The Joy of EX

A guide to the
Yamaha EX5, EX5R, and EX7
Synthesisers

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1

About this book ...

The purpose of this book is to provide an easy introduction to the features and potential of the Yamaha EX5, EX5R, and EX7 synthesisers. The manual tends to treat every subject with equal weight and is thus incredibly boring and difficult to read. This book assumes you know how to power a synth on and off, and that you have a basic understanding of what MIDI is or does. You may need to consult the manual if you're unfamiliar with certain terms, but generally this guide should be sufficient to get you using the EX to it's fullest, programming your own sounds, and making music.

The EX is driven by one of the most powerful synthesis engines in the world, and this guide aims to show you how to make the best use of it. There are several different types of synthesis available :

1. AWM	Yamaha's wave playback synthesis engine, which provides access to hundreds of superb digital waves stored on 16 MB of internal ROM.
2. AN	Yamaha's Virtual Analogue Model algorithm, the heart of the Yamaha AN1x synthesiser, oscillators, filters, etc...
3. VL	Yamaha's Virtual Acoustic Model, the engine behind the Yamaha VL-1, VL1-m, and VL70m synthesisers *(this algorithm is not available in the EX7 model)*
4. FDSP	Formulated DSP, which is a new collection of algorithms offering synthesis and processing techniques, new to the EX series. Some of these are quite mad, and deserve your abuse.
5. Sampling	You can sample your own stuff, load samples from disks, and include these waveforms as elements in your voices.

Included with this guide is a floppy disk with a selection of voices. These voices will be referred to in the guide, because they demonstrate specific aspects of the EX synthesis engine. If you want to follow through the guide start to finish, then you should load these voices into your EX. Don't worry about your other internal voices, they should be on a disk somewhere. You kept those other disks, right? Basically, the voices and performances of this floppy disk are the same as the EX7 UK Factory Set #1, except that the first few Performances and Voices are changed for our examples.

We will also cover some of the compositional tools included in your EX, such as the fully programmable Arpeggiator, Sampling, the Keymap mode, and the built-in sequencer.

But remember, mash everything up, make your own sound, and eventually try to forget any of the rules you might learn in this book. Only a wimp uses factory presets, right?

Alright then...

2

Setting Up

If you’re not using your EX with an external sequencer, then setting up is just a matter of plugging it in and turning it on. Connect the audio outputs to your mixer or amplifier and you’re away.

Connecting to a Sequencer

If you’re incorporating your EX into an existing MIDI setup, or connecting it to any MIDI sequencer or computer, you’ll have to connect both the MIDI IN and MIDI OUT to the sequencer or computer’s MIDI interface. (On the EX5 use MIDI A ports for the time being.)

Usually in this case, you’ll set the EX to Local OFF mode, since the sequencer will route the MIDI from the keyboard to the currently selected instrument. (Local ON/OFF is in the Utility/MIDI page). If Local is OFF, then the EX is only responding to MIDI arriving at the MIDI IN port, so you must configure the sequencer to send the MIDI Thru, back to the EX again, on the desired channel. If you do this while Local is ON, you will hear each note flanging slightly, since it is being played twice, by the Local Keyboard AND from MIDI. You’ve probably encountered this before.

You can set the Global MIDI Transmit channel and the Global MIDI Receive channel in the Utility/MIDI page of your EX.

Voice Banks

There are four banks of Voices in the EX. Two of these are fixed in ROM (Preset 1 and Preset 2) and two of these are free for your use (Internal 1 and Internal 2). Each bank has 128 Voices in it.

Enter Voice mode (Voice button, top left, left of the LCD). The name of a voice should be in the LCD. At the bottom of the LCD you will see P1, P2, I1, and I2 above the first four F-Keys. These are your bank switches.

Go to Bank Internal 1, Voice 1, by hitting the F-Key for I1, then A-1 on the Program buttons at the right end of the front panel. If this says “Init Voice” there’s a good chance your EX Internal banks are empty. If there’s something other than “Init Voice” there, then someone has already loaded a set of voices into your Internal banks.

There should be some floppy disks that arrived with your EX. These each contain a file that you can load into the EX. They contain various collections of voices, including many duplicates, and some demo sequences. If you want to follow along with the examples used in this guide, load the floppy disk that came with it (it’s a Synth ALL file). You can always load the other voice sets again later anyway.

Performances

Beside the Voice button is Performance. Hit this one. You are now in Performance mode.

Performance mode can be used to make a multi (up to 16 voices, on different MIDI channels) - but it can also be used to make layers of voices, keyboard splits, or master keyboard-type setups that will allow you to control other synthesisers more efficiently.

There are 128 Performance locations, all of them user re-writable.

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The EX Architecture

Basic Structure

The EX has two basic “play” modes, Voice and Performance.

Voice is used for a single voice, which can contain up to 4 elements. Performance is used for designing a multi setup to use several voices simultaneously, or else to make voice layers, splits, etc. and provides Master Keyboard functions that allow you to control many synthesisers from the EX front panel.

DSP Limitations

Sampling and AWM elements are not limited. Using just this technology you can access the full polyphony of your machine (64 on EX7, 128 on EX5) without any restriction.

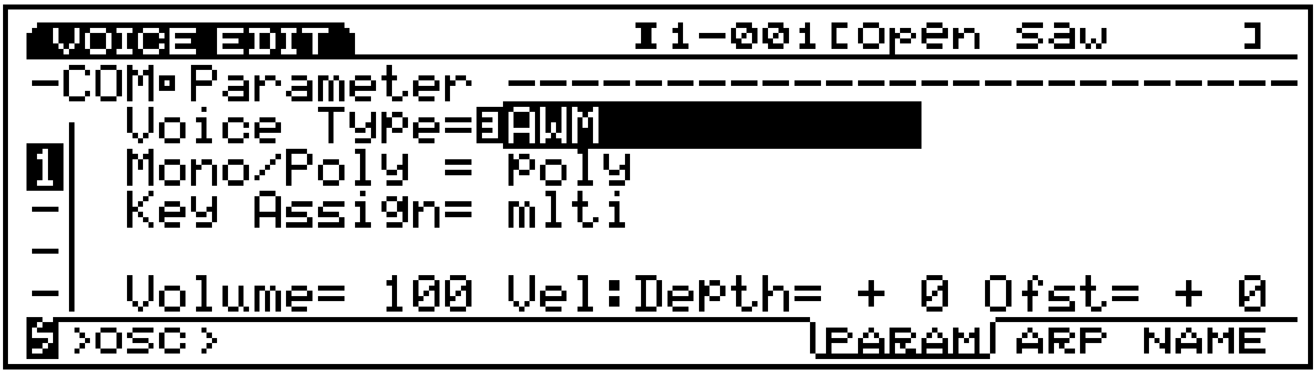
However, the insert effects, the VL algorithm, AN algorithm, and FDSP algorithms are all sharing the same area of DSP. When an element of this type is selected, the DSP area is sent the formula to perform that specific function. Since these “formulae” are of various sizes, it is hard to predict how many AN elements, or insert effects you may be able to use in a certain situation. Your EX will tell you when the DSP area is full, though, with a very irritating message that you’ll see over and over again when trying to make multi setups on your EX:

“DSP Resource Full” means you can’t turn on any more DSP algorithms without first going back and turning another off. It is up to you to manage the DSP and to decide how best to use it within any song setup. As you might imagine, the VL algorithm is very large, but a chorus effect is comparatively small.

Voice Types and Structure

There are several different TYPES of voices.

Put your machine in Voice mode, and hit EDIT/COM/Param. (By the way, we will continue to use this method of describing a page, in which the sequence of keys pressed are separated by a forward slash.) This is one of the Voice Common edit pages.



The first parameter on this page is your Voice Type.

The first choice of voice type is “AWM”. This is the common “wave playback” type of voice, and lets you include up to 4 elements, which can each include either ROM waveforms or your sampled RAM/FlashROM waveforms.

On the EX5 and EX5R you will find the second Voice Type “VL+AWM”. This means that element #1 is now a VL algorithm (the other 3 elements are still AWM). Unfortunately the EX7 can’t do this one, because the DSP area is half the size of the EX5.

Next you will find the Voice Type “FDSP”. This Formulated DSP voice is a collection of processes to which you must send an AWM element. So in this case, your elements are still AWM, but each can be sent (or not) through the chosen FDSP algorithm. If you’re still in the Voice Common page, you’ll notice that when you selected FDSP a new F-key appeared called FDSP (on F-key #5). Push this one, and you can decide what type of FDSP to use, and adjust it’s parameters. To the left of this page, you can see the “ElmSw” list of the 4 possible elements. Each of these can be turned ON or OFF to be either sent, or not sent, to the FDSP algorithm you’ve chosen. We’ll get further into FDSP in a little bit.

The following Voice Type is “AN+AWM” in which element #1 becomes an AN algorithm. The AN algorithm is the analogue model, very similar to that used in the AN1x synthesiser. In the EX5 and EX5R you can choose either AN (Poly) and AN (Layer). AN (Poly) is two notes of polyphony in one element, AN (Layer) is two elements that are each monophonic. The EX7 has only monphonic AN.

You can also choose Voice Type “AN+FDSP” *(EX5 and EX5R only),* in which the AN algorithm is sent through the FDSP algorithm.

Lastly, you can choose type “Drum” in which each key can access a different element, or group of elements. You can have 128 Elements in a Drum Voice.

Each of these voice types will be described in more detail in the Voice section.

Performance Structure

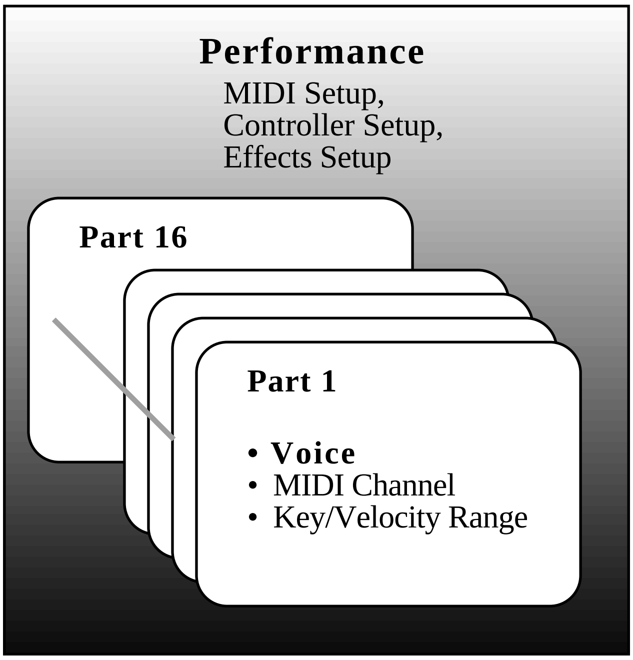
Perhaps the most common use of the Performance will be as a multi setup. In this case you will probably want each “Part” of the multi to be on a different MIDI channel. This is the default, and can be restored by initializing a Performance.   You can have 16 Parts in a Performance.

There are two ways to view the Parts. Go into Performance/EDIT. The first is the MLT view (F-Key #3). In this mode you see columns of parameters. Each column represents a Part, and all the parameters are listed beneath. You can navigate the rather long list more quickly using F-Keys #4 through #8. The first item in the list is the Bank, then the Voice, then it’s Volume, etc. The Voice type parameters in here are offsets, and don’t actually change the Voice as it’s stored in memory.

The other view is the Part view (F-Key #2).   This view shows you more information at once about a single part. Sometimes this is preferable when tweaking a part within a mix, since you can see more parameters at one time.

In both these views, you can navigate the selected parts using the Program buttons 1-16 on the right of the front panel. Remember that these views are only “views” and have no functional or mode change associated with them.

The first F-Key contains the Common parameters, parameters that are applied to the entire Performance, such as the Arpeggiator setup, Effects setup, etc.



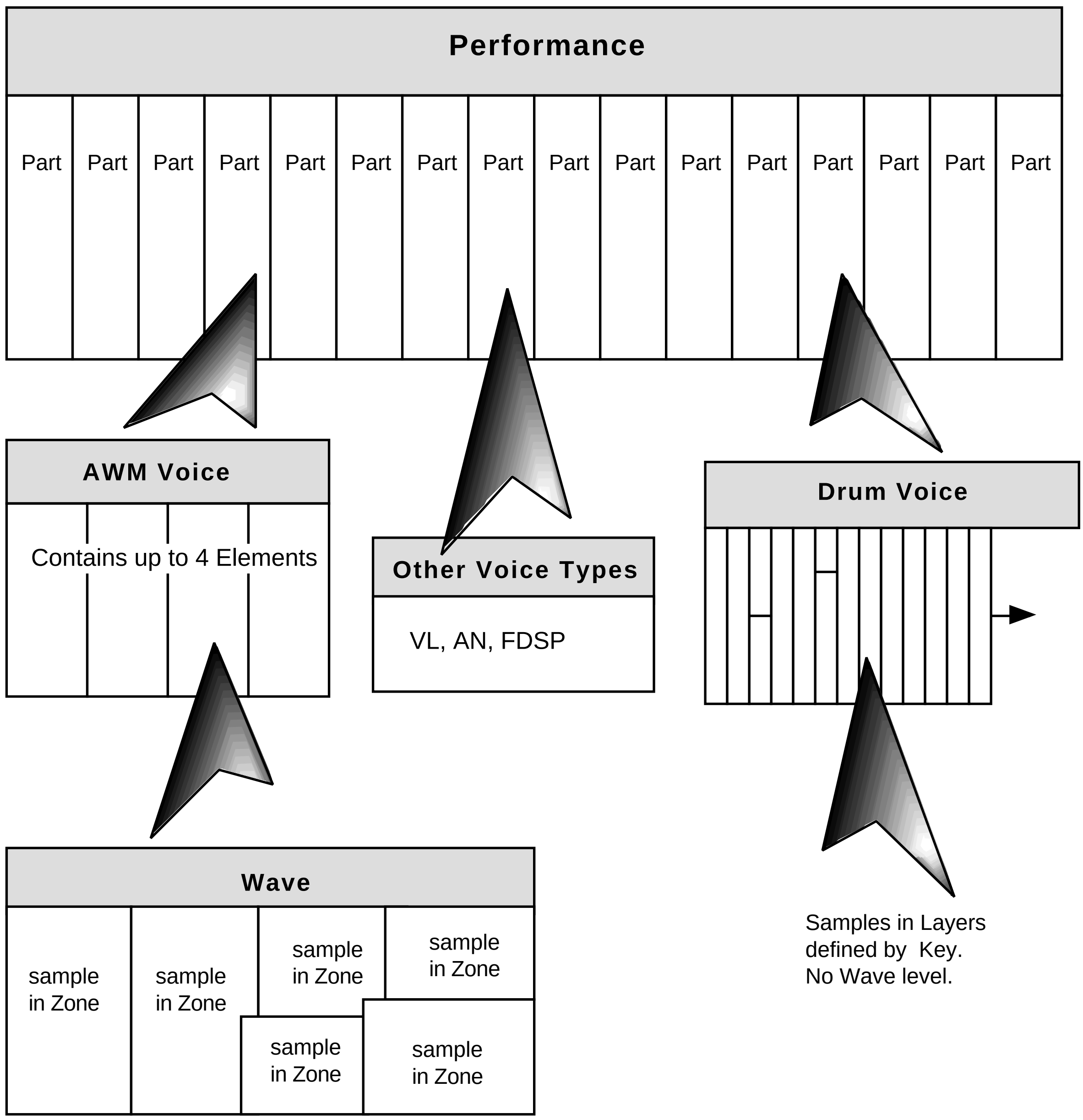
Effects

There are two kinds of effects in the EX synthesiser, global effects and insert effects.

The global effects are called Rev and Cho, and these are the send/return type. Each element or voice can be sent in different amounts to these, but at any moment there are only the same two Rev and Cho in the machine. Rev and Cho are stored with a voice or performance.

The insert effects are called INS1 and INS2, and these can be plugged into a voice and will be saved with this voice. There will be a limit to how many of these you can turn on before you fill the DSP Resource (see the section in the Performance chapter on DSP Resource Full).   The Insert effects are perhaps more interesting than the global effects - there are far more variations available within them.

AWM Structure in EX



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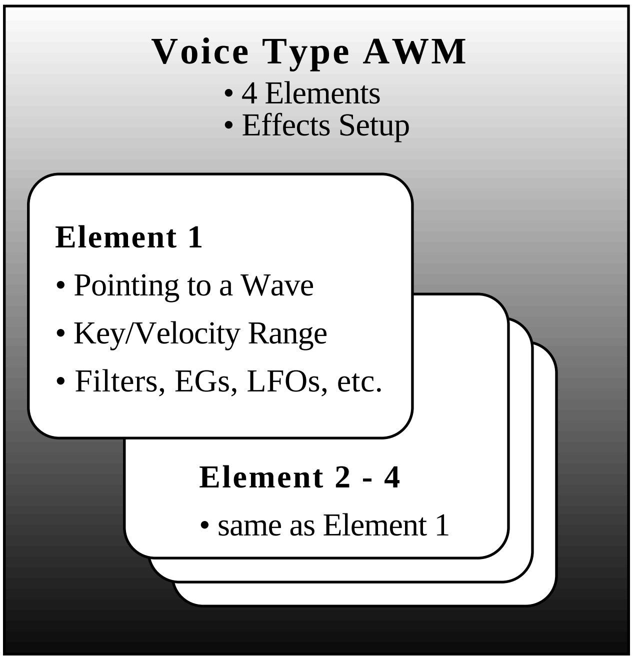
Voices

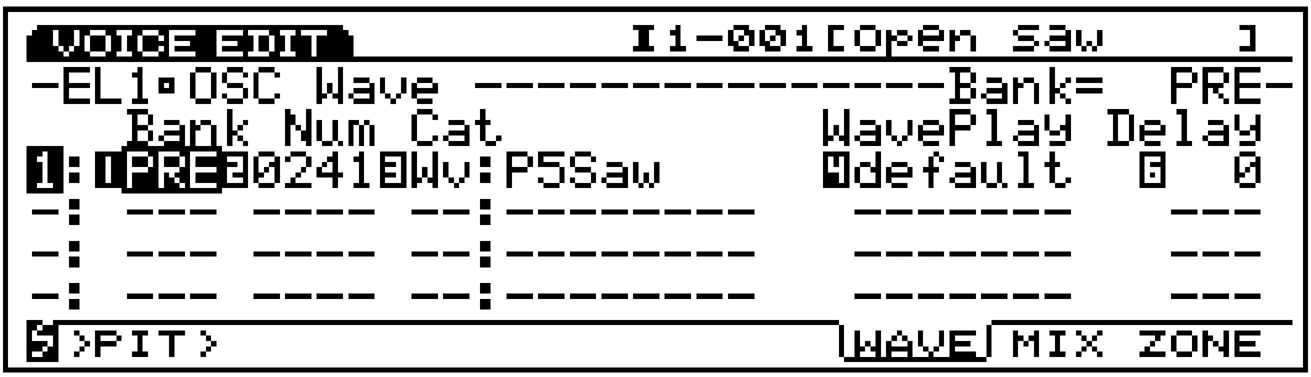
For the following section it is recommended that you Load Synth All from the file on the floppy disk.

AWM Voice



AWM is Yamaha’s word for PCM synthesis. Basically, this is the playback of sampled waveforms, stored inside the synthesiser, and manipulated with envelopes, filters, LFOs and so on. This technique is used in the majority of contemporary synthesisers.

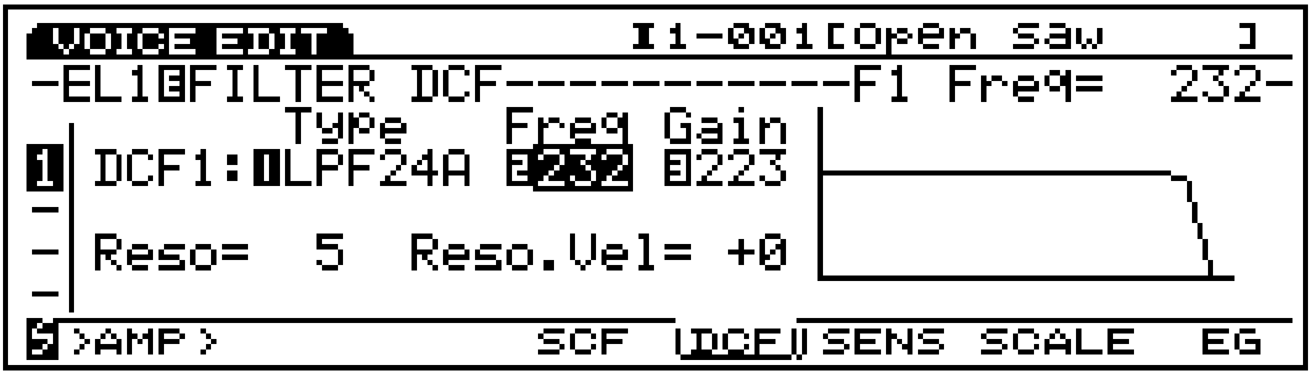
If you’ve loaded the floppy that came with this book, choose Voice mode, and Voice I1:001: Open Saw. This is basically an open sawtooth wave in a single element. If you look at the page EDIT/OSC/WAVE you will see this:



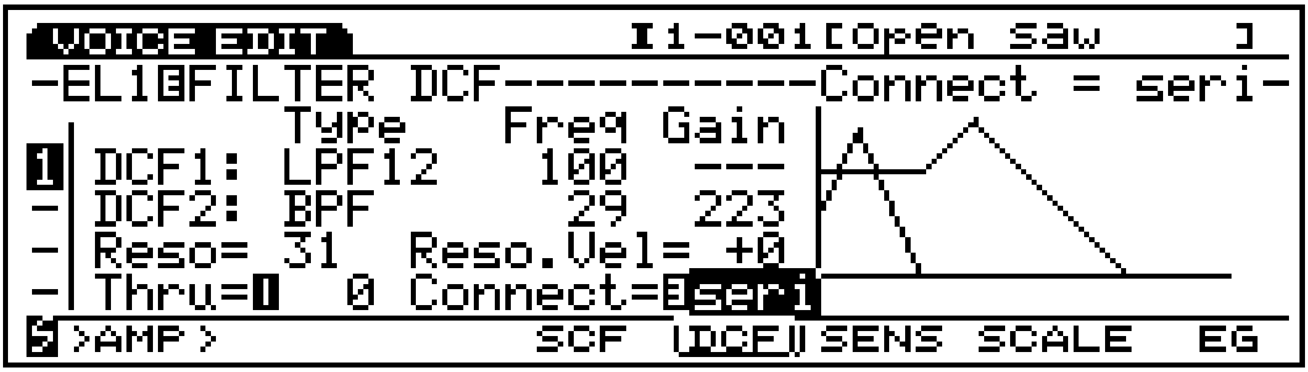
This is the oscillator page. The PRE means that this wave is from the Preset bank (stored in ROM, you cannot erase these). You can choose RAM or FLS here also, which will then look at any samples you have currently in your RAM or Flash memory (if you have some installed). The next number is the wave number, and in the PRE bank you have a choice of 419 different waveforms. Since there are so many, the Cat column (category) can be used to quickly jump to certain types. For example, the waveform, or synthesiser types of waves are categorised by Wv, for Wave. If you change this to Fx, you will jump to the first wave in the Effect category of waves.

