recording, you must optimize the inevitable trade-off that occurs between the noise floor of your system and its available headroom (also known as distortion when you run out of it!). The optimal situation occurs when your system achieves a state known as "unity gain." In a nutshell, this means that an input signal of 0db remains a 0db signal as it passes through the various gain stages (EQ, master faders, DP/4 inputs and outputs) of your system when those gain stages are in a "neutral" position.

We will start calibrating levels with the DP/4's inputs and output levels set to zero (knobs fully counter clockwise). We will also leave all the DP/4 processing units set to "dry," eliminating any variance that might arise from the way the DP/4's effect algorithms themselves impact the source signal's level. First, we require an input signal. If you have a sequence ready, go ahead and play it. The demo sequences that come with many synthesizers will also work. Set the channel input trims and/or preamp gains on your mixer so that the input signal hovers around 0db on your mixer's channel meters with the channel faders in their 0db position. (You may need to adjust the output level at the sound module itself to achieve this.) Now, turn the aux send faders feeding inputs 1 and 2 of the DP/4 to their 0db setting (usually the 12 o-clock position). You are now ready to turn up the input knobs on the DP/4 itself. As you rotate input knobs 1 and 2 clockwise from their fully off positions, you should see the green input LED's on the DP/4 begin to flash, registering the input signal. Keep bringing up the level until the red clip LED's start flashing. Now, back off the level just a tiny bit. Your input signal is optimized when there are intermittent, momentary flashes of red. Sustained or frequent red flashes indicate clipping. If your levels are correctly set, you should find the optimal position for the DP/4's input knobs somewhere around the 12 o-clock position. If this is not the case, check all the gain stages prior to the DP/4 until you find the culprit(s).

We can follow a similar procedure for the DP/4's output levels. Raise the level of the aux return fader being fed by the DP/4's outputs to it's 0db position. (Most mixer aux returns are set up as stereo pairs with a single level fader or knob.) Solo the aux returns on your mixer so that you hear only the signal coming from the DP/4. Now, rotate output knobs 1 and 2 on the DP/4 clockwise to raise its output level. You will begin hearing your source signal, dry since the DP/4 is not yet applying any effects algorithms. Stop increasing the DP/4's output level when the signal is again hovering around 0db on your mixer's meters. This will probably occur when the DP/4's output knobs are somewhere between the 12 and 3 o-clock positions.

If you wish, you can extrapolate from the procedures above to set the remaining input and output levels on the DP/4. The shortcut, of course, is to set knobs 3 and 4 in the same positions as knobs 1 and 2.

Congratulations, you have now optimized input and output levels in your system, at least as far as the DP/4 is concerned! Your system is now at unity gain, meaning that without further tweaking, a signal will leave your system the same as when it entered. This is, of course, a rather boring musical goal. In future articles, we will turn our attention to tweaking. Lots of tweaking! ▬

Bio: Michael Harvey earns his living as the Business Manager for Consumer Products for Microsoft Sweden. He spends his living on grown-up toys from Ensoniq and other companies in the military-industrial-musical complex.

The DP/4 and MIDI

Part I — Modulating Effect Parameters

Steve Byhurst

In this article we'll be exploring how algorithms can be programmed to use MIDI data to alter (or *modulate*, as Ensoniq calls it) its effect parameters. Then we'll examine the different types of algorithms and check out some of the ways that MIDI commands can be used to control the effects they produce.

Programming Modulators

Every algorithm has a number of programmable parameters which control the way it will sound and the way in which it will respond to modulators. These vary according to the type and complexity of effect, but there are eleven which are common to all algorithms, the first three (Name, Mix and Volume), and the last eight (Modulation).

The DP/4 is configured so that two different modulation sources can be used per algorithm. This means that if you are using all four units you can use up to eight modulators in total, two per unit. These can be all the same (one source controlling all), all different (eight separate sources controlling one parameter each), or somewhere in between (perhaps one source controlling one group of units, another controlling a separate group). This configuration has been criticized for not going far enough by some users and it can be limiting in certain situations. However, for less demanding applications it's fine.

The eight modulation parameters are split into two identical sets of four, one lot for the first modulator, Mod1, and the other (you've guessed it) for Mod2. Use the first parameter, SOURCE, to choose one of the eight modulation sources already programmed in the system global pages. Footpedals/switches connected to the DP/4 can be selected as modulators but I will concentrate on using MIDI data. DESTINATION selects which algorithm parameter you want the source to modulate. Any two sources can be programmed to control any two parameters apart from number 00, the algorithm name (no great loss there). This means that amongst other things you can use Mod1 to modulate Mod2 and vice versa.

The final two parameters, RANGE MIN and RANGE MAX, decide by how much your source is going to affect its destination in percentage terms, the range being 0-99%. Note that you are able to invert the values and have a higher minimum range than the maximum. In practice this would reverse the effect a modulator has, meaning that a mod wheel, for example, would appear to send its highest value at the start of its travel rather than at its end.

You may want to confirm that modulator data is actually altering a chosen parameter. This can be done in edit mode by first scrolling the DP/4 display down to the chosen parameter. Now if you send the data you should see the flashing value change accordingly.

There are a couple of general things to consider when programming and using modulators. Pay particular attention to the modulation range parameters. You will need to experiment a bit to get these right for each effect and it is usually best to start with a narrow range unless you want dramatic results. Also, you might find that some parameters (particularly delay times, LFO widths and others which involve rapid reprocessing) are prone to glitching when parameters are quickly "swept" in real-time by a modulator. This may happen because of the combined effect of other parameters and some experimentation with these can sometimes eliminate the problem.

Mix and Volume

It is true to say that not all parameters are suited to real-time MIDI control. However, Mix and Volume, common to all algorithms, provide a very useful way to modify signal levels for all effects.

The MIX parameter regulates how wet or dry the signal will be, a value of 00 being fully dry, 99 fully wet. Use of this depends a lot on your equipment setup. If your source signal is entering the DP/4 via an external mixer you will most probably want Mix to remain fully wet as the mixer will control the dry level. Howèver, if a direct input is used, the DP/4 could be used to control its level and also remotely or automatically change the wet/dry balance by assigning a MIDI controller to the Mix parameter. Very handy if you don't have a MIDI-controllable external mixer to do the same job.

VOLUME controls the level of algorithm output signal. Set to 00 for no output, 99 for full volume. A reminder here that if the output is linked in series to another's input, a setting of 00 will mean that no signal is passed on and therefore no sound will be produced from either algorithm. We have already seen that system global command #57 can enable MIDI-controlled mixing using controller #7 data, but with this parameter any of the eight system controllers can be used instead. This is useful if you are unable or do not wish to use controller #7.

Now for some specifics on the different algorithm types and their associated parameters, starting with...

Reverbs

There are eleven of these divided into standard and non-standard. Both types benefit from the use of Mix and Volume to dynamically change how much reverb is heard. Modulators can be used on standard types to control the depth and width of sound, and also add tonal variation. Non-standards are less suited to real-time control of their most detailed parameters and, once carefully programmed, these are probably best left alone.

Small Room, Large Room and Hall are the most complicated standard reverbs and all three share the same twenty parameters. Whilst it is certainly possible to modulate any of these, I usually leave alone the values that shape the basic character of the reverb as most of the time there is no advantage in using real-time control to alter them. However, the DECAY parameter is very useful for controlling the "bigness" of the reverb during various points of a song and a continuous controller like a mod wheel or pedal works well as a modulator for this purpose. One thing to watch — make sure that you set the modulation range to a reasonable amount, 80% being about maximum. Hall reverb can decay for up to 250 seconds! Also, try using the HF BANDWIDTH and LF DECAYTIME parameters to add some tonal variation.

The Small Plate and Large Plate reverbs are less complicated and have fewer parameters, but once again the DECAY and HF BANDWIDTH commands are the most useful. There is also a LEFT/RIGHT BALANCE value which can be used as a pan control. The Plates work especially well for vocals and you could try changing the emphasis of certain phrases by altering values as a track progresses.

Looking at the non-standards, the two Reverse reverbs and the Gated reverb both feature HOLD TIME, used to control the length of effect, and HF DAMPING to alter tone. The Gated reverb also has a LEFT/RIGHT BALANCE parameter. The Non Lin reverbs are the most specialized type available and have the familiar HF DAMPING, HF BANDWIDTH and LEFT/RIGHT BALANCE values for tonal and pan control.

Next time we will have yet more MIDI fun with algorithms, including DDLs, choruses, flangers and phasers. Beam me up Scotty! ▬

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Putting the DP/4 to Work

Rhythm Tracks

Michael Harvey

The first step in recording a typical pop song is to lay down the basic rhythm tracks. With a band, the members simply troop into the studio, pick up their instruments and play. Depending on the quality of the performances, some of these parts may survive to the final mix. Unless the artist and producer are aiming for an unadulterated live feel, however, the basic rhythm tracks are usually supplemented or even replaced by subsequent overdubs. It is not uncommon for every part but the drum kit to be re-recorded during the course of a typical commercial project.

In future articles, we will explore uses for the DP/4 in band situations and at how to best record specific instruments. Initially, though, we'll assume the most basic arrangement possible: a one-man-band-producer-engineer working with a sequenced rhythm section. Rhythm tracking in this case consists of recording a premix version of a sequence to serve as a guide for the overdubs to follow: vocals, acoustic instruments, solos. At this point, the sequence can remain a sketch, leaving some "creative space" allows for ideas to develop and be incorporated as the project unfolds. Remember, though, once you start committing music to tape, you are locked into your arrangement and tempos. Do your experimenting with length, song structure, tempos, and time signatures now. (This would correspond to "pre-production" for a band.)

Unless your sequence is already fully realized, it may sound flat or sterile at this point. Perhaps not all your sound modules have onboard effects. Or perhaps the mix needs to be just a little punchier, more in your face. Since we will be discarding these tracks later, we don't want to invest too much time tweaking them. Nevertheless, there are some simple ways to give them life, an investment that will help inspire your vocalists and musicians (in this case, you) to deliver strong, informed performances. It should come as no surprise that the DP/4 is perfect for the job.

Last time (Issue #105), we set up the DP/4 for this task: we calibrated levels, wired it into our system, and built a framework for the necessary patch. Since this patch essentially split the DP/4 in two — into an effects processor and a signal processor — we called the patch "Schizoid." Assuming you have used your DP/4 since the last article, go ahead and load "Schizoid" as the active patch. (Remember, it's a Config patch. Hint: If you want patches to load automatically as soon as you stop scrolling, press the SYSTEM/MIDI button and advance to parameter #55. Use the jog wheel to toggle "Auto-Load Preset (SelectMode) =" from "Off" to "On.")

The premix can be improved both by enhancing individual parts and by enhancing the entire mix. Sounds simple. In reality, the DP/4 can be used in limitless ways to achieve these goals. Alas, paper and ink are not limitless; this article allows space to examine only one approach: we will use the processing group formed by units A and B (group AB) on the DP/4 to apply chorusing and some delay to "widen" individual parts in our premix. Next time, we'll use group CD to apply compression and enhancement to the entire mix.

Effect algorithms for group AB can be selected in one of two ways: we can choose directly from existing two-unit patches; alternatively, we can load one-unit algorithms into each unit individually. To save work, let's see if there are any two-unit patches we can use as a starting point. Press Unit button A. The jog wheel will now scroll through all 100 two-unit presets stored in the DP/4. (Refer to the manual to see a complete list.) To hear the different patches, start your sequence playing and use your mixer's effects loop to send signal to the DP/4. You may want to solo a few parts to hear the effect clearly — bass, drums and piano cover the spectrum nicely. As you scroll through the algorithms (assuming this wasn't the very first thing you did when you unpacked your DP/4!), you will be amazed at the range of possible effects.

Keep scrolling until you come to patch #41, "Detune & Spread." If you don't have "Auto-load" turned on, press SELECT to load the patch. The LCD window reveals that this algorithm applies chorus and delay to incoming signals. If you are monitoring your sequence, you will hear the bass spread out to occupy the entire stereo spectrum. You may also hear some increased sibilance or even a discrete echo if you are applying this effect to drums. Let's begin working with the chorus algorithm. Bypass all four units by pressing the CONFIG button twice. Now press Unit button A twice to reactivate it. Press the EDIT button to enter editing mode. (As the edits accumulate, be sure to save your work regularly. You can toggle between the most recently saved patch and your newly edited version by repeatedly pressing the EDIT button. This allows you to compare the two versions before committing to the altered patch by saving it.)

Chorusing is a pitch-based effect that splits the input signal in two, rhythmically detuning and retuning one half, then recombining the dry and effected portions. The result is a "fattening" of the original signal. The DP/4's "8 Voice Chorus" algorithm offers a few additional twists. Instead of splitting the input signal in two, this algorithm effectively creates 8 copies of the input (4 on the left, 4 on the right), applying individual detuning and retuning to each copy. The algorithm also features a stereo delay that permits feedback between the left and right portions of the signal. The possible effects generated by this rich algorithm range from simulating an ensemble of players from a single source, to our goal: adding body to a sound.

As noted above, chorusing depends on a balance between the wet and dry signals. The effect is strongest when both signals are near the same level. Accordingly, most chorus presets on the DP/4 are programmed with a "Mix" setting somewhere around 50. Since we are using our mixer's effects loop to apply the chorusing, we will actually control the mixture between the wet and dry signals at the mixer itself. Normally, this would imply a fully wet Mix setting of 99. Since we are applying delay in series with the chorus, however, and since we want the delay to affect both the chorused and unchorused portions of the signal, we need to send a moderate amount of the dry signal "around" the chorus directly to the delay; a Mix setting around 70 will do the trick.

The next parameter page lets us set the rate and depth of detuning applied. The "Rate" determines how fast the chorus oscillates back and forth between tunings, while "Depth" determines how far apart those tunings are. These two parameters comprise the heart of the chorus effect. Play with the settings to explore their effect: a slow rate with a wide depth produces the aural equivalent of open-ocean swells; a fast rate with a wide depth is akin to riding a sonic roller coaster; and so on. For subtle fattening of sounds, I find that a moderately slow rate of about 10, combined with a moderately narrow depth of about 20 produces a pleasing effect across a wide variety of instruments.

Parameter "Stereo Spread" superimposes a stereo field on the signal. The higher the setting, the wider the field. Use this parameter to increase or decrease the amount of horizontal space an instrument takes up on your song's virtual sound stage (the simulated 3D space you create by panning instruments within the stereo field). For fattening, I find that a setting around 60 produces a pleasing "heft" to bass sounds without creating a wall of mud, while giving pianos and organs good presence. The remaining parameters control the behavior of the built-in delay. Since we will be applying delay separately, moderate settings can be used within the chorus. Try settings around 15, 10ms, 20ms, and 20 for "Regen," "Left Regen Time," "Right Regen Time" and "Delay Regen," respectively, to add interest to the chorus, without creating distinct delays.

We are now ready for the delay algorithm. Press Unit Button B twice in order to reactivate it. The LCD reminds us that this is a "Dual Delay." Since Unit B is the final stage in this processing chain, we want its output as hot as possible to maximize our signal-to-noise ratio. Start by setting the Volume parameter to 99. Remember that Units A and B are connected in series. Unit B will therefore apply its algorithm to the full output from Unit A. Since we gave Unit A a Mix setting of 70, this output is a mixture of wet (in this case, chorused) and dry signal. For the finished patch, we are striving for a complex sonic stew consisting of chorused copies of the original signal, delayed copies of the original signal, and delayed copies of the chorused signal... We can use the Mix parameter for Unit B to control the balance between these various elements. If we set the Mix value to 99, the entire signal will be delayed, an interesting result but not quite what we're after. A Mix setting around 60 produces the desired effect.

The "Dual Delay" algorithm allows us complete control of two independent delay lines, one for the left channel and one for the right. The two lines, furthermore, can interact, with the output of one line feeding back into the input of the other. With long delay times, this "cross-regeneration" can be used to create swirling, stereo ping-pong effects. For our purposes, we can use short delay times to simultaneously add width and transparency to selected sounds.

Somewhere between 15 and 30 milliseconds lies the threshold at which a delayed sound is heard (by humans, at least) as a distinct copy rather than part of the original signal. Delay times under this threshold lead to a perceived widening of the original sound and can actually be used to place a sound within the stereo field much more precisely than level-based panning. For the current patch, delay times of 20ms and 30ms work well for the left and right channels respectively, with "Regen" amounts of 10 and 15. Leave the "Pan" settings to -99 and +99 for a wide, airy sound. A modest setting of 20 for the "Cross Regen" parameter adds a barely perceptible sense of depth. The "Regen Damping" parameter can be left at "0" since the feedback signals our patch creates are too short to reveal any high frequency roll-off.

If you haven't done so already, now is a good time to save your work. (Be sure to save the entire patch as a Config preset.) Since group AB also works well as a stand-alone patch for synth pads and bass sounds, you might want to save it as a separate two-unit preset. To do so, press Unit Button A and then the WRITE/COPY button. Pick a revealing name like "Punchy Bass" or the like for your new patch.

Assuming that you are still listening to your sequence, begin applying small amounts of "Schizoid" to individual parts in your mix via your mixer's effect sends. Bass, kick drum, synths, even piano sounds, can all benefit. The adage, "All good things in moderation," applies particularly well to effects processing, however. Be selective. Improving one or two parts can create a surprising improvement in the entire mix. Speaking of which, next time we'll turn the DP/4 loose on the entire mix itself and hear how this amazing box can bring snap and sparkle to your song. ▬

Bio: Michael Harvey earns his living as the Business Manager for Consumer Products for Microsoft Sweden. He spends his living on grown-up toys from Ensoniq and other companies in the military-industrial-musical complex.

The DP/4 and MIDI

Modulating Effect Parameters — Part II

Steve Byhurst

Welcome back DP/4ers! Last time we were looking at reverb algorithms. Now let's check out some more effects and see what their parameters have to offer us in terms of modulation.

Delays

The DP/4 features five delay algorithms and these offer us more opportunities for MIDI control than most of the other types. As with the reverbs, Mix and Volume are very useful here, but the other parameters give full control over the processed effect.

Multi Tap Delay supplies us with four separate taps, each with four parameters. TIME defines the length of delay, LEVEL fades taps in or out and REGEN controls the echo depth. PAN speaks for itself. Try using LEVEL to bring in extra taps when required. This flexible effect demonstrates the limitations of the two modulators per algorithm system because we can only choose to modulate two parameters of any one tap or share two modulators between two taps.

The **Dual Delay** offers two identical DDLs, one fed by the left input the other by the right. A stereo input could be used, or, as a full set of parameters is provided for each DDL (TIME, REGEN and PAN), two mono signals could be processed separately. Both modulators using the same source could be used to change the same destination parameter for each channel of a stereo signal, or, two different modulators could be used to change parameters of the mono channels.

EQ-DDL-with LFO adds an LFO option to a stereo DDL. This is one of my favorite algorithms as it's great to experiment with. The LFO allows you to produce all sorts of weird Sci-Fi noises as well as standard modulated delay effects like chorus. Try using LFO RATE and LFO WIDTH to control the amount of modulation, and DELAY REGEN and CROSS REGEN to control the delay. Avoid high values if you don't want your ears to suffer from a signal blow up.

Like many of the other multi-effects, this algorithm also features an EQ section. There are five parameters, two bass controls, two treble, and an input level trim to eliminate clipping of boosted signals. The EQ is placed before anything else and therefore decides the tone of the input before it is processed. This can alter the final result quite appreciably, especially when using "swept" effects like flangers and phasers.

Tempo Delay is a standard stereo DDL having a delay time measured in note values rather than the more usual milliseconds. This delay rate can then be controlled by a programmable internal tempo or by MIDI clocks. If the TEMPO CONTROL parameter is programmed to receive MIDI Clocks your sequencer clock rate will govern speed and the DP/4 will respond to any tempo changes. This is excellent for sequenced synth riffs and you can also modulate FINE TUNE, REGEN and PAN to get the exact echo effect you require.

The **3.3 Sec DDL** algorithm uses two units to give longer delay times and also features an "Instant Replay" mode. It is possible to record (or sample) whatever audio goes into the mono input for 3.6 seconds and then play it back in a loop. Set DELAY MODE to Loop/Mute and use the DELAY SET parameter to choose one of the eight modulation sources to instigate record or playback. An on/off controller is best as on triggers record, off triggers play. This is the closest the DP/4 gets to sampling and, whilst obviously a limited option, it is a nice extra to play with. Of course you also have parameters like TIME, REGEN and PAN to control the delay line.

Choruses and Flangers

There are two chorus algorithms, each suited to different applications. The **8 Voice Chorus** supplies what Ensoniq calls a "symphonic" effect and is specifically programmed to add eight voices to a single source by using LFOs and Delays. It certainly does give a big sound but needs to be handled carefully to avoid ending up sounding too wobbly (who needs technical terms!). There are a number of useful modulation parameters apart from the usual LFO RATE and LFO WIDTH, including DELAY REGEN to add an echo element and STEREO SPREAD to sweep from mono to stereo.

The **EQ-Chorus-DDL** algorithm gives us the modulated detune effect, generally used to fatten-up thin sounds, that choruses are famous for. All the normal chorus parameters are available but the ones specifically affecting the DDL like DELAY REGEN and ECHO LEVEL are also worth experimenting with. For example, a controller like aftertouch could be assigned to expressively add an echo on certain phrases.

There are two flanger algorithms and, like the choruses, one is specialized and the other is a multi-effect. The specialized **Flanger** algorithm has all the basic parameters you need to create that classic flanging whoosh sound. Use LFO RATE, LFO WIDTH and REGEN to define and alter the quality of what can be a very strong effect.