The MIX page allows you to define the fine tuning, level, and panning of this element. The ZONE page allows you to define the high and low key of the element, or the high and low velocity of the element. This can be useful for making keyboard or velocity splits.

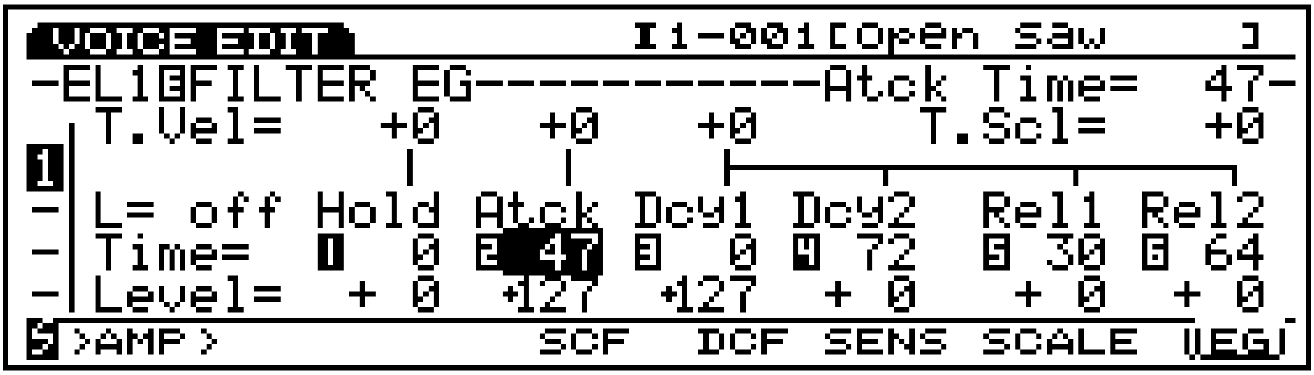
EXIT this page, and let's jump to the Filter (FILT) page. You will see that there are five sub menus in the Filter page called : SCF, DCF, SENS, SCALE, and EG. Go to DCF first.



DCF is the Dynamic filter. This is your typical synthesiser filter, with different modes and resonance, etc. You can hear it if you lower the Freq (Cutoff Frequency). But let's change the filter Type. You can see that there are first of all several different Low Pass Filters of different slopes. As soon as you reach the 12dB slope ANOTHER filter appears. Cool! Now you've got two. You can change the type of number two as well. And create weird stuff by combining a Band Pass with a Low Pass, etc. You can even change whether the two filters are placed in parallel or in series.

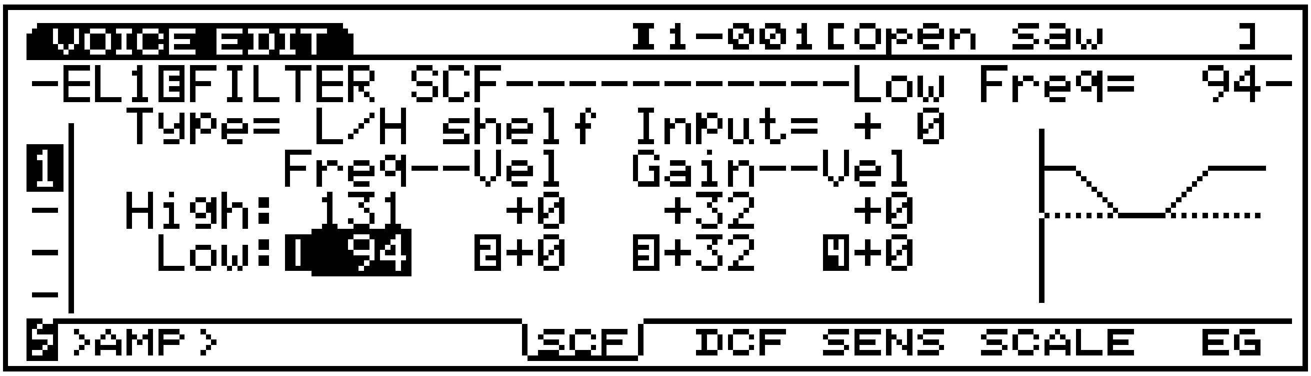


Okay, so leave this is some contorted filter space, and let's jump to the EG page. In the EX synth, EG always means Envelope Generator. This will be the Filter Envelope. Take level number 4 down to zero and start increasing time number 4. (Position 4 is the sustain portion of the envelope, this is often the best place to begin designing your EG.) Now take time 2 and increase it slightly until you have an audible attack phase. Now hit the EG switch again, your screen should change from the EG graphic display to this:



As well as putting names on the time and level points, this has now opened up several real-time control possibilities. You can use velocity to adjust the attack time, for example. More on these when we get to the Amplitude section of the editor.

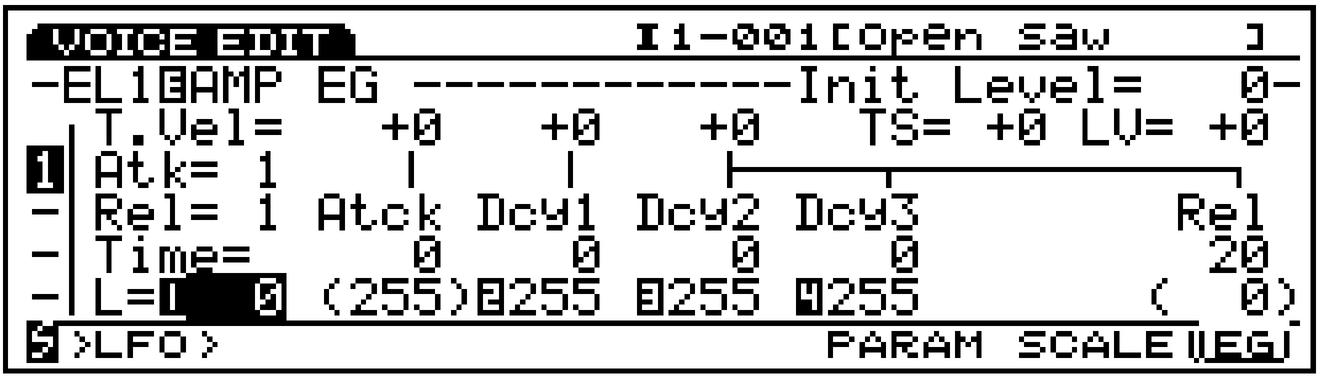
For now let's go to the SCF page. This is the "static" filter. Here you can do several EQ type filters, or even just add some extra gain. Perhaps one of the most useful of these is number 1, the Low/High Shelf.



Notice that the filter isn't completely "static", you can adjust the cutoff or the gain parameters with MIDI velocity.

The remaining two filter pages are the SENS page, in which you can adjust the DCF's sensitivity to the envelope, velocity, and key scaling. The Key Scaling curve is defined in the SCALE page.

Let's have a look at the AMP section. Enter the EG section here first.

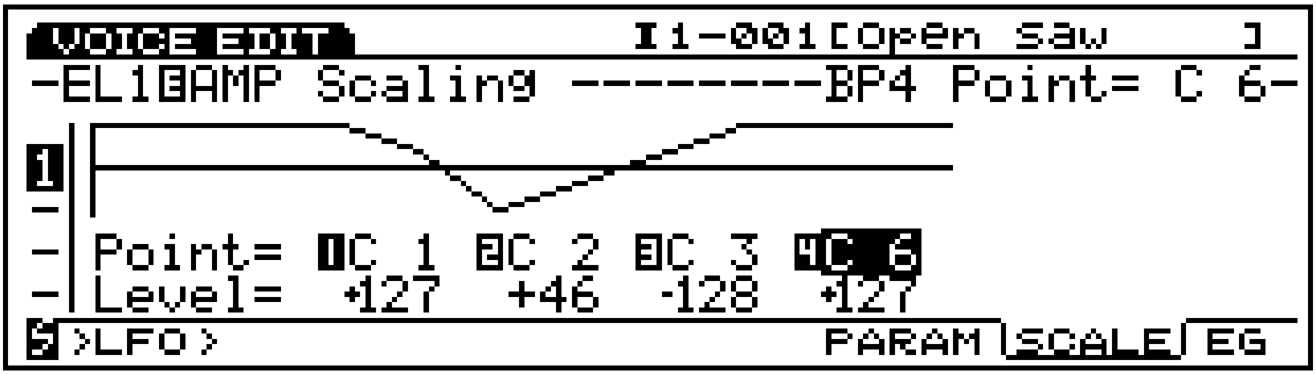


Again, the EG f-key toggles the display between the graphic view of the envelope or the extra parameters. You have several ways of manipulating the envelope in real time. From left to right, the top row allows you to:

• use velocity to adjust the Attack rate
• use velocity to adjust the first Decay rate
• use velocity to adjust the other rates together
• scale the speed of the entire envelope by MIDI note (rate scaling)
• use velocity to adjust the Decay Level

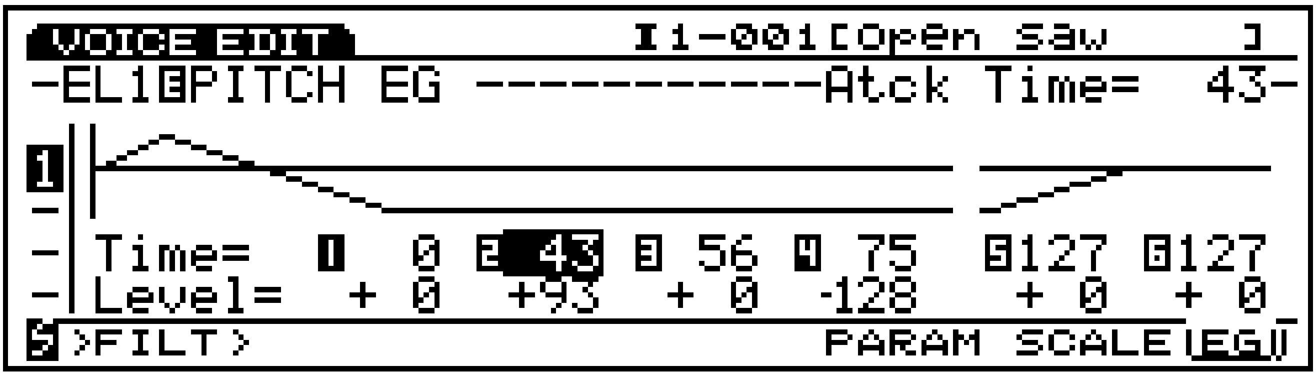
Hit the PARAM f-key. This includes some more basic Amplitude related parameters. Here you can adjust the overall Level, the Key Scaling of the level (KeyFlw), the basic Velocity Sensitivity, the Panning, and the ExpLowL. The what? What's ExpLowL? This is the "Expression Lower Limit". If you use the "Expression" controller for volume control (cc 11, not cc 7), then this lets you set a bottom limit on how quiet the element will get. Hmmm. Useful, eh?

The SCALE page lets you scale the Level of the Element in a more precise way than you could do with the KeyFlw parameter. For example, using the SCALE page you could scale the Element like this:

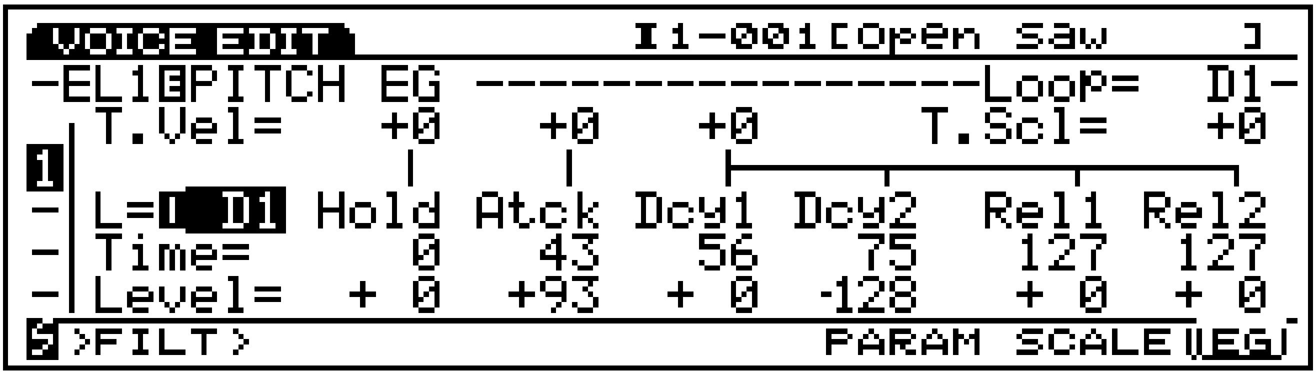


With this kind of Amp Scaling, the Element will disappear in the middle of the keyboard, but be at full level at the lower and upper ends.

Let's now check the PITCH section of the Editor. In the Pitch EG you have many similar parameters to the other EGs. Let's try making a wacky pitch envelope, like this:

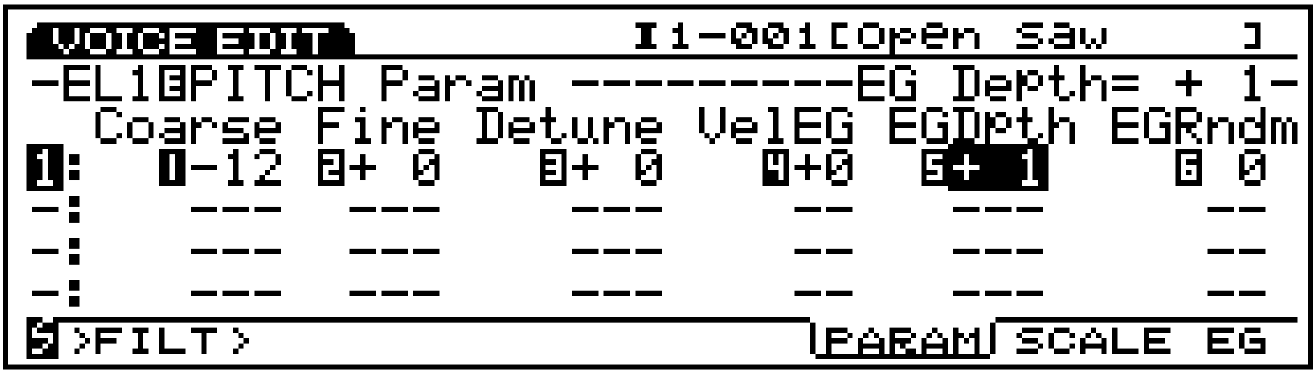


Hit the EG button again to get the extra parameters. Notice to the left there's a little parameter called L. It's currently set to OFF. Set this to "D1". Listen.



“L” means loop. You can loop the envelope at three different points: the first point “Hold”, the second point “Attack”, or the third point “Decay 1”. Whenever the envelope hits the Decay 2 point it loops around to the point you’ve set in the “L” parameter. The Filter Envelope can loop also, by the way.

This kind of pitch sweep is rather special-effect-ish. However, if we jump over to the PARAM page we can reduce the depth of this envelope, so that it’s more subtle. Bring it down to 1. Now it just sounds like it’s slightly broken. Mmmm, nice. Try playing “Amazing Grace” with it.



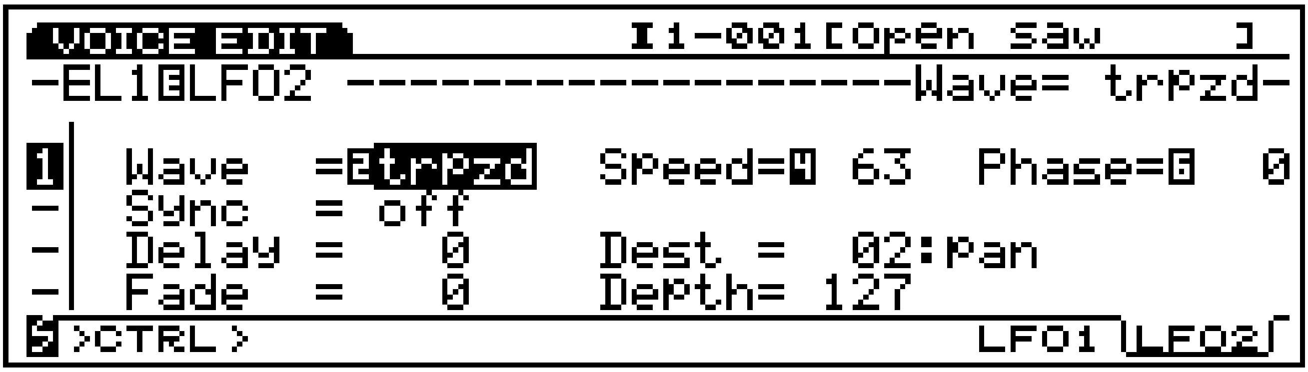
If it still sounds too deep, even on 1, then you need to reduce the levels in the EG itself.

The last page in the Pitch section is again called SCALE, and of course, let’s you scale the pitch.

There are a couple different ways to think about scaling pitch. One of them is Microtuning, in which the actual pitches of the scale are altered to something other than our standard milky white “Equal Tempered” tuning. The other way is to stretch or compress the tuning of the entire instrument. This is called Key Follow in the EX. When it’s set to 100% an octave played on the keyboard sounds an octave in pitch (unless you’ve got a wacky pitch EG, but anyway). If you set it to 50% you have to play two octaves on the keyboard to hear a one octave change in pitch. The “Centre” is the key around which this slope changes. What’s it good for? Excellent for percussion sounds, and can be very cool used with the Ring modulator or self FM in the FDSP voice algorithm.

Let’s jump to the LFO page now. Exit the page your in, and hit LFO.

You’ll see immediately that there are two LFOs. The first one has fixed destinations, but can be routed to the pitch, amplitude, and filter cutoff in varying amounts. LFO2, on the other hand, can only be routed to one destination, but this destination can include aspects of LFO1. You can make some weird and wonderful modulations by adjusting the speed of LFO1 with LFO2. LFO2 also has some unusual wave types, such as trapezoid, and sample and hold.



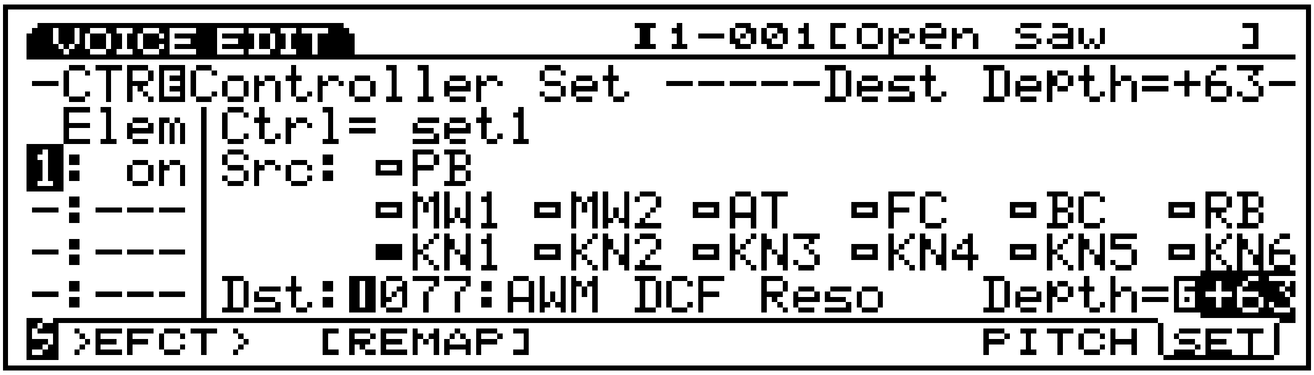
Probably by now, your "Open Saw" is starting to get a little less than useful. You might want to consider adding some real-time control to the voice, so that you can adjust the filter cutoffs, resonance, LFO speeds, etc. Let's exit the LFO page, and drop back down into CTRL (Control).

There are two sections to the Control page, the first is called Pitch.

In pitch you can set the pitch bend range, and the upper and lower ranges can be set seperately. The lower range can be set to a maximum of 4 octaves (48 semitones). This page also lets you set the portamento time, and mode. The two modes are "fingered" and "full time". If you set it to "fingered" portamento only happens when notes overlap. In "full time" portamento is always active.

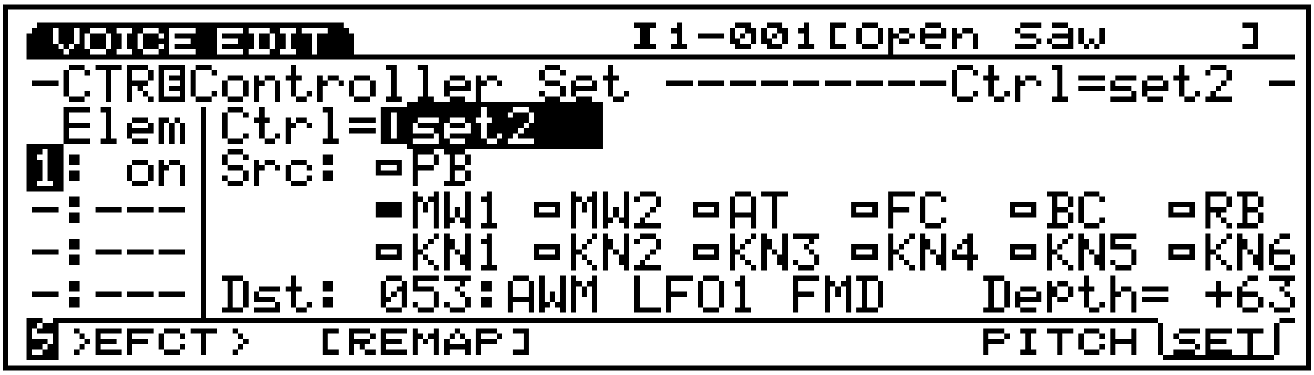
Now, let's check the second of the CTRL pages, called SET.

SET is basically a controller matrix. You have 16 sets, which is a little like having 16 patch cords. You can assign 16 controller+destination combinations. Okay? You can turn each of these sets on or off for each element (to the left of the screen), although at the moment we're working with a single element voice. So let's turn set 1 ON for Element 1, and the turn Knob 1 on, and then down at the bottom set the Destination (DST) to 077: AWM DCF Reso. The DEPTH of this control can be set to the right, let's set it to maximum for now.