EQ-Flanger-DDL features a slightly less powerful flanger but gives us quite a few extra parameters and a built-in delay line. Apart from LFO RATE and WIDTH you can also try playing with FEEDBACK and NOTCH WIDTH to alter the flanger quality, and DELAY FEEDBACK and ECHO LEVEL to change the amount of DDL used. The SAMPLE & HOLD RATE parameter is one found in quite a number of other al-

gorithms and allows a rhythmic element to be introduced in the sound by applying a sampling rate to the LFO. Just modulate the rate to change the number of samples. This is very effective if used sparingly.

Phaser and Panner

I suspect few of us actually understand the difference between a flanger and a phaser. The DP/4 manual offers an explanation but, as with most things, it's best to use your ears to decide if a particular effect can give you what you want or not. Personally, I think of a phaser as being able to provide a stronger, more intense whooshy sound than a flanger. As such, it is not an everyday effect, but it does have its uses. The **Phaser-DDL** algorithm certainly offers many variations if you do need to use one. Not surprisingly, the parameters are very similar to the **Flanger-DDL** and can be used in the same way.

Another of my favorite algorithms for experimentation is the **EQ-Panner-DDL**. The combination of a panner facility with a delay line gives lots of scope for creating great stereo effects, from slow smooth sweeps to fast rhythmic jumps. There are lots of useful parameters including nearly all the ones provided by the Flanger-DDL algorithm. Most of them respond well to modulation, especially RATE, SAMPLE & HOLD and DELAY REGEN.

Tremolo and Vibrato

The **EQ-Tremolo-DDL** and **EQ-Vibrato-DDL** algorithms have nearly identical parameters and only differ in the way they apply modulation (tremolo to volume, vibrato to pitch). Their architecture is very similar to the Flanger/Panner-DDL algorithms and as such they offer many opportunities for creative effecting. I find them particularly useful for giving life to a previously static and boring synth sound. The parameters lend themselves easily to real-time expression, the best ones being RATE, WIDTH and DELAY REGEN.

That's it for now. Can I just put in a reminder that I would love to hear from anyone who uses the DP/4 with MIDI in a different or unusual way that has not been covered before. Please get in touch — Britain is not that far away by post!

See you soon for Part Three. ▬

Bio: Steve Byhurst is a British composer of synthesizer-based instrumental music. He is an aspiring soundtrack writer who would love to make a living from the results of using his (mainly USA-made!) gear. Write to him at 1 Oaklands, Oakhill Road, Horsham, West Sussex, RH13 5LG, U.K.

Back to (Ensoniq) School

DP/4 Basic Concepts

Robby Berman and Roy Elkins

This article is the first in a series based on Ensoniq's instructional videotape for the DP/4. The videotape, *EVS-2*, was created by Roy Elkins, the Director of Ensoniq's award-winning training program — chances are that's where the salespeople at your local music store were taught how to use Ensoniq gear. Every new DP/4 is now shipped with a copy of *EVS-2*, but for those of you who haven't benefited from this freebie, we humbly submit these explications for your perusal.

A Highly Placed Source

Suppose you have a '59 Les Paul and you've got it plugged into some kind of box. It could be a chorus box, a distortion box, or a cereal box for that matter. Well, then, how many audio sources do you have plugged into that box? *This is not a trick question.* You got it: one, yer guitar.

Now let's say that the box has two inputs, so you can also plug in your accordion (go ahead, think us uncool). How many things would then be plugged in? Yup, two.

And the bonus round: the box has sprouted yet another input and in goes your electric kazoo. How many audio sources have we got now? Three, of course.

If plug in one more thing — how about a vocal microphone? — we'll have four sources going into the box.

We've just described the four possible configurations of the DP/4. The key to using the DP/4 is this simple: always remember whether you've got one thing coming into it — known as a "1-Source Config" — or two ("2-Source Config"), or three or four. If you know this, everything else about the DP/4's routing will follow. Here's what we mean.

The DP/4 has four effects processors inside its unassuming black case, each one known as a "Unit." The Units are labeled as A, B, C and D. How many Units — or effects processors — you'll be able to use for any particular sound you've pumped into the DP/4 is determined by how many things it is you're planning on pumping.

The First Picture Show

Take a look at the "input configurations" graphic stenciled onto the upper right part of the DP/4's front panel. This thing is one handy cheatsheet.

Actually, it's more like four. There's a diagram for a 1-Source Config — for situations like we had earlier where we just had our Les Paul plugged in. There's one for a 2-Source Config, like when we had the guitar and accordion; a 3-Source Config and a 4-Source.

Let's look more closely at the diagram for the 1-Source Config. Okay, that 1 on the left side of the picture is our one source — our guitar — coming into the big box. Inside that box we've got, what, A, B, C and D? Hey, that's all four of our Units, or effects processors. So, we're seeing that when we plug one source into the DP/4, we can use all four Units. That's simple enough. By the way, ignore the little numbers on the right side of the big box for now — they're outputs, and a whole 'nother story.

Check out the 2-Source graphic. Alright, we've got two things coming into the DP/4. Don't be confused by the fact that they're labeled 1 and 3, they're still our ever-popular guitar and accordion (never mind how we'll play them both at the same time). The reason for the numbering scheme will become clear later on. So, our guitar — 1 — is going into a box with A and B inside it. This tells us that Units A and B will be set up to do their digital dirtywork to our Les Paul. The other source, our accordion (represented by the 3), goes into another box with C and D entrapped therein. This must mean that our second source will have Units C and D at its disposal. Altogether, the diagram shows us that when we're using a 2-Source Config, Units A and B will be applied to the first source, and Units C and D will be used for the second source. Since we don't see any lines or anything else between the two boxes, we can further assume that we've got two distinct and separate signal paths going on here. Got it?

If you examine the next illustration, you'll see that, in a 3-Source Config, the first source will use Unit A, the second Unit B, and the third will have C and D at its disposal. The fourth diagram shows us that when we've got four sources coming into the DP/4, each one goes to its own single Unit.

These diagrams show the four different ways the DP/4 can be set up. That's all there is to it: How many things you have plugged into the DP/4 will ultimately determine how it behaves. As we said earlier, it's the single most important thing to keep straight in your mind in order to tame this powerful beastie. Later on, we'll discuss the way to tell the DP/4 how many sources we'll be using.

Starting from Scratch — Reinitializing

Before we go any further, let's reinitialize the DP/4 to make sure we're all starting from the same point. If you've already created some of your own edits, you'll have to offload your data to some external Sys-Ex storage device via MIDI. All the

stuff that was in the DP/4 when it first came from the factory will be restored when we do the deed.

To reinitialize your DP/4, hold the front panel's System button down, and while holding it, press the Unit B button. The display will read "Hit <Write> To Init RAM Presets." This is really a warning screen to give you a chance to bail out — after all, anything you've saved to memory will be wiped out when you initialize. If you've got the will to wipe, press the Right arrow button once. Now the screen will more emphatically pronounce "Hit <Write> To Reinitialize!!!!" When you now press the Write button, the display will flash off for a second before returning with, first, the DP/4 startup screen, and then the "Select 4U Psets" display.

Your DP/4 is now reinitialized, refreshed and ready to do our bidding, which we'll get to next time the class meets. ▬

Bios: Berman and Elkins were one of the hottest comedy teams in a little thing we used to call "vaudeville." Lionized from Pushmataw County to Kalamazoo, the road was their home and audiences their family. Or the road was their life and audiences their heaven. Or something.

Putting the DP/4 To Work

Part III — Rhythm Tracks

Michael Harvey

There is a famous thought experiment about the nature of change and reality, a variation of which goes as follows: After the battle of Trafalgar, Admiral Nelson's flagship, the *HMS Victory*, is moored at the harbor in Portsmouth and turned into a tourist attraction. The boat, being wooden, suffers in the elements. Occasionally, a plank rots and must be replaced with a new piece. After several centuries of upkeep, every piece of wood from the original boat is replaced; some planks below the water-line are replaced multiple times. The sign over the entrance to the ship still reads *"HMS Victory."* But is it? Is it really still Nelson's flagship?

The philosophers can debate whether the boat is or is not Nelson's ship, and if not, when exactly it stopped being so. One fact is apparent though: The original construction of the *Victory* was incredibly important in determining the shape, quality, and structure of the actual ship (whatever it may be) currently moored in Portsmouth Harbor. Assuming the craftsmen who replaced her various parts were skillful, the *Victory* one visits today should be true in spirit and form to the ship launched from the Manchester dry-docks some 300 years ago.

The basic rhythm tracks recorded as the first step in a typical recording project are a lot like the original *Victory*. Though every part may be replaced by overdubs, the original tracks live on in the finished recording. If the tempo of the original tracks wandered, the tempo in the final piece will wander; if the tuning was off, later performances will suffer. And if the original tracks were uninspired, subsequent performers will find it hard to deliver inspired performances of their own. Without going overboard, it pays to invest some time upfront to make our initial scratch tracks as exciting as possible. This investment will pay off in quality overdubs down the road.

Several installments ago (Issue #105), we set up the DP/4 for rhythm tracking: we calibrated levels, wired it into our system, and built the necessary Config patch. We then used two of the DP/4s' processing units to apply chorus and delay to individual parts in our song, a mini version of the mix-down process. This month, we want to "punch-up" the song. To do this, we'll approximate the work done by a mastering engineer, using signal processing — compression, equalization, and enhancement — to add presence and sizzle to our tracks.

Let's start with compression. To hear the compressor without any other signal processing, bypass Unit D by pressing its Unit Button twice. Now press Unit Button C and then the EDIT button. Scroll to algorithm #83, "EQ-Compressor."

Compressors are incredibly versatile and can be used for everything from removing excessive sibilance from vocals to automatically lowering a music track when a DJ begins to speak. They also excel at our goal — adding punch to final mixes — and are used heavily by mastering engineers and radio stations to that end. In pop music, compression has become an effect itself; the silky smooth texture of radio-ready pop is due in part to judicious use of compression at all stages of the recording project, from tracking to mastering.

To apply compression to our entire mix, patch Units C and D inline with your mixer's main outputs (refer to Issue #105 for step-by-step patching instructions) and leave the MIX parameter set to 99. Since group CD is the final output stage before tape, we want to keep levels hot; increase VOLUME to 99. Set COMPRESSOR GAIN to 0.

The next two parameters comprise the guts of any compressor: the compression ratio and threshold. The compression ratio is expressed as, you guessed it, a ratio. Compression ratios on the DP/4 range from 1.1:1 to infinity. A ratio of 4:1, to take a typical setting, will compress the processed signal such that a level increase of 4db at the input will be reduced to a 1db level increase at the output. (A compression ratio of 1:1, then, applies no compression; a ratio of infinity turns a compressor into a brick-wall limiter.) The second parameter, the threshold, draws a line in the audio sand. Signals that remain behind the line are unaffected. Signals that cross the line, however, are compressed according to the compression ratio. If the threshold is set relatively low, virtually the entire input signal will be compressed. Conversely, if the threshold is set very high, only peaks in signal level will be compressed.

To set these two parameters, start with the compression ratio and then use your ears to set the threshold. We can set the ratio by first thinking about our musical goals. Are we trying to prevent a digital multitrack from clipping? In that case, we want a high ratio to keep all peaks below the clipping threshold. Are we trying to help an uneven vocal performance stay afloat in the mix? Then a low ratio that won't destroy the singer's natural dynamics is called for. In the case at hand, we want to increase the energy level of an entire song by squashing the loudest and quietest parts closer together; a modest ratio of "3:1" works well for that task. Now, with your song playing, use the jog-wheel to slowly increase the threshold, starting from "0." Listen closely to your music. At some point, you will begin to hear the output level of your song decrease. The ideal threshold level lies in the region

where the compressed audio just begins to dip in volume. Depending on your input levels, you will probably find this region lying around 20db.

Scroll ahead one parameter. A simulated LED display shows the gain reduction applied by the compressor. If you used the above procedure and your ears aren't blown, you will probably see signal peaks being reduced by 3-5db.

The compressor's character is determined by the next two parameters: attack and release times. The attack time determines how quickly the compressor acts once a signal crosses the threshold. It should be set slow enough to allow instrument transients to pass through relatively uncompressed but fast enough to catch the body of the sound. Proper settings for this parameter depend heavily on source material. A snare drum (with a quick transient and short sound envelope) needs different handling than a saxophone (with a slow transient and a long envelope). Try a setting of 10ms on song mixes. The release time determines how long the compressor remains active after the input signal drops below the threshold. Set the release time too fast, and an annoying pumping action will be heard as the compressor releases loud signals prematurely; set it too slow, and the compressor will still be active when the next attack transient materializes. A setting of 50ms yields good results when compressing a mix.

The next three parameters control a built-in noise gate. For a song mix, we basically want the gate to stay completely out of the way, that is, open. Unless your song has unusually quiet bits, the default settings for the gate parameters should work fine.

We are now ready to apply equalization. Speaker designers have known for years that music "sounds better" with the bottom and top-ends boosted and have designed their consumer speakers to oblige. The "loudness" button found on most home-amplifiers performs a similar function. The DP/4 doesn't come with a loudness switch, but we can give our music a similar boost via the EQ section of the EQ-Compressor algorithm. For some serious punch without mud, try boosting bass frequencies in the range of 80Hz by about 6db. To add presence, try boosting mid-range frequencies in the range of 6kHz by 6db. Use the DP/4's "compare" feature to hear your song with and without the EQ-Compressor algorithm.

Before strapping on the headphones for overdubs, there is one final step we can take to really give our tracks some sizzle. Despite its name, the DP/4's Van Der Pol filter is an exciting algorithm. More accurately, it is an "exciter" algorithm. In a nutshell, an exciter adds high-frequency distortion to a signal. (Remember, distortion is any artifact not present in the original signal.) The resulting output, even though distorted, displays greater transparency and sparkle. Overuse, however, results in harsh, tinny tones. A word of warning: this particular algorithm is addictive. Your ears will quickly adjust to the excited signal. Give your ears a break and re-check your mix before printing to tape or you may be stuck with shrill, brittle recordings.

Press Unit Button D to reactivate it and press the EDIT button. Scroll to parameter #89, "Van Der Pol Filter." This algorithm works by isolating a portion of the input signal via a bandpass filter, generating high-frequency overtones based on the filtered signal, and then recombining the processed and dry signals. The amount of processed signal is controlled by the MIX parameter. Start conservatively; try a setting of 22. Leave the VOLUME parameter set to 99.

The next two parameters define the width of the bandpass filter. Most of the body of your song probably resides in the frequencies below 5kHz. Try increasing HIGHpASS FC to 4000. Leave LOWPASS FC set to 16. Since a highpass and lowpass filter in series cause a drop in level, it's necessary to boost the filtered signal level somewhat. Too large a boost, however, will lead to annoying, grainy distortion. Try a setting of +15 for the FILTER GAIN parameter.

At this point, our flat and lifeless scratch sequence should be spacious and punchy, ready to inspire some great overdub performances. Unless you don't mind waiting for your sequencer to synch up to your multitrack every time you press the record button, it is helpful to record the scratch sequence to two tracks of your multitrack. (Don't forget to record a two-bar count-off.) These tracks can then be monitored as guide tracks during overdubbing. This step marks your final opportunity to change song structure and tempo: Once you start committing performances to tape, you are locked into your song arrangement, just as the builders of the Victory were locked into a final hull design once they set her keel.

With a pumped-up set of guide tracks on tape, we are ready to turn our attention to the real task at hand: recording a killer song. Just as a ship is carefully built, plank by plank, a good recording is built performance by performance. High quality workmanship and materials are required in both cases. And, of course, state-of-the-art tools. As we will see, the DP/4 provides a veritable tool chest in two rack-spaces. ▬

Bio: Michael Harvey earns his living as the Business Manager for Consumer Products for Microsoft Sweden. He spends his living on grown-up toys from Ensoniq and other companies in the military-industrial-musical complex.

The DP/4 and MIDI

Part III — Modulating Effect Parameters

Steve Byhurst

Hello and welcome to the final part of this series for those of us who just can't stop modulating our parameters.

Pitch Shifters

The DP/4 offers four different pitch shifter algorithms, all having two pitch-shifted voices. Each algorithm is designed with a specific purpose in mind. The Pitch Shifter is the standard and is best used to provide a doubling effect. There are tuning, level and pan parameters for each of the voices, all of which provide opportunities for real-time modulation. In addition you can also make use of the MIX parameter to fade the effect in or out as required and there are LFO rate and width controls to bring in a chorus type effect if you want one. Pitch Shift 2U offers the same parameters but uses its extra processing power to provide smoother pitch shifts for higher quality effects. Use it in the same way as the 1U.

PitchShift-DDL is the algorithm that allows the spectacular spiraling up and down effect as demonstrated in a couple of the original presets. It gives us the same controls over the two voices as before but replaces the LFO with a comprehensive DDL section. Experiment with the DELAY TIMEs, MIX and REGEN values to give some really unusual sounds.

Finally, FastPitchShift is the same as the standard pitch shifter apart from having a much narrower range which allows us to make fast pitch corrections where maybe our vocals are not quite as in tune as we would like. Input your recorded vocal track to the DP/4, assign a continuous controller like the mod wheel to control the FINE parameters, and you're all set. When you get a drift out of tune use the mod wheel to put the vocal back on track. It does take some practice to get this right but then, if you have the audio track in sync with your sequencer, record the controller data onto a sequencer track and you can have repeated playbacks with perfect vocals!

Amps and Speakers

On the whole I feel that this group of effects benefit least from modulation control as they are basically emulations which, once set up to the particular sound you want, are not usually changed during a performance. Generally, it is the instrument feeding the amp or speaker that is changed. You can, of course, use a more convenient controller (like a pedal) to set up parameter values rather than using the DP/4 keypad and this is one more way of making use of the modulation facilities.

All three guitar amps feature similar parameters but are optimized for different levels of distortion. The first is for hard rock, the second rhythm 'n' blues and the third heavy metal. It is really the combination of settings that specify the sound so experimentation is the order of the day.

There are three programmable speaker emulations. Apart from changing the OUTPUT GAIN there is very little else you can do with the Speaker Cabinet. Tunable Speaker is much more flexible and supplies no less than eleven EQ parameters to get just the speaker sound you want. The last, Rotating Speaker, is the exception to the rule for this group. It cries out for real-time modulation, especially when playing programs that are named after a certain well-known organ model. The SPEED and INERTIA parameters are the main controllers of the effect, but also try using DISTORTION LEVEL OUT and DISTORTION TONE to achieve the exact timbre you require.

EQ and Filters

I have already mentioned in a previous article that a basic EQ section is available within some of the combination algorithms. However, there are occasions when more detailed control of tone is required and the Parametric EQ meets this requirement very nicely by providing four separate bands of EQ. Use the GAIN parameters to apply cut and boost to chosen frequencies.

VCF-Distortion provides, amongst other things, a Wah-wah sound with PRE/POST-DISTORTION VCF Fc controlling the level of Wah-wah modulation. To replicate the famous pedal effect use a CV pedal to control these parameters. Distortion and Auto-wah effects can also be obtained.

There are two remaining filter effects. Despite its name the Rumble Filter can dispose of hissy high frequencies as well as nasty low frequencies and is particularly useful in feedback routings. The VandrPol Filter is a type of exciter effect and brightens up dull recordings. Both of these algorithms need careful setting-up and as such are not really suitable for modulation control.

Signal Shapers

I include under this heading algorithms which process the envelope shape of a sound. All of them need to be programmed very carefully to get the right effect but, once they have been set up, modulation can be used to bring in the effect at different points of a song or to dynamically vary the results. The EQ-Compressor, Expander, Keyed Expander, Inverse Expander and De-esser all have parameters which can be used for these purposes, although you do need to have a full understanding of what each one does to use them successfully.

The Ducker/Gate algorithm comes in useful for creative experimentation as well as the traditional ducking function. Interesting results can be obtained by gating a sustained sound (like a synth string chord) with a transient source (maybe a snare drum) to provide rhythmic staccato effects. Try playing a sustained sound over the top of a transient sound sequence and use the mod wheel to change the effect by altering the various parameters available. Some great results can be achieved.

The Vocoder

This is another algorithm where most of the effect parameters need to be adjusted in advance of a real-time performance. However, once that is out of the way use MIX to vary the amount of vocoded signal. Remember that this is the only 4U effect in the DP/4 and so you need to make sure that any modulation signals are routed to all four units if you want the overall sound to be changed. Alternatively, each unit could be modulated independently, perhaps using VOLUME to create your own mix of the four frequency bands.

Utilities

There are two other algorithms we haven't looked at yet that are very useful for certain situations. The first, Sine/Noise Gen, offers a choice of sine wave or noise generator output (BALANCE controls a mix should you want one), and really comes into its own when using modulation control.

We all know that there is very little you can do to a sine wave, but we do have access to SINE FREQUENCY and this could be swept by modulators for effects use. It could also be used for tuning purposes and to provide an additional source of LFO modulation. You can do a lot more with the noise generator. Filter parameters are provided which can be used to create various sound effects like surf, wind, rain, etc. It can be used as an additional source for a synthesizer which doesn't provide this facility or perhaps to free voices for purely musical uses, leaving the sound effects to the DP/4. Sending the output to other algorithms (try reverbs or delays) provides even more scope for natural and unnatural sound effect creation.

The second utility, as its name implies, is not an effect at all but rather a bypass facility. Exactly what No Effect does once selected depends on the setting of the Bypass Kill parameters of the current config. If the unit containing No Effect is set to Bypass, audio signals will be passed, if set to Kill, they won't. What this gives us is another way of controlling Bypass mode other than the program change map and controller source options within the System MIDI pages. Also, as the Mix and Volume parameters used by other algorithms are provided, No Effect can be remotely controlled by a MIDI source to change the volume of external signals feeding the unit which may not require any effect processing.

And Finally...

Deciding which parameters you want to modulate with which source and by how much, is largely a matter of working out what you want to achieve and then changing values until it sounds right. However, I hope that this series has provided you with a few pointers to guide your way around the DP/4's algorithms, and maybe some ideas for experimentation.

The DP/4 and MIDI articles will continue with a deeper exploration of the MIDI commands and some more ideas for using them. Bye for now! ▬

Bio: Steve Byhurst is a British composer of synthesizer-based instrumental music. He is an aspiring soundtrack writer who would love to make a living from the results of using his (mainly USA-made!) gear. Write to him at 1 Oaklands, Oakhill Road, Horsham, West Sussex, RH13 5LG, U.K.

DP/4 Trekking - Creating a 6.6 Second Delay

(Transoniq Hacker #112)

DP/4 Trekking

Creating a 6.6 Second Delay

Alan Blake

I recently received a call from a radio station asking if there was a way to have the DP/4 act as a 7 second delay for incoming calls. He told me that an ex-employee had configured it to do this but he's no longer with the station. Well, there IS a way to do this!

First, you must set up a 2-Unit Preset. Go to SELECT/CONFIG. Dial with Data Entry Knob to parameter 51 (2Unit Preset). Hit Unit A and use data entry to scroll to parameter 50 which should be "3.3 Sec. Delay." Press SELECT. Now press Unit C and scroll to parameter 50 again. This should also say "3.3 Sec. Delay." Now we have both 2Unit presets with the same algorithm. We now have to reset regeneration parameters to zero. Go to EDIT/Unit A and scroll with right arrow to REGEN=06. Reset this to 00. Now right arrow over to REGEN DAMPING and set this to 00. Go to EDIT/Unit C and reset its Regeneration parameters the same way. We can then save this as a Config. Preset if you want. Just press the WRITE button and rename and/or save to a particular Config. location.

Since there is no connection between AB and CD we still have a problem. But we can circumvent this barrier by using audio cables to our advantage. To get a 6.6 sec. delay, we need to run an audio cable (mono shielded) from OUTPUT #1 to INPUT #3 of the DP/4. Then run a cable from OUTPUT #3 to your mixer or other monitor. Of course your signal source will be coming into INPUT #1. Any signals coming in will now be delayed by 6.6 seconds! ▬

Bio: Alan and his brother John Blake (World renowned jazz violinist) have been involved in composing songs for the Kinocraft Film Co. Kinocraft produces documentaries largely of historical content. Alan has written for and/or performed with Grover Washington, George Howard, Rachell Ferrell, and Kevin Eubanks.

DP/4: Program Changes Without MIDI Channels

Johnny Klonaris

If you're like me and you have more than one multi-timbral MIDI device, you're likely trying to squeeze a lot into a mere 16 channels. Yet if you've got a DP/4 and want to be able to call up any sort of config or preset, you may need to tie up five different channels. This article is about a set of System Exclusive (SysEx) commands you can use to let you get at any of the configs or presets. You should be warned that this is pretty much hacking, in the older sense of the word (or "truer" sense if you're one of us old people) of getting down into the lower levels of a machine to do something clever and at least a little bit pointless.

I should probably mention that you're going to need some way of entering SysEx commands into your sequencer (or similar tool, likely computer based). Otherwise, entertaining reading though this may be, it won't be very helpful.

The basic idea is to turn off MIDI for everything (or most everything) and send virtual button presses via SysEx to "operate" the DP/4. Note that Ensoniq does not include the SysEx messages with the DP/4; you have to mail away for them. I suspect that this is in part to prevent "casual" use of SysEx: folks trying it out just because it's in the manual (starting to sound like a disclaimer, huh?). You can make most any MIDI device goofy by sending bogus SysEx commands, so it's good advice to be careful. That said, one good way of learning how to use something is by example.

I've tried to make sure that these examples work and won't cause problems. If you start having problems, please don't immediately run to the phone and call Ensoniq (or me for that matter). There are lots of chances for something to go wrong, but I don't think we're going to make your DP/4 goofy. At least not by accident.

Setup

This grew out of my need to be able to call up a preset, in this case the 4U preset 68, "MetalMaster Gtr" and disable unit A for a song, so that my puny mind would not have to actually remember something (or perhaps it grew out of my need to kill a couple of hours, I'm not sure). In any case, my DP/4 is set up to accept program changes on channel 16 for the config presets. This is the setup I'll be using for the example I'll describe. I'll mention a little later how you might experiment with actually calling up programs with no channels dedicated to the DP/4.

Note that you don't have to set this up exactly the way I do. This is just one way of setting up the DP/4, so that only program changes are received on channel 16. The rest of the article will still apply if you receive data on other channels, it will just be a bit more pointless.

To set up your DP/4 to receive Config preset changes on channel 16, do the following steps:

- Press the System button.
- Press the A button. This takes you directly to 00.
- Use the right arrow to advance to 01.
- Disable MIDI (if necessary) by turning the knob counter-clockwise.
- Use the right arrow and knob to repeat the process for parameters: 08 (Unit B), 15 (Unit C) and 21 (Unit D).
- Use the right arrow to select 28 and set the config channel to 16 (or whatever you prefer).
- Set parameter 29 to "Enabled."
- Set parameter 30 to "Received."
- Press the System button until it shows 50. Set 50 to 01 with the knob.
- Set parm 51 to "Enabled."
- Set 53 to "On."

At this point the DP/4 should be set up ok to accept the data we're going to send.

Now we're ready to start sending data to it. The data consist of a single program change to call up the 4U Presets, some button presses, knob twiddling, and more knob presses. Fun all around. Specifically, for the example case, we're going to do this:

- Call up config "program" 50 (Select 4U Psets)
- Press "Select"
- Press "A"
- Turn the knob to "68," "MetalMaster Gtr"
- Press "Select"
- Press "A" to disable that unit

I used my sequencer (Bars and Pipes Pro on my Amiga 500) to enter the necessary data directly into a track. This is where you're going to need to understand how to get your sequencer to send SysEx data, preferably as part of a sequence. You're going to need to leave some time between some of the commands or the DP/4 will pout. I'll describe what I did with Bars and Pipes Pro (which runs on the wonder of wonders, the Amiga and that "windows" thing too). I'll try to keep it general enough for most sequencers.

The first thing you want from above is the program change to

call up the "Select 4U Psets" config: number 50. This should be simple enough, edit in a program change to 50 in your track (attached, presumably to channel 16). If you play the track, you should see a big red 50, and "Select 4U Psets" on the top line of the character display.

If you were doing this manually, the next thing you would do would be to press the Select button. This is where we finally get into the SysEx. What you need to do is find a way to make your sequencer send out SysEx messages. In B&P Pro, I simply make sure that I'm displaying SysEx for the track and click with the pencil. I then enter SysEx bytes one at a time (I also have to remember to turn on "System Exclusive" in the MIDI out tool).

You need to send two messages, one for the button down, one for the button up. They look like this, including the SysEx head and tail (in hex):

	Header				ID	Msg	Cmd		Button		EOX
Button Down:	F0	0F	40	00	00	01	00	02	00	05	F7
Button Up:	F0	0F	40	00	00	01	00	02	08	05	F7

The only difference is the next to last data byte is a 00 for down and 08 for up. The 05 in the last data byte is the button number for Select. Sending this sequence to your DP/4 is equivalent to pressing the select button (cool!). However, I find I need to leave a fraction of a second between the messages or I get an error. If two messages are too close together, the DP/4 will show:

SysEx Message 01
Receive Error 03

This means you need to insert more time between messages. An eighth note at a moderate tempo should suffice. A quarter to half second seems to work ok for me. Experiment with the spacing until you don't see the error.

If you've got this working, the Select button light should now be on. Next is to press button A so that we'll actually be selecting a 4U preset when we turn the knob. The easiest way is to copy the previous pair of SysEx messages and then modify them. I found it convenient to display lyrics and enter a single word that described what the SysEx messages were, like "select" and "buttonA," etc. Came in mighty handy when copying and moving things around.

In any case, you want to copy and modify the previous pair of messages to change the last data byte from 5 to 0 to reflect the A button. The messages should look like:

	Header				ID	Msg	Cmd		Button		EOX
Button Down:	F0	0F	40	00	00	01	00	02	00	00	F7
Button Up:	F0	0F	40	00	00	01	00	02	08	00	F7

After you've sent this pair of messages, the letters of the first unit in the display should be all CAPS. We're almost there.

The next SysEx message is the knob turning command. The VIRTUAL knob turning command — cool as it might be, the knob doesn't actually turn. If you thought it would, you're being way too imaginative and that probably explains why you're involved in music!

Anyway, the command looks very much like the one above. The differences are in the last three data bytes. The command changes from 02 to 03 and the last two bytes represent the direction and amount of the knob change. In our example case, we want 4U preset 68. Since we're still at 50 from the sent program change, we want to turn the knob clockwise, 18 positions (12 hex). The command is:

	Header				ID	Msg	Cmd		Knob		EOX
Knob turn:	F0	0F	40	00	00	01	00	03	08	12	F7

More on the format of this command in a moment.

After this has been sent out, the select button should be blinking (or if you have Auto-Load Preset [55] set to Yes, the preset will self-select after a second). All that's left to do is to "press" the Select and then A buttons by duplicating the SysEx messages from above, and we've completed our example. If you have not changed 4U preset number 68, you should see:

Metal Master Gtr
AMP>spk>pit>rev

With a red Bypass light on over the unit A button.

Generalizing

You can use the program change and the virtual button and virtual knob SysEx messages to pull up most any combination of presets you want on the DP/4. The general form of the commands are below as follows:

	Header				ID	Msg	Cmd		Button		EOX
Button down:	F0	0F	40	00	00	01	00	02	00	xx	F7
Button up:	F0	0F	40	00	00	01	00	02	08	xx	F7

Where xx is the button number. In hex, the numbers are:

00	A (Unit A)	07	System
01	B (Unit B)	08	Left Arrow
02	C (Unit C)	09	Right Arrow
03	D (Unit D)	0A	Cancel
04	Config	0B	Write
05	Select	0C	Foot switch 1
06	Edit	0D	Foot switch 2

Simply plug in the correct number as the last data byte, and

send the messages. Remember to always send both the button down and the button up commands.

The virtual knob command is very similar, but requires some honest to goodness second grade math.

```
                Header      ID  Msg   Cmd     Knob     EOX
Knob turn:   F0  0F  40  00  00   01   00  03   xx  yy   F7
```

The knob "value" is coded into two numbers xx and yy above and represents a direction, and an amount from 0 to 63. Once you've decided on the value for the knob and the direction, you want to divide the value by sixteen; use the quotient for xx and the remainder for yy. If you want the knob to be turned clockwise, add 8 to xx.

Or, stated more in the bit domain: send the high order nybble, then low order nybble of the byte: d0nnnnn — where d=1 is clockwise, and d=0 is counter- clockwise, and where nnnnn represents the number of knob clicks in the range of 0 to 63.

Customizing

One thing left as an exercise for the student is to make this work without tying up any MIDI channels. What you'd have to do is select the type of preset from the config completely by virtual buttons and virtual knobs. Keep in mind that you'll have much less of an idea what state the DP/4 is in, since it didn't just receive a program change like above. Also, when selecting things with the virtual knob command, remember that you probably don't know where the knob "is" when you start. One way would be to rotate the knob at least 100 positions in one direction or the other to know that it is at one end of the range. This means a knob change could involve 3 virtual knob commands: two to get you to one end, and one to position you where you want to be.

Trouble shooting

There's lots that can go wrong. With SysEx messages, one character typed wrong can make all sorts of things happen. The most common problem for me was keeping the messages spaced out enough so that I didn't get the error 03.

The errors are displayed on the panel for about a second. Try to keep the DP/4 in sight when the messages are going out to see if one of these turns up. The message will be as shown above, with a receive error that is one of these codes:

```
Code Meaning
 01  Receiver time out (>1 second since last byte of message was
     received.
 02  EOX (End of Exclusive F7h) was received when command code data
     was expected
 03  DP/4 was still processing previous command.
 04  EOX received when other data was expected
 05  The byte received after the data block was not EOX.
 06  Command message contained an invalid or illegal argument.
 07  Parameter value in command was illegal.
 08  Button number in command was illegal.
 09  Knob value in command was illegal.
 10  The preset received by the DP/4 was not automatically loaded be-
     cause its type was incompatible with the current configuration.
```

Other than that, just be careful and you should be able to call things up that you maybe thought you couldn't.

Note that when you're trying to select a value for the knob, remember that it's not an absolute value, if you send a value of five to try to get your preset, and discover the correct value is six and send that, you've just turned the knob 11. You'll need to resend the whole sequence to try it out again.

Another problem I had is that Bars and Pipes Pro likes to send out the "current" program change on a track when the track stops. This works fine in almost all situations, but this is a bit goofy — the lone program change in my track winds up sending me back to the Config Preset 50 "Select 4U Psets." I worked around it by playing the track and turning it off while the track was still playing.

Other than that, between working up the sequences and writing this article, I've managed to get out of trying to lay down that solo guitar track for one more night! Ah, diversions...

Plug

While I've got your attention, I thought I'd put in another plug for the Mirage-Net. This is a mail based "network" of Mirage aficionados. It's low traffic and mostly just a way to keep connected to any Mirage info that might be handy. To join, you need to be able to send and receive Internet mail. If you're interested, send a message to me at: mirage-request@hpdsojk.cup.hp.com, and include your address. I'll send you back some goodies. From CompuServe, you can access the Internet via: >INTERNET:mirage-request@hpdsojk.cup.hp.com. Sorry, no World-Wide Web server just yet — we have a firewall! ▬

Current Ensoniq O.S. (Disk/EPROM)

EPS	2.49/2.40	SQ-R	1.02
EPS-M	2.49/2.41	SQ-R 32	2.03
EPS-16 PLUS	1.3/1.00F	SQ-R PLUS	1.15
MASOS	2.0	SQ-2	1.2
MIRAGE	3.2	SQ-2 32	2.03
ESQ	3.5	SD-1/SD-1 32	4.10/4.10
ESQ-M	1.2	DP/4	1.14
SQ-80	1.8	KS-32	3.01
VFX	2.30	ASR-10	2.5/1.5
VFX-SD	2.1/2.00	KMX-8	2.00
SQ-1	1.11	KMX-16	1.50
SQ-1 32	2.03	TS-10/12	3.04
SQ-1 PLUS	1.1	KT-76/88	1.60

DP/4 Hackerpatches

Dave Kelly

I've had my DP/4 for a couple years now. I was probably one of the first to own one. Knowing that the DP/4 can do many things was a bonus but I purchased it primarily for its reverb and the ability to be able to get almost any sound out of the DP/4 that was in my head. I feel that the quality of reverb is what separates okay effects processors from truly excellent units. The DP/4 doesn't have to take a back seat to anything near its price range when it comes to reverb quality. I've used some of the better 'verbs including the Dynacord DRP20 which I love and most of the high end Lexicon gear. The DP/4 can more than hold its own with these units. In fact I can get sounds from the DP/4 that I can't get with any other processor.

I'm including two patches that have been developed the past couple of years in recording sessions. My favorite is "Birds." It's a 4-unit, 1-source reverb that has tremendous stereo spread and creates a ghost like vocal effect. It's not for use on every song but will add a very haunting effect to a lead vocal. Try it on a ballad that has sparse instrumentation and see how it widens the feel of the entire mix.

The other patch, "The Room" combines moderate decay times with pre delay and a good amount of room detuning to produce a very smooth sounding room reverb.

Most of my patches are designed for vocal use. One thing I do is use the room detune parameters of the hall and room algorithms to really smooth out the decay. These parameters can be used to create a pitch shift type of effect if not over used. Be careful, because if you detune too much it becomes very hard to use the patch on a guitar or other instrument. I've found that using the detune parameters work best on vocals and snares. I've reprogrammed every RAM patch in my DP/4. I think the factory reverbs tend to be very generic sounding and don't really show what the unit can do.

The DP/4 is a programmer's dream. I don't have any real

Name: Birds
Use: Vocals

Alg	Para Eq	Alg	Lg Plt	Alg	Sm Plt	Alg	8 Vce Ch
01	99	01	77	01	99	01	78
02	99	02	99	02	99	02	99
03	83 Hz	03	2.16	03	2.97	03	06
04	00 db	04	0	04	6	04	15
05	302 Hz	05	34	05	38	05	99
06	+09 db	06	30	06	54	06	00
07	05	07	65	07	84	07	16ms
08	5000 Hz	08	82	08	77	08	25ms
09	0 db	09	33	09	32	09	17
10	06	10	00	10	+08	10	
11	9k	11	00	11	-02	11	
12	+04 db	12	00	12	+05	12	
13	-5	13	00	13	-17	13	
14		14	00	14	00	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	

Notes: Configs: (00) 1 (01) Ser (02) Ser (03) Ser (04) 00 (05) 00 (06) Mono (07) b (08) b (09) b (10) b

Name: The Room
Use: Vocals

Alg	Lg Room	Alg	Lg Room	Alg	Lg Room	Alg	Lg Room
01	75	01	79	01	74	01	75
02	99	02	99	02	99	02	99
03	2.06	03	2.77	03	2.84	03	2.92
04	63	04	61	04	69	04	67
05	+03	05	+06	05	+05	05	+08
06	31	06	44	06	31	06	32
07	46	07	60	07	55	07	46
08	74	08	79	08	55	08	91
09	92	09	99	09	36	09	67
10	18	10	28	10	21	10	25
11	41	11	31	11	33	11	63
12	45	12	32	12	33	12	62
13	+85	13	+99	13	+72	13	+87
14	5ms	14	11ms	14	4ms	14	7ms
15	14	15	00	15	14	15	16
16	59	16	16	16	65	16	40
17	21ms	17	15ms	17	5ms	17	26ms
18	04	18	01	18	11	18	11
19	70	19	21	19	22	19	36
20	+79	20	+22	20	+83	20	+81
21	+35	21	+35	21	+35	21	+31
22	+23	22	+30	22	+12	22	+16
23		23		23		23	

Notes: Configs: (00) 1 (01) Ser (02) Ser (03) Par (04) 00 (05) 00 (06) Mono (07) b (08) b (09) b (10) b

secret to programming. I just start with an idea while tracking or mixing and just keep tweaking the parameters 'til it sounds the way I like it. There're few things more musically pleasing to an engineer than a smooth decaying, warm reverb. I've listened to most of the newer mid-priced reverbs out there and they just don't compare. I have a few things I wish Ensoniq would have built into the unit like better meters and balanced ins/outs but this would have increased the price quite a bit. If you want to hear what a $2000+ reverb sounds like.... go out and buy a DP/4. You won't be disappointed! ▬

Bio: Dave owns and operates Luna Recording in Glen Burnie Md, a 24-track studio. His band is PASSIONFIX. He's been doing music since he was 5 (37 now but don't tell) and has engineering seriously for something like 10-12 years. He spends most of my conscious hours and even many of the unconscious ones thinking about ways to improve sound.

Dynamic Parametric Equalization with the DP/4

Or — How I Saved My Shorts

Johnny Klonaris

They say that Mr. Watson, for years after the invention of the telephone, often expressed regret that the process of making Bell's telephone work didn't have a clear, defining moment, but was in fact an arduous and incremental task. Only in later years did he start telling the story of the acid spill and the famous "Mr. Watson, come here, I want you." Our culture embraced this more interesting story and made history of it. To his credit, Mr. Watson did invent the anti-tinkle circuit and that telephone thing seems to have caught on big time.

I, on the other-hand, was able to turn a musical limitation into an article that might be of some use to a portion of the *Transoniq Hacker's* esteemed readership. We each do what we can.

This article tells the story of how I used the DP/4 to remove some pretty bad guitar finger squeaks and save a song, not

to mention a few hair cells in my inner ear.

It was a dark and stormy recording session. The engineer and producer were collapsed in the corner — it was just me and my old twelve string going at it. I was playing flawlessly when a massive earthquake caused my finger to slip, creating that horror of horrors: a finger squeak.

Something like that. In fact I'd been struggling with what, for me, is a challenging finger picking part on twelve string guitar. When I got one with no major clams in it, I was pretty eager to accept it; except for those squeaks. I left the part as it was, hoping to be able to fix it in the mix with the DP/4.

Months later it was time to see if my guess was right. The plan was to use a notch filter, tuned to take out most of the sound energy of the squeak, without seriously affecting the sound of the 12-string guitar. There were several small squeaks, and I wanted at least a little bit of finger and fret noise to make it sound like a real guitar performance. The problem was, there was one clunker that I called "Death at 3:04." This was a super-chirp with harmonics that would dry clean your shorts. I knew that any filtering solution would have to be dynamic, cutting in only on the problem areas and getting out of the way the rest of the time.

I wound up setting up a two-unit, stereo setup for the guitar. This was part of a four-unit config preset that also had a pair of reverb units for the rest of the mix. For the guitar I used unit C set to a Parametric EQ and unit D to a Vander-Pol Filter for the thrill of excitement (wow). I had my Amiga with Bars and Pipes Pro synched to the tape deck, so I set up a track to contain control change information for changing the filter gain dynamically.

The EQ on the DP/4 unit C was set up as follows:

# Parm	Value	Comments
00	Parametric EQ	
01 Mix	99	
02 Volume	99	
03 Fc	160Hz	Bass low shelving filter
04 Gain	-5dB	Reduce booming from close mic-ing
05 Fc	1200Hz	Mid1 bandpass filter
06 Gain	-10dB	Mid-range notch to improve clarity
07 Q	3	
08 Fc	4500Hz	Mid2 bandpass filter (this is the one)
09 Gain	+02dB	default is a slight boost for shimmer
10 Q	6	Fairly narrow filter
11 Fc	9kHz	Treble high shelving filter
12 Gain	+4dB	Hey, it's a 12-string, this is where it lives
13 Pad	-4dB	(EQ Input Level Attenuation)

Mod1 Parameters:

14 Src	Cntrl-8	Mod wheel — this is the controller of interest
15 Dest	009	Sets mod parameter to filter gain
16 Min	00%	This works out to -48dB
17 Max	70%	This works out to +2dB

Mod2 Parameters (not used)

18 Src	Off
19-21	Don't care

This setup did double duty of taking out the squeaks and EQing the guitar with a bit of cut in the bass and midrange and a bit of a boost in the treble. The notch filter at 4500Hz is the variable filter. It has a relatively narrow Q and is variable from a slight boost of +2dB with the mod wheel up full, down to a rather severe -48dB when the wheel is all the way down.