Of course, when you're in EDIT mode, the knobs are functioning as data entry for the edits you make, so if you want to check the results of this assignment without leaving the editor, hit the hardware switch called KNOB MODE to test the results of Knob 1. Hit it again to turn it off.

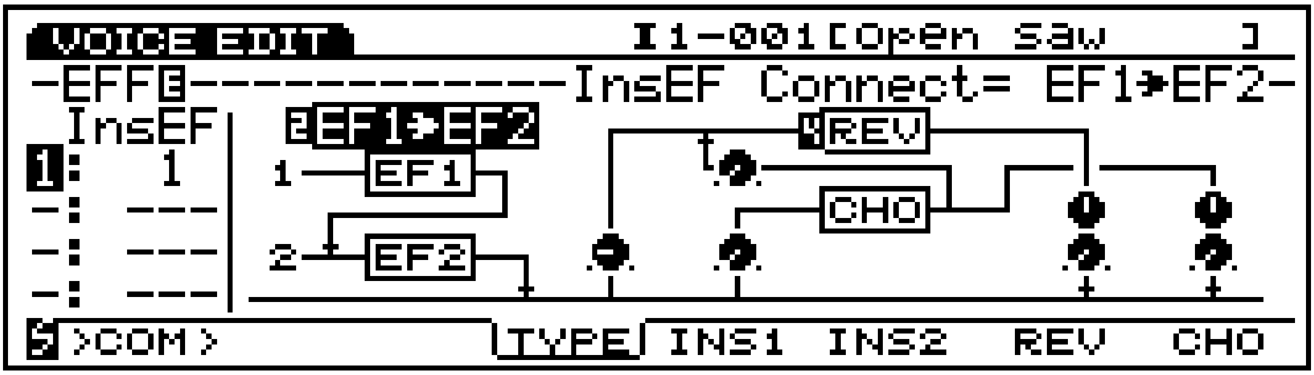
Now let's say we want to use the Mod Wheel to control the amount LFO1 modulates the Filter Cutoff. Go up to the top of the screen and increment set1 to set2. Turn it ON for the element. Then turn on MW1. Set the Destination to 053: AWM LFO1 FMD (filter modulation depth), and set the depth again to maximum, like this:



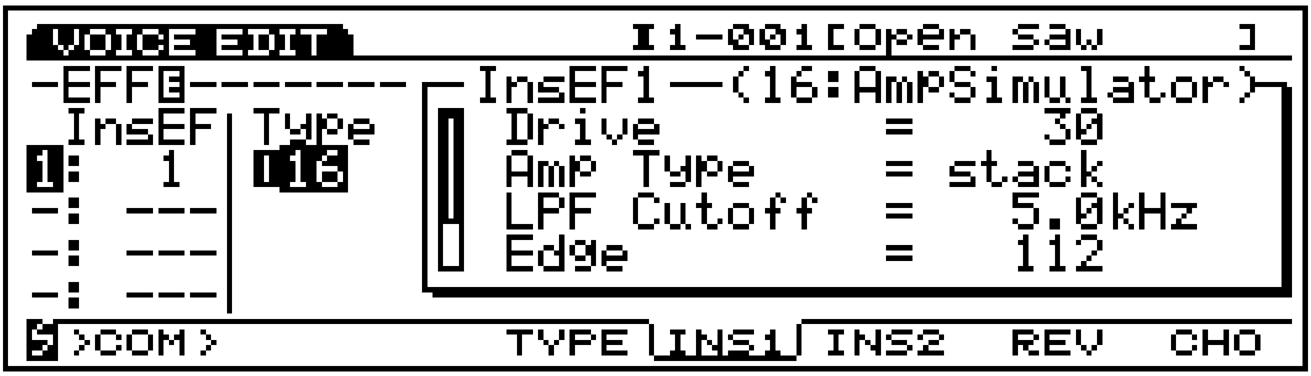
Got it?

Okay, let's have a look at the effects.

As we mentioned earlier, there are two types of effects: insert effects and send effects. The insert effects can be configured in three ways; parallel, or serial (with either INS1 or INS2 first in line). You can set this on the front page of the EFCT section. This page also lets you set up the send and return levels for each of the send effects (REV and CHO) :



Of course, we're not hearing this yet because the effects are currently off. Hit the INS 1 key to enter the first insert effect. Set the effect type to 16: Amp Simulator. Again, you choose which element is going into which insert effect on the left, where each element can be OFF, 1, or 2. Like this:



You can cursor through the parameters within the effect.

Now try INS2. There are 79 different effects in here. Try 64 the "Talking Mod", go back out to the controller set, and assign set3 so that something modulates Dest:020: EF2 Vowel... mad stuff.

If your voice sounds great, store it somewhere using the STORE button.

Let’s move on to another voice type.

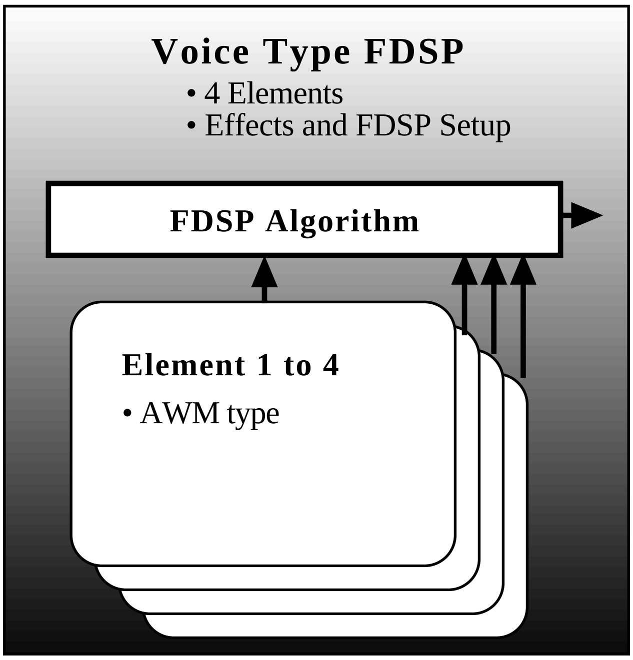
RECIPE ONE

• Getting more gain •

If you need to get some extra gain onto an element, use the SCF (static filter). The last three “types” are just gain boost.

Also check Common Volume, Oscillator Mix Level, Amplitude Parameter Level, and the Filter DCF Gain.

FDSP Voice

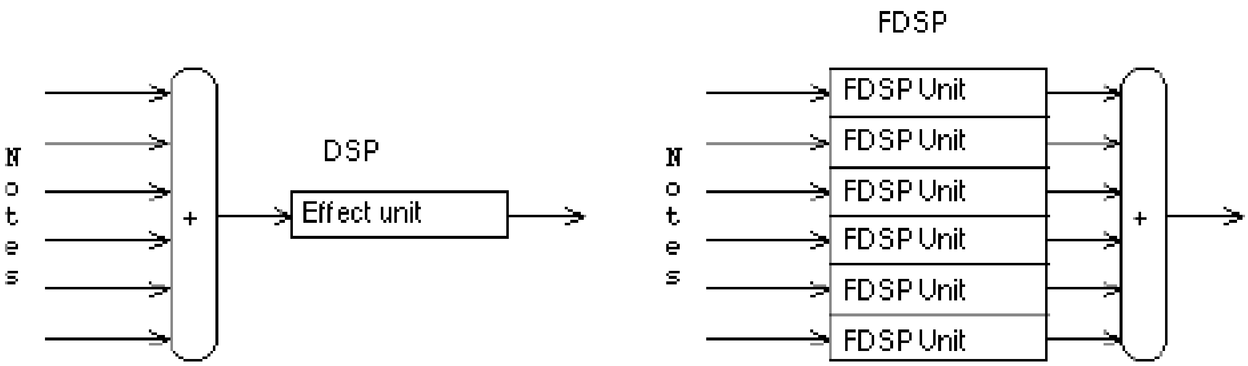


FDSP is a kind of process. As you see in the diagram above, it needs an input. In an FDSP voice, one or more of the elements must be routed through it in order to hear any effect of the FDSP. Generally, we route AWM elements through the FDSP. In the EX5 and EX5R and AN element can also be routed through the FDSP.

There are ten FDSP algorithms. Let's have a quick look at what FDSP is, and what each of the algorithms do.

FDSP is very similar to an effect processor, in that it takes an audio source and processes it. The main difference between FDSP and a traditional Effect is that the FDSP is polyphonic. Each note going through it is processed individually, thus allowing pickup simulators and so on to be modelled more accurately, and keyscaling and velocity to be used within the algorithms. Due to the limitations of the DSP used in the EX, an FDSP voice has a polyphony of 8.

Look at this diagram:



Here is a brief explanation of the ten FDSP Algorithms in your EX:

1. EP Pickup — This is a model of the electro-magnetic pickup used in electric pianos.

2. EG Pickup — This is a model of the electro-magnetic pickup used in electric guitars.

3. Water — This is an original effect in which sample&hold filter modulation is passed through a resonating string algorithm. Sounds kind of watery.

4. PWM — This realizes a Pulse Width Modulation sound on any input wave.

5. Flange — A Flanging effect with envelope control for each note individually.

6. Phaser — A Phaser effect with envelope control for each note individually.

7. Self FM — The input is frequency modulated by itself.

8. Tornado — This is another kind of FM synthesis in which the signal is modulating a frequency of 0 frequency (DC). By varying the modulation intensity and feedback, FM type sounds or analog sync sounds can be created.

9. RindMod — Ring Modulation.
This modulates the input signal’s amplitude with an oscillator and creates mad sidebands.
(in fact there are two oscillators: main and sub)

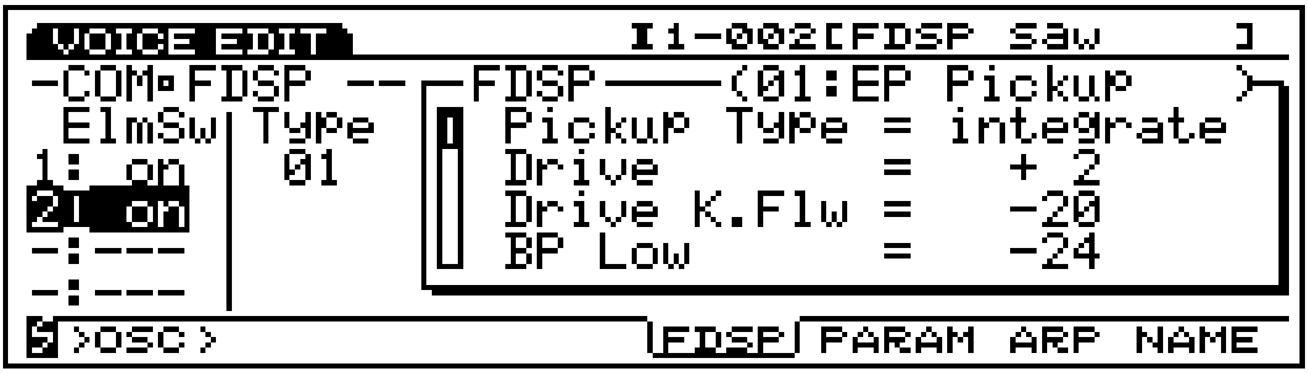
10. Seismic — This is an overdriving, time-variant low-boost filter. Good for distortions and for fattening sounds. The attack transients can be clipped resulting in emphasized or compressor-like sounds.

So let’s check an example.

Choose voice 002 in the I1 bank. It’s called “FDSP Saw”.

This voice is using the first FDSP Algorithm called “EP Pickup”, meaning “electric piano pickup”. Hit EDIT/COM/FDSP. As with other parameters in the EX, each element can be routed via the left hand of the screen. This voice uses two elements, and both of them are switched ON for the FDSP algorithm. If you turn them off, you can hear that the voice is composed of two, detuned, synth waves.

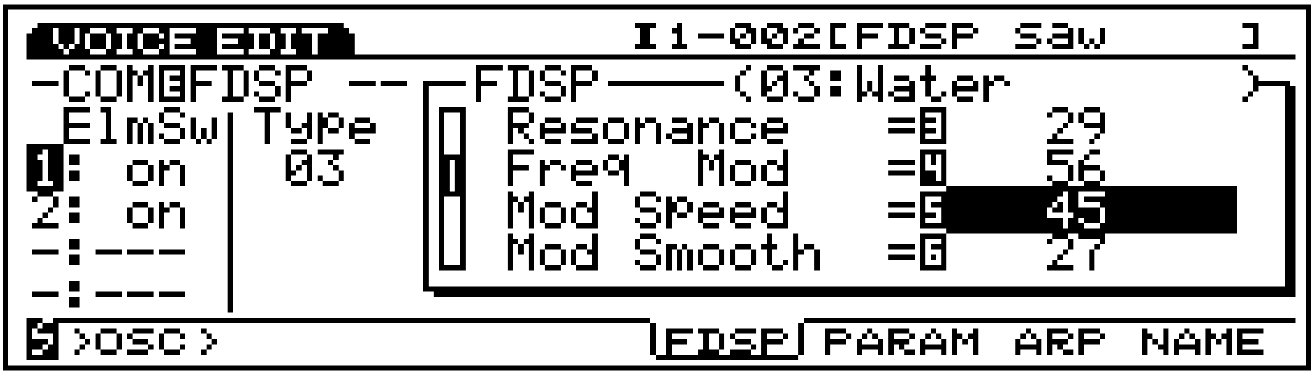
Turn them back on again and start looking through the FDSP parameters.



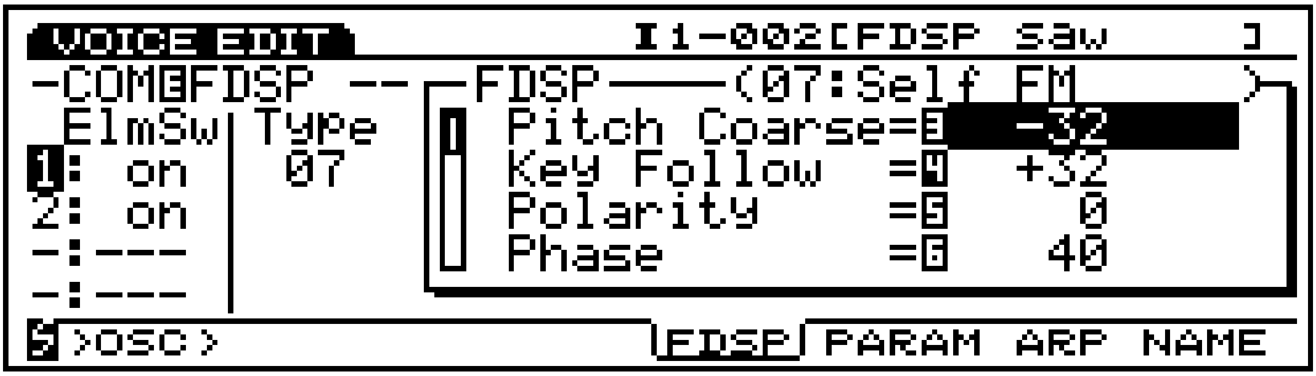
The three types result in progressively more aggressive settings. The middle “integrate” type, is most useful for actual pickup simulations. Try adjusting the drive parameter to see what happens. This is like moving the pickup closer to the tine of the electric piano. You’ll notice that the higher the drive, the greater the “dynamic range” of the pickup, meaning that the changes in timbre from low velocity to high velocity are exaggerated.

Let’s try some other algorithms. Choose number 3, “Water”. Scroll to the last of the Water parameters and set the Dry Level to zero, so you’re hearing only the effect. Now go back to the top. The “pitch” parameters refer to pitch of the virtual string or resonator. Key follow refers to how this pitch changes across the keyboard. (Note: With all FDSP key follow parameters, a value of 32 represents an even octave scaling. ie. a filter scaled at 32 will scale such that one octave on the keyboard moves the cutoff frequency one octave. This is useful if you’re using high resonance filters, and want to keep them in “tune”.).

Scroll down further and experiment with Resonance, Freq Mod, and Mod Speed. Resonance is that of the filter being modulated. If you scroll further, the Feedback parameter is the “resonance” amount of the virtual string being triggered by this filter modulation.



Now try algorithm 7, “Self FM”. Again, first scroll to the end and turn off the dry signal. Essentially, what we have here is a single oscillator FM system. The envelope can control the drive of the modulation. This envelope can be either in attack mode or decay mode (it’s very simple). In decay mode, it starts at maximum, and you have control over the time until zero. In attack mode, it starts at zero and you have control over the time until maximum. Set it to Decay with a time of about 70. Now go up to drive EG and set to maximum (63), and finally go to the top and start lowering the pitch coarse.



Getting the idea?

Notice also that you can control some of the FDSP parameters in real time using the knobs or other MIDI controllers. The “FDSP Saw” voice has Knob 1 assigned to the pickup “drive” parameter.

A full explanation of all the algorithms, and each parameter, begins on page 124 of the manual.

Recipe Two

• How to use Scenes •

You've probably noticed, maybe tried, the Scene switches above Modulation Wheel 2.

So what exactly is going on here, how do we control this?

First of all, remember what a Scene is. It's the current state of the six knobs. That's all.

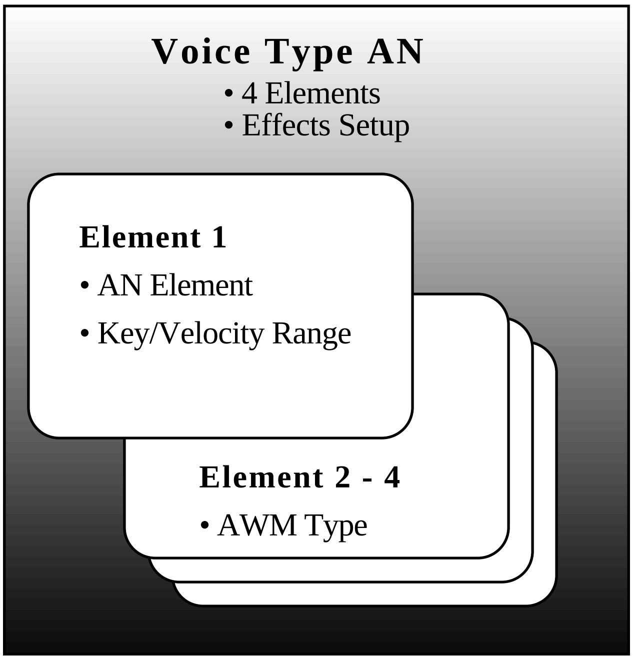
So if I choose a voice, set the knobs such that I like it, and hold the STORE button down while I hit a Scene switch, I store those settings there. Then I can change the knobs to a completely different configuration, and hold STORE while I hit the other Scene switch.

Now I can instantly recall either of these by hitting the appropriate Scene switch. I can also hit both of them together, and MORPH between them using the Mod Wheel 2 or the Foot Controller.

The Scene settings are stored with the Voice.

You can choose whether the Mod Wheel 2 or the Foot Controller is the master Scene Controller in Utility/Control.

AN Voice

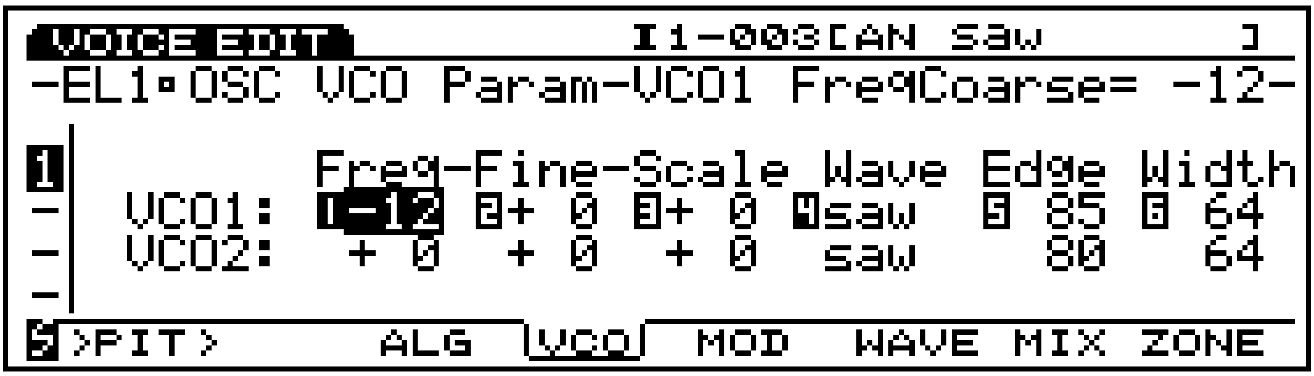


The AN algorithm is an analogue synthesis model, based on the same algorithm used in the Yamaha AN1x synthesiser. Again, the AN algorithm in the EX uses the DSP resource, and is therefore limited by the number of other DSP functions you're trying to do simultaneously.

The EX5 and EX5R version of AN can have a polyphony of two. They can be configured as two "Poly" or two "Layer". The EX7 only has a single polyphony version of AN.

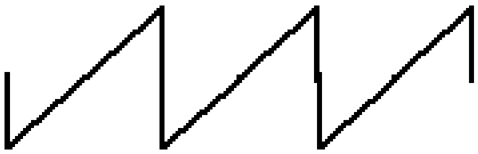
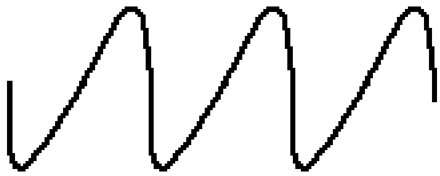
Let's examine an AN voice. The next voice from our floppy is I1:003: AN Saw.

Hit EDIT, and enter the OSC page. Then choose the VCO page.



You can see here that there are two "VCOs" (voltage controlled oscillators), which can be tuned, and scaled independantly (although, this voice is only using VCO1). You can choose different waveforms for each oscillator, and you can adjust the Edge parameter.

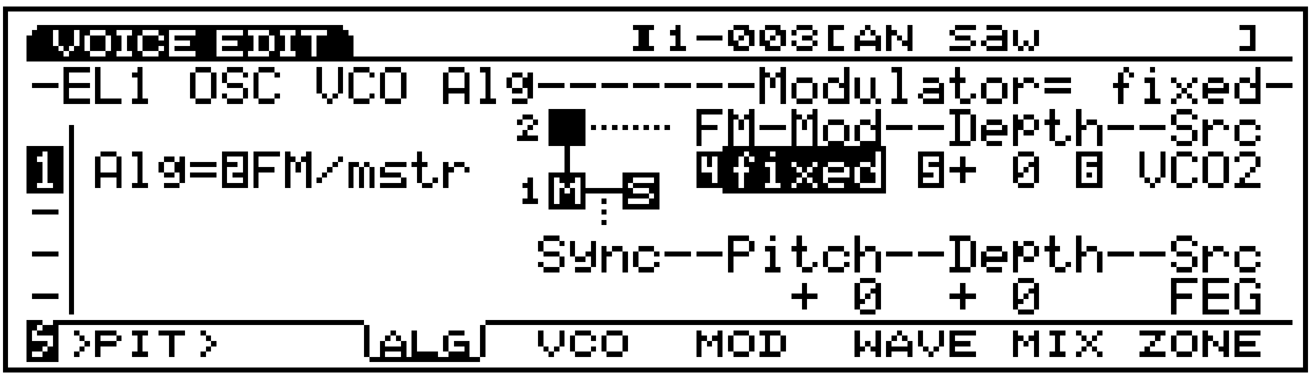
The Edge parameter describes the "sharpness" of the waveform, or the amount of harmonic content. For example, at maximum (127) the edge parameter would make the saw wave a perfect digital saw shape, and as you lower the edge, it becomes more rounded, like this:

If you bring the edge parameter all the way down to zero, the sharpness of the saw descreases so much that the result is almost a sine wave. In our tests, we've found that edge values in the mid-eighties to be most similar to typical Prophet 5 or MiniMoog waveforms.

The last parameter, Width, is the pulse width. Although since this is a "virtual" analogue, we can vary the pulse width on any of the waveforms. Pulse Width modulation is assigned in the MOD page.

The other pages in the OSC section are the MOD page (which is the LFO > VCO patching), the WAV, MIX, and ZONE pages, which let's you layer the AN element with other AWM elements, and the ALG page. The Algorithm is basically to allow the oscillators to be used for FM or Sync functions in different ways. I find the FM/Mstr setting the most useful, but you can find a full explanation of these algorithms in the manual at page 113. Let's have a quick look at what happens in this setting:



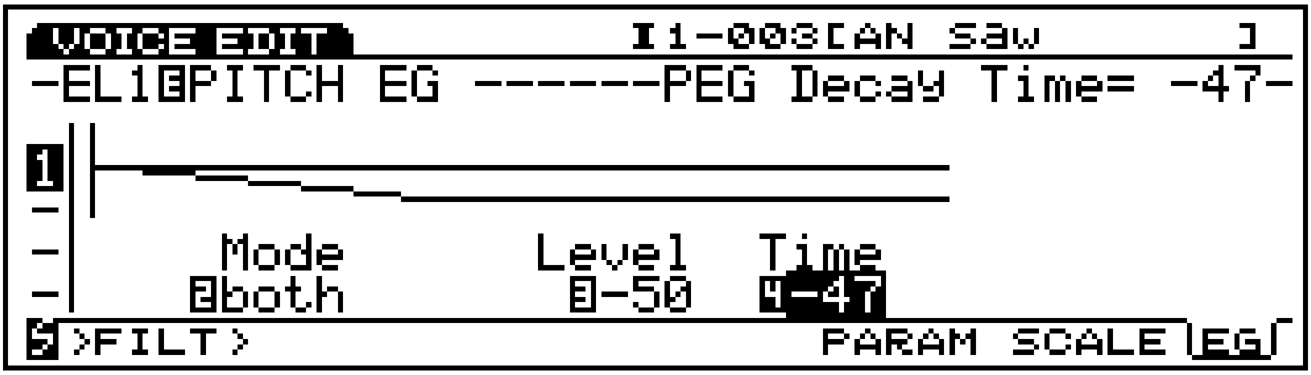
In this mode, the Sync is active and the first oscillator (VCO1) is being frequency modulated by two sources. Usually, once of these will be VCO2 (under Src). Try bringing up the depth on the FM, so you can hear what happens. The timbre get's more complex, and the perceived pitch changes. Now change the Src to VCO1, so that this oscillator is modulating itself. Now, under the FM Mod column, change the "fixed" to FEG (the filter envelope). Now the depth of the FM is being controlled by the filter envelope. Zero the FM depth. Let's try the Sync.

Sync is the result of the one oscillator being slaved to another oscillator's cycle. On analogue machines it was an nice way of creating more complex sounds. In our AN algorithm, the Sync is created within the one oscillator, so you don't have to worry about VCO2. The Slave oscillator is virtual.

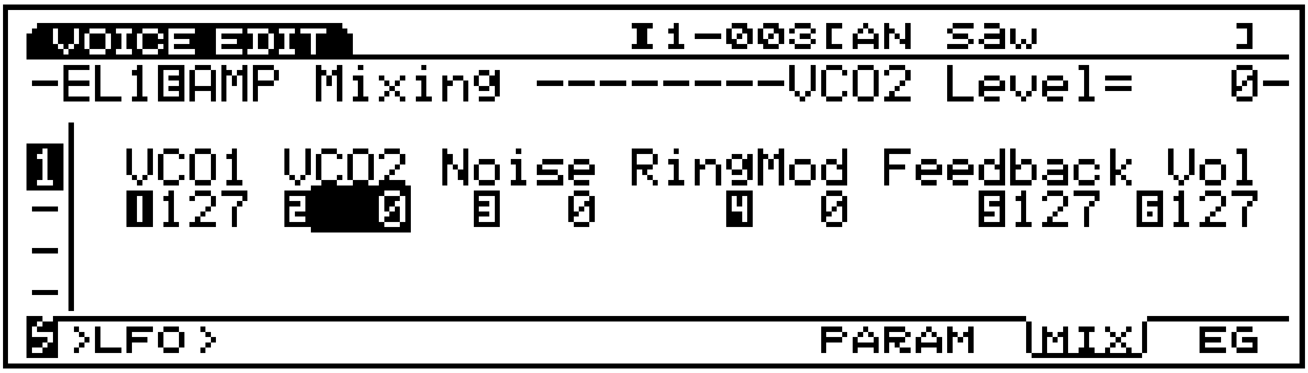
When you change the Sync Pitch, the harmonic structure of the main oscillator changes quite radically, but the percieved pitch stays the same. The depth in the Sync section is again to modulate the Sync oscillator over time, you can use the filter envelope or other sources to do this. Try it out.

The filter section of an AN element is slightly different from that of the AWM element. There are not so many filter types, but there is always a seperate HPF (High Pass Filter) to roll off some of the bottom end if necessary (and makes an interesting controller destination, by the way). Also, the EG Depth control is more similar to an actual analogue machine, allowing you to modulate the cutoff frequency in either a positive or negative direction.

The pitch EG is slightly different also. It's not the full ADSR envelope found in the other element types, but just an attack portion. However it can be positive or negative in depth, and have a positive or negative setting (see page 117 of the manual). Try increasing the time parameter. You'll find that our voice takes more time to come UP to the pitch. Try decreasing it, take it below zero. Now the EG has reversed it's setting so that we begin on the original pitch and slide away from it.



Let's have a look at the AMP page. Page EDIT/AMP/MIX.



Here we find the level for the mysterious VCO2 which we've heard about, but not heard yet. You also have access to a noise generator, a Ring Modulator (between VCO1 and VCO2), and the Feedback control. The Feedback parameter is the strange one, for a couple of reasons. Although it's set at 127, you're not hearing it yet. And this is because it's assigned to a controller (Knob 4). Hit KNOB if you want to try it (see Recipe box at the end of this section). Careful, it's dangerous. What this does is route the output back into the mixer again. It's great for fattening or distorting basses, etc.

One last point about the AN voice. If you check the control settings page (EDIT/CTRL/SET) you'll see that the destinations for cutoff, or resonance, etc. are different from those in an AWM element. For example, the AWM Resonance is destination number 77, but the AN Resonance is destination 111.

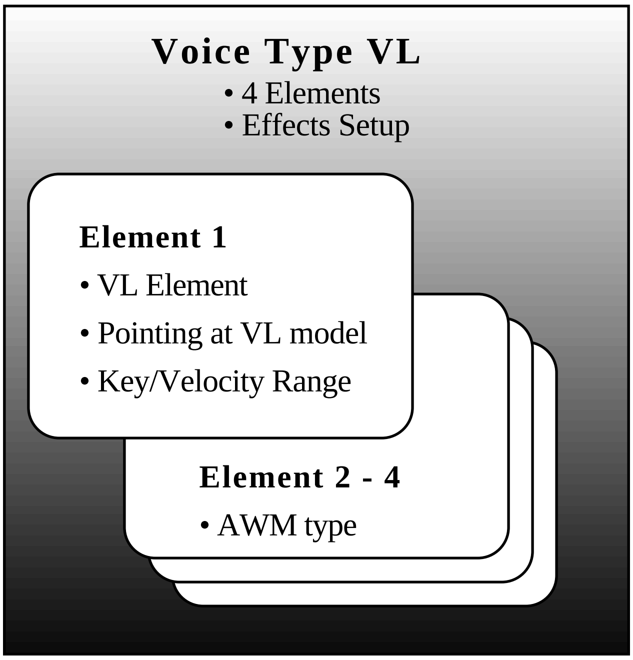
Recipe Three

Some parameters are controlled "upside down"

There are several parameters that behave "upside down" when assigning controllers to them. Most parameters are offset by the controller value (filter cutoff, EG times, etc.), and the controller adds or subtracts from the default value. However, some strange exceptions to this rule exist, and Feedback is one of them. The Feedback default is the MAXIMUM that the knob can take it at full depth. FM Depth also works this way around. It's weird, but true. And we don't know why.

VL Voice

(EX5 and EX5R Only)



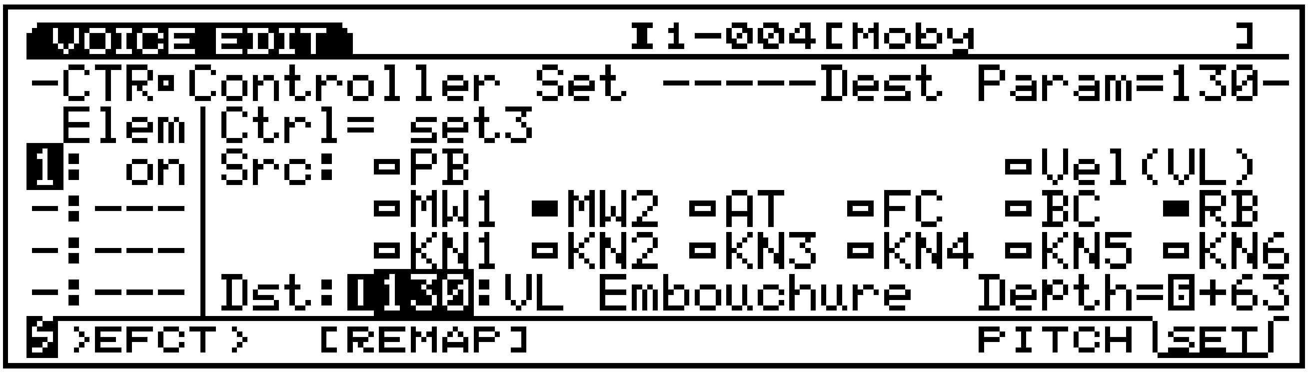
Voice I1:004: Moby is a demonstration of the VL algorithm used in the EX5 and EX5R.

You don't have access to the actual editing of the physical model within the EX, so instead there is a supply of basic models (256 of them, found in the OSC page). You can then offset parameters using the EX editing system. This isn't necessarily a bad thing. Firstly, the VL model is extraordinarily complex, and editing it through a little screen would be a nightmare. Secondly, the interesting side of VL is not the way it sounds, but the way it moves. You have full access to the controller map, and this is what makes VL interesting.

Let's check out Moby.

The concept behind Moby is that it's a mutated brass type of instrument, much bigger than anything you could blow with your skinny little lips. Thus, when you blow into it, you can barely control the embouchure enough to keep it in one mode at a time. Try this, move Mod Wheel 2 all the way back to zero, and play very lightly in the middle of the keyboard. Just hold the note down, and try not to trigger the aftertouch. The sounding pitch should be quite high. It's a ridiculously high harmonic or overtone. Now slowly bring Mod Wheel 2 up, and you'll hear the mode shift down through the overtone series until it finally reaches the fundamental.

If you open the controller page, you'll see a similar layout to that in the other voice types, but again, notice that the destinations are VL specific.



VL Embouchure is probably the coolest of all the VL destinations. This approximates lip tension in a brass or reed instrument. If you don't care about that, then just remember that it's the parameter that can force the VL model into different harmonic modes. In Moby, it's assigned to both the Ribbon, Aftertouch, AND Mod Wheel 2.

Aftertouch is also adding to the VL Pressure destination, which is the amount of energy being forced into the VL model. If you want to get realistic, this is best done with a breath controller, but Moby is pretty far from realistic. The VL Pressure and VL Embouchure are not fixed discrete paths, as you find in most machines, but influence each other in a non-linear, organic kind of way. For example, if the pressure is higher, the embouchure will tend towards more stable modes.

VL is a mysterious creature. Other interesting VL destinations to explore are Tonguing, Scream, Growl, Throat, Damping, and Absorption.

Recipe Four

Program Change in the EX

Since there are five banks of presets, we must use bank select to change patches by MIDI. Some sequencers have bank select tools, which may work. If they don't, then you must enter by hand the following information:

Controller 0, value 63
Controller 32, value XX
Program Change within the bank.

Nothing will happen until all three bytes are received. They should be seperated by one or two MIDI ticks, to ensure the correct order.

The five banks (for value XX) are:

Performance	64
Voice Preset 1	0
Voice Preset 2	1
Voice Internal 1	2
Voice Internal 2	3

For example, if on ANY MIDI channel, you send Bank select 64, then Program Change 1, you will be selecting the First Performance.

Then, if you send Bank Select 1, Program Change 1 on MIDI channel 5, that part of the Performance you just selected will change to Voice Preset 2, number 1 (PW Pad).

So, what about Drum Voice? Eh, matey?

(It's in the Sample Section, Chapter 6, because that's where it belongs.)

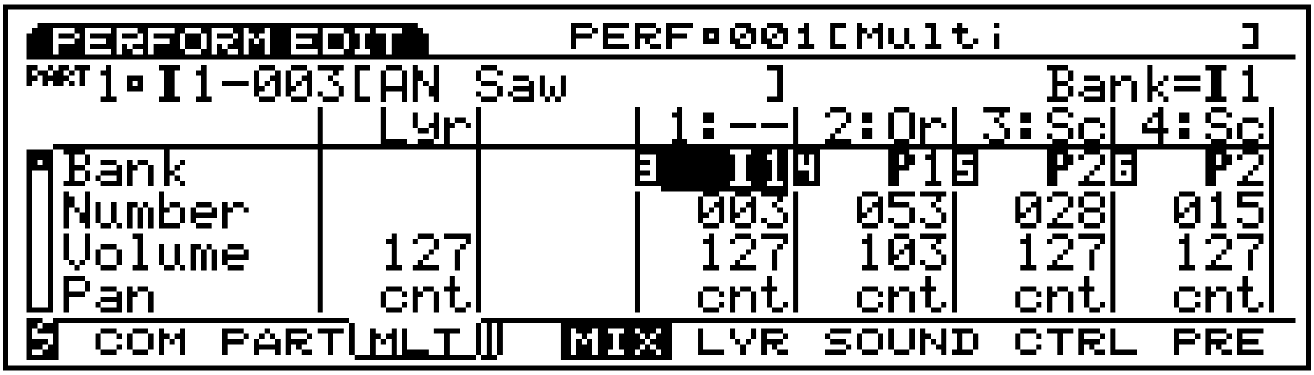
5

Performances

Performances can be used for several different things on the EX. You can make a Multi setup, in which you want to play several different sounds on different MIDI channels. You can make Layer and Split type Performances to combine voices in certain ways. And you can use the Performances as MIDI Master Control setups, since each Performance can have an entirely unique MIDI setup.

Performance as Multi

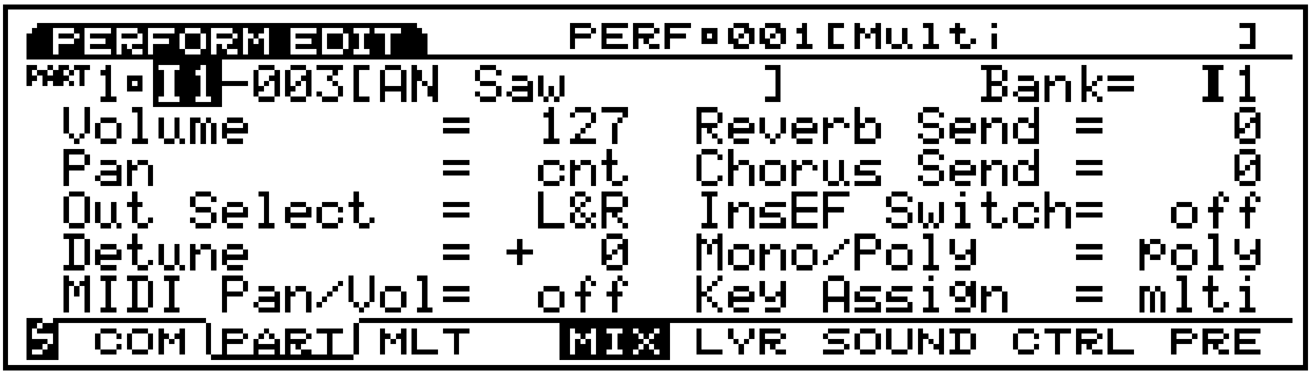
Choose the first Performance on your EX. If you loaded the floppy disk file, this should be Perf 001: Multi. Hit EDIT/MLT/MIX. This is the “multi” view.



In this view, each channel or part is viewed as a vertical strip. At the top is the Bank and Voice number, then it’s Volume, etc. As you move down the list you see all the available parameters for that Part. None of these parameters actually change the voice data itself, they are just offsets to that data.

You can navigate to various sections of this list using the MIX, LYR, SOUND, CTRL, and PRE f-keys. Using the arrow keys, or the Program keys, you can jump through the sixteen parts laid out in the sixteen columns.

Now hit the PART f-key to the left of MLT. In this view, you see many more parameters at once, but for one Part at a time. This is often more convenient when tweaking a sound within your multi setup. You can still navigate parts using the Program keys, and you can still navigate to various sections using the MIX, LYR, SOUND, CTRL, and PRE f-keys.



Let's have a quick look at these various sections. The above page (MIX) is basic voice selection, level, pan, output, etc. as well as the effects sends. There are only two global effects for the entire multi, so you have to choose these carefully if you're trying to reproduce a Voice as it sounds in Voice Mode.

Recipe Five

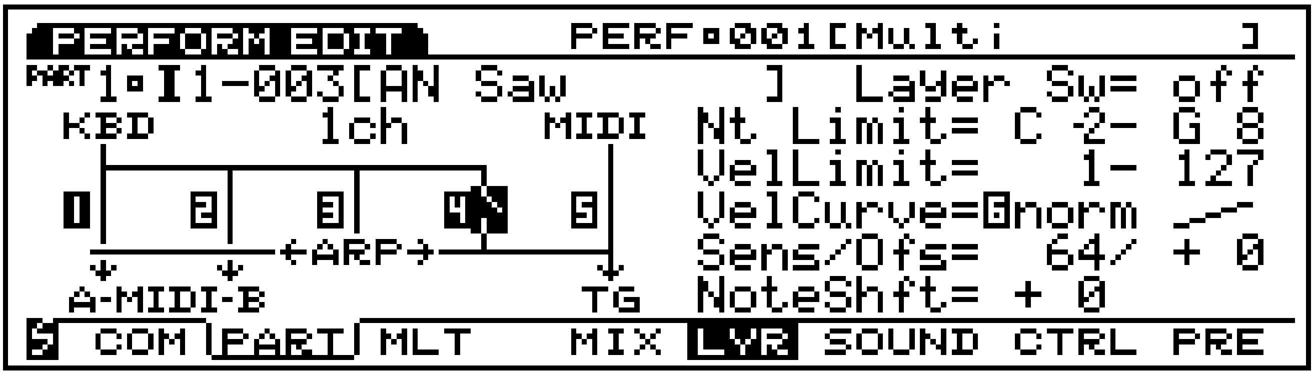
Copy Effect

When you use a Voice in a Performance, it often doesn't sound quite the same as it did in Voice Mode. This is because the effects may be different. If you want to make sure that the Voice sounds the same as it did in Voice Mode, you must turn on the insert effects and duplicate the send effects (Rev and Cho).

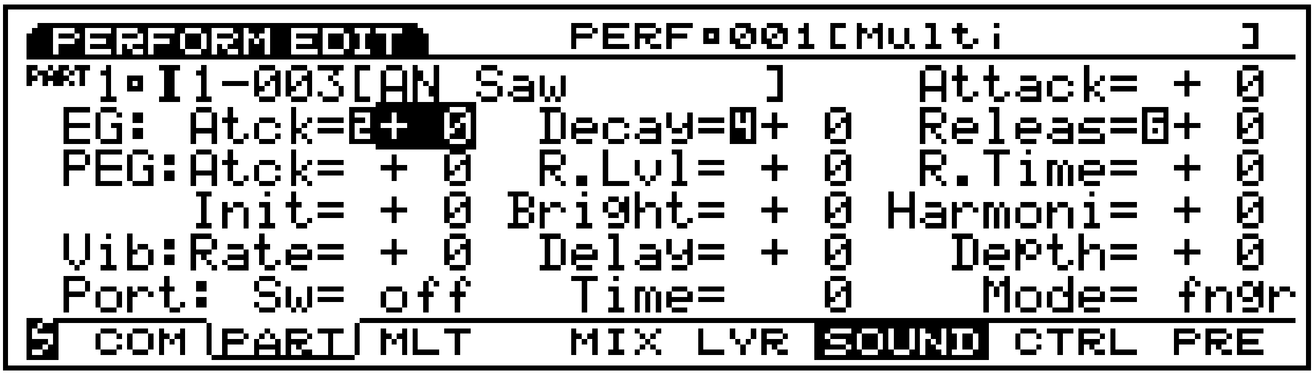
The easiest way to copy the send effects is to enter the Effect Editor (EDIT/COM/EFCT/). Choose Rev or Cho and hit COPY (on f-key 5). This will give you the option to copy the effect of the currently selected part into the Performance Effect.

The Insert Effect can be turned on or off for each part, but of course, due to the limited DSP resource, you can't turn on very many at once. The Insert Effect will be that which was stored in the Voice itself.

The next page, LYR (Layer) include the Layer Switch (more on that in moment), and other Layer type parameters, key range, velocity range, note shift, and several switches for the MIDI transmit and Arpeggio settings.



The SOUND page is a collection of offsets to the Voice synthesis parameters themselves, such as filter and envelope settings.



The CTRL (Control) page lets you adjust the pitch bend range, and turn the controllers (knobs, etc.) on or off for each Voice. You cannot re-assign the controllers from within the multi.

The PRE (Preset) page lets you setup whether or not the controllers and knobs are transmitted over MIDI. And also their default value which is sent when the Performance is selected. This can be very useful for Master Keyboard or Live Applications in which the EX is used to control other synthesisers.

The COM (Common) page lets you set up parameters for the entire performance, such as the arpeggiator and controller numbers and effects settings.