So much for the unit setup, but there's more. I keep MIDI disabled for individual units and use channel 16 for general DP/4 control. You can tailor this to your needs by changing the parameters specific to MIDI control parameters. Here's what I use in the way of relevant system parameters:

# Parm	Value
35 Control Chan	16
36 MIDI is	Enabled
44 Controller8	Mod Wheel #001
61 Mod Resp Rate	30

This last parameter is important. If it's set to the minimum, 1, it takes a little over two seconds to get from the maximum value to the minimum, and vice-versa. If it's set to the maximum, 30, the response is very fast — fast enough that you can hear clicks in the output if the controller changes too fast. I wound up using an actual mod wheel to enter the control data so that gave me enough steps to smooth the transitions. This setup worked wonders on my guitar part to remove the unwanted slides and leave the sound pretty much untouched: once I got the control track right.

Getting the control track right was a whole 'nother adventure. This wound up being more of a trial and error process than anything else. The Q on the filter (Parm #10) is fairly narrow — you might not hear the effects of the filter easily. You might want to experiment with the filter Q set to a lower value at first to hear the effect, then later set the value as high as you need to get rid of the nasty parts, leaving the sacred bits for the world to hear.

Eventually, once I was sure it was working (see Troubleshooting below), getting rid of the horrid noises became a matter of timing. I eventually got the knack of whipping the mod wheel from max to min, holding it there for about a

quarter note and zipping it back up to the max. With a little cut and paste on my sequencer, I was able to build a track that contained the control information to engage the filter at all the right parts.

Helpful hint: Work from the beginning to the end of the song. This let me make cuts and pastes to first position the beginning of the notch to the beginning of the offending sound, then I adjusted the length of the notch to catch all of the squeak.

Troubleshooting:

As neat an idea as this seemed, it took me about as long to make it work as it did to track down and eliminate the dozen or so squeaks worth bothering with. I recommend setting the Q value to a low value (like 1) at first and using a mod wheel on a keyboard to verify that in fact the filter is affecting the sound. Also, use edit mode on the unit in question and pull up parameter 9. The displayed value should change while it's highlighted so that you can know if your MIDI controller info is arriving at its destination.

Depending on just about every variable you can imagine, including your sound source, microphones, EQ and whatever, the filter settings above may need to change to suit your needs.

After all that, I was happy to find that what had once been the aural equivalent of a ten-penny nail being driving into each ear, had become a mostly tolerable guitar part. Ain't technology wonderful?

As for interesting stories; mine wasn't. It was just me at home, it was a sunny afternoon, and if there was an earthquake, I didn't feel it. But hey, my squeaks are gone! ▬

DP/4 Hackerpatches

Dave Kelly

These are a couple more 4-unit/1-source patches that have been developed the past couple of years in recording sessions. Most of these patches are designed for vocal use. One thing I do is use the room detune parameters of the hall and room algorithms to really smooth out the decay. These parameters can be used to create a pitch shift type of effect if not over used. I've found that using the detune parameters works best on vocals and snares.

Staggered: This patch uses the plate algorithm. It gives the feel of large ambience while having short decay times. This is achieved through the delay parameters on the chorus and the serial routing. Best for vocals.

Power Room: While using the large room algorithm, this patch gives the feel of a very tight room while still creating an openess around the instruments it's used on. Best for "in your face" vocals and snares. ▬

Bio: Dave owns and operates Luna Recording, a 24-track studio, in Glen Burnie Md. His band is PASSIONFIX. He's been doing music since he was 5 (37 now but don't tell) and has been engineering seriously for something like 10-12 years.

Name: Staggered

Use: Vocals

Alg	Sm Plt	Alg	8 Vce Ch	Alg	Sm Plt	Alg	Sm Plt
01	99	01	81	01	90	01	89
02	99	02	99	02	99	02	99
03	0.95	03	07	03	1.05	03	1.13
04	0	04	51	04	3	04	0
05	19	05	99	05	34	05	19
06	54	06	00	06	54	06	39
07	42	07	250 ms	07	47	07	63
08	63	08	500 ms	08	58	08	79
09	31	09	99	09	29	09	32
10	+09	10		10	+11	10	+05
11	-06	11		11	+05	11	-08
12	+07	12		12	+05	12	+05
13	+04	13		13	-05	13	-11
14	00	14		14	00	14	00
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	

Configs: (00) 1 (01) Ser (02) Ser (03) Ser (04) 00 (05) 00
(06) Mono (07) b (08) b (09) b (10) b

Name: Power Room

Use: Vocals/Snare

Alg	Lg Room	Alg	Lg Room	Alg	Lg Room	Alg	Lg Room
01	66	01	63	01	63	01	44
02	99	02	99	02	99	02	99
03	0.96	03	1.07	03	1.07	03	1.24
04	1	04	2	04	6	04	15
05	+05	05	+03	05	+04	05	+03
06	48	06	56	06	55	06	41
07	49	07	57	07	56	07	42
08	85	08	76	08	75	08	99
09	68	09	42	09	31	09	80
10	23	10	25	10	28	10	28
11	28	11	16	11	22	11	43
12	18	12	24	12	25	12	45
13	+69	13	+77	13	+74	13	+77
14	14 ms	14	15 ms	14	8 ms	14	13 ms
15	14	15	20	15	57	15	41
16	23	16	16	16	69	16	90
17	27 ms	17	36 ms	17	28 ms	17	24 ms
18	09	18	19	18	37	18	90
19	14	19	21	19	51	19	71
20	+60	20	+77	20	+78	20	+70
21	+50	21	+40	21	+43	21	+48
22	+19	22	+16	22	+15	22	+18
23		23		23		23	

Configs: (00) 1 (01) Par (02) Ser (03) Ser (04) 00 (05) 00
(06) Mono (07) b (08) b (09) b (10) b

The DP/4 Little Golden Book

Introduction and Primer

Ray Legnini

Hello out there in Hackerland. Your fine hosts, the editors of the Transoniq Hacker, have asked me to do a series of articles on the DP/4. What we'll be doing is going through the DP/4 from the ground up with an emphasis on real world applications. That is, the stuff you can actually use in your work, whether that be live playing, song writing, sound design, post production, or music mixing. The DP/4 is an incredibly deep machine, and there's plenty of jobs that it can handle in the course of making your music. We'll explore some of the obvious uses, and some of the more esoteric functions of this unique processor. It is truly the Swiss Army knife of the audio world. I take mine to every session that I do.

Getting Oriented

One of the first things you encounter when you are new to the DP/4 is the *Config*, short for configuration. Remember, the DP/4 is unique in this respect; it can be set up (configured) to be almost any combination of four processors that you could imagine, running in *series* the way a guitar player chains together his effects boxes on the floor (the first effect feeds into the second effect, then the third, etc.) or in *parallel*, the way mixing console aux effect sends are set up (each available aux send knob sends to a separate effect.) Add to that the possibility of having a mono or stereo *input* as well as either a stereo or mono *output* and you could get very confused, very fast.

But wait... It's easy, if you just think in terms of what you already know about the gear you already own. Visualize the processing job you are trying to accomplish. Do you want to process *one* guitar sound, *one* vocal? Try a 1-source, 4-Unit Config. All 4 of the processors go to work on your *one* input signal. Do you need to add digital delay *and* reverb to a vocal track simultaneously from only one aux send on your mixer? A 2-source, 2-Unit Config will do it. The output of the first effect feeds into the second effect. (You'll still have 2 Units left over for another job with this scenario.) Are you mixing down a track and need four discreet effects (say, a small room for percussion, a large hall for vocal tracks, some digital delays for the guitar solo, and a nice chorus to add to the electric piano part)? Well, then you need a to select the 4-source, 1-Unit effects.

Our first exploration will involve the 1-Unit algorithms found in the 4-source Config. I'll presume you're mixing some music. This setup will work whether you have one aux send or as many as four available. I'll refer to only one aux send. The process can be repeated for as many aux sends as your situation allows. Connect your mixer's aux send to the DP/4 Input #1. We'll deal with only a simple stereo output from the DP/4 for the first examples. Plug the DP/4's audio outs #1 and #2 into your audio mixer, panning them left and right to hear the stereo processing. Set the DP/4 Input to 12 o'clock and the Output 1 and 2 volumes at maximum. Select Config #52. Now hit the Unit A button. This selects from the list of available 1-Unit presets in RAM (#00 through #49) and ROM (#50 through #99). Dial down to RAM #00 Vocal Plate 1 and hit *Select* to install this effect. Send your music source from your tape recorder or sequencer via the aux send on the mixing console to the DP/4. You should hear a smooth reverb added to the sound. Very nice but...

Yes, you too can edit

Hit *Edit* and you will enter the world where the effects parameters live, waiting to be tweaked. Very often I start with a sound that's close to what I want and then tweak it in context. The last part (*in context*) is very important. Many novices make the mistake of adding EQ and/or effects in solo mode without listening to the other instruments and tracks of the mix they're working on. The whole song is supposed to be working together.

Scroll through the editable parameters of the Large Plate algorithm until you get to parameter #03, the reverb decay time. This goes from 0.40 seconds (very tight) up to 140 seconds (extremely ambient). Move the data knob to select a new value while listening to your mix. The song's tempo, the instrument being processed, the complexity of the orchestration and the musical style all come into play here.

Getting specific, let's say you're processing some percussion for a ballad. If the part is sparse, a long reverb setting of 3 to 5 seconds might add a nice tail to the clave, for example. If the percussion is busy, even at the same slow tempo this long reverb setting could create unneeded wash and mush in your mix. A tighter setting of 0.5 to 1.5 seconds of decay might be more appropriate. Next, try the pre-delay time found at parameter #04. This delays the onset of the reverb time you just worked on. Small amounts can add a little separation between the source instrument

and the reverb, while long pre-delays can add a slapback echo effect. A few subtle changes in these two parameters can really make a difference in the total sound of your track. As always, save any new setting you like to RAM or a computer based librarian program.

In future articles we'll explore other editable reverb parameters and what to do with them. We'll be covering the DP/4's many other algorithms, too. Stay tuned.

During the course of the series feel free to send in your comments and suggestions. I welcome your input on subjects that you feel need to be covered. We'll try to tailor specific articles to your needs wherever possible. ▬

Bio: Ray Legnini is a sound designer working at Ensoniq in his own padded room in the Loud Weird Noises Department.

The DP/4 Little Golden Book

Part II — Understanding Reverb Parameters

Ray Legnini

Hellooooooooo.... (Insert long Grand-Canyon-sized reverb sound effect here). Today, class, we're going to delve into the much misunderstood world of the reverb parameter. I'll try to cover some basic concepts involved when editing digital reverb processors. Along the way I'll try to give you examples, tips and experiments that you can try in the privacy of your home studio. If you've been following this series of DP/4 articles, you'll know that my main focus is on getting readers to explore the inner workings of the gear you own as it relates to your personal music creating activities rather than just to pass along theories and commentary.

Set It Up

For the examples, you'll need your DP/4 set up to receive a percussive signal at Input #1 from an aux send on your mixer. You could also use any instrument capable of creating some drum sounds plugged directly into Input #1. Set up the outputs #1 and #2 to send in stereo to your mixer. Now for the DP/4:

Select Config #52, 1 Unit presets. To do this, hit the *Select* button, then the *Config* button, then scroll with the data knob to #52. Now hit the Unit A button and select ROM preset #54 Large Plate Reverb with the data knob. Leave the other units bypassed, we won't need them for this discussion. I suggest using a percussion instrument such as a clave or cowbell as the sound source for this experiment because it allows you to focus on what the reverb is doing rather that what the sound is doing as would happen if you were using a complicated groove or a TS-10 Hyperwave pad, etc. Once you understand the parameters you're editing here, you should move on to other sound sources.

The Pre-delay Parameter

Hit the *Edit* button to enter edit mode. Start sending your percussive sound through the DP/4. Scroll to parameter 03, the decay time. This is the one parameter that probably needs no explanation. Leave it set between 2 and 4 seconds for this experiment. Move on to parameter 04, the pre-delay time. The pre-delay time is a delay line, just like any other digital delay; it delays whatever gets sent to it by an amount determined by the time setting. In this case what it's set up to delay is the source sound getting into the reverb tank. Very short pre-delay times are typical (anywhere from 1-15 msec) for percussive stuff . 40-60 msec is useful for something like strings. Try it. The longer the pre-delay time is before the onset of the reverb, the more obvious the gap between the two sounds becomes.

The tempo of the music will play a part in this, too. Longer pre-delay times sound good on sounds with a slower, less percussive attack (like vocal aahs or string pads) because it thickens the sound rather that create additional distracting percussive hits. In fact, you can use a longer pre-delay to your advantage if you time the pre-delay amount to the tempo of the music you're working on. Take the number 60,000 and divide it by the tempo of your music. This will give you the time in milliseconds for one beat of music at your tempo. Simply multiply or divide that number get other musical intervals. Why 60,000? That's the number of milliseconds in one minute (1000 msec = 1 second). Try this example to get you started: Go back to the reverb time parameter and set it to .81 seconds; set the pre-delay to 200 msec. Now play some percussion. You get a slapback echo made of reverb!

The HF Damping Parameter

Recall the original settings for reverb time and pre-delay time; then scroll to parameter 05 HF Damping. This parameter controls the way that high frequencies will be filtered out over time. As you increase the value of the High Frequency Damping the sound will get darker faster and faster. The highest settings are the most extreme, high frequency information decays very quickly. Think of it this way: reverb is actually a series of very quick echoes. A sound goes into the processor, it gets delayed, sent to additional delay units to be smeared and delayed again, then the signal is sent back to the beginning of the path to be processed again, and so on and so on. As you can imagine from this very simplified picture, it's just a big circle, like a toy train. The output of the delay eventually winds up at the beginning of the chain to be processed over and over. This would repeat forever if there were no controls like the HF Damping parameter. Imagine this: Each time your little delay train goes around the circle, a little bit of the high frequency information gets off until eventually it's all gone. That's HF Damping, the higher the setting, the more high frequency information is removed at every repeat. An average setting would be between 20-60. Again, the context and the source have a lot to do with the settings you might choose.

Experiment with these parameters until you feel comfortable with them. Remember, if you have a librarian program running on your computer you can save unlimited variations while you're learning. Stay tuned for more DP/4 fun... ▬

Bio: Ray Legnini is a racecar driver, an ex-CIA operative, movie stuntman and brain surgeon who likes to play with digital reverbs in his spare time.

More on DP/4 Reverb Parameters

Ray Legnini

If you've been following this series of DP/4 articles, you know that last time we touched on some of the basic parameters that you need to understand in order to tweak existing DP/4 presets or your own reverb creations. We used a plate reverb last time. This time we'll continue, using the room and hall algorithms. We'll be focusing on using the parameters that are different in this type of reverb. Any changes we make apply also to the DP/4+. The parameters are identical.

Set up your DP/4 and console. We need an aux send connected to Input #1 of the DP/4. Connect the stereo outs of the DP/4 to two channels of the console for monitoring. A sound source capable of sending a percussive instrument such as a cowbell should also be set up. As I mentioned last time, it is easier to hear the differences in the parameter changes you make while learning if you use a short percussive sound. In fact, sometimes when I'm editing, I'll use a workstation keyboard and sequencer (TS-10 — plug, plug) to repeat a short phrase so that my hands are free to tweak reverb settings.

On the DP/4, select Config #52, 1 Unit Presets, by hitting the *Select* button and then scrolling with the data knob. Then, hit the *Unit A* button, and select ROM preset #52 Hall Reverb. The parameters and the experiments in this lesson apply to the Small Room, Large Room and Hall Reverb algorithms.

Last time we discussed the Reverb time, pre-delay time, and High Frequency Damping parameters. They function in the same manner. Take a moment now to scroll through the editable parameters available in Hall Reverb. The first new one you'll see is *LF Decay Time*. This parameter will set the rate at which low frequencies will decay. Higher values mean that low frequency information will take longer to decay. Try it; send a tom or a kick drum to the DP/4 while increasing this parameter. You should hear a definite increase in bottom "boom." At some point it will start to sound like you're in an airplane hanger. A negative setting on this parameter will have the opposite effect; the low frequencies will decay faster. Return the *LF Decay Time* setting to "00."

Moving on, we come to the Diffusion settings. There are two of these, one for high frequencies, the other for low frequencies. These are followed by the Decay Definition parameter. Send a cowbell to the reverb while you change the value for Diffusion 1. With lower values you will hear more of the quick echoes that make up the reverb. Here's a tip: if you're editing and want to quickly compare a value you've chosen with the original, hit the "Cancel/Undo" button located to the left of the cursor arrows. This will undo the last edit you made and return it to the previous value as long as you still have that parameter flashing in the edit window. Recall the Hall Reverb preset before continuing.

Moving on, the signal passes through a detune section. This helps to smooth out the decay even more. Edit the rate and depth parameters. In this case it's easier to hear the effect with a sustained sound. This detune should be easy to understand if you are familiar with a synth LFO altering pitch. The pitch moves slowly above and below the center starting point at a steady *Rate*. Higher settings of *Depth* will make the effect more obvious. Recall the original preset before continuing.

Hit *Edit* and scroll to parameter #14. The next group of parameters control a delay line used to simulate the sound source bouncing around the room, creating the early slap echoes you hear in a hall. Two sets of these delay parameters are provided. Use your percussive source again while you move the values for *Time* and *Level*. Again, try an extreme setting (Time = 120 msec and Level = 50-100) to get your bearings and then back off into more subtle settings as you get more comfortable.

Recall the Hall Reverb preset before continuing. Hit *Edit* and scroll to parameter #20, *Position Balance*. The three settings available here allow you to change your listening perspective, from closer to the stage to farther back in the hall. You can blend the three position amounts to suit the context of your music. Zero out the factory settings while sending a percussive source through the DP/4. No reverb! This is actually a mixer of three areas within your hall, close to far. With your percussion sequence playing, listen to each of the position taps separately; then start to combine them.

That's all for now. Editing reverbs may seem difficult at first, but with a little practice you too can create useful presets. As always, save the best variations to a librarian program for easy recall. ▬

Bio: Ray Legnini is a world renowned television host. Watch for his new TV show "Reverbs of the Rich and Famous" *coming this fall.*

Tune Your Bass with a DP/4!

Tom Tracy

Imagine what it would be like if you could create a preset in your DP/4 that could be used to tune your four-string bass. I know what you're thinking — how can this be done? The DP/4+ has a 2-Unit Guitar Tuner algorithm, but not the DP/4! Let's be a little creative here.

Although the DP/4 does not have a tuning algorithm, it does have a Sine/Noise Generator, and that's the key. Think about it. Your bass has four strings. The DP/4 has 4 Units (A, B, C, and D). If each string were "connected" to a unit, you could use a Sine/Noise Generator in each unit, and set the appropriate frequency as a tone for tuning each string. Unit A will be used to tune the E string on the bass, Unit B would be used for the A string, Unit C for the D string, and Unit D for the G string.

Set the Config

We will make the Bass Tuner a Config Preset. Sure this could be saved as a 4 Unit Preset, but the benefit of saving this as a Config Preset is that a Config preset will remember the bypass/unbypass status of each unit. If all four sine/noise generator frequencies are ringing out (unbypassed), it would sound worse than a Spinal Tap concert. To Set the Config:

1. Press Edit, then Config.

2. Press the Left Arrow button until the red LED display shows "00."

3. Move the Data Entry Slider until the LCD displays "1 Source Config."

4. Press the Right Arrow button once, and set this to "AB — CD Routing=AB + CD parallel."

5. Press the Right Arrow button again, and set this to "AB Unit Routing=[A+B] parallel."

6. Press the Right Arrow button again, and set this to "CD Unit Routing=[C+D] parallel." We've just made parallel connections between all units.

7. Press the Right Arrow button three times, and set this to "AB Input Select=(1) Mono." If you have a cool expensive stereo bass, set this to "AB Input Select=(1,2) Stereo."

8. Press the Right Arrow button once, and set the remaining four parameters to (b)ypass (k)ill A=k B=k C=k D=k."

9. If all four red unit LEDs are not lit, press the Config button until they are.

Set the Algorithm Parameters

1. While in Edit mode (the Edit LED should still be lit), press the Unit A button.

2. Press the Left Arrow button until the red LED display shows "00."

3. Move the Data Entry Slider to select preset #90 "Signal Generator."

4. Using the Data Entry Controls set the remaining parameters to:

Mix=30	Volume=99
Sine/Noise Gen	Sine Freq=165 Hz
Sine/Noise Gen	Balance=00
Noise Filter LowPass Fc=16 K	
Bass Fc= 200 Hz	EQ Gain=+00 dB

Treble Fc= 05KHz EQ Gain=+00 dB
EQ Input Level Trim=+00
All Modulation parameters=Off

We've just set the parameters in Unit A to tune the E string on the bass.

Use the Copy Feature

This feature (explained so elegantly in the Storage Section of the *Musician's Manual*) will save time in copying the algorithm and parameter settings in Unit A to the other units.

1. While in Edit mode (the Edit LED should still be lit), press the Write button.

2. Press the Unit A button. The top line shows "Write to 1U Pset."

3. While holding the Unit A button, press the Unit B button.

4. Turn the Data Entry Knob clockwise until the display shows "Hit <WRITE> To Copy Unit A to B."

5. Press Write.

Repeat the above steps, copying Unit A to Unit C and D. Now the Sine/Noise Gen algorithm is in all four units.