DSP Resource Full ! - DAMMIT!

Yes, we know this message will drive you crazy. But try to be cool and lets sort out what's going on and how to cope with this.

First of all, remember all the things that use up our DSP resource:

• VL Algorithms *(EX5 and EX5R only)*
• AN Algorithms
• FDSP Algorithms
• Insert Effects

Any time any of these are in use, some amount of DSP is being eaten up.

In the EX5 you may have 4 insert effects, or else 1 insert effect plus 1 synthesis algorithm. If you decide to turn on the insert effect on the Tube Crunch organ (channel 2), this will be fine. But if you now want to turn on the insert effect on anything else, you'll get the dreaded message. However, if you decide you really don't need that AN voice after all, change that voice to a non-DSP voice, and then you can turn on 4 different insert effects.

In the EX7 you may have 1 insert effect OR 1 synthesis algorithm. On the Tube Crunch organ (channel 2 of our multi) you cannot turn on the insert effect at all until you select a non-DSP voice on channel 1.

In all the EX synths, if you try to turn ON an insert effect and the DSP resource is full, you'll get the message and the insert effect will stay OFF. If you try to select a voice that requires DSP and the DSP resource is full, the following non-DSP voice will be selected instead.

There's only so much power, and you must decide how to best distribute it.

Layering within a Multi

There are two ways to layer sounds within a multi setup. You can turn the Layer Switch ON for any two voices, and this will link them so that either of their MIDI channels will trigger the layer. The limitation of this technique is that you can only layer two voices like this. The strength of this method is that it will work in a live situation, without a MIDI sequencer.

If you're using a sequencer to re-route your MIDI data, then the other way to layer sounds is to set them to the same MIDI channel. You can theoretically set all 16 parts to the same MIDI channel if necessary. The disadvantage of this method is that you cannot hear the result directly from the keyboard, since the LOCAL ON routing is by Part, not by MIDI channel.

Recipe Six

Editing a Voice in Multi Mode

Often it's convenient to be able to edit the Voice while your song is playing, so that you hear it "in context", and can adjust the parameters to suit the song. It's easy enough with the EX.

Just hit the Performance and Voice keys together (so that both light up). Choose which Part you want to Edit the voice within, using the first two f-keys. Then hit EDIT. You are now in a normal Voice Edit mode, but the multi is still active, so you can Edit the voice within the track you're working on.

The only difference between this Edit mode and normal Voice Edit is that you cannot access the Voice Type or the Effects page. You know why, right? Yep. DSP Resource Hazard.

If a voice uses a sample, you can go down into the wave edit level, without leaving the multi, and without stopping your song. This is very handy when trimming the front of a sample or fine tuning it to a track.

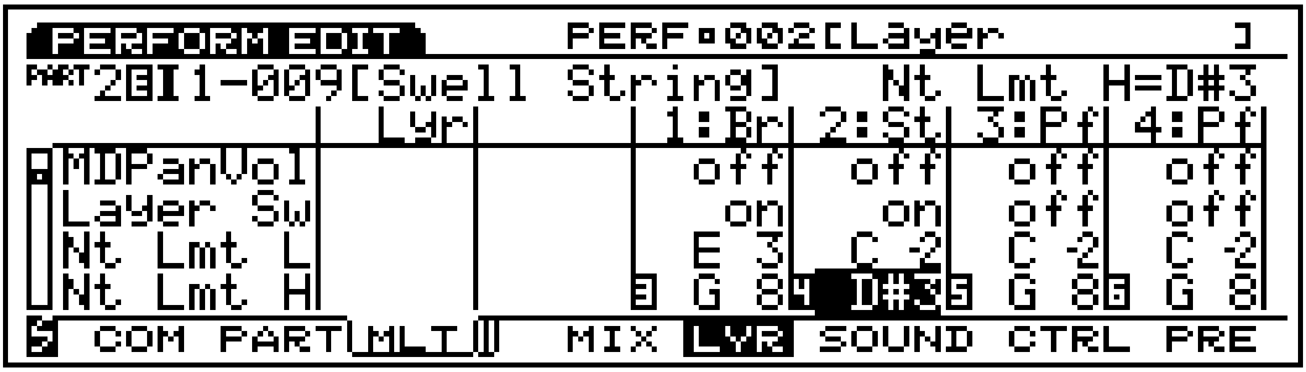
Performance as Layer or Split

If you want to make Performance Layers and Splits for use in live performance, or in a non-Multi way, then the Layer Switch is the key. Choose the second Performance from our example set, called “Layer”. Hit Edit.

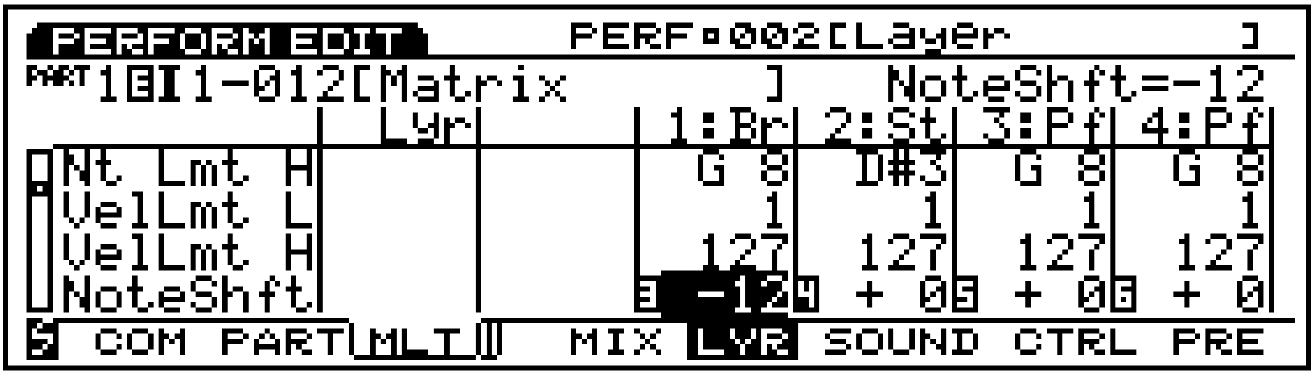
If you scroll down the MLT view, you will see that Parts 1 and 2 have the Layer Switch turned to ON. It doesn’t matter which of these two parts is selected, you will be playing both because they are linked by this switch.

You might be saying to yourself, this patch is ridiculous, I don’t need to Layer a Matrix Brass sound with a String sound. So let’s convert it to a Keyboard Split.

Right underneath the Layer Switch are the Note Limits. Set the break point wherever you like. It makes sense to me to split around E3, and to keep the strings in the left hand since there’s a good chance the strings will be slower than the brass. Whatever. Try it out.



Now the Matrix Brass sounds a little too high, so I’m going to shift the range down an octave.



I’ll crank the reverb a little on the Matrix, and store it into Performance position 3 after renaming it “Split”. (You can check that one out, if you like.)

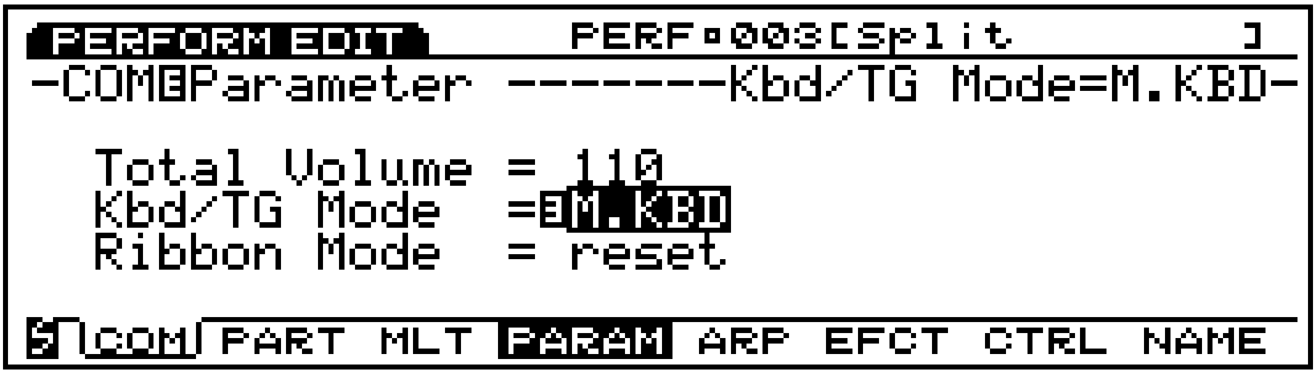
Performance as Master Keyboard Setup

If you want to use the EX as a Master Keyboard for your studio or live rig, there's a wealth of possibilities. Each Performance can have a unique MIDI setup, a unique Knob assignment, and even a default controller "position" which can be sent when the Performance is called. If you're controlling a number of synthesisers, you can send up to sixteen Program Change messages upon calling up a Performance. There are two MIDI ports, A and B, that can be assigned individually. Basically, from the EX you should be able to control your entire setup from song to song.

If you want to set up Performances to control other synthesisers, then the first thing you should change is the Keyboard/ToneGenerator mode (Kbd/TG Mode) in the Performance Common section. Let's take our "Split" Peformance and adapt it so that the "Matrix" sound plays my real Matrix 12 instead of the EX sampled version. (Actually, I don't have a Matrix 12 either, but pretend.)

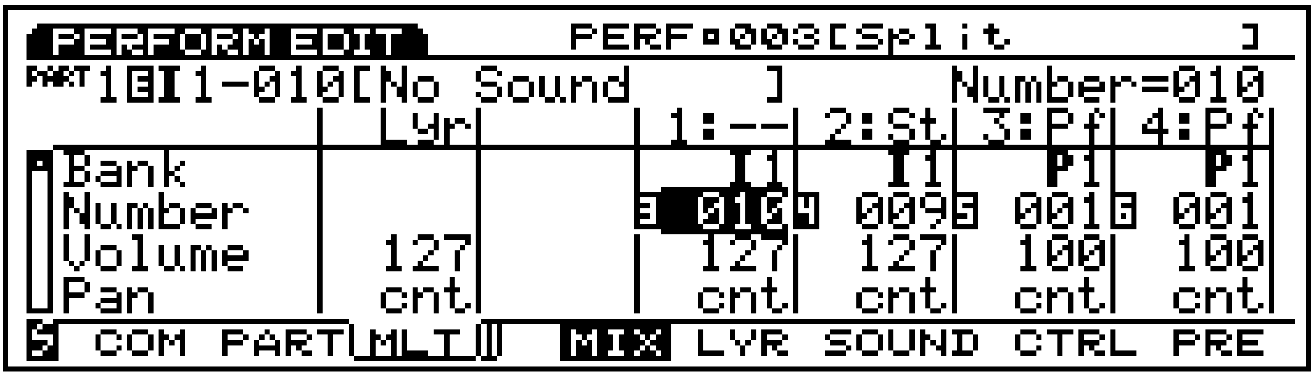
Go to Performance Common. EDIT/COM.

Change the Keyboard Mode to M.KBD (Master Keyboard).

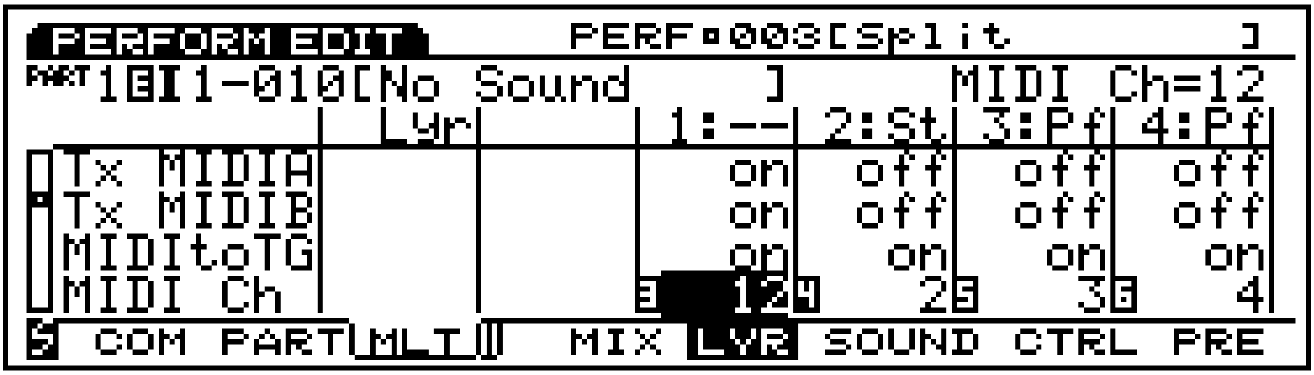


Essentially, this just changes the routing of the EX's keyboard so that some Performance parameters (velocity range, key range, transpositions, curves, etc.) happen BEFORE the internal tone generator.

Change Part 1 so that it's playing nothing. (I keep an empty voice for this purpose, it's called "No Sound" and is in I1:010).

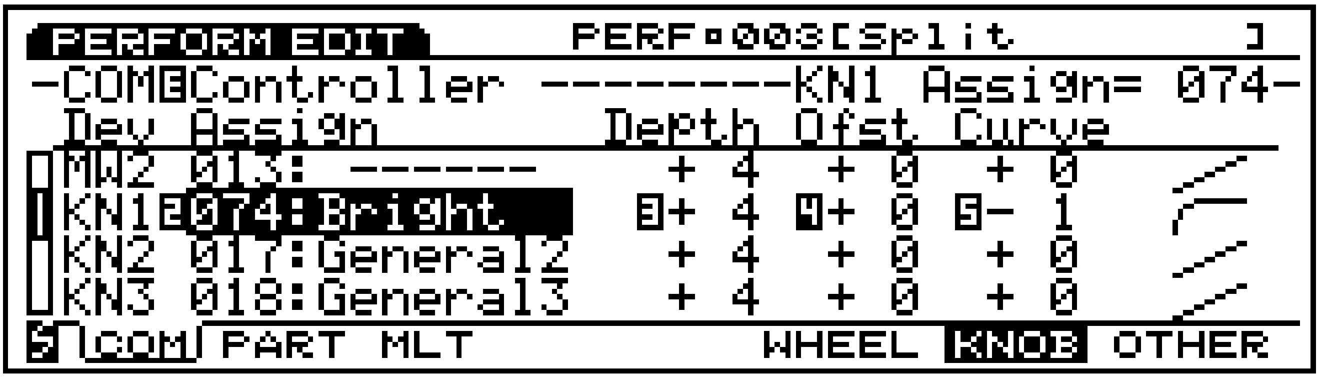


Now scroll down to the LYR section, near the end, to change the MIDI channel of Part 1 to whatever you Matrix 12 is set to receive.



If the MIDI output of the EX is going to the Matrix, it should be making sound. If not, check that "Transmit MIDI" is ON for the port your using. (Just above MIDI Ch, where it says Tx MIDIA.)

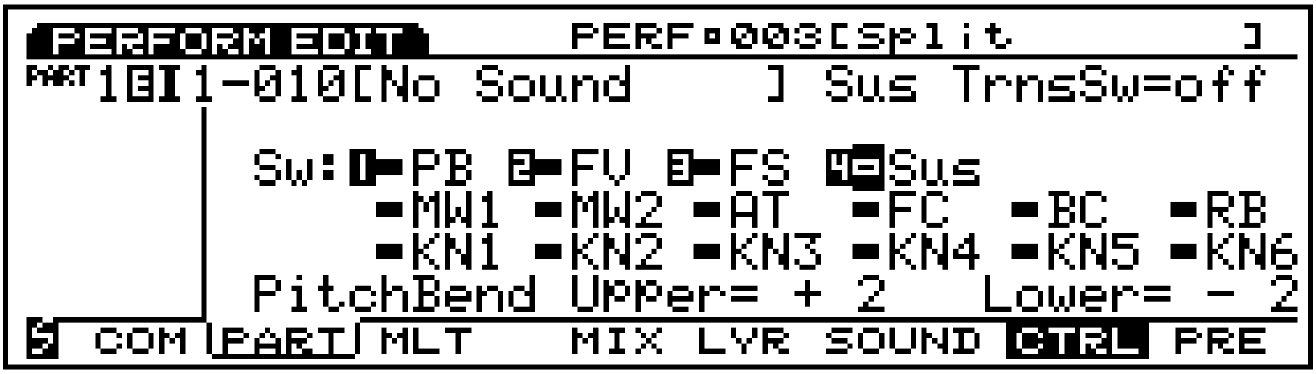
Now let's say we want to use Knob 1 to control the filter cutoff on the Matrix. Set up the Matrix so that it has a controller routed to it's filter cutoff, and set the EX so that the Knob transmits this controller. My "imaginary" Matrix is receiving controller 74, which is routed to the filter. Go into the Performance EDIT/COM/KNOB page, and change the controller of Knob 1 to 74.



You can also change the depth, offset, and curve of this Knob if you like.

Hit KNOB Mode to check it's working.

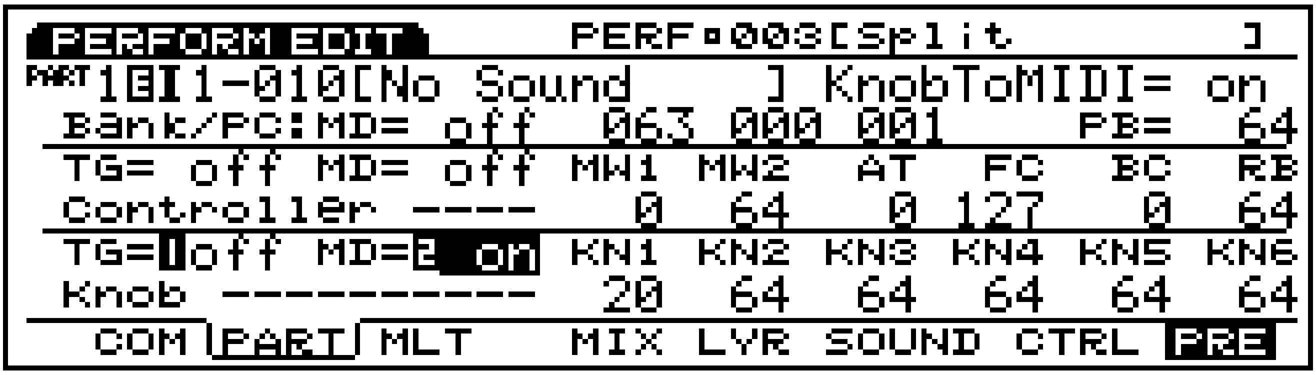
Yep, it's working. But I don't like the way the Sustain pedal is affecting the Matrix, since I want to just sustain some Strings, and then play on the Matrix above. So I go to the Controller Transmit page, EDIT/PART/CTRL, and turn OFF the Sustain pedal (Sus). This stops the Sustain pedal from being transmitted, but it will still function locally.



You can have up to 16 zones on the keyboard this way, all playing different instruments if you like. Notice that the “Layer Switch” is only relevant for the Local sound of the EX, you can turn ON the Tx MIDI A or Tx MIDI B for any other Part at the same time.

If you decide to explore the Master Keyboard possibilites, and start building your own custom Performances for this purpose, you should also be aware of the Preset page (PRE). What this gives you is an INITIAL STATE for each Part or MIDI channel. When the Performance is selected, each Part can send a Bank Select, Program Change, Pitch Bend Value, and an intial value for all the controllers and knobs on the EX.

Here, I’ve set up the Performance so that it sends a default value of 20 to my Matrix 12 on controller #74 (KNOB 1). Note that you can choose whether the default is sent to the internal TG, external MIDI, or both.



This can be extremely useful for live performance. Each song in the set can be set up completely with a single Performance selection.

Recipe Seven

Edit Compare

When Editing a Voice or Performance you can quickly check the existing “Stored” version by hitting the EDIT button. Hit the button again, and you return to your current state.

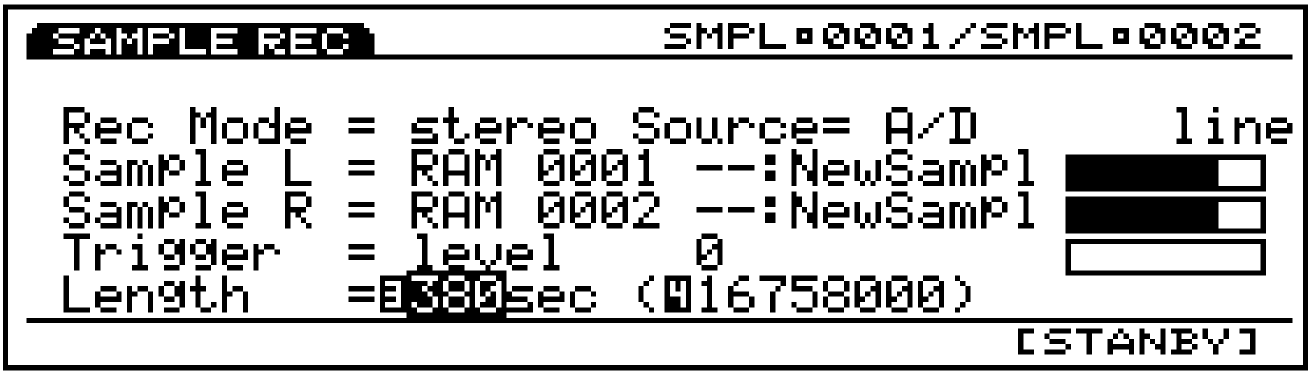
6

Sampling

There are several ways to get sample data into the EX. You can record samples from the analog inputs, and you can sample sounds the EX makes itself (internal resampling). You can load WAV, AIFF, or Akai files from floppy disks. And you can send sample data into the EX from a computer using SMDI.

Recording a Sample

If you want to record something directly with the EX, enter the SAMPLE page, and hit the REC f-key. You should see something resembling this:



The record mode can be set to several Left/Right combinations in the EX5 and EX5R, but in the EX7 you can only record mono samples (although you can play back stereo samples).

The Source parameter can be switched to A/D (analog to digital converter, otherwise known as your audio inputs). You can choose line or mic level input. And you have a hardware input gain on a knob above the master volume fader. If you want to resample the EX itself, then set the source to “Internal”.