Setting the Frequencies

We've just copied the algorithm and parameter settings in Unit A to Units B, C, and D. Now all of the units are set to tune the E string. That's not what we want. The frequency needs to be set for each unit to match the appropriate string. Based on a tuning reference of A=440 Hz using an equal-temperament pitch table, set the Sine Frequency parameter (parameter 03) for each unit as follows:

Unit A (the E string) - Sine/Noise Gen Sine Freq=165 Hz
Unit B (the A string) - Sine/Noise Gen Sine Freq=220 Hz
Unit C (the D string) - Sine/Noise Gen Sine Freq=294 Hz
Unit D (the G string) - Sine/Noise Gen Sine Freq=392 Hz

Note: Sine values are rounded off to the closest number. Absolute values are E=164.81; A=220.00; D=293.67; G=392.00. Sine values are also one octave higher than true pitch, making it easier to hear the tone.

Save Your Config Preset

1. While in Edit mode (the Edit LED should still be lit), press the Config button.

2. If all four red unit LEDs are not lit, press the Config button until they are. You shouldn't hear any tones.

3. Press Write. The top line of the LCD display shows "Write to Config." This is what we want.

4. Use the Data Entry Knob to find a Config location to copy you Bass Tuner into.

5. Press Write again. Now using the Arrow buttons and the Data Entry Knob, name your preset something like "Bass Tone Tuner."

6. Press Write one more time to complete the process.

How to Use the Bass Tone Tuner

Now that you've finished making it, let's use it.

1. Connect the DP/4 to an audio source, and your bass to Input 1 (or 1 and 2, see earlier comment).

2. Press Select, then Config, and locate your Bass Tone Tuner preset.

3. Press Select again to select it. (Makes sense, huh?)

4. Press the Unit A button twice. This is the E tone. Match it with your bass.

5. Press the Unit A button again to bypass it (red LED lit).

6. Press the Unit B button twice. This is the A tone. Match it with your bass.

7. Press the Unit B button again to bypass it (red LED lit).

8. Press the Unit C button twice. This is the D tone. Match it with your bass.

9. Press the Unit C button again to bypass it (red LED lit).

10. Press the Unit D button twice. This is the G tone. Match it with your bass.

11. Press the Unit D button again to bypass it (red LED lit).

You have just successfully used the DP/4 as a bass tuner! Variations of this can be used for different pitches and different instruments. You might want to unbypass any two consecutive units as a "tone drone" for your next musical composition. The possibilities are endless. Much thanks goes to Dave Cook, a fine bass player in the Reading PA area, for coming up with the "Bass Tuner" idea. ▬

Bio: Tom Tracy is an Ensoniq corporate citizen and writes documents like Sound Manuals, Musician's Manuals, *and an illustrious bevy of things that nobody reads.*

Listening in on Customer Service

Ensoniq's DP/4+

Dennie Edwards

With the introduction of any new product comes the onslaught of calls to Customer Service about all the new features. Ensoniq's latest product, the DP/4+, is no exception.

The following is a brief summary of the differences between the DP/4+ and the original DP/4. Since there are some real physical differences between them, upgrade is impossible. For the sake of this article, the differences will be divided into the following: Hardware, System Software, Seamless Effects Switching, and Effects Algorithms.

Hardware

Balanced TRS Connectors

The DP/4+ has balanced inputs and outputs. A 1/4-inch jack utilizing TRS (Tip/Ring/Sleeve) connectors carries the balanced signal. Balanced signals allow for better signal to noise characteristics. A balanced signal is carried over three wires. One wire is the ground. The other two wires carry the audio signal 180 degrees out of phase with each other. These signals have the same potential voltages in relation to ground. The signals are balanced with each other. The input device re-aligns the phase of the two signal wires and allows the signal to flow with very low noise. When noise cuts across the cable, it will appear equally with the same phase on both audio signal wires. Remember that these are out of phase with each other. The input device re-aligns the signal phase, thus putting the noise 180 degrees out of phase on the two lines. When the signals are combined, the noise cancels itself out.

Ground Compensated Outputs

The outputs of the DP/4+ are ground compensated which improves the signal as well. This electrical ground and the audio ground are isolated. There is also a +4/-10 output level switch. The +4 should be used when using balanced gear and the -10 should be used when using regular line level equipment.

Headphone Out Jack

There is now a Headphone preamp on the front panel. This is great for editing, using as a practice amp, or live sound effect auditioning. The output level knobs will adjust the mix in the headphones. There is also an Output Mute button on the front panel. This allows the headphone signal to be "soloed" so it will be easier to edit processed the sounds or for practicing.

Neutrik XLR/ 1/4" Connector

Also on the front panel is a special Neutrik XLR/ 1/4" combination connector. This connector duplicates input 1 which is located on the rear panel. When it is used it disables input 1 in the rear of the unit. When the Mic Gain switch is enabled (LED On) both the XLR and the 1/4" connectors are enabled. If the Mic Gain switch is disabled, the 1/4" jack still is enabled but the XLR is now disabled.

Three Segment Input Signal / Peak LEDs.

The DP/4+ has three Signal/Peak LEDs. The DP/4 only has two. The first green light activates when the signal is at least -30dB. The second yellow LED will light when the signal is -12dB. The last red will light when the signal is -6dB below the Analog to Digital Converter clipping level. It is best to have the signal barely light this LED during the peak of the signal level.

Input Configuration LEDs

Above the input configurations diagram on the right side of the front panel are four LEDs. These light to show what input configuration the DP/4+ is currently in. This can be really useful when editing and selecting presets, knowing what input configuration the DP/4+ is in.

Additional Foot Switch Jack

On the back panel is an additional foot switch jack, labeled Foot Switch 2. This will allow the use of up to 2 SW-10s. These foot switches can be used as either mom-

entary or toggle foot switches allowing you to have up to 4 foot switch controllers for effect parameters. Do not use single pedal foot switches that have a mono tip/sleeve 1/4" jack with any DP/4 or DP/4+.

Intelligent Jack Switching (System Parameter 60)

There are 2 new features in regard to input/output jacks and intelligent switching. Outputs 3 and 4 can be routed to 1 and 2 with the change of System Parameter 60. Change this parameter to YES. This is useful when the DP/4+ is in a rack with a patch bay and all outputs are plugged into the patch bay and the output of 3 and 4 need to be mixed into outputs 1 and 2. Previously, you had to unplug the DP/4 output jacks 3 and 4 — which could have been inconvenient.

The next feature deals with the inputs. When there are cables plugged into inputs 1 and 3, and the input configuration is 4 Source, then the signal of input 1 is split in parallel and will be directed to both unit A and B. The signal present on input 3 will be split parallel and sent to both units C and D. When in a 3 source mode, input 3 will not do this. The input will be routed as usual. This is useful when there are only 2 sends from the mixer or other inputting device. This can be used to create a 2 source input config with units A and B and C and D in parallel with Dual Mono Outputs.

System Software

System Parameter 58 and 59 (Auto-Load Preset and Remain in Select Config Mode)

Following is an explanation of the new System/MIDI parameters that have been added to the DP/4+. Parameter 59 has been replaced with a similar feature.

When the DP/4+ system parameter 59 is set to NO, the DP/4+ will behave as if unit A is pressed after a config preset is selected. This is overridden if System parameter 58 = YES. This is an auto load preset feature. It would be very confusing if after every config preset loaded, the unit would enter select unit mode with A active.

System Parameters 45-48
(Footswitch 1-L, 1-R, 2-L, and 2-R)

The DP/4+ now has four possible foot switch controllers. They can be defined as controllers on System Parameters 45-48. The ranges vary between controllers and other functions. This is where their functions are determined.

System Parameter 56
(Unit Chan PrChgs Get 1U Psets)

Now with the use of System Parameter 56, the DP/4+ can receive program changes differently than the DP/4. This parameter allows you to select 1 unit preset independent of which input configuration the DP/4+ is in. When set to NO, it behaves exactly like the original DP/4, and program changes select presets according to the input configuration. For more information about Active MIDI Channels, see the System MIDI section in the manual. This is great for using a MIDI foot controller to select individual 1 unit presets one at a time.

System Parameter 60 (Mix Outputs 3/4 into Out 1/2)

System Parameter 60 was discussed in the above section under hardware. This parameter helps those who have the DP/4+ in a patch bay. When using a patch bay, all outputs usually have jacks inserted into them. If there is no jack into output 3, then outs 3 and 4 bounce down to 1 and 2. If there are jacks in 3, setting this parameter to YES allows the signal to be routed to 1 and 2 without having to physically unplug outputs 3 and 4.

Seamless Effects Switching

Hold on to your digital signal processor chips, because this part of the DP/4+ is really cool. When selecting different algorithms, there is no longer an interruption in the signal. There is a digital dry path around the processors that is controlled by software. As you select a new preset there is a crossfade to the dry signal (around the currently selected preset). When the effect download is complete, there is a crossfade back to the newly selected preset. The result is that there is no signal loss or muting. As you select 4 unit presets, you will hear the signal crossfade to dry and then back again. This is okay for vocals and drums, but what about guitar? Well, grab a guitar and let's see how the unit can seamlessly switch between two effects. After you have a guitar, do the following;

- Plug the audio of the guitar into the front panel jack on the DP/4+

- Next, press the config button.

- Then, using the data entry knob, select the config press 53 1 SRC: Mono In

- Press Unit A, and using the data entry knob select 4 Unit Preset Monster Lead (69).

• While playing a signal note lead, press Edit, Unit C, and using the data entry knob select 1 unit preset 85.

This will demonstrate how effective the crossfade can be. Remember, you can now select 1 unit presets over MIDI no matter what configuration you are in with System Parameter 56 set to yes.

New Effects Algorithms

2U Guitar Tuner

Yes, there is now no excuse to be out of tune. When selecting this preset it automatically put you into edit mode on parameter 03 which allows you to see the tuning deviation of the input. Parameter 04 has the ranges of Guitar and Bass. Anyone who has tried to tune a bass with most guitar tuners can testify to the need of the two ranges. The mix and volume can be set to allow no signal to pass through the tuner or for it to pass through to two other units.

Guitar Amp 4

This is a new guitar algorithm. It is designed to better emulate a Class "A" tube amp. This is the purest form of a classic tube amp. It is very efficient and will not produce too much amp distortion unless the tube bias is turned high (then it sounds like tubes are going bad). Tube bias should be set from 10-25 for an "EC" sound, 20-45 for an "EJ" sound and settings of 45-60 for a "SRV" sound. New parameters include Amp level Detect Attack and Release. This determines how fast the "bite of the amp" tracks. Experiment with the parameters at the extremes and the differences will be more obvious.

Digital Tube Amp and Dynamic Tube Amp

Guitar Amps 1-4 have 1 Waveshaping Table. Both Digital Tube Amp and Dynamic Tube Amp not only have different parameters but also have 8 different waveshaping tables available. A waveshaping table determines how the even-order distortion harmonics are produced. These are harmonics that are not produced by hard clipping. They are the natural harmonics created by a real tube. The most important part is that you can dynamically change which wavetable is being used by how hard or soft the guitar is played.

VCF-Distort 2

This algorithm is the same as VCF-Distort 1 with the addition of an additional feedback after the Post-Distortion Envelope follower. This feedback loop also passes through a damping filter that is used to control the amount of high end frequencies in the feedback signal.

Tunable Spkr 2

This algorithm is intended to have a warmer sound than the original Tunable Speaker algorithm. The parameters are similar with the addition of a Noise Gate. It is recommended to be used with the new guitar algorithms in the DP/4+.

Vocal Remover

This algorithm will remove any signal that is both present in the left and right audio channels. It is important that they are equally in both channels. There is a parameter (03, Vocal pos) which will aid in the amplitude adjustment of the channels, in the left to right balance. There is also a parameter (04, L/R Delay) to adjust time phasing differences between the left and right channels, which should correct any timing or phasing alignment problems. By the way, this algorithm can be adjusted and may be used for other frequency ranges other than vocals. Try this new algorithm on *"Back in the U.S.S.R."* on the Beatles' *White* album.

Summary

Ensoniq has taken a hard look at suggestions from their customers and has improved the already powerful DP/4. The best way to measure the DP/4+ is to hear one at your local dealer. It is a true audio Swiss Army Knife. There are new algorithms, new and improved features such as Seamless Effects Switching, and it sure is a lot lighter than a Marshall. ▬

Bio: Dennie Edwards is an Ensoniq Product soldier who is currently working on a new project for his band, Dream Research. He is attempting to record an album entitled, "The Major League of Kraftless Guitarists: 16 Singles." *It promises to be balanced in all food groups and each track will consist of over four ounces of milk...*

DP Stuff

Other Types of Reverbs

Ray Legnini

In this installment, we are going to explore two of the other reverb types available in the DP/4. Of course, this also applies to the DP/4+ and the DP/2. The DP/4 has Reverse and Gated algorithms available in addition to the standard complement of halls, rooms and plates. As usual, you'll need to set up your sound system with the DP/4 being monitored in stereo. A mono send from your sound source or console will be sufficient. Hook that up to Input #1. Again I'll suggest that you use a sequencer for sending test material into the reverb to allow hands free tweaking. Select the Config "Select 1U Presets," #52. Now hit the Unit A button to select the first preset for our experiments today.

breveR esreveR

The Reverse Reverb algo, found in ROM at #55, employs an envelope generator similar to those found in a synth. (Sometimes called an "ADSR," for *Attack, Decay, Sustain* and *Release,* or an "ADR" if the envelope has no dedicated Decay segment.) In this case we have an Attack, a Hold (the Sustain portion) and a Release. So, that means that we have an ADR; there is no Decay segment in the DP/4 envelope. If you understand programming a synth envelope, then you are well on your way to mastering the reverse reverb in the DP/4. Here's how it works: When your signal crosses the *threshold,* the envelope begins. This fades in the reverb according to the time you set for parameter #04, *Rev Attack.* The envelope then proceeds as it would if it were in a synth. It goes to the next stage, the *Hold Time,* again another user defined time. After this time has passed, the envelope immediately goes into its *Release,* which is parameter #05. This may sound like a long amount of time, but remember, these times are in milliseconds.

While reverse effects are great on drums, don't rule out this algo for other creative purposes. If you need an eerie vocal timbre, try sending the vocal into Reverse Reverb, with an envelope set up to complement the tempo of the music. This can be done by setting the times to the numbers you get from a Beats to Milliseconds conversion chart or by the "divide the tempo into 60,000 method." [Divide 60,000 (one minute's worth of milliseconds) by the tempo to get the value of one quarter note in msec. Then divide or multiply that number as needed to get other musical note values.]

Suppose your music is at 100 BPM. Simple division says that one quarter note equals 600 msec. We'll use the value for an eighth note, 300 msec. Go into edit mode by hitting the EDIT button. Enter this time into the parameter values for *hold time, attack* and *release.* Set parameter #06, Reverse Trigger Threshold, to -30db for starters. Change the level on parameter #12 to "00." This will eliminate the slapback delay effect also present in this effect algo. Play some parts into this effect. Vocals take on a spooky quality; drums and percussion have a techno edge to them. Move the value of the threshold, #06, lower (around -50 dB) and try some sustained guitar chords. The effect will trigger multiple times as the guitar decays to below the threshold. Experiment further with the setting of the threshold parameter while a test sequence is playing. This will need to be tweaked to give the desired result according to the context of the part being sent to the DP/4.

Gated Reverb

The gated reverb effect has been popular for a number of years. Some guy named Phil used it on his tunes. It caught on. The old fashioned way to make gated reverb was by applying a noise gate to the output signal of an analog plate reverb, thereby drastically and abruptly shortening the reverb decay. The same concept lives in the DP/4, but in one algorithm. You need to understand that there are two independent components here, too, just as in the version above. From the Reverse Reverb explanation above, you should understand the concept of an envelope's hold and release times. A similar thing happens here. We have two parameters that you would typically adjust to taste after calling up this reverb: *Hold Time* and *Release Time.* Call up preset #57. While playing a percussive source into the DP/4, move the value of parameter 06, Release Time, from its minimum of 1 msec (very abrupt) to 400 msec (smooth decay, like a short plate reverb). Set this back to 100 msec. when you're done. Do the same exercise while moving parameter #04, Hold Time. The tempo conversion tips from above apply here as well. A Hold Time of 250 msec and a Release Time of 50 msec would be relevant to a tempo of 120 BPM; try it. Save your edit!

That's all for this installment. Send in a cool patch, maybe we can publish it in a future *DP/4 Hackerpatch.* ▬

Bio: Ray Legnini is the former President of Brazil. He now works as a sound jockey at the Ensoniq factory in Malvern.

The DP/4 and MIDI

Part I: System.Midi Parameters

Steve Byhurst

Hello again and welcome along to a couple of articles looking at the DP/4 MIDI commands. This time I will "boldly go" to each of the MIDI parameters in System.Midi mode and suggest some ways of using them.

If you are new to the DP/4, or haven't tried using its MIDI functions yet, press the System.Midi button to start looking at the available parameters.

System.Midi Structure

Once in System.Midi mode you have access to 64 system and MIDI parameters. Many of these directly or indirectly relate to MIDI use and it is the setting of these that I will specifically cover. The remainder are foot switch/pedal setups and simple preference parameters which control the way the user interface works. None of the System.Midi parameters are affected by preset changes — once set they become your default setup.

There are two categories of parameters in System.Midi mode, those that are specific to units and those that have a global or system-wide function. The first 35 parameters are made up of 5 sets of 7 identical commands. Each effect unit (A-D) has a set as well as Config, which here functions as a "virtual unit" because it has its own set of MIDI parameters just like one of the units. The remaining parameters are all global.

We'll start by looking at the unit-specific parameters. You can press one of the unit buttons or the config button as a short-cut to access the relevant set of commands. Remember, there is one of the following commands for each of the units and their settings are completely independent of one another.

MIDI Channel/MIDI Enable

These two parameters allow you to set a MIDI channel from 1 to 16 and to enable reception of volume and program change data on that channel (a separate channel is used for other controller data which we will look at in part two). There is no restriction on which numbers can be used by the units A-D, but the config unit must have a different number to the others to ensure correct reception of data.

Many factors can affect the number of different channels you need to achieve what you want, and it can be difficult to decide on a default. I find it best to work out your most usual way of working and set the parameters accordingly. They can always be changed later and there is a System Exclusive Dump option that can be used to save different setups for recall at the appropriate time (also detailed in *Part Two*).

If you are short of spare channels and/or do not wish to send different MIDI data to each unit, use just two channels by setting units A-D to the same number and the config unit to another. There is still much you can achieve with this setup. Alternatively, if you need absolute control, program each unit with a different number.

The configs you normally use are also a factor in deciding how to use different channels. All four MIDI channels are only active when a 4-source config is in current use. This means it would be a waste of channels to have each unit set to a different number if you hardly ever make use of 4-source configs. Also, if you only use the DP/4 as a single processing unit you will only need one channel number.

The Enable parameter can be used to control usage of individual units sharing the same channel number. A simple trick I sometimes use is to program the same number for each unit in a 4-source config and set them all to disabled. Then, when sending program change or volume data on that same channel, you can choose to enable only the unit or units you wish to receive that data.

Program Change

This parameter simply governs whether a unit will ignore or receive any program changes sent to it on its channel number. If set to receive it will enable presets to be changed and will select 1/2/4 unit presets depending on the current config's source setup.

In a similar way to the MIDI Enable parameter, this allows you to choose which unit, or units, will respond to program changes sent on a particular channel even if all units share the same number.

Note that there is also a global program change master switch which must be turned on if any program changes are to be received.

Program Change Map

There are three parameters which govern the mapping of incoming MIDI program numbers to DP/4 preset numbers for each unit. The first simply turns the mapping on or off, whilst the second and third are used to input the actual numbers to be mapped. When set to "Off" a default preset map is used.

There is one thing to note before using this facility. The DP/4 displays program change numbers as 1-128 whilst the actual numbers received are MIDI standard 0-127. This means that if your MIDI controller (and this includes some Ensoniq models!) displays standard numbers the two displays will always be out by one. This may be confusing but is not a problem once you understand why it happens.

The main application for mapping is to allow a MIDI controller's program change to select an appropriate effect in the DP/4 without having to get program numbers in both pieces of equipment to match. There is no problem if you want a favourite effect to be used by many of your programs as the same preset can be used to map onto multiple program numbers. You can set individual program numbers to be ignored and also make a program number bypass, un-bypass or kill any particular unit you wish. These functions are great for when you are using a workstation-type keyboard to control the effects used on is sequencer tracks. They allow you to add, delete or change the effect just by issuing a program change. Bear in mind though that the DP/4 offers other ways of doing this which may be more suited to a given situation.

Unit Bypass

The last of the unit-specific parameters, this lets you select a controller source which will act as a bypass switch for each unit. Among the MIDI options are pitch bend, note number, note velocity, aftertouch and controller numbers 0-127. I usually make use of controller numbers which are not being used for modulation purposes.

This parameter comes into its own if you want to bypass units separately but do want to use any of the other bypass options the DP/4 provides. These may already be in use or may be used for another purpose, and because there is a wide choice of controller you can always find something which is not being used elsewhere.

Prepare yourselves for next time when we'll go global! ▬

Bio: Steve Byhurst is a British composer of electronic music. He is still trying to convince people that his work is serious enough to be paid for.

The DP/4 and MIDI

System • MIDI Parameters — Part II

Steve Byhurst

As promised last time, we are now going even further into the depths of System.Midi Land. Hold on tight! (I do my best to make this stuff more exciting than it really is you know).

Global Parameters

Right, we've seen what can be done to program the four effect units and the config unit for MIDI, what about parameters that affect the DP/4 as a whole, irrespective of individual settings? Well, these can be found from parameter number 35 onwards. Pressing the System.Midi button repeatedly will get you there quickly and then cycle through groups of related commands. Let's start at the beginning — numbers 35/36.