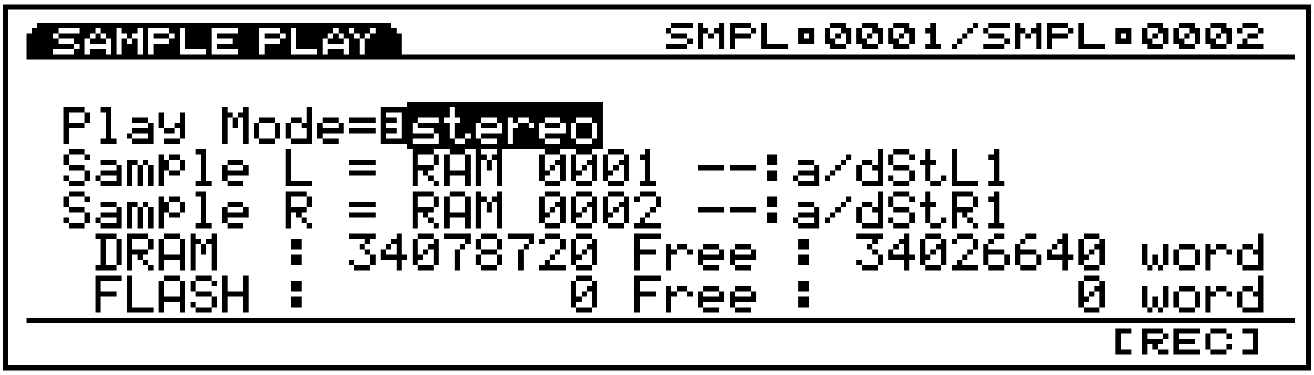
You can trigger the sampling by level or manual command. When set to “level” the sampling will begin whenever any input occurs above the threshold you set.

You can set the maximum size of a sample below that, and also use this to see how much memory you have available.

Hit Standby to activate the recording stage.

Go.

After recording, exit the Rec page, and you should see something like this:



Play the keyboard, and you should hear the sample you just made.

Will get to editing it in a moment, let’s first just check the other methods of getting sample data inside your EX.

Loading a Foreign Sample from Floppy

This one is easy enough. Make sure your floppy disk is DOS format (format it inside the EX if you want to be sure), copy a WAV file onto it. (There’s one on our floppy if you want to try it.)

If you’re a Mac user you must remember to name the file in an DOS manner. That is, 8 (or less) characters, a dot, and WAV. For example: SAMPLE.WAV Or in the case of an AIFF file, it must be named SAMPLE.AIF

On the EX, go to Disk mode, File Load, and choose WAVE. The WAV file on your disk will appear as a WAVE to the EX. Load it. Your done.

With an Akai floppy disk, you will see the Akai Program. So make sure that any Akai samples you want are included in the program.

Loading a sample via SMDI

You must have a SCSI board in your EX for this transfer.

If you have sample data inside your PC or Mac, you can send this data over SCSI directly to the EX RAM area. SMDI is a standard protocol for sending samples via SCSI like this.

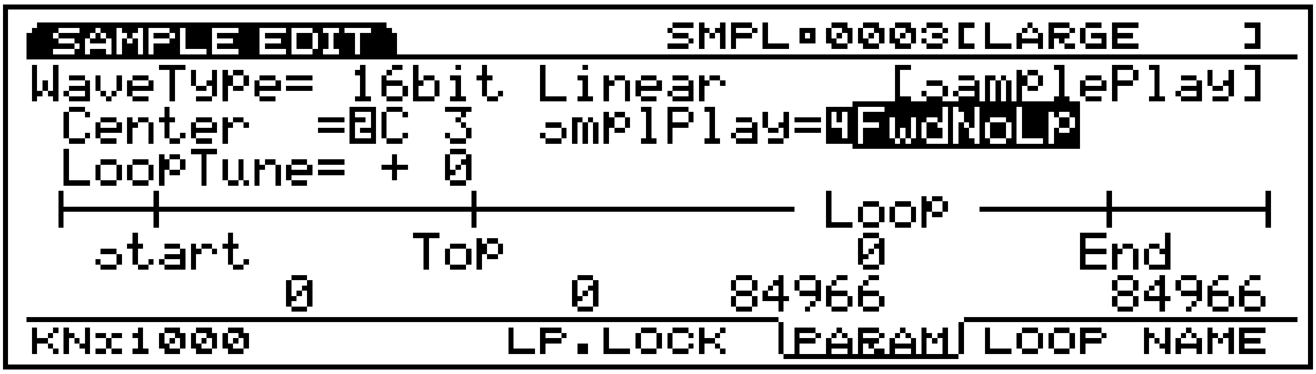
Check the SCSI ID of the EX (in Utility/Other), since your computer’s sample editor will want to know this. Open your sample in the editor, and set it up to send to an SMDI device on the correct SCSI ID. Be careful that you don’t duplicate any SCSI IDs with other devices on your SCSI bus.

Basically, you just SEND from the computer. If all is correct the EX will show an SMDI Data message while it’s receiving the data.

Be careful not to overwrite existing sample data. SMDI requires a sample position to be designated. If you tell it to send to sample position #1, any existing data in the EX at sample postion #1 will be overwritten.

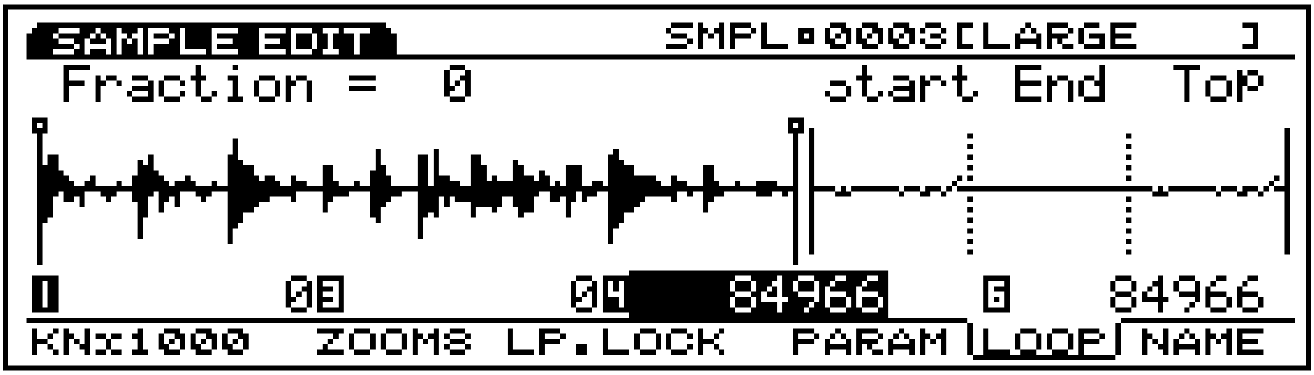
Sample Editing

So now your sample is inside the EX. (If not, load the LARGE.WAV) from our floppy disk. Enter the SAMPLE Mode, and choose the sample number with our sample in it (very likely, it's number 1). So hit EDIT and we'll enter the Sample Editor. There are three main pages in this section, PARAM, LOOP, and NAME. NAME is quite obvious. Choose PARAM first.



Here you can set some defaults for the playback of this sample. Many of these things can be changed from the Voice using this sample, so it's not necessary to make any permanent decisions here. You can choose a default original key (Center), and a default playback method. Let's turn the Loop ON for a moment. (FwdLp = Forward Loop)

The bottom three numbers are the sample start, loop start, loop length, and sample end points. But let's hit the LOOP page while we look at these.



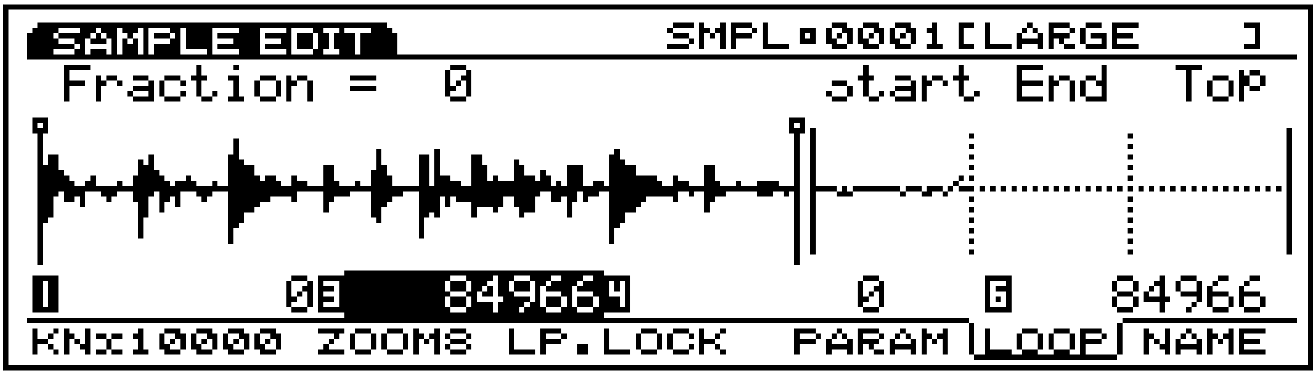
The currently selected parameter in the above picture is the Loop Length. You can scroll this with Knob 4. However, you might find that the scrolling is very slow. Notice in the bottom left corner there's an "KNx1000" written. If you repeatedly tap this f-key, you change the number of samples you jump when you turn the knob one tick. Set this to 1000 and you can move the points much more quickly. When you want to fine tune the placement of the points, you can set it smaller again. Notice the loop cross point zoom view to the right of the screen. If the Zoom is set to 1, this displays individual sample positions so that you can set the loop points very precisely.

Notice that you must re-trigger the sample to hear the change in any points.

Notice also that you cannot start a sample after the loop start point. Nor end a sample before the loop end point. The loop is always WITHIN the sample, even if it includes all of it.

If you trim the front or the end of the sample, you may want to crop or "extract" it (ie. throw away the stuff outside your selection). You can do this with the JOB commands. In this editor, when you hit JOB you have the choice to Extract or Normalize the current sample data. You can also delete, copy, or append two samples using these JOBs.

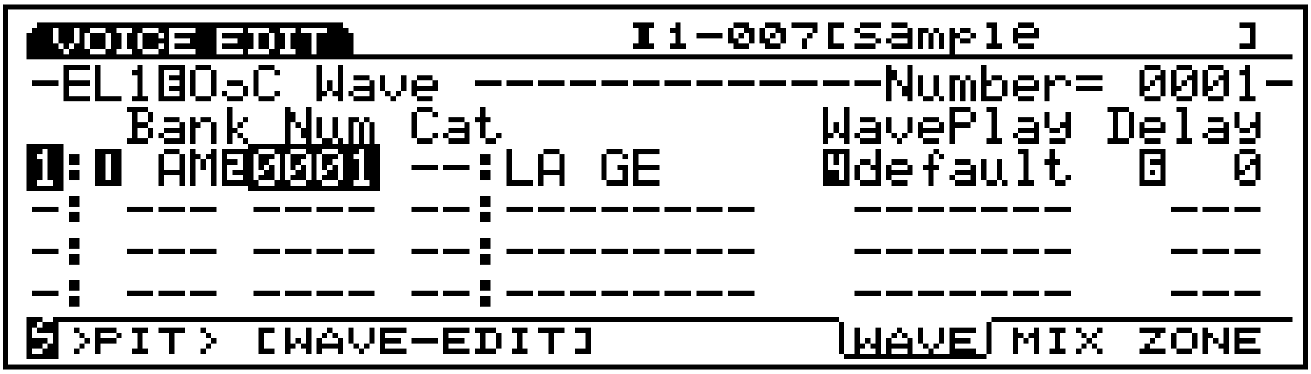
Now do me a favour before we leave this section, set the sample to FwdNoLp (Forward, No Loop) and set the Loop In point to the end of the sample, 84966. We’ll see why later.



Using the Sample in a Voice

Go back to Voice Mode, and choose voice I1:007:”Sample”. In fact, this voice is just an Init Voice right now, but we will use it to make our Sample Voice.

Hit EDIT/OSC and you’ll see that the current Oscillator is PRE:001:Pf:Grnd1, a standard piano sample in the Preset ROM collection. We want to access the RAM area, so let’s change the PRE to RAM, the sample number is still 001, and hopefully you’ll see our LARGE wave written there. Like this:



But, something’s wrong, you still can’t hear it, right?

Right.

You still can’t hear it because this WAVE is currently empty. It stole the name “LARGE” from the sample, but in fact there still is no WAVE as yet using this sample.

Jeez, you might say, I’ve got to make a WAVE just to use a sample?

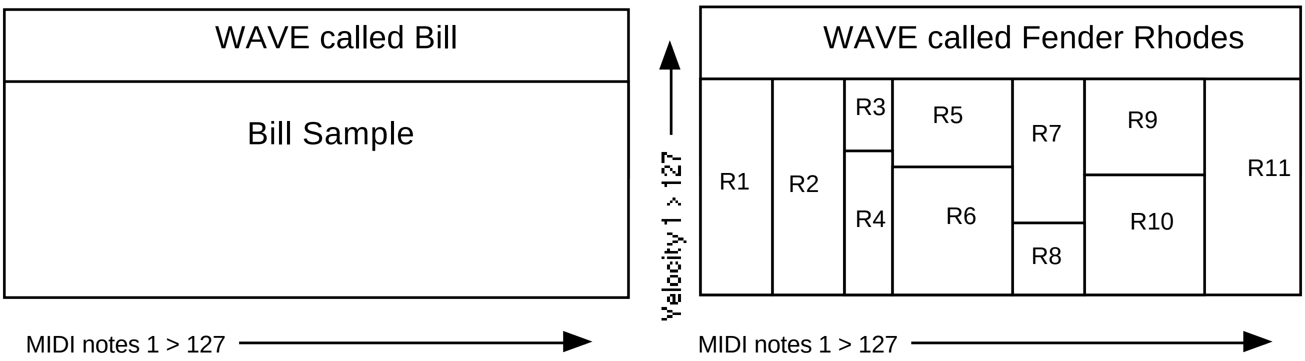
Well, you do and you don’t. You do if you want to use the sample in a Voice Element. (You don’t if you only need to access the sample from a Drum Voice.)

It’s not difficult, though, and you’ll see in a second, it opens up some possibilites.

What exactly IS a WAVE?

A WAVE is a group of one or more samples. The WAVE is great because it can contain 1 sample or 100 samples, and still be handled as one unit.

Look at the following two diagrams:



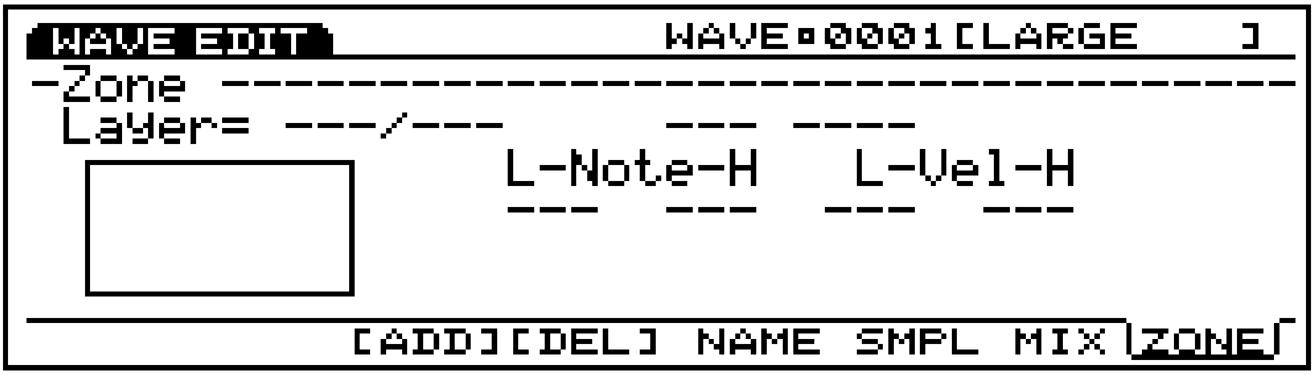
Wave Bill contains only one sample across the entire note and velocity range.

Wave Fender Rhodes contains 11 different samples strung out all over the place to build up a complete multi-sample of the Fender Rhodes.

Once these samples are grouped as a WAVE, you don't have to think about managing all those bits. You can load the wave as a single unit, point to it from an Element as a single unit. It can be very useful. And you can forget about what samples were used to make the WAVE.

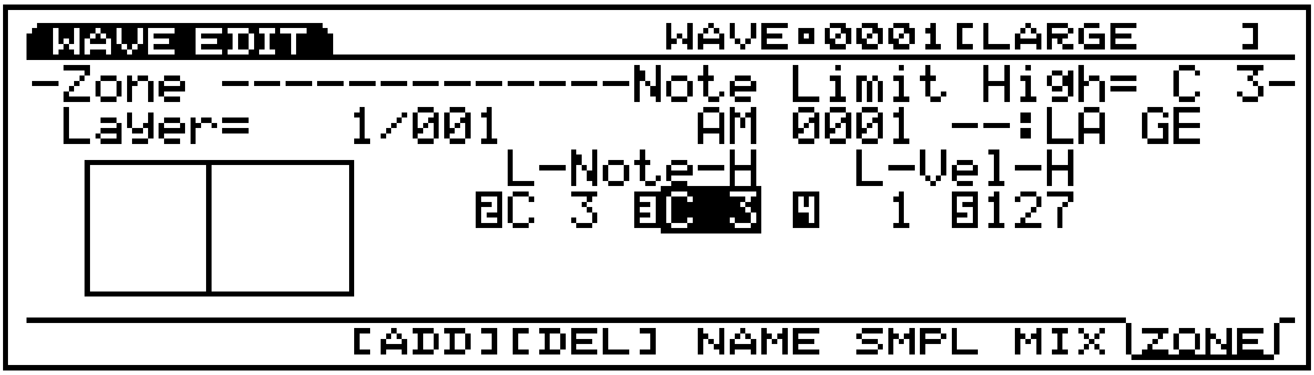
In fact, many of the Preset ROM WAVES are complex multi-samples like this.

So, let’s hit the F-key that says WAVE-EDIT.

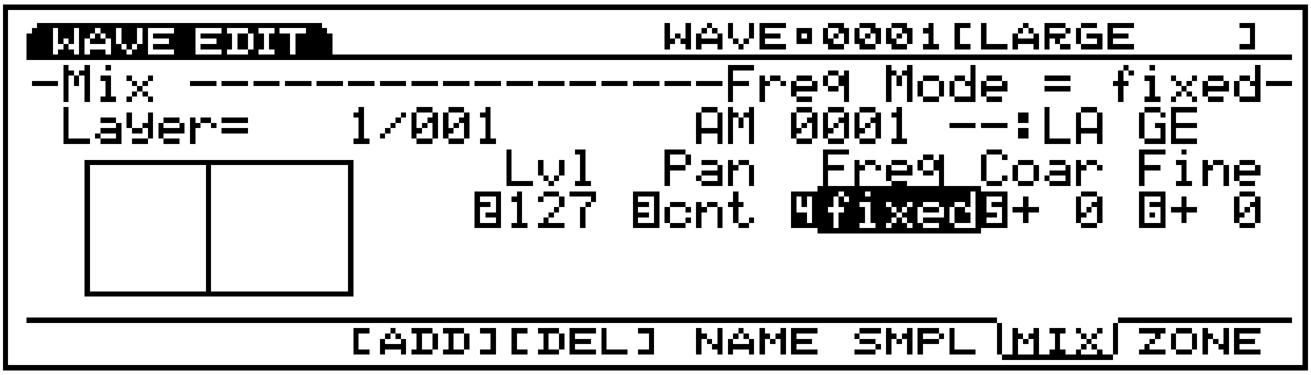


Sure enough, it’s empty. That’s why it was so quiet. Hit the f-key that says ADD, and this will ADD a new “Layer” to your WAVE. It put that Piano in there again, didn’t it? No problem, just change the PRE to RAM, and the LARGE drum beat should show up.

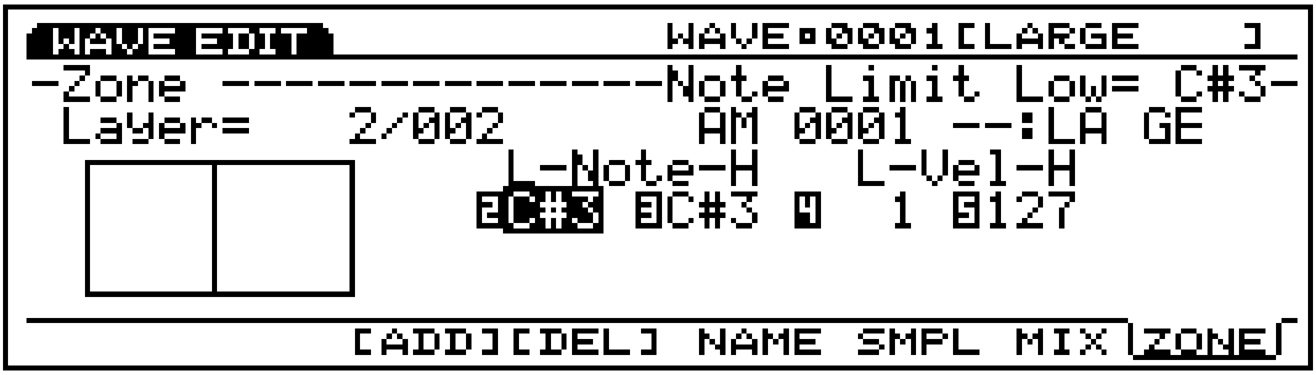
Now you can see your ZONE. For reasons that will soon become apparent, let’s limit the zone to one key, C3 (60). Set the low and high Note to C3.



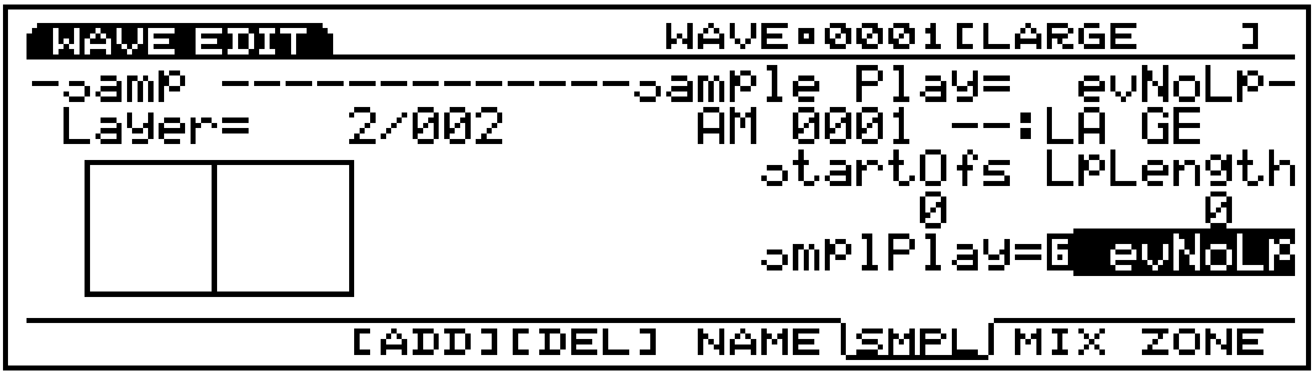
Hit the MIX f-key, and let’s set the Freq to fixed. We’ll do this so that no matter where on the keyboard we play the sample, it will play at the same pitch. It doesn’t stop you from transposing or pitch bending the sample, so don’t worry. You can coarse and fine tune freely even when Freq is set to Fixed. You can Pan it here too if you like. Make sure the Level is Maximum.