Control Channel/Enable

This page allows you to choose which channel number from 1-16 you want to function as a controller channel and whether it is enabled or not. As I said in part one, all modulation sources are received on this channel as well as bypass controllers. How the modulation sources change effects in individual units is part of the preset programming. Here we just select which channel the modulation is received on.

The chosen channel can be the same as any or all of the units, or completely separate. How it is used depends on your normal working method. If you are driving the DP/4 from a sequencer you may wish to have a separate controller track with associated unique channel number, keeping it apart from other data. However, in most cases there are advantages in using the same channel you have allocated for the config unit. With this method, as long as you don't wish to control units A-D separately, all your MIDI data can be sent down one channel.

The Enable parameter can be used to filter out all modulation to the DP/4 by simply setting it to "Disabled." Thus you can eliminate the effect of the modulation without changing the modulation parameters of the preset or the parameters set within your MIDI controller.

DP/4 Controllers 1-8

These parameters are used to select a list of eight DP/4 system controller sources. Any two of these may then be chosen to modulate any two parameters of an algorithm. There are non-MIDI as well as MIDI sources. The MIDI ones are the same as those on offer as bypass controllers (listed in part one).

If you are really into MIDI programming you will probably find a total of only eight system controllers quite restrictive, but the rest of us will probably be happy enough with a few old favorites like pitch bend and mod

wheel. In fact continuous controllers like these are the most useful modulators, but also try to include a switch (like the patch buttons on some Ensoniq keyboards) because these can be very useful for certain effects. For instance, if you have programmed a restricted modulator range for an algorithm parameter you can instantly switch from off to a preset depth of modulation.

Program Change Master Switch

This on or off parameter governs whether any program change messages will be recognized irrespective of individual unit setups. It must be set to "On" if you wish any of the units to receive program changes.

One use of this function is to act as a temporary override, stopping the DP/4 from receiving program change commands without having to change the individual unit parameters.

Control 7 Reception

MIDI controller 7 messages are used to send volume data. If this parameter is set to "Yes" the DP/4 will receive this data and direct it to a unit with a matching channel number, otherwise it will be ignored.

If any of the units share a channel number all of those units will be affected by volume changes from a matching channel. For complete MIDI controlled effects mixing give each unit a different channel number and each one can then receive different volume changes on matching channels.

Send Program Change/Controllers

The DP/4, as a multi-effects generator, is geared toward being a MIDI reception unit. By using a computer software program or a workstation-type keyboard, it can be easily incorporated into a complete system and controlled remotely and/or automatically. If this is how it is normally used in your setup it is unlikely that you will want anything coming out of the Midi Out socket, so most of the time this parameter will be set to "No."

Having said this, there may well be MIDI setups where you do want to send out data from the DP/4 and this parameter caters for that possibility. Program changes will be sent out on the preset's primary unit channel and also any controller data generated by a connected foot switch or pedal.

Modulation Response Rate

When using controllers to modulate parameters it is possible that a parameter will be changed either too quickly or too slowly when modulation is applied. This control supplies some help by changing how quickly modulation destination values are altered by incoming data. The slowest response of 1 has the highest resolution with the fastest response of 30 giving a rougher modulation.

This is very much a try-it-and-see parameter. If the sound of your modulation is not what you want, try using this control to see if it can make an improvement. Each effect/modulator combination can require a different rate but generally a fine resolution is more useful, especially when controlling volume.

System Exclusive Dumps

Outgoing system exclusive dumps are controlled by a facility which is accessed by pressing the Write.Copy button whilst System.Midi mode is selected. A screen with two parameters lets us send system dumps from the DP/4 to an external storage device. Combinations of the two parameters allow individual preset or bank dumps of any 1, 2, 4, or config units, as well as all preset banks, all system/midi parameters, or a complete dump of all data in the DP/4 (except ROM presets). Once a type of dump has been selected another press of the Write.Copy button actually sends the data. The screen indicates that a transfer is being made and then tells you if it was okay or not.

Incoming dumps are automatic with the screen displaying a message to say what kind of dump has been received or if there was an error. However, there are two System.Midi parameters which let us set a system exclusive ID and allow or disallow reception of system exclusive messages. The first only needs changing if you are using multiple DP/4s with something like a computer librarian program, and the second only needs to be set to "Disabled" if you have the unlikely problem of stray MIDI dumps affecting the DP/4.

Maybe these parameters cannot be described as particularly exciting, but given a bit of imagination they can be utilized to give some exciting results. Ta-ta for now. ▬

Bio: Steve Byhurst is a British composer of electronic music. He is still trying to convince people that his work is serious enough to be paid for.

DP/4 — Part Cinco

The Non-Linear Reverbs

Ray Legnini

Welcome, reverb fans. Our task today is to explore the DP/4 non-linear reverb algorithms. The three non-lin reverbs available in the DP/4 are unique in their structure. They are typically used to create a more unusual reverb space than could be achieved when using a conventional reverb algo. There are nine Envelope Levels that get combined in various proportions to give these reverbs their unique sound. As you look through the preset's editable parameters you will probably notice some familiar names, like "bandwidth" and "diffusion." They operate much the same; we've covered their functions in previous articles.

Setup

Again, we'll start with the basic items you'll need to have on hand to work with this tutorial. You'll need a source instrument to be processed, preferably percussive, sent to the DP/4's input one, rear panel. I suggest a sequencer be used to allow the source to repeat and leave your hands free to do some tweaking. Monitor the DP/4 output in stereo, of course. You won't need to monitor the source through your system, we'll do that mix inside the DP/4. Also, you'll need three 1/2-inch toggle bolts, glue, nails, and a chainsaw. (Just checking to see if you're paying attention.) Also, it's handy to have a computer librarian program running while experimenting. Save all of your variations as you work.

Select DP/4 Config preset #52 "Select 1U Psets." Hit the Unit A button to enter Select mode. Now, dial in 1U preset #60, "Non Lin Reverb 3." We're going to strip this down to the parameters given below. Take a moment to enter these values before proceeding. Save your work as "My Non Lin."

MIX = 50
VOLUME = 99
ENV LEVEL 1 = 50
ENV LEVEL 2 = 50
ENV LEVEL 3 = 50
ENV LEVEL 4 = 50
ENV LEVEL 5 = 50
ENV LEVEL 6 = 50
ENV LEVEL 7 = 50
ENV LEVEL 8 = 50
ENV LEVEL 9 = 50
HF DAMPING = 30
HF BANDWIDTH = 80
DIFFUSION 1 = 80
DIFFUSION 2 = 80
DENSITY 1 = 50
DENSITY 2 = 50
PRIMARY SEND = +80
REF 1 TIME = 250 MS
SEND = +00
REF 2 TIME = 350 MS
SEND = +00
LEFT/RIGHT BALANCE = +00

SET ALL MOD SOURCES AND AMOUNTS TO "OFF" OR "00."

Fun

Okay, now send your percussive source through this reverb. You'll hear a semi-gated sound; all the envelope levels are the same. For the first experiment, start at parameter 03, the first of the level controls. While your source is playing, start setting the values for this and the next 4 levels to "00" as you listen to the changes. The reverb will have a more obvious pre-delay, as if you were hearing the sound bounce off the rear wall of a room. This can be a useful component of a more complex reverb using multiple units; we'll get to that in a future article. Save this variation as "My Non Lin 2."

Recall the first example. This time edit the values for envelope times 5 through 9, setting them all to "00." You get a tight, close ambient sound. Again, useful as is or as a building block for something bigger. Save it. Continue editing this variation by changing the levels of the first four envelope values. Try descending values of 50, 40, 30, and 20, just to keep the numbers nice and round. You get a more natural decay shape because the reverb is fading away, like it does in real life. Save it.

Another variation: recall the original "My Non Lin" that you programmed. Try setting the envelope levels to 10, 15, 20, 25, 30, 35, 40, 45, 50. (Again, I'm using easy round numbers just for simplicity's sake.) You should hear a smooth crescendo of reverb. Here's a good programming tip. While you're listening, hit the "Edit/Compare" button to toggle between the original version and the new variation. The difference should be obvious. The flashing LED indicates the edited version, a solidly lit LED indicated the original you are comparing it to. Remember you can toggle this anytime during the editing process without damaging the work you've done. Get back to the edited version and save it.

There are many useful variations to be found by experimenting with these envelope values. Take some time to tweak them and build some presets of your own. The three Non Lin algorithms share the same parameters, but give different tonal results. See the manual for more details.

Fun, Fun, Fun

You may have noticed while programming the initial preset for this lesson that there values entered for parameters 19 and 21, but no levels. Why? For the next part, of course. Recall the original "My Non Lin" preset that you programmed. Go to parameter #20 and set it to a value of +25. This sends an echo of your source signal, delayed by 250 milliseconds into the reverb. You should hear a slight slap back effect within the reverb. Try other values; return this to "+25" when you're done. Now, go to parameter #22 and set it to a value of "+25." This sends yet another echo of your source signal, delayed by 350 milliseconds into the reverb. Now you have a swinging slapback echoing reverb. Setting the Reflection times to other values will allow you to time these to any tempo. Save your work. Lastly, try sloping the values the envelope levels 6 through 9 downward to something like 40, 30, 20, and 10. This will give a more natural decay. Save it.

You should have a nice collection of starting points on which to build, now. Send in your favorite preset to the *Hacker Patch Head Patch Hack.* (Say that 5 times in a row fast and win a free car!) That's all for now. ▬

Bio: Ray Legnini is one of the original Apollo 13 astronauts. Look for his cameo appearance in the hit movie.

The DP Delay Algorithms

Ray Legnini

Delays have been a popular effect ever since the first cave man yelled into his cave and heard his voice rumbling around its huge spaces. The first true analog delay!

In more modern times we musical humans have come to love our delays. It seems that every musical style has its own unique uses for echoes and delays. Yes, even classical music. It is not uncommon to hear a theme in a classical work acoustically echoed by the use of dynamics or by having the motif played by another instrument several beats after the original. Ambient music wouldn't be as ambient without the DDL. Live singers love delay lines when used to thicken a vocal performance. Guitar players add them to their solo sounds. Studio engineers make clever use of musically timed delays to enhance the groove of a tune. You get the idea..

We are going to look at some of the ways you can use your DP/4 delays to enhance your music. This will also apply to the DP/4+ as well as the DP/2. The algorithms are the same in all of these processors. Many of the delay ideas can also be applied to any effects algo that can display delay times in milliseconds including those found in the TS Series and the ASR-10.

We are going to focus on the delay-only algorithms in the next series of articles. There also effects algorithms in the DP product family that are multi-effects which also happen to include a delay line. The ideas presented here can also be applied to them.

Today's Math Lesson

It is a good idea to have the formulas for determining musical interval delay times memorized. I've stated this in previous articles, but a quick re-cap is in order for new readers. Here's what you do: Take the number of milliseconds (thousandths of a second) in one minute (1000 x 60 seconds = 60,000) and divide that by the tempo of your music, expressed in Beats Per Minute (BPM). The result of that calculation is the number of millisecond that it takes for one beat of your music (a quarter note) to pass by. It is then a simple matter of multiplying or dividing *that* number to get any other musical relationship you need. That means that if your tempo is 100 BPM, 60,000/100 BPM equals 600 milliseconds (commonly abbreviated as "ms" or "msec"). If you need to know what an eighth note is, then divide the answer by two; you get 300 ms.

The Tempo Delay

First we'll look at the Tempo Delay. The layout of this effect is relatively simple. You can adjust the tempo with Parameter 04 to suit your song. You adjust the musical interval with parameter 03. This is great when you are mixing a song and need the vocal to echo rhythmically with the music. If you generally sequence your tunes, simply get the tempo from the sequencer and enter it into the DP, no need to use a calculator. Another cool feature of this effect is its ability to receive incoming MIDI clocks and sync the delays to the clocks. This means that you can have a sequence playing that has multiple tempos with or without accelerandos/ritards and the delays will automatically be adjusted to keep up with the music! To do this you need to set Parameter 06 to MIDI Clocks, and also hook up a MIDI cable from your sequencer to the DP's MIDI in. Make sure that your sequencer is set to send MIDI clocks. (Some PC based sequencers allow you to turn this function on or off as needed.) Be aware though, that the DP is averaging the MIDI clock beats as received in order to determine the tempo. On the DP/4+ and DP/2 there is an additional parameter called "Tempo Delay Smoothing." This determines how fast the unit will respond to incoming tempo changes, whether from a foot switch tapping or from MIDI clocks as in our example. Lower settings respond faster while higher settings make a more gradual transition to the new tempo. If your tempo changes are happening quickly and vary in speed by large amounts, you may find that a lower setting is best for the parameter. Experiment.

Applications

Here's how to create a wide rhythmic echo effect: Only one DP output is needed for this. Try panning the source instrument to one side of your mix and send the delayed signal (set to an eighth note for starters), with no regeneration, to the opposite side of the mix. Regeneration is the amount of additional repeats you will hear created by an

internal feedback loop which sends the delayed signal back into the delay line to be delayed again. It is Parameter 07; set the amount to "0" for this example.

Our second example will create a '90s version of the old analog tape delay line, minus the ratty recording quality. Recall the original Tempo Delay preset. We can better simulate the old tape style delay machines with the Regen Damping parameter. This feature takes each subsequent repeat and filters it, making it a little darker as it passes back into the delay lines. The higher you set this parameter, the faster the delays will darken. Set the Tempo Delay Time to an eighth note and the Delay Regeneration amount to about "50." Set the value of parameter 09 to "00." Now send a repeating signal into the delay line as you move the value for Regen Damping upward. The repeats will be darker, more "damped" as the parameter name suggests. As you move this parameter higher, notice also how the number of audible repeats seems to change. This is because as you start filtering out the high end of the repeats you will get to a point where they are so dark and filtered that they disappear after only a couple of repeats. To hear more repeats when the damping is set to a high number, raise the value for Parameter 07, the regeneration. Be careful with this because as you change the damping to a lower number, high regeneration settings can cause the echoes to "run away," repeating endlessly. Since you have MIDI hooked up, remember that you can assign a MIDI controller to change any parameter, allowing for real-time creative changes. These moves could also be recorded into your sequencer for an automated mixdown.

Try different settings and save the variations that are useful to you. Of course if you are using a librarian program like *Galaxy* (tm) from Opcode or a parameter editor like *Unisyn* (tm) from Mark of the Unicorn, you can save unlimited variations. You can also save DP/4 sys-ex to any device that can record it, including a computer based sequencer program or an instrument such as an ASR-10. ▬

Bio: Ray Legnini is this year's winner of the Nobel Prize for sampling. He plans to spend the one million dollar prize foolishly.

Automated Mixing with the DP/4

Robert Schulze Lutum

With the information contained in this article, you will be able to automate the volume, EQ and FX settings of up to four tape tracks with an Ensoniq DP/4 parallel effects processor controlled by a MIDI sequencer.

This is especially useful if you are working with a four or 8 track tape deck. Using the DP/4 as an automation means that you can pack vocals and guitar and saxophone on to a single track, because the DP/4 can automatically change each track's volume, EQ, pan and FX setting at any time.

Analog Audio Connections

In this model setup, the DP/4 will mix four tracks to stereo. Connect each tapedeck track you want to automate directly to a DP/4 input. Connect Outputs 1 and 2 to your mixer.

Press "Edit," "Config" and select "4 Source Config." Scroll right and select AB Output Select "1-2 Mixed Stereo." Scroll right again and select CD Output Select "3-4 Mixed Stereo." If you have not stuck any plugs into output sockets 3 and 4, all input will be mixed to outputs 1 and 2. Scroll right again and set each unit to (b)ypass instead of (k)ill.

MIDI Control

The DP/4 provides eight internal controllers, called DP/4 controllers. These can control selectable parameters in each unit. A single unit can respond to two of these eight DP/4 controllers.

Each DP/4 controller can be controlled by a MIDI controller. Press "System" and scroll until the red LED displays (page)"37." The yellow display reads: "DP/4 Controller1=MIDI Control#004" This means that incoming MIDI "controller 4" messages will influence DP/4 controller 1, but only if the DP/4 is receiving them on its "Control Channel." Scroll left to page "36" and check that "Control Chan" is set to "07" and "MIDI is Enabled." Scroll right again and on pages 37 to 44 set "MIDI Control# 061 to 068 respectively. Now MIDI controller 61 controls DP/4 control 1, 62 controls DP/4 control 2 and so on.

Each unit's bypass/kill function can be assigned its own MIDI controller. The DP/4 defaults are MIDI Control# 75, 76, 77 and 78 for units A, B, C, D respectively. Check "system" pages "6," "13," "20," "27." We'll leave them like that. When the DP/4 receives "MIDI controller # 75" with the value "0" on its controller channel (channel 7, remember?) it will switch unit A to "bypass," while value "127" switches "bypass" off.

Volume

Remember: Each unit also has its own MIDI channel. Volume control is a special case, because each unit's volume can be controlled via the unit channel, not only via the DP/4 control channel! Unit channels are assigned on pages "00," "07," "14" and "21." Defaults are channel 1 to 4 for units A to D. Set "system" page 57 to "Receive Control 7 On Unit channel" to "Yes."

Example for Unit A

So now we have made the necessary settings we can control each unit's volume via the unit channel (1, 2, 3, 4), and two additional parameters of a unit's effects algorithm via the control channel (7).

Let's select the preset 52, "Hall Reverb" for unit A. A's volume is already being controlled by MIDI controller # 7 received on channel 1. How about controlling the reverb "Mix" from a sequencer? Easy: Scroll to page "23." Set "Mod1 Src=" to "Cntrl-1." The second line now reads: "MIDI Control#061." Scroll to page "24." It reads: "Mod1 Destination Parameter=001." Scroll back to page "01." The "Mix" parameter is flashing. You see? The parameter numbers correspond to the page numbers. Scroll to page "25." It reads: "Mod1 Param Range Min=34% Max= 99%." For the widest possible control range set "Min=" to "00%."

Scroll to page 27. The settings for the second controller are made here. MIDI control #067 is controlling parameter 03, Decay time. That's neat. Sending MIDI controller #75 on channel 7 will toggle bypass depending on the value. Use this for radical effects. You can mute a tape track by turning the unit's volume to "0" or by setting "bypass" to "kill."

All you have to do now is make your sequencer send these controller messages, and you will see: It really works! Now think of all the tricks you've always wanted to do but couldn't, because you couldn't automate tape tracks: Fading out while turning up the reverb mix, cleaning up tracks by muting the rough spots, controlled panning, dynamic EQ'ing etc.. Now you can. ▬

DP/4 Custom Configs

Part I — Design and Preparation

Steve Byhurst

It may well be that many users of the DP/4 only use the factory programmed config presets, or minor edits thereof, to get the effects processing they require. In a lot of cases this method will generally get you fairly close to what you want to achieve, but it is no replacement for a custom-built config that gives you exactly what you want. It is well worth learning how to program configs because they are the key to getting the most out of the DP/4.

Of course many of us who feel that they would rather spend their time playing music than button-pushing and knob-turning, myself included, might think that designing and programming a completely new config would not be worth the investment of time required. However, I hope to show you that this isn't the case and that by following through a set list of stages you can quickly learn to program your own config presets and find out more about the capabilities of the DP/4 at the same time.

For the purposes of this article I am assuming some knowledge of how the select and edit modes of the DP/4 are used so that I don't have to mention every single button push. I also assume that you have a rough idea of how configs work (you certainly should have if you own or regularly use a DP/4!).

Stage 1: Design Plan

Now I know that the term "design plan" tends to present visions of reams and reams of paper with tons of scribbles on, but it doesn't always mean spending ages writing out detailed plans. As with any design project it is well worth sitting down with pen and paper (or keyboard and word processor) and working out exactly what you want to achieve. This stage is important because just taking a few minutes to actually think through what you need can save a lot of wasted time later when it comes to using those digits on programming buttons and knobs.

Apart from making clear to yourself what the general purpose of the effects processing is, in relation to programming config presets there are several key questions you need to ask yourself. They go something like this:

(a) Which effects/algorithms do I need to have? Check in the manual to remind you of what is available.

(b) Can I make use of multi-effect algorithms? Using one of these may free up room for additional effects without losing any flexibility. Despite constant use of the DP/4 I still find that I forget that some algorithms contain useful little extras like EQ and noise gates.

(c) How do I want the inputs to be configured? This depends very much on how many sound sources you have and whether you want to process mono or stereo inputs. Again, check with the manual to remind yourself of the various input configurations available.

(d) How do I want the outputs to be configured? This mainly depends on the way your system is set up. In the majority of cases outputs will be assigned to one stereo pair but all four outputs may be required if you have your DP/4 routed to a mixer.

(e) How do I want the individual units to be routed to one another? You should work out which links, parallel, serial, or feedback, will be needed between each of the units.

(f) How do I want to set the parameters of the algorithms I have chosen? Consider the settings of parameters like mix, volume, depth of effect, modulation, etc. You only need to roughly think this through at this point as tweaking will almost certainly be needed at a later stage.

(g) Will I need the bypass status of each of the units to be set to bypass or kill? This may be important, especially if you are setting up a config that will be controlled over MIDI to turn individual unit effects on and off.

Working through these questions will force you to make initial programming decisions which, even though you may change them later, will give you a firm base to start with.

Now gear yourself up for the start of button-pushing!

Stage 2: Preparation

First, just a quick reminder of what a config actually does. A config is a type of preset which, when selected, loads unit presets into the four units, sets up the signal routing between the units, and reconfigures the inputs and outputs of the DP/4. There are a total of 100 presets containing 50 RAM locations in which to store your own creations and 50 selectable ROM presets. Any of the 100 presets can be used as a base for programming new configs by changing the unit presets it uses (and therefore the algorithms they contain) and/or altering the parameters of the config itself. Like the unit presets it is then saved by writing over one of the existing RAM locations.