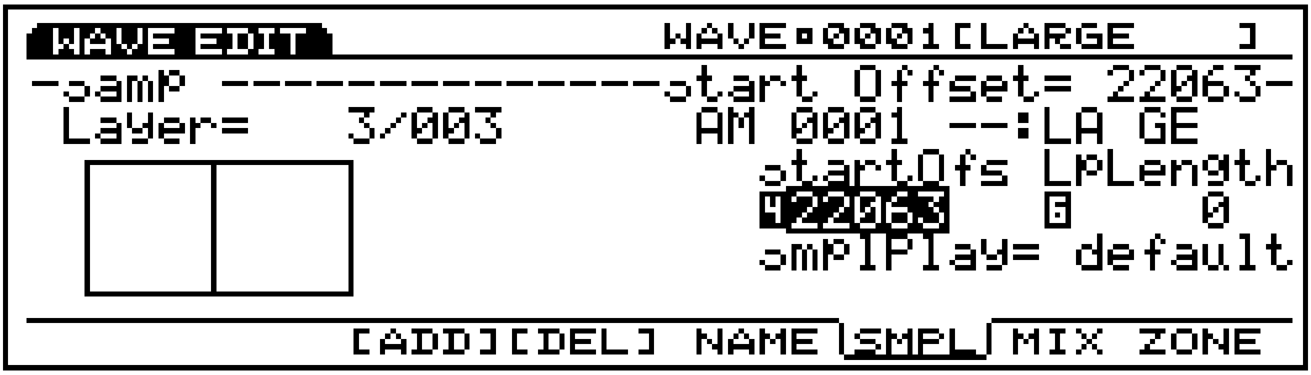
Now hit ADD again. Then the f-key ZONE. Notice that the “Layer= 2/002” now. This is layer 2 of a total of 2. But conveniently, the second layer defaulted to whatever was set in the last layer. Let’s just change the key to C#3.



Hit the SMPL page. Where it says SmplPlay, change it from “default” to “RevNoLp”. Now you’ve got a key to play the loop in reverse.



Hit ADD one more time, and go back to ZONE. Define it’s note limits as D3. Go back to SMPL again and switch to to forward play. Now you can offset the Start Time of this sample, set it to 22000 (use the ten-key pad), or so.



Now you have a section of the loop beginning at the snare drum. (This was the reason we turned off the loop back in the sample editor, and moved the Loop In point down to the end. The Start Time Offset CANNOT move inside the Loop In point. So if this is something you like to do, remember that limitation.)

What you’re building here is a WAVE. You’ve already got three versions of this loop within the one wave. Now a RAM wave is like a RAM sample. There’s no need to STORE after editing, but there is a serious need to SAVE TO DISK.

WARNING

Samples and Waves disappear
when you power down.
Back ‘em up.

Why Not Put ALL My Samples In One Big Wave?

Well, because a WAVE is used by an Element. An Element has filters and envelopes and so on, but ONLY ONE SET. So if you wanted a different filter on the snare drum than on the kick drum, for example, you’d be screwed if they were in the same WAVE. WAVES are for SAMPLES THAT BELONG TOGETHER.

How Can I Use More Than Four Samples In A Voice?

Use Drum Voice, coming up next.

Recipe Eight

Hardware Shortcuts

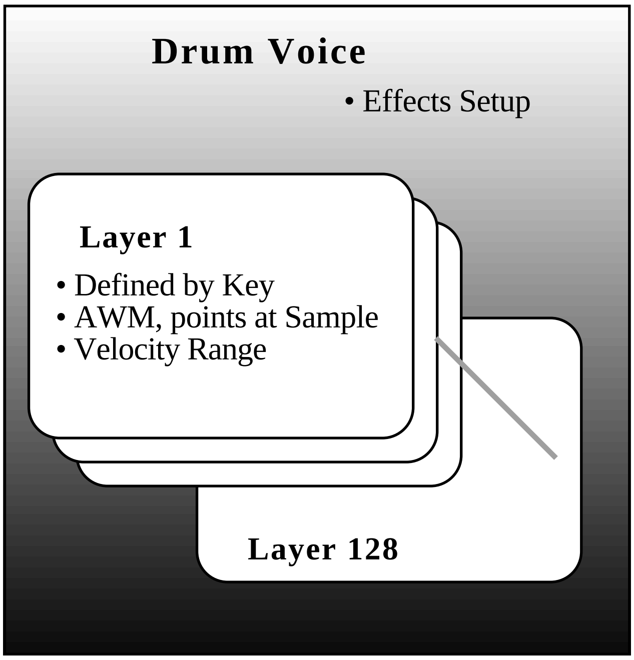
1 Learn to use the hardware buttons for Part and Track selection when in Performance mode, or any of the MIDI modes (patterns, arpeggiator, sequencer).

2 Also, notice the Element selection and Element On/Off function of the Bank Switches. This is useful in Voice Edit.

3 The Ten-key pad can be used for entering large values, such as sample addresses or delay times, but can also be used for quick entry of note values and velocity values in any of the MIDI modes. It can also be used for naming.

4 Whenever you're in an edit mode, the Knobs change function to become data entry. If you need to check the Knobs play function without leaving the editor (such as when assigning control destinations in a voice), just hit the hardware KNOB button.

Drum Voice



Drum Voice is pretty important when you work with your own samples. If you want to have many different samples in one voice, with different envelopes and filter settings on each, then this is the type of voice to use.

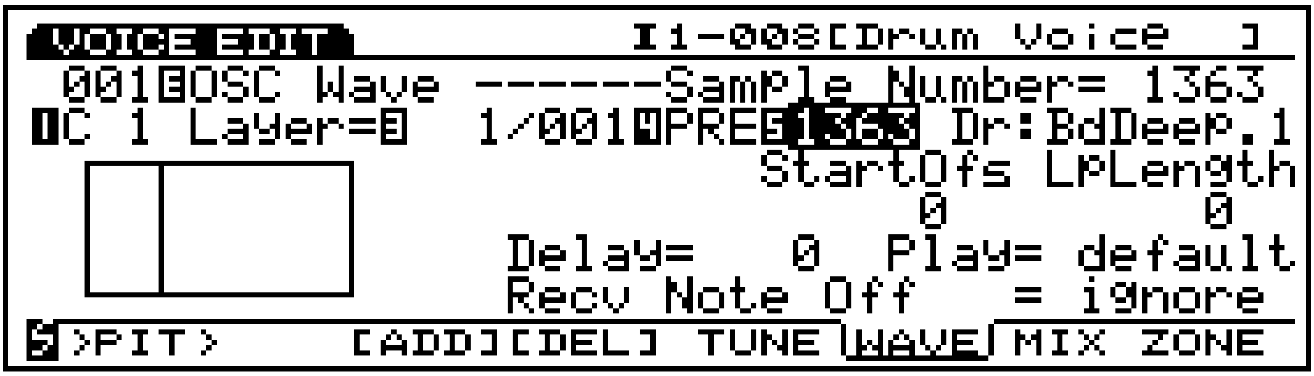
Essentially, Drum Voice is very similar to a “normal” Voice in the various sections of the Editor. The main difference is that Drum Voice is built of “Layers” instead of “Elements”. In Voice, you could have 4 Elements, and each of them had editing of filter, envelopes, etc. In Drum Voice, you can have up to 128 Layers, and each of these has it’s own filter, envelopes, etc. Each Layer, however, is associated with only one Key. Although, you can have more than one Layer on the same key if you wish. And of course, you can have several keys using the same sample.

The Layers are created much like the Layers in Wave Edit.

Let’s use “Drum Voice” I1:008, as an example. It’s completely empty at the moment.

Choose it, hit EDIT/OSC/WAVE.

In the upper left corner, you see written here C1. If it’s not on C1, choose C1 (you can type the number 36 on the ten-key pad if you want). There’s nothing on C1 yet, so let’s hit ADD. This immediately creates a layer and defaults to a Preset Snare Drum wave. Scroll the wave back to BdDeep.1 (wave #1363). You screen should now look like this:



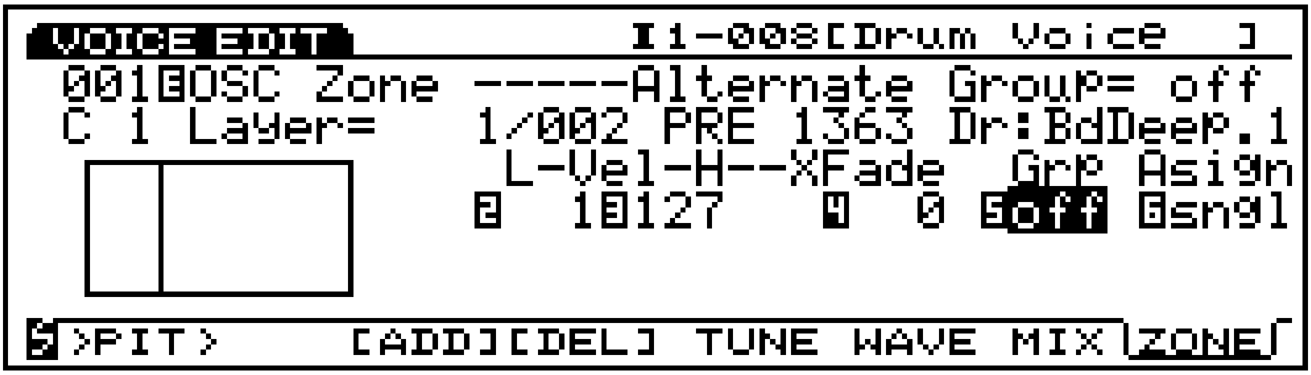
Let’s imagine we want more of a click on this sound, so we need another wave on the same key. Hit ADD again. This will immediately copy the previous layer to a new layer, and the display will now say Layer= 2/002 (2 of 2, ignore the little black 3, that’s the knob). Scroll the wave of this second layer to wave #1391, Stick.1. You now have two waves on the same key.

If you choose the Play=default parameter, you can reverse the wave here if you like. You can also use a note-on delay, or alter the start time offset of the sample in this page. And most importantly, notice the parameter called “Recv Note Off”. This is set to “ignore” at the moment. This means that no matter how long you hold the note, the sound will play the same length. In truth, it’s using the amplitude envelope, but ignoring the “sustain” portion. If you are using samples, and want to have a “sustain” portion, and then a “release” portion, you should change this parameter to “receive”, in which case that layer will behave as a normal ADSR-type trigger. Note that each Layer has it’s own settings for all these parameters, even if there’s more than one Layer on the Key.

Now lets hit the MIX f-key to change to the MIX page. Here you find parameters to set the level and pan of the layers, the sends of the layers, as well as a choice of Insert Effect placement. Any drum layer can be in either insert effect (although there are only the two insert effects for the entire Drum Voice). Choose InsEF number 1 for both Layer 1 AND Layer 2 of our C1 key.

Hit Exit. Then hit EFCT/INS1, and set it to type 15:Overdrive. You can now overdrive any drum in your kit if necessary. Set up the Overdrive to your liking, and let’s go back out to the Oscillator page again.

Look in the ZONE page. EDIT/OSC/ZONE.

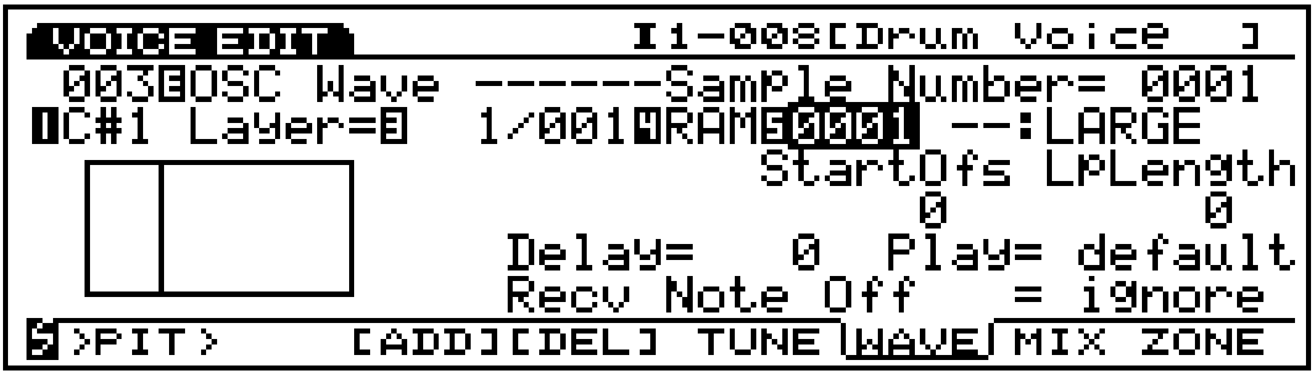


Here you can set a Velocity lower and upper limit for any Layer. In this way you can build velocity splits. If you use the XFade amount, you will create a velocity crossfade. Also in this page, you can choose a Layer to be a member of a Group (Grp). If two layers are members of the same group, they will mute each other when they play. This is often used between high hat samples so that the closed hat will always mute the open hat. But you can use it for anything you find interesting. You can have 128 groups if necessary.

The last parameter in this page is called Assign. In single mode, if the Layer is re-triggered it stops the previous one and begins the same sound again. If it’s in multi-mode it completes all of them, even if they overlap.

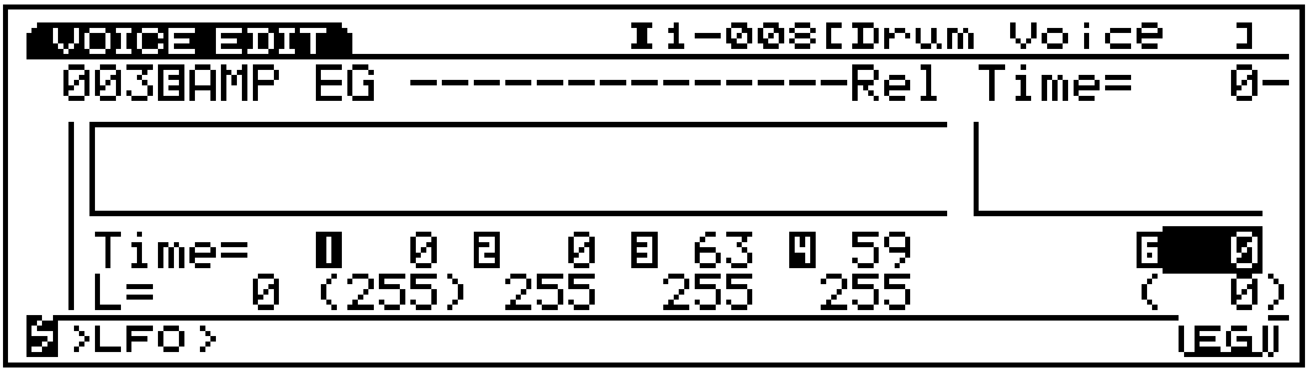
The TUNE page let’s you tune the Layers, as you might expect.

Now let’s choose a different key for the next experiment. Choose C#1, hit ADD, and choose the RAM sample we have. (If you’ve powered down since the sampling section, you may have to load the LARGE.WAV again.) It should be in position RAM:001. Try it.

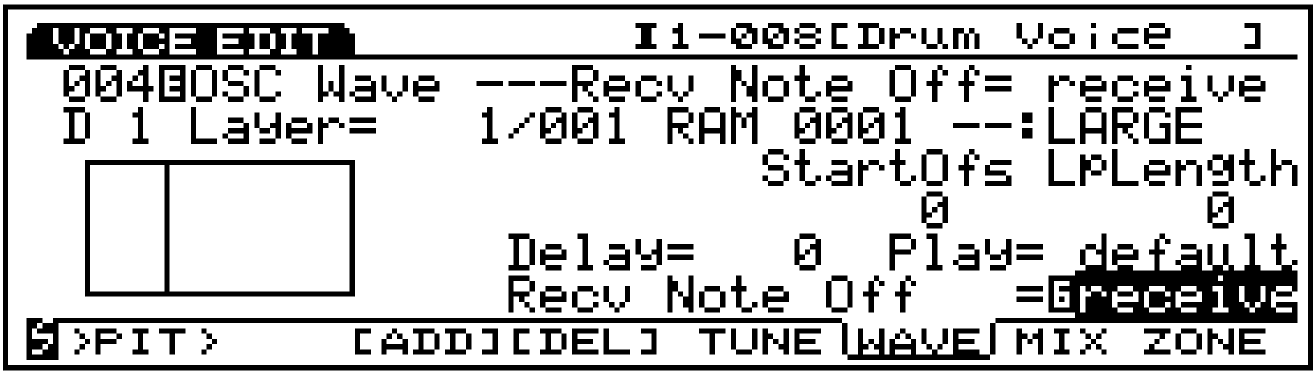


You should hear the first couple beats of the drum loop. (This is an ideal sound to hear the difference between Assign single and Assign Multi in the ZONE page, by the way. Hit the key twice quickly, and you can hear what's happening in the "multi" setting.)

Now, this loop is currently incomplete. You probably want to be able to control the length of the loop from the keyboard or MIDI, so we need to change the "Recv Note Off" parameter to "Receive". Do this. Now you can hold the entire loop if you want. You might notice that the release time is rather long, and you can't stop the loop very quickly. This release time remains from the trigger mode that the drum voice defaulted to. We can fix it easily by jumping to the AMP/EG page and reducing the release time. Try this.



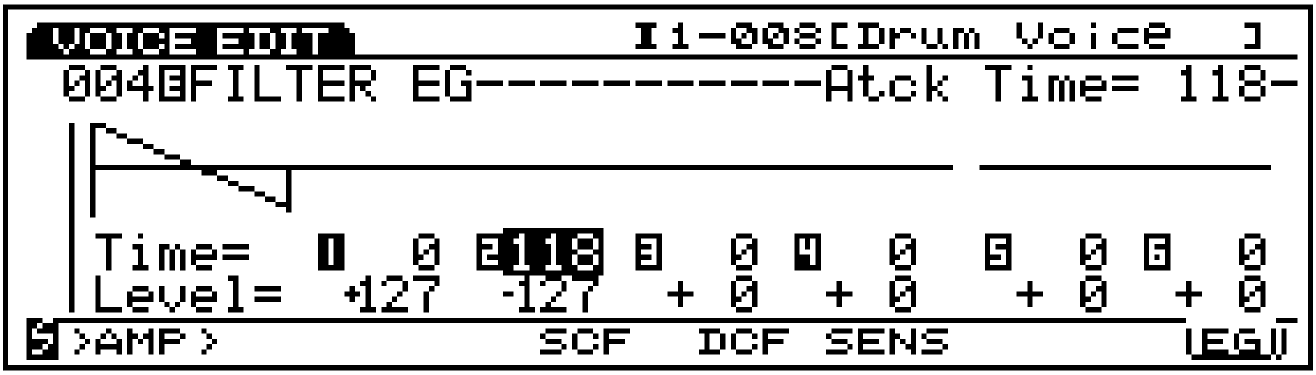
Let's make another Key, D1, and ADD the same sample to it. RAM:001:LARGE. Let's set the Amp release to 0, and the Recv Note Off to "receive". Come back to the WAVE page:



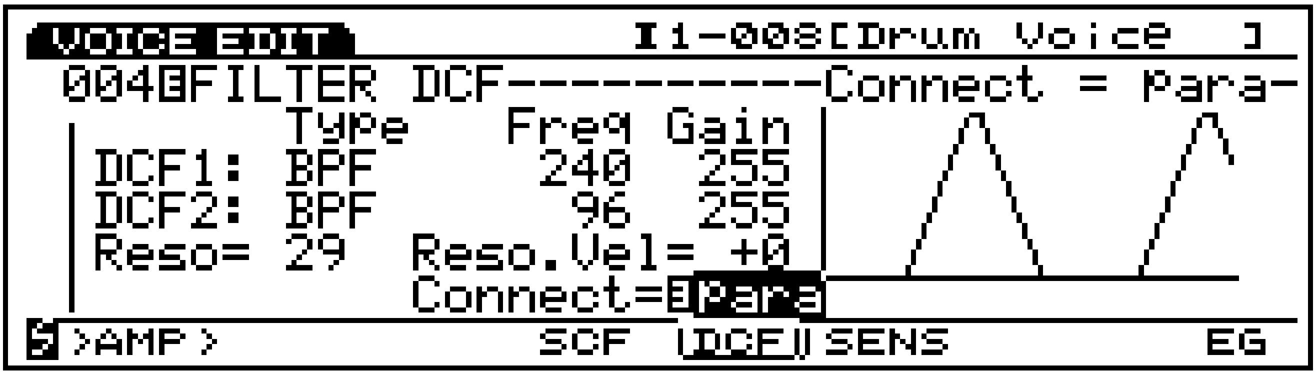
C#1 and D1 should now sound the same.

But let's change the Start Offset of D1 to somewhere around 22000 (use the ten-key pad), so that we're triggering the loop at the point where the first snare drum starts. With several copies of a loop like this, you can transform breakbeats into your own new patterns very easily.

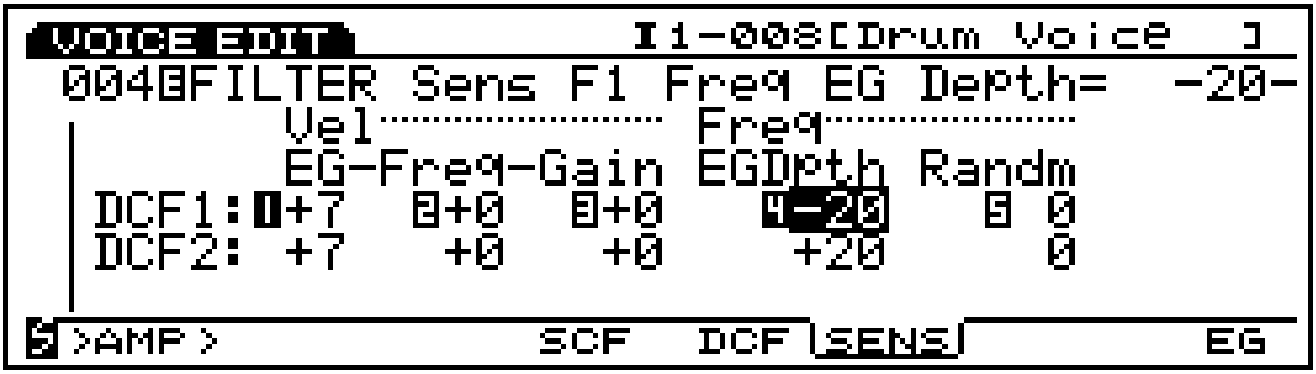
Now let's mess it up further. Go out of OSC and into FILT, enter the EG section of the filter page and set up a downward sweep, like this:



Go to the DCF section and choose the BPF (band pass filter) on both filters. Set the resonance quite high. And put the filters in parallel mode. Like this:



Now set the envelope sensitivities so that they're opposite. This way the sweep will move the two band pass filters in opposite directions.