The very first thing to do is to check that you don't already have a config preset that matches your design plan. It is unlikely that you will find a perfect match (hopefully you won't have gone to all the trouble of writing a design plan only to find that what you want was there all the time!), but you might find a preset which is not too far off your plan to use as a base for editing. If you do find one you have to decide which way to go: edit or start from scratch.

Remember that even though a config may contain the unit presets you want, you may still need to significantly change the config and algorithm parameters. I have found that it is usually easier and less confusing to go for the "start from scratch" option, especially when there are very useful config templates in ROM just waiting to be put into edit mode (as you will see in part two).

Whichever route you choose, before starting to edit your config preset template it is helpful to make sure that you have the right unit presets ready to be loaded into it. This is because other programming is best done after the process of selecting a unit preset or presets for the config.

The type of unit preset(s) you are able to select will depend on the type of source config you will be using. A 1 source config can only select 4 unit presets, a 2 source config selects only 2 unit presets, a 3 source config selects 1 or 2 unit presets, and 4 source configs only 1 unit presets. Although you can change individual unit presets in edit mode after you have selected them in a config, it makes things easier if the correct ones are in there to start with.

Now check that you have ROM/RAM unit presets which contain the algorithms you require. Factory programmed 1 unit presets cover all the different algorithm types, but you may not find all the combinations you need for 2 or 4 unit presets. If this is the case you will need to edit existing presets and then update them ready for use by your config. At this stage don't worry about programming the algorithm parameters, just make sure you have the right algorithms saved in the right type of unit preset.

There is just one more thing to mention. If you are not able to save presets to a computer librarian, and are therefore dependent on internal RAM for storage, make sure that you identify presets which you will not mind writing over. When the time comes to save your new presets you don't want to lose old ones which you may have wanted to keep.

Hope to see in part two where I will go on to look at the config parameters themselves, algorithm programming, and testing of the finished config. Bye for now. ▬

Bio: Steve Byhurst is a thirty-something British composer of electronic music who is quite adamant that his compositions should not be classed as "New Age."

DP/4 Custom Configs

Part II — Config and Algorithm Parameters

Steve Byhurst

In a previous article we looked at drawing up a config design plan and the preparation needed before programming can begin. Those being the first two stages of creating a custom-built config, let's now move on to the remaining ones.

Stage 3: Config Programming

If you decided in Stage 2 not to use an existing config preset as an editing base because it didn't come close enough to your design plan, you have the alternative of using one of the templates mentioned in part one. Those thoughtful, generous guys at Ensoniq have provided eight ROM config presets (numbers 53-60) which cover all the input source and mono/stereo variations possible. For example, preset 53 will give you a "1 source: mono input" set-up for 4-unit processing of a single mono source, and preset 60 a "4 source: 4 mono out" set-up for single unit processing of four individual mono signals. Look at your design plan to see which type of source config you require as this will dictate which of the templates to use.

Once you have decided on your template you are ready to select the unit preset(s) it will use. The ROM templates have a dry "no effect" algorithm in each of the units so these need to be replaced with the preset(s) you have (hopefully!) already prepared. So, select your config template, press relevant unit buttons as required, and use

the data entry knob to select your unit preset(s).

If you are not using one of the ROM templates but one of your own previously programmed configs instead, you may want to replace some or all of the existing algorithms. This means you either need to change to a different preset that contains the required effects, or change individual algorithms loaded into the units of the current preset.

In Select mode you can only choose replacement presets that are appropriate to the config type. This is not a problem for 1-source configs where you are only dealing with single units. 2-Source configs can only select 2-unit presets, and with 3-source configs 1- or 2-unit presets are available according to whether a single unit or pair of units is selected. 1-Source configs, being the most complicated, have the most restrictions as they can only select 4-unit presets. However, any individual units in any preset can be easily changed by going into Edit mode, selecting a unit, and then choosing a different algorithm. You will be choosing from the 1-unit presets you already have in memory so make sure that you have an appropriate one available.

One trick to mention here which is tucked away at the back of the manual, is a way to load a 2-unit preset whilst in a 1-source config. In Edit mode press the A and B, or C and D, buttons at the same time and you will have access to the 2 unit presets in memory. You can then select one and place it your chosen pair of units.

Now you can move on to edit the template's config parameters. Make sure the yellow config LED is on (if not, press the Config button) and then press the Edit button.

The first parameter, number 00, doesn't need to be changed because you have already selected an appropriate template, but if you are editing an existing config this is where you can change the input configuration. This governs what the remaining parameters will be and depending on the type there are six, eight, or ten parameters to program. Most of the parameters refer to signal routing between the units.

For 1-source configs you can select routing between units AB to CD, units A to B, and units C to D. There are also two config dependent parameters setting dry path or feedback amounts and a choice of mono or stereo input to the AB units. The last four parameters are the same for all four source types. They decide what happens when a unit bypass button is pressed: bypass lets a dry signal through only, kill lets no signal through.

2-Source configs lose the AB to CD routing, which is not applicable, but gains a mono or stereo input choice for the CD units. Slightly different again are the 3-source config parameters. There is a choice of whether AB unit outputs will be dual mono or mixed stereo, CD unit routing, CD input select, and a config dependent parameter for dry path/feedback amounts.

The 4-source config is the simplest of the four types. Apart from the bypass settings there are only two other parameters: AB and CD units output selection.

Unless you need to have complicated feedback routings, your design plan should make the programming of the config parameters fairly quick and easy.

Stage 4: Algorithm Programming and Testing

This stage is probably the one you will spend most time on as a lot of the algorithms have a large number of para-

meters and you can spend ages tweaking them to get as close as possible to the result you want.

Using your notes from the design plan, start to edit each of the unit's algorithms using the rough parameter values you decided upon in your plan. If the units are serial or feedback routed you should pay particular attention to the parameters which govern how much signal is passed between them, as these can greatly affect the quality of the output signal.

At this point it may be possible that you need to swap around the order of the loaded algorithms in your config, and there is an easy way of doing this by swapping any two units. Press Edit, Write, and one of the unit buttons you want to swap, and then press and hold that unit button whilst simultaneously pressing the unit button of the second unit. The display will confirm what you are about to do before you finally press Write to do the swap. You can use this method to quickly hear what would happen if the order of units is changed.

Next, put some sample source material through your config and see how close you are to achieving what you want. By using the bypass buttons you can isolate each of the units to check out the way they are processing your sound source. Doing this should very quickly show up any things you might have forgotten in the planning stage. Once any major anomalies have been corrected all you have to do is make the fine tuning needed to make the config match your exact requirements. Remember that when editing algorithm parameters you can use the Edit button to toggle between the version in the edit buffer and the original to hear what changes your edits have made.

When you are completely happy with your config, make sure that it, and the relevant unit preset(s), are safely written to new presets. It is worth thinking up a good descriptive name for it which you will remember later. Don't forget to use the unit buttons as a shortcut to get to the various groups of alphanumeric characters.

Also, I know it can be a bit of a chore but it really is a good idea to try and keep a note of what each of your custom-built configs does. You never know if you might be able to use one of them as a template for a future config which needs a similar design. It doesn't take that long to jot down a few notes and, after all, why duplicate programming that already exists. You can very quickly build-up a whole library of useful configs in this way.

In Part Three I will provide a real-life example which puts all this theory into practice. See you then. ▬

Bio: Steve Byhurst is a thirty-something British composer of electronic music who is quite adamant that his compositions should not be classed as "New Age" or "New Instrumental."

Automated Delays

Ray Legnini

Hello again delay fans. In today's lesson we will explore more practical applications using Tempo Delay. The information provided here applies to the DP/4, DP/4+ and the DP/2. I'll note any changes that apply specifically to one unit or the other as we go.

In our hypothetical example, we'll presume that your song has a chorus section where you'd like to have quarter note echoes and a verse section where you'd prefer sixteenth note echoes. Let's presume as well that you want to have this change automated so that your MIDI sequencer sends the modulation commands to the DP.

For simplicity, let's send a Mod Wheel (MIDI Controller #1) on Channel 1 to the DP as the command to change the delay units from quarter notes to sixteenth notes. A Mod Wheel value of "0" will select the minimum value and a value of "127" will select the max. You don't actually need a keyboard from which to send the controller information, you can just add the commands to your sequence at the appropriate spots. Once you understand the basic concept, you can customize this setup to suit your individual needs. You can select any controller and any MIDI channel for the messages to be sent on. If you need more info on this topic, see your manual.

You'll need to hook up a MIDI cable from your sequencer's MIDI Out to the DP MIDI In. Set up your sound system with a single aux send connected to the DP input and patch the stereo outputs to your mixer. Select the "Select 1U Preset" Configuration on your DP.

If you call up the 1-Unit Tempo Delay algorithm into Unit A (1U preset #63 in the DP/4 and DP/4+, #88 in the DP/2) you will find that the default version has some modulation routings already set up for you (they are slightly different between the three models). The first modulation is the one we're interested in now. Check with your particular unit that there is nothing being modulated by the second set of parameters. Do this by hitting the "Edit" button, and then use the left/right arrows to move through the editable parameters. Disable the second modulation by setting the parameter "Mod2 Src" to "Off."

The DPs have eight available controllers that can be used as modulators. The sources of these are selectable by the user. They can be any legal MIDI controller. The settings for these are found in the System pages. Hit the "System" button and scroll to parameter #37 on the DP/4's or #23 on the DP/2. The list of eight DP controllers starts here. As you scroll through you'll notice that they have been set at the factory to some of the most used MIDI controllers. Since we can't presume the state that your unit is in from previous uses, let's select "Controller 1" and set the value of this parameter to "Mod Wheel #001." Remember "Controller 1." This is the info you'll need when setting up the modulation in the Tempo Delay. You'll enter this into the "Mod Src" field. While you are still in the System parameters go to parameter #35 on the DP/4's and #22 on the DP/2 (Control Chan =___). This selects the MIDI channel on which controller commands will be received. Again, for simplicity, set your Control Channel to "01" and check to see that the lower portion of the display reads "MIDI is Enabled." We're done with the System parameters. Hit the "Edit" and then the Unit "A" button to return to editing the Tempo Delay.

Select the "Tempo Control" parameter, #06. Set this to "MIDI clocks" — this will allow your sequencer to control the tempo. Scroll to the "Mod1 Src" parameter. Set this to read "Cntrl-1." The lower portion of the display will now read "Mod Wheel #001" assuming you did everything correctly. Set the "Mod Destination" to read "Parameter 003." This is the parameter that controls which musical unit the delay will produce.

But what values do you need to enter into the modulation Minimum and Maximum fields to achieve the desired effect proposed for our lesson? Read on. The available values for modulation amount go from 00% to 99%. To determine the relationship between percentage and the actual note value, set a mod amount for the "Min" parameter (parameter #12 on the DP/4, parameter #13 on the DP/4+ and the DP/2) and then scroll back to parameter 03 and look at the value as you move the controller to the minimum value or send the value from your sequencer. A handy feature to remember when dealing with any modulated parameters: the value for a modulated parameter will update in real time if you are on that parameter's edit page.

There are twelve musical divisions available in the DP's

Tempo Delay, from 1/32nd notes to 1/2 notes. With a calculator, you can determine that approximately every 8 units of modulation you add will give you the next musical division in the list. I've already done the work for you. Here are the modulation values and the corresponding musical unit that they will select:

00% selects 1/32 note
08% selects 1/16 triplet
17% selects 1/16 note
25% selects 1/16 dotted
34% selects 1/8 triplet
42% selects 1/8 notes
50% selects 1/8 dotted
59% selects 1/4 triplet
67% selects 1/4 note
75% selects 1/4 dotted
84% selects 1/2 triplet
92% selects 1/2 note

For our example, we select modulation amounts of 17% and 67% to get the two delay units we want. Enter "17%" into the "Mod 1 Param Range Min" field and "67%" into the "Mod 1 Param Range Max" field. Now it's a simple matter of adding a Mod Wheel into your sequence at the measure where you want the changes to happen. Insert a value of "0" in your sequencer and you'll get the 1/16 note delay; insert a mod wheel value of "127" and you'll get the 1/4 note delay. Confirm the changes by watching the display update while you are still in edit mode with parameter 03 showing. Armed with the table above, it's easy to change values to suit any song. Since MIDI is controlling the tempo, this setup can work with any song in your library. As always, save this variation. ▬

Bio: Ray Legnini is the founder of the S.P.C.A. (The Society for the Prevention of Cruelty to Analog). Contributions to this worthy fund are always welcomed.

The DP/4 as Rain Maker

Johnny Klonaris

Thunderstorms in northern California? Yup, as I write this, it's pouring down rain. Our relationship with rain has been more intense than usual these last few years — the thunder is just an added bonus. Still, the sound of a thunderstorm is my favorite sonic experience. I recall my youth in Michigan — lying awake on a hot summer night with the wind in the elms, rain, thunder distant and sometimes surprisingly near, add that to the sound of freight trains and freight planes, crickets, traffic; just an amazing cacophony. But I digress before I start.

I admit to loving the sounds of rain. Overworked, maybe, but so it was that the patch CV-Wet Thunder! caught my ear when I first got my DP/4 home and stepped through the patches. But it was a long time before I got around to really looking into how this patch works. Once I did, it surprised me; there's a lot more to this patch than you might think.

For me, the CV-Wet Thunder! patch was the one where I suspected something might be wrong with the DP/4. When I called it up, I got what at first sounded like just hiss. Then I realized it sounded more like rain. I tried running some sound through it, but it didn't respond. When I looked into the patch a bit, I saw that the first unit (A) was set up as a noise source. Ah — processed white noise, no big deal, so I just left it for a while. I did notice however, that the sound wasn't just wide band noise, somehow it sounded more like rain.

Just what is it about rain? Like wind and surf, it's one of the classic relaxation sounds. One reason would be the masking power of these sounds. They cover a wide range of frequencies which can cover other sounds much like a fog obscures vision. But there is more. Listening to an FM radio tuned off-station is pretty close to random noise. There is a similarity to rain, but if you asked people, "Is that sound, rain?" not too many would say yes. It's the differences between pure noise and the natural noises that are the interesting parts. Actual rain has some structure to the sound that makes it different to our ears from other sounds.

It was that structure that drew me back to this unassuming, four unit patch on the DP/4. I decided to find the "Thunder" part. It wasn't too tough for a human. I eventually figured out that the "CV" in the name referred to the use of the control voltage pedal to modify the balance between a sine wave and noise. I switched this to use the mod wheel instead and played around a bit. Sure enough, rain at one end, and at the other end, a deep rumbling not all that different from distant thunder. I decided to figure out just how a sine wave gets turned into thunder.

My first guess was completely wrong. I expected some super-low frequency, nearly subsonic sine wave. Nope: 136Hz — low, but hardly subsonic. That's when I decided to take this small mystery apart. If you've got a DP/4, you might want to call up the patch (4U preset 49). If you've got a CV pedal, plug it in and you're ready to go. If not, you'll need some other modulation source. Mod wheel is pretty common and easy enough to set up. Select Unit A and press Edit. Use the arrow keys to pull up parameter 11 and change the modulation source to something convenient for you. Now you can fade between a gentle rain, and rain with distant thunder (the modulation range parameters are set to make this so).

The patch uses all four units configured as:

Unit A	Unit B	Unit C	Unit D
Sine/Noise Generator	Phaser	DDL	Reverb

By disabling units B, C and D, you hear the unaffected noise or sine wave depending on the modulation you're using. Try turning the other units on one at a time and you'll get an idea of how the last three units manage to randomize the sound and give it some structure. The phaser sweeps up and down the sound, adding a bit of structure to it, but nothing much by itself. The DDL is what does the majority of the mangling. It takes the sine wave and pitch shifts up and down every second or

so. It also adds several "random" echoes since the LFO is cycling at a period that is close to the delay of the DDL. Still, with the Phaser and DDL, the sound is more like an overprocessed sound than rain and thunder. The Reverb finished the job. It smooths out most of the discernible artifacts of the Phaser and DDL and you're left with a sound that might not be what you would expect just by looking at the configuration.

So what?

Kinda my thought. But then I thought to bypass Unit A. Now I've got a patch that will turn any sound into a rainy day. (See what too much rain can do to you?) But this is just the start. Any good reverb will randomize a sound to some extent. Now we can take that to an extreme, or anywhere in between.

It's the in between area that I usually find most interesting.

The rest of this article will discuss the ideas behind some changes that are possible. The side bars have more detailed instructions. In addition, patch files are available for downloading to your DP/4. More on all this later.

Rain Verb

It seemed to me that it should be possible to do a sort of a sea shell for rain. Something that sounded like rain, but only when there are other sounds around. You might first try playing music through the modified Wet-Thunder patch to see what it does to musical sounds; simply bypass Unit A and play some sound through the first input. I think you'll notice that to change the patch so that musical sounds would sound like rain requires that a few things happen. For one, percussive sounds will definitely peek through a DDL and your ear will immediately pick up on the distinct echoes and recognize the sound as not rain. Higher pitched sounds will also need to be dealt with as these come through quite recognizably, especially in the first few seconds of the sound.

The first thing I did was to replace the GEN module with a pitch shifter. This allowed me to broaden the pitch a bit more going into the rest of the processing so that higher pitched sounds would be scattered further, faster. What really made this work was to change the input configuration so that units A and B were in a feedback loop. I set the pitch shifters to -.90 and +1.50 semitones (which works out to a sort of 3:5 ratio. With the feedback loop this meant that each pass through, all tones would get split into two tones one slightly lower, and one slightly more higher, and a complex tree of pitches, branching higher and lower comes out, with a slight tendency to rise, since there is more up shift than down shift. This particular setup also has a slight cricket tendency; a pure tone through the AB pair will produce a chirping pattern on the order of 1-2 Hz that doesn't hurt the illusion too much. This mostly solved the high pitched sound problem.

I next tightened up the delay; the separate echoes are more obvious with musical sounds than with pure pitch or noise: with music you've got more cues to help you pick out the individual echoes. I changed the left and right delay times on unit C from 700ms and 400ms to 210ms and 160ms, keeping the same 7:4 ratio. This shortened the times between echoes so that they were likely to be lost in the reverb decay.

Bumping up the reverb decay to a hefty 11.4 seconds finishes the plastering-over job. Most any sound run though this 4 unit patch will come out sounding quite a bit different than when it went in.

Surf Verb

A simple modification can produce another special purpose patch. By changing the predominance of the pitch shifting to downward rather than upward, we come up with a patch that, to these ears at least, sound more like crashing surf. I simply modified both pitch shifters to shift down by five to six semi-tones.

Admittedly, the applications for both of these patches are rare: they are definitely for special purposes, but sometimes that's just what you need.

In the spirit of exploring the middle ground mentioned earlier, I decided to see if I could create more musically useful patches.

Cave Verb

Going back to Rain Verb and reducing some of the pitch modulation and other mangling, we can convert this into a patch that retains some of the musical struc-

Creating the Patches

Here are some step-by-step instructions for creating the various patches mentioned in the article. In general, when I capitalize an entire word, I mean that you should press that button. This allows me to use the syntax of the DP/4 and the syntax of English at the same time — when that works.

Modified CV-Wet Thunder!

To call up CV-Wet Thunder, dial up CONFIG 50 and SELECT it. Now press A and dial up number 49 and SELECT it. You should start hearing a gentle rain. To modify this patch to use the mod wheel instead of the control voltage pedal, EDIT parameter 11 (use the arrow keys) and change it to a value you can use for modulation. (Cntrl-8 Mod Wheel works well for me).

To bypass the Noise/Sine Generator, simply press A twice. Now you can play sounds through the Wet Thunder patch to see what it does. Try bypassing units B, C and D to find out what each contributes to the patch. At this point you can play with the sin/noise balance and/or bypass Unit A to hear what the patch does with musical sounds.

Rain Verb

To create the Rain Verb patch, set up for 4 Unit preset: SELECT, CONFIG, dial-up 50, SELECT. To get the patch, press A, dial back one to 49, SELECT. The display should show "CV-Wet Thunder!" With A selected, press EDIT and dial-up 80: Pitchshifter. Using the arrow keys to select parameters and the dial to change the values, change the following parameters:

Arrows	Dial
03 Semi	+02
04 Fine	+50
07 Semi	+00
08 Fine	-90
12 Rate	48
13 Width	30

Press B and change

01 Mix	75
02 Volume	99

Press C and change

03 DelayTime	210 ms
04 DelayTime	160 ms
05 Rate	66
06 Width	25
08 Delay Regen	-34
09 Cross Regen	+44

Press D and change

03 Delay	11.4 sec

Press CONFIG and change

02 [A<>B] feedback	2
04 Amount	75

At this point you have Rain Verb. You can save this away as a 4U preset or a Config preset — your choice.

Surf Verb

To create Surf Verb, start with the Rain Verb you just created, and make the following EDITs:

Arrows	Dial

Press A and change

03 Semi	-05 *
07 Semi	-06

*Note, this gives a pitch change of -5 semi tones +50. A bit odd, but it works.

That's it. Hang ten.

Cave Verb

For Cave Verb, once again call up Rain Verb, and EDIT, then:

Arrows	Dial

Press A and change

03 Semi	+01
13 Width	10

Press B and change

04 Width	50

Press C and change

06 Width	10

Press D and change

03 Decay	5.50 sec

This should inspire some spelunking music, perhaps on the order of Pink Floyd. Or not.

Downloading Patches

The realities of transferring data in these times requires us to consider a small number of computer types and a similar number of transfer methods. Between the ubiquitous (oh, look it up, you'll see it again) PCs and Macs, as well as Unix systems, Amigas, Ataris and others as well as transfer methods such as ftp, email, and the current be all and end all, "Da Web," it could seem like it's all too complicated. Well, sure — get use to it. Still, there are some islands of common ground. The arc program and its various incarnations such as lharc, lzh and others, do exist in some form on every computer platform I've run across. So, that (specifically .lzh format) is my personal choice for providing these patches to the world.

Details on actually downloading the file are left up to you. Mostly. The file is named "rainmakr.lzh" and is located at the *Hacker's* ftp site. If you're using a browser, this is linked to the *Hacker's* home page at http://www.transoniq.com/~trnsoniq. The ftp site itself is at ftp.transoniq.com/~trnsoniq.

The file, once on your computer, can be broken into a group of files that can be sent to your DP/4. What you'll need is a computer, a program capable of reading and dissolving .lzh files, and some way of sending a SysEx file from your computer to your DP/4; many sequencers can do this. There are also public domain and shareware programs like MIDI-TERM and others that will send the contents of the SysEx files to the MIDI interface. The files contain the actual System Exclusive messages: they start with hex 0xF0 and end with 0xF7.

On my VFX, if you save a single patch with SysEx and send that patch back to the VFX, it goes into the edit buffer and you can save it where you want. Unfortunately, that feature is not implemented in the DP/4. The files I provide will overwrite the patches on your DP/4 in the same locations as they were on my DP/4. This particular conundrum was much more of a challenge than it seemed it should be, but I settled on a solution that I suspect will work for most.

The patches that I'm making available will overwrite the Config presets in the last three memory locations 47-49. I figure if you have the means to transfer these files to your DP/4, you should have the means to backup what is in those locations. I realize that my not be true for everyone, but it seems a reasonable compromise.

So, the file: rainmakr.lzh contains three files:

Filename	Preset # and name
rainverb.syx	47 Rain Verb
surfverb.syx	48 Surf Verb
caveverb.syx	49 Cave Verb

Each file contains one SysEx message that will put the named Config Preset in the preset number shown.

To make sure your DP/4 will receive and understand these messages, make sure you have the following parameters set:

Press System (until 50 shows up):

50 MIDI SysEx ID=01
51 Receive Enabled
52 Preset Memory Protect=Off

Set up this way, if your DP/4 receives the SysEx message in the file rainverb.syx the display should show:

Cfg Preset 47
Dump Received

If so, you're on your way. Keep dry.

ture that goes into it. I reduced the range of the pitch shifting, and it's LFO range, the depth of the phase shifter, and the range of the LFO on the DDL to reduce the amount of randomization of the sound, and I reduced the reverb decay to lessen the amount of confusion added at the end. The result is a patch that still retains some of the rain-like structure of Rain Verb, but that lets more of the original sound through, and is thus more like a regular reverb. This patch could find some use in a more musical, and less special effect setting.

More Verb

Okay, I admit it, I'm basically lazy. The idea of having to come up with one more patch name, four-letter "Verb" was more than I could stand, so I gave up creating more patches. Seriously, the idea here is that you can continue the process. By reducing the amount that each of the units changes the sound, you should be able to come up with a kind of effect that twists the sound around by the amount you want, to create the balance you want between some sort of reverb, and some sort of special effect like changing a beautiful guitar solo into a nightmare of a tidal wave. The fact is you get to choose. And I suppose that's the point of these wonderful toys that technology, Ensoniq, and our occasional ability to afford them, produce.

Free Verb

You can follow along in the step-by-step instructions and create your own copies of the patches I've described. If you want to learn more about how these things work, breaking them down and playing around with them is a great way to do that. At the same time, different folks have different needs. To that end, the patches above are also available for download, ready to turn your DP/4 into your own, personal rain maker. (See the sidebar on downloading patches.)

Last Verb

I hope this little exploration into the wetter side of reverberation has given you some ideas for ways to get into your DP/4 and do more with the possibilities it offers. I had fun making everything just a little bit wetter, but now I've got to cut the grass. ▬

DP/4 Custom Configs

Part III — An Example: The Rejuvenator

Steve Byhurst

In Parts One and Two I looked at my own methodology for creating custom-built configs. For this final part I will produce a real-life example following through the stages I have already presented.

Design and Preparation

A few months back I came across some early recordings of mine which I had mastered onto compact cassette tape over ten years ago. Some of the tape was already showing signs of deterioration and so I was eager to transfer the recordings onto DAT so as not to permanently lose them.

The poor sound quality of the recordings was such that I wanted to process them to tidy-up and improve the quality before re-mastering. I decided that this was a job for my

trusty DP/4 and, as I didn't already have a config that did what I needed, I had to program a new one which I would call, for obvious reasons, the Rejuvenator.

The general purpose of the config was to make the DP/4 act as one multi-effect processor using all four units to process the stereo signal from a cassette player and send the improved stereo output to the DAT player. The answers in my design plan were as follows:

(a) The algorithms required will be: a noise gate to clear up noise before the start and after the end of recording; a parametric EQ to get rid of any unwanted hums, clicks, etc.; an exciter to compensate for loss of frequencies in the original sound source; and a reverb to add some ambience to the somewhat dried-out recordings.

(b) As I want to use the DP/4 as one integrated processor I can have one unit per effect. No multi-effect algorithms are required. I will use the Expander because it has the best noise gate, the comprehensive Parametric EQ, the Van Der Pol Filter for excitement(!), and the Hall Reverb as a good quality ambience provider.

(c) I need to process stereo inputs so the cassette outputs will go into inputs 1 and 2.

(d) The output to DAT will be in stereo from outputs 1 and 2.

(e) The signal first of all needs to be processed by the Expander, then fed into the Parametric EQ, the Van Der Pol Filter, and finally the Hall Reverb. As I want a chain of effects, the units will be serially linked in that order with no feedback.

(f) Because of the nature of this particular config, most of the algorithm settings need to be experimented with at the testing stage. Default values will be fine as a starting point.

(g) It will be more useful to have the bypass mode set to Bypass rather than Kill for all four units.

I started my preparation by deciding that I would use one of the ROM templates as my base for programming as I didn't have an existing config that would be worth using. The design plan told me that I needed to set up a 1 source config using one 4 unit preset, so the first thing I had to do was create a new preset with all the required algorithms loaded in the correct processing order.

I created a 4-unit preset called "Rejuvenator" (to be consistent) by using a "no effect" template I had in my preset library and loading in the algorithms from the standard 1 unit preset list one by one. Now I was ready for the creation of the config.

Config Programming and Testing

Config programming was, in this case, fairly simple. First, my plan told me I needed to select ROM preset 54, a "1 source:stereo in" type, to act as my template. Then I loaded up my prepared 4 unit preset and, as the preset was custom-made and needed no editing, went straight into config edit.

The type of source was, of course, already correct so no need to change that. I made links between all the units serial and set dry signal amounts to zero. The type of input to the units was set to stereo and all four units were set to bypass status so that I could test out the effect each algorithm had on the original signal. Finally, I christened the config with its proper name and wrote it to a spare preset.

The algorithm programming stage was especially important with this particular config as the parameters of each effect needed to be programmed mainly by alternately listening and then adjusting until the right result was achieved. They also depended on the variance in quality of the source signal and even the initial programmed values would need to be altered according to the changes in sound quality from individual recordings. I therefore decided to set up values which I thought would be right most of the time. I could always tweak them in real-time when it actually came to the re-mastering process.

First, I set all of the volume parameters to a full 99 so that each unit processed the full output of the unit that was feeding it. For mix values I set the first two units to a full 99 for full processing and the last two to an initial 00 as the exciter and reverb was not always required. I also decided that I would use MIDI to control the volume setting of unit D and thus the final volume amount sent to the DAT recorder. This provided me with a means to control overall volume by using something like a modulation wheel, and thus the ability to make smooth and more easily controlled fades.

The expander was carefully set up to get rid of unwanted noise at the beginning and end of tracks. It has parameters which enable quite sophisticated control of the way the noise gate functions but needs to be programmed according to the dynamics of any one piece of music. I was also able to override it in real- time by using the bypass button if I wanted manual control of the volume.

The parametric EQ was initially set up to have no effect on signals but when required could be used in a very subtle way to either cut noisy or boomy sounds, or boost bass or treble as required.

The Van Der Pol filter is normally used on individual sounds and you have to be very careful when using it on an entire mix, using only small amounts. It appears to be a very subtle effect but using the bypass switch to toggle between the original and effected sound can help to check what it is adding to signals. I set it up to give a mild effect when I needed it.

The reverb was required because some of my source recordings were very dry and desperately needed some ambience to bring them alive. Small amounts of a slightly-tweaked version of the hall reverb gave me what I wanted, adding some space to the sound without making it too smudged.

With all the parameters set to initial values I played one of my recordings through the DP/4 and tweaked away until I was happy, which actually didn't take too long. I found the bypass and edit buttons very useful in checking out the processing of each algorithm. The end result was achieved and the quality of sound from the re-mastered recording was much improved.

That's It

If you are actually interested in using the Rejuvenator on your own recordings, I haven't provided a complete set of programming values because, as I have said, the settings of the algorithms will change according to the type of source signal. However, by following what I have written you should be able to program the unit preset and config parameters, and I leave the algorithm programming to you so that you can customize them to your own circumstances. Why not try out this method for yourself? Have a nice config! ▬

Bio: Steve Byhurst is a thirty-something British composer of electronic music who is quite adamant that his compositions should not be classed as "New Age," "New Instrumental," or placed in any other meaningless category. So there!

Making the Most of the DP/4

Efficient Unit Usage

Steve Byhurst

Now some of you out there may be lucky enough to have more than one DP/4 or other dedicated effects processor, but I suspect a fair number, like myself, do not. Even if you do, most will not be able to afford the luxury of not having to worry about using each one to its full extent. Here are some ideas for using the units of the DP/4 as efficiently as possible.

The Unit's Place in the Scheme of Things

As we should all know by now, the DP/4 offers us four independent effects processing units. These are known by the rather logical, though not very sexy, names of A, B, C, and, yeah, D. They may be used singly or in various combinations if more power is needed to produce a particularly complex effect.

Each of the units has exactly the same processing power and can be programmed to use any one of a specific list of algorithms contained within the DP/4. An algorithm is simply a term used to describe a single effect or set of effects within one processing block. Most of the algorithms concentrate on doing one particular job but may include extras where appropriate. Others may have two or three functions which can be used singly or in conjunction with one another. Programmable parameters within each unit preset control all the effects of the algorithm it has been loaded with.

At the next level up, config presets decide if and how the units are connected to each other, and which inputs and outputs of the DP/4 are to be used.

The Efficient Unit Strategy

The first thing you need to do is to go through the list of available algorithms and make sure that you know what each offers. Some will be single-effect algorithms with fine control over the function they provide. Others will be multi-effects which will concentrate on giving you two or three effects with only essential control parameters. In particular, notice those which give extras like a noise gate or an EQ section. Through everyday use you will automatically get to know which algorithm may be suitable for any given situation.

Once you have a good knowledge of potential algorithms, you need to look at the limitations on actually using them. There is a lot of flexibility inherent in the operating system of the DP/4, but with a limited amount of processing power available some of the possible set-ups will inevitably have restrictions. You need to know these to make efficient programming decisions. The major factor governing the flexible use of units is the type of source config used.

It's Those Source Configs Again

The choice of source config for any particular project

will in turn have a bearing on how much choice you will have in deciding which type of algorithm to use.

Selecting a 1-source config means that you will be processing a single signal and can therefore use the maximum power of all four units. This gives you more freedom than the other types and much more scope in programming. If you want an extremely complicated multi-effect you could use four combinations of multi-effect algorithms, though it is unlikely that you will need this level of processing. Probably one or two multi-effect algorithms coupled with two or three single ones will give you the required power. Single-effect algorithms in all four units will give you less effects but more control over how each of the ones you do have sound. Alternatively, if you only want one very complicated single effect (like a complex reverb), load algorithms of the same type into each unit and vary their parameters and routings to produce your own unique sound.

A 2-source config divides your processing power between two sets of two units. With this setup you could say that you only have half of the flexibility of a 1-source config for each of your inputs, but there is still a lot you can achieve. Multi-effect algorithms are more important here. Make as much use of them as you can. For instance, try using the DDL and EQ functions contained in most of the multi-effect algorithms instead of using the single EQ and Delay algorithms. They compromise on programmable parameters but most of the time you will find that they have enough power to get you close to what you want.

I find that 3-source configs are used less often than the other types because the facility they offer is for quite specialized set-ups. However, the same principle of making choices between single- and multi-effect algorithms, according to need, still applies.

The 4-source config, despite being one of the most useful configs, is the least flexible in terms of using units. Here, if you want anything more than a simple effect, you simply have to choose a multi-effect algorithm. For most purposes this isn't as bad as it sounds as it is worth a little bit of compromise in your choice of effects to have in return the facility to independently process four different inputs.

I should just mention here that a few of the algorithms are ganged, i.e., they need more than the power of a single unit and therefore can only be loaded into multiple units. The options for using them with other effects are limited (or in the case of the Vocoder, zero). However, these are quite specialized effects which are not required in the majority of multi-effect situations and so the question of efficiency is not really relevant.

Making Use of Unemployed Units

If you find that you have unused capacity in the DP/4 (one or two units not being used), there are a number of ways in which you could make your parallel effects processor feel more fully employed.

Firstly, and most obviously, check to see if you could improve upon your setup by using those spare units, maybe by replacing multi-effect algorithms with single ones. With some of the time delay algorithms you could try duplicating an effect in another unit and then slightly change the parameters to get a thicker, richer sound.

Should you have signal inputs that are not being used, and the type of config you have loaded allows you to use them, why not connect up another sound source and put some effects on that input? This could be used to augment the sound source you have on the other inputs or could be something entirely new. Alternatively, split the input so that the same sound source goes into two separate units (or pairs of units), and you could toggle between the two to get totally different effect treatments. Using MIDI to do this remotely or automatically via sequencer commands could yield some interesting results.

If you do not have or cannot use more signal inputs, you could still direct your input signal to the spare units for processing that is totally separate from what the other units are doing. Again you could toggle between the two by using the bypass controllers, or perhaps send the output of the alternative effects to be recorded. In this way you could record two different effect mixes at the same time, maybe to play back later to decide on which is best. ▬

Bio: Steve Byhurst is a thirty-something British composer of electronic music who would really welcome generous offers of financial sponsorship. Go on, surprise me!

ly some of the fore-mentioned products do not have disk drives (SQ family, ESQ-1, VFX, KS-32, KT-76/88 do not have disk drives). They use two other means of storage, RAM cards/cartridges or Sysex. RAM cards and cartridges are becoming more and more scarce because of the age of this technology. They can still be purchased through Syntaur, 500 West Prairie Ave., Eagle Lake, TX 77434. Phone: 409-234-2700, 800-334-1288. Net: http//www.fatsnake.com/syntaur.

The advantage of using RAM cards is that the transfer is very fast and they allow you to access an additional bank of sounds.

It may be better to use Sysex for storage if you have to save a lot of files or very large files. There are products on the market such as the Yamaha MDF-2 which will allow you to send and receive data via Sysex and save the data to a standard floppy disk. You can also do a Sysex dump to a computer, provided you have a MIDI interface and software installed on the computer.

As you already know this, consider it a reminder. Sysex is short for System Exclusive. Before anyone asks, no, you cannot transfer Sysex info to or from two different model units, like from the SQ-80 to a SQ-1. It does not work that way. The data is "exclusive" to that particular product. Keep in mind though that it will work between two of the same product (ESQ-1 to another ESQ).

I hope this will help when you are trying to use your keyboard with an external sequencer. In each case you can see that this is a simple procedure and does not require you to have a degree in phisiomiscaboobalation. ▬

Bio: Eric Montgomery's work has appeared in Ensoniq products and videos and in projects from Integrity Music and Salt Records.

A Via Musictone

Tom Tracy

This DP/4 preset, for the perspiring guitarist, pays tribute to Eric Johnson. That is not to say that Eric uses this DP/4 preset. To the best of my knowledge, Eric does not use or endorse any Ensoniq products. This was created as a way to recreate his "sound." Thanks, Eric, for the inspiration.

History

One of the benefits of being an Ensoniq employee (specifically the in-house technical writer) is that you often get to try new products before anybody else does. That was the case with the DP/4. As it was being developed, many people in-house (as well as "the chosen few" out of house) were involved in the testing, troubleshooting and critiquing of the DP/4. Not only did I get to write about how this great processor works, but I got to make suggestions (some that were implemented) in regard to the architecture, the ease of use, *and* I got to create many presets that were actually released in the final product! Obviously, since it was a complete departure from any other Ensoniq product, there was no other Ensoniq product to base this one on, and there were little to no presets in the alpha units (most locations said *BLANK*).

As you can imagine, knowing that the DP/4 needed 400 killer presets before it could be released, everybody involved worked diligently to create what they thought were the "ultimate" DP/4 presets. In order to create guitar-based presets for this new effects processor, I decided to first build a "generic" guitar for beta testing, and writing presets. The in-house "code" for the DP/4 was "Quaker." Every new product needed a code-name in case there were "spies" just waiting to steal information (which is always a possibility in this industry). Therefore, the guitar was affectionately called the "Quakercaster," and at one time, it was for sale exclusively through the pages of the *Transoniq Hacker* — but alas, nobody wanted it, so I still own it. Many presets that still existing in the DP/4 (and DP/4+) were designed using this very guitar.

Because everyone involved in the DP/4 beta program loved this product and worked hard to present it in its best light, Ensoniq's Sound Development was SWAMPED with presets — many of which were never used (there were just too many). This 4-unit preset falls into that category. Fortunately, I found that I had archived it via Sys-Ex to a TS-10 floppy and was glad I saved it. Here it is, for the first time in public — a bit of Ensoniq history.

The Tone Thing

As a guitarist striving to improve my chops, I borrowed an Eric Johnson instructional video from a fellow employee. Eric had (has) a unique tone that I thought would be a challenge to try and capture as a preset on the DP/4. When he played his Strat delicately, his sound was sweetly chorused; as he dug in, it was harder, with more guts and sustain. I think I came mighty close to capturing this sound (I'm still working on the technique, though).

Making the Preset

Avoiding all the usage of data entry controls (and press this, press that verbiage) to keep this article concise. Press Edit, and then set the following parameters:

Unit A= Parametric EQ
Mix= 99
Volume= 99
Bass Fc= 506 Hz
Gain= -25 dB LoShv
Mid1Fc= 778 Hz
Gain= -01 dB
Q= 05
Mid2Fc= 1450 Hz
Gain= -15 dB
Q= 01
TrebleFc= 11 kHz
Gain= +10 dB HiShv
EQ Input Level Attenuation= +00

Modulation
All modulation is disabled. For all four units, set the modulation parameters this way:
Mod1 Src=OFF
Mod1 Destination Parameter=OFF
Mod1 Param Range Min=00%
Mod1 Param Range Max=99%
Mod2 Src=OFF
Mod2 Destination Parameter=OFF
Mod2 Param Range Min=00%
Mod2 Param Range Max=99%

Unit B= Guitar Amp 4
Mix= 99
Volume= 43
Amp PreAmp Gain= +15 dB
Amp Output Level= 66
Amp Level Detect Attack= 1 ms
Amp Level Detect Release= 800 ms
Amp Tube Bias= 06
PreEQ Input Level Trim= +00 dB
PreEQ HighPass Cutoff= 150 Hz
Pre-EQ Fc= 600 Hz
Gain= +20 dB
Q= 06
Noise Gate Off Below= -70dB

Gate Release Time= 40 ms
Speaker High Pass Cutoff= 26 Hz
OutEQ1 Fc=120 Hz
Gain= +00 dB
Q= 05
OutEQ2 Fc=2500 Hz
Gain= +00 dB
Q= 06
Speaker LowPass Cutoff= 2.0 kHz

Unit C= EQ-Chorus-DDL
Mix=66
Volume=99
LFO Rate= 24
Width= 23
Chorus Center= 99
Left/Right LFO= In-Phase
Chorus Left Delay Time= 600 ms
Chorus Right Delay Time= 300 ms
Chorus Delay Regen= +14
Chorus Left Echo Time= 1000 ms
Chorus Right Echo Time= 750 ms
Chorus Echo Level= 00
Bass Fc=506 Hz
EQ Gain= +00 dB
Treble Fc= 11 kHz
EQ Gain= +10 dB
EQ Input Level Trim= +00 dB

Unit D= Hall Reverb
Mix= 47
Volume= 99
Hall Decay= 2.81 sec
Hall Predelay Time= 10
Hall LF Decay Time= +15
Hall HF Damping= 18
Hall HF Bandwidth= 99
Hall Diffusion1= 71
Hall Diffusion2= 64
Hall Decay Definition= 25
Hall Detune Rate= 25
Detune Depth= 15
Hall Primary Send= +50
Ref1 Time= 105 ms
Level= 00
Send= 00
Ref2 Time= 120 ms
Level= 00
Send= 00
Position Balance= +94 -17 +12

Config Parameters
1 Source Config 1,2 > ABCD
AB Input Select= (1) Mono
AB Unit Routing= [A->B] serial
CD Unit Routing= [C->D] serial
AB-CD Routing= AB->CD serial
Dry Path Around AB Amt= 00
Dry Path Around CD Amt= 00
(b)ypass (k)ill= A=b B=b C=b D=b

Save It

If you like this preset, name it and claim it (save it as explained in the musician's manual). Press Write and select a location using the big silver knob. Press Write again, and use the arrows and silver knob to name the preset. I named mine "Ah Via Musictone." Press Write once more. You've just saved the best guitar preset ever.

The Conclusion

The key to the best sound is that the Input Level knob is set properly. Make sure that when you play hard, it only occasionally goes into the red. If I may paraphrase Jack Nicholson from the film *As Good as it Gets*, "(this preset) makes me want to be a better player." Thanks for reading. If you've got any improvements or suggestions, please drop me a line (through the *Hacker*).

Bio: Tom Tracy is weathering the journey, taking the path less traveled. To Fehrion, wherever you are.