It's starting to get messed up pretty good now. If you want, go back to the OSC/MIX page, and turn on the insert effect for this layer.

Remember that the samples are lost when you power down (unless you've used Flash RAM), but your Drum Voice can be stored in the EX, and will use the same samples if they're loaded again. The only requirement is that the samples be in the same ORDER next time, since any voice using the RAM samples will do so by it's sample position.

This is easy enough to do. If you save WAVE, it will save all the WAVES in memory. And when you Load WAVE you can load ALL of them at once, and they will be in the same positions by default. You can also load a single WAVE from a file of many WAVES if necessary, more on this in the chapter called “Data Management”.

RECIPE NINE

Floppy Disk Format

If you find that the floppy disk formatting is rather slow, use your computer. It’s just a standard DOS disk.

7

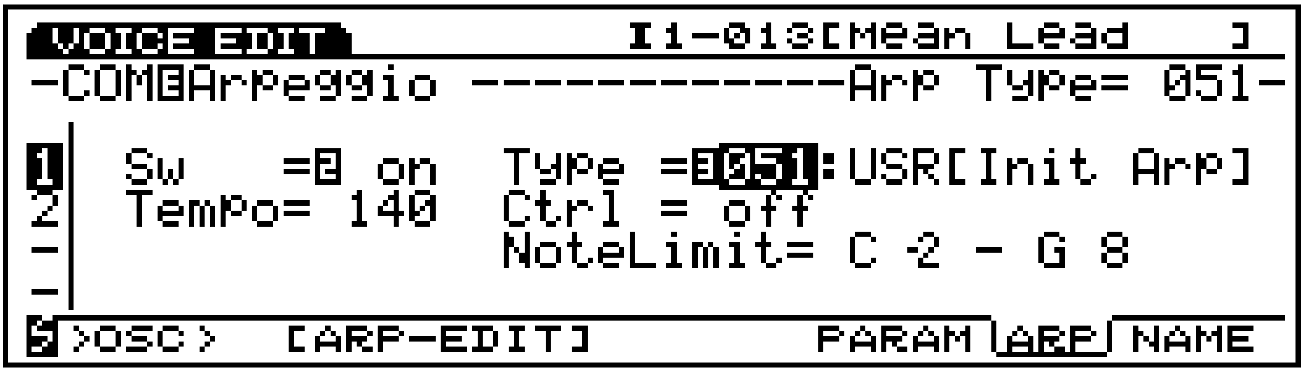
The Arpeggiator

One of the great features of the EX synthesisers is the fully programmable, 4 track, arpeggiator. For an extensive explanation of everything the arpeggiator can do, check your manual at the section beginning on page 238. For the purposes of this guide, we'll show you the basics of making your own arpeggiator pattern, using an imported MIDI file as a template.

Let's assume you've got a pattern in your sequencer that you think would make an excellent arpeggio pattern. Save it out as a MIDI file, copy it to a DOS formatted floppy, and import it into the EX. (The example we're using is on your floppy disk, and is called Arp.MID).

By default this MIDI file will be copied to the EX SONG file. If you then hit Play the onboard sequencer will play this file. Check this to make sure you've got the data.

Now choose a voice to test the arpeggiator with. Turn the Arpeggio switch ON, and hit EDIT / COM / ARP. Scroll through the parameter called "Type".



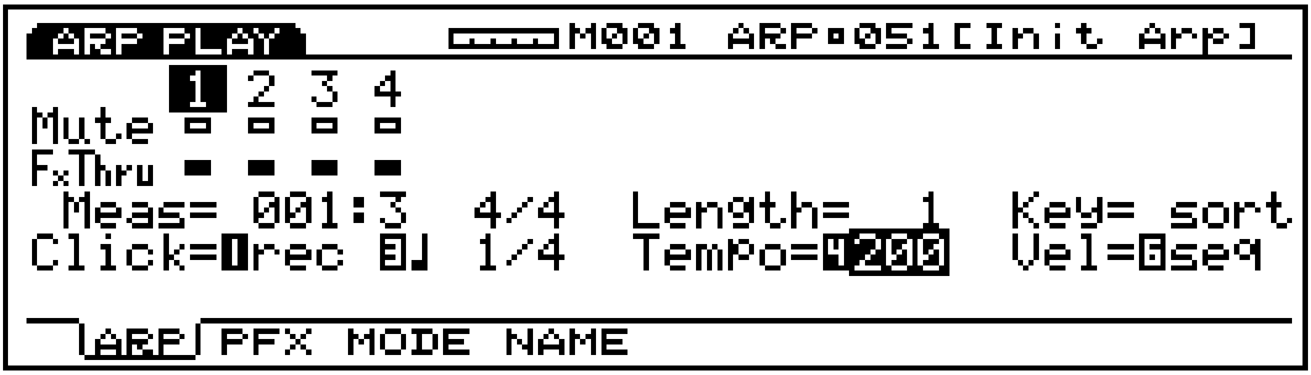
The first 50 are preset, you can't change them. At 51 they become User patterns, and you can choose any of these to make your own arpeggio pattern. Choose one of the User patterns, and hit F-3, Arp-Edit.

Now you have to copy your MIDI file into this location. Hit JOB (beside the main Edit switch), and then section Job 2 / 3. Get Phrase. You'll now have a choice of what measure and track in the Song to copy into the Arpeggiator. Our example is in Track 1, Measure 001 - 002.

Hit Enter and execute the Copy. Then hit exit a couple of times to get out of the Job mode and back into the Arpeggio editor.

If you hold down some notes, it should make some attempt to play an arpeggio, but will probably sound completely ridiculous. We've still got to set it up. Now it gets interesting.

First, change the Arpeggio Length to 1 (since our MIDI file is only 1 measure), and set the tempo to whatever you want.

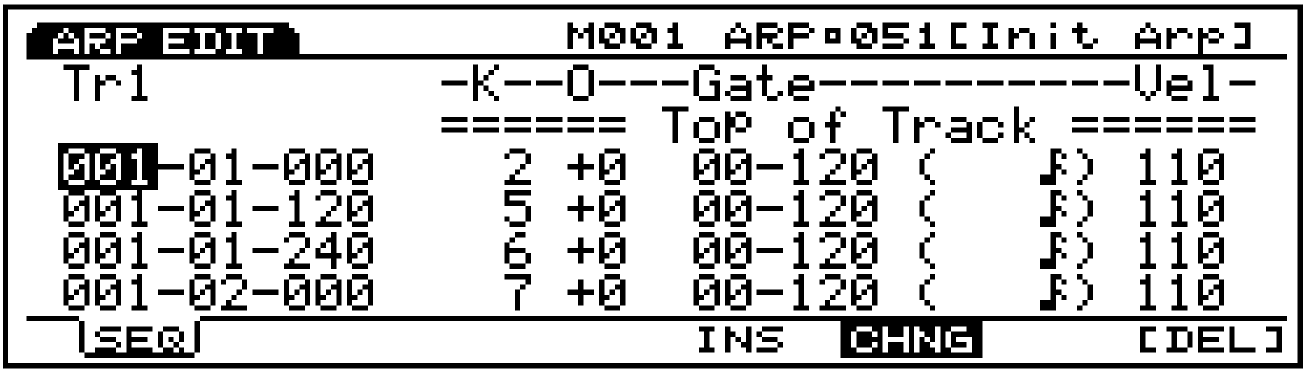


Now, go to the MODE page, and change track 1 to Search Low. Basically this is one mode that causes the arpeggiator to always play a note for each note of your pattern. There are many different modes here, which are explained in the manual, but for now let's just use Search Low.

In our example, there are 7 *different* notes, even though some are repeated. If I play less than seven keys on the keyboard, Search Low will repeat a note even if it doesn't have seven different notes to work with. If you set the MODE to Non-Search, it will leave gaps when unable to find seven different notes.

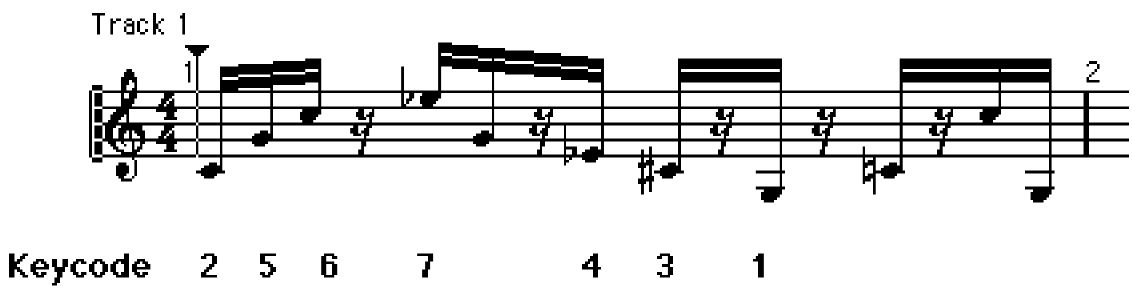
Now hit EDIT again.

You should see something similar to this:



The first column here represents the timing of the pattern.

The second column represents the KEY CODE. Now the key code is not the note itself, but is a representation of the position of the notes. For example, the key codes were assigned by the arpeggiator to our original sequence like this:



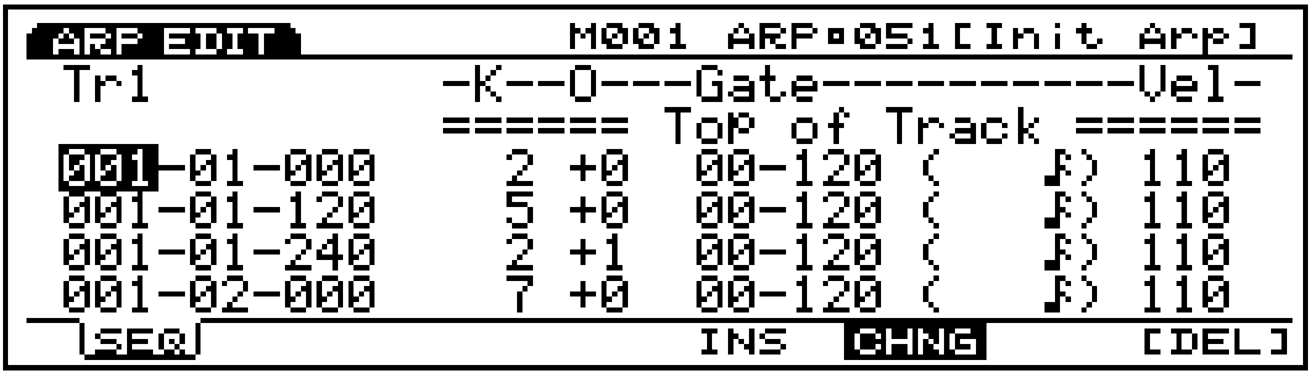
The lowest note is 1, the next lowest is 2, and so on.

When you hold down keys on the keyboard, it replaces these CODES with the lowest note you play, the second lowest you play, etc.

Got it?

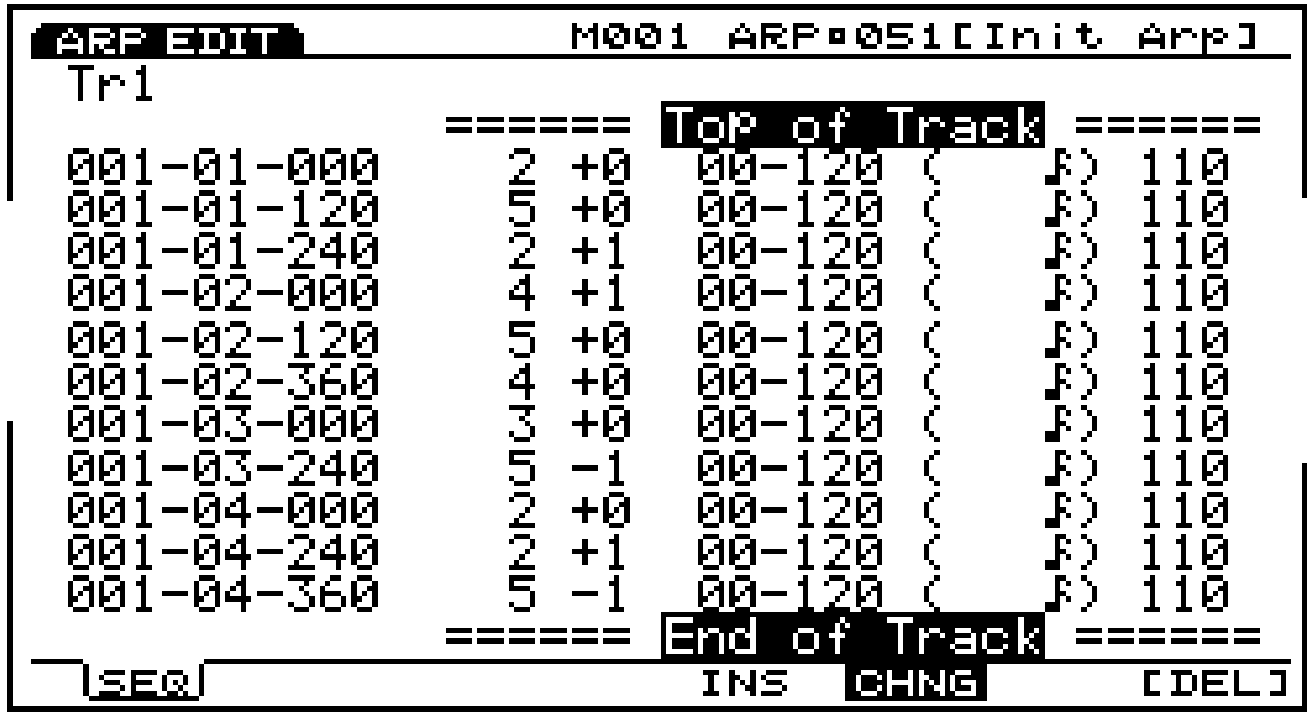
Now, you could theoretically leave the sequence as is, and use it. But, if we tweak it out further, it can be much more useful, or more musical. The main problem with it at the moment, is that if I put my fist down on the keyboard, the pattern that results doesn't have any of the "internal relationships" or vibe of the original. Why? Because the arpeggiator called the third note CODE 6, whereas my interpretation is that it's actually the first note up an octave.

So let's fix it. In the ARP EDIT window, let's go in and change the third event to CODE 2, but in the "O" column (octave column) let's put +1. Let's also do this for the tenth (second last) event.



Let's do the same for the 8th and 11th events. Let's call these CODE 5 (second note) but with an octave setting of -1. And while we're at it, we'll call the fourth event CODE 4, +1. Basically, I'm just removing all excessive key codes, and replacing them with octaves.

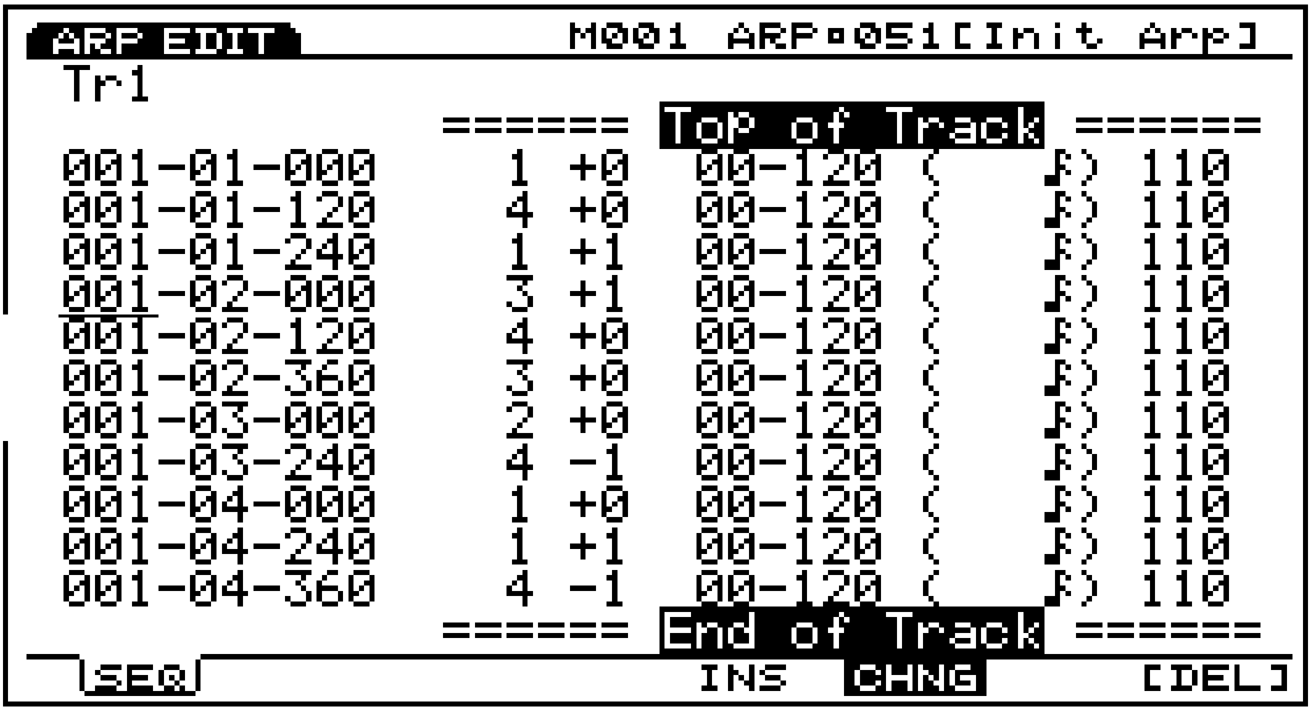
Now my entire pattern looks like this:



There are only four key codes remaining, 2,3,4, and 5.

So now, I only have to hold down four notes on the keyboard to supply enough information to get the arpeggiator to crank out something resembling my original idea.

And to tidy it up further, I'm going to rename all CODE 2 to CODE 1, rename the CODE 3 to CODE 2, rename CODE 4 to CODE 3, and rename CODE 5 to CODE 4. This just makes it easier to read, and understand what is going on when you hold down 4 notes on the keyboard. So finally, my pattern looks like this:



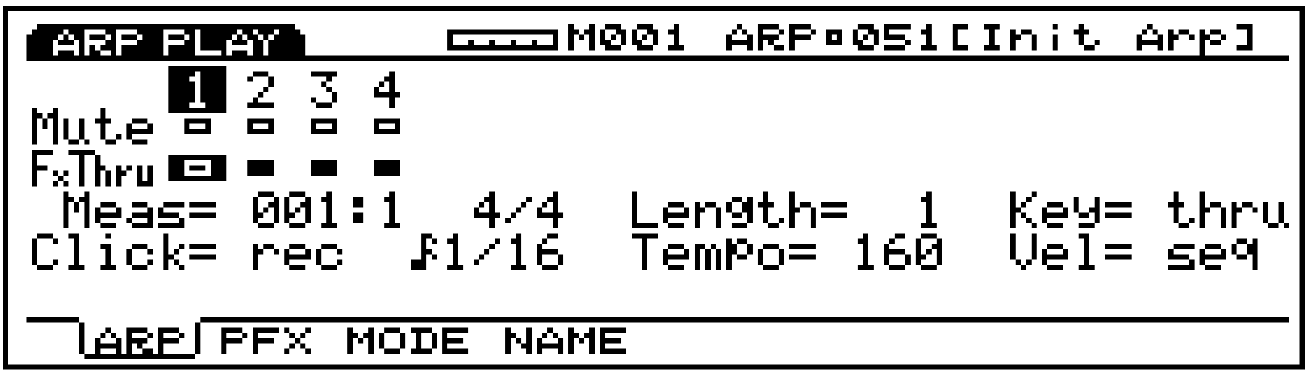
Of course, you don't have to begin with a MIDI file. If you know what you want you could enter it directly in this editor (using the INS function). You could import any data from a track or pattern on the internal sequencer. Or you could record the basic pattern in real time in the Arp Editor itself.

Exit this page to go back to the top of the Arpeggiator section.

You can name your arpeggio pattern in the NAME page.

And before we leave this, let's have a look at the Play Effects (PFX) page.

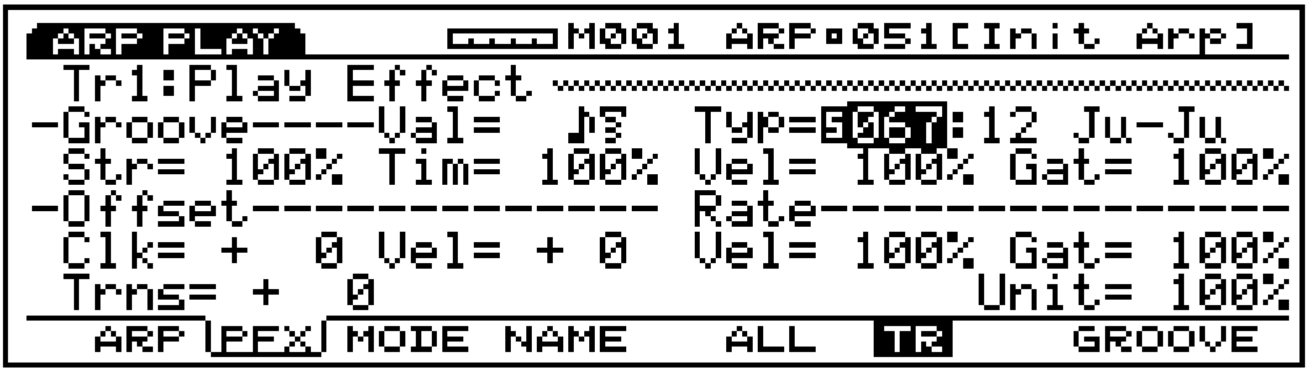
First of all, turn OFF the FxThru switch on the main ARP page:



These default to FxThru ON (which means bypassing the play effects).

Now enter the PFX page itself. Play Effects are offsets to the MIDI data, much like the inspector parameters in Cubase. You can offset velocity, note length, etc. and you can impose quantize grids (called Groove) on the data, without actually changing the data itself.

Hold down some keys, and start scrolling through the Groove Type list.



You should be able to hear how each groove is altering the feel of your arpeggio. The last groove is called 101:user, and you can design your own groove in this space. You can also edit the existing grooves if you want.

Below the groove stuff are some basic offsets, so you can alter the Velocity, Gate time, etc. of your pattern as well.

WARNING

Arpeggios disappear
when you power down.
Back 'em up.

Yes, this is a drag. Sorry, but it uses the same RAM area that the patterns and songs are in.

You can save the Arpeggios to disk.
And you can load them again, individually if necessary,
to build up your own library of patterns.

In the Utility/Voice Mode page you can set the Arpeggiator to HOLD or not, and you can choose a MIDI channel to transmit the Arpeggiator to another synthesiser.

8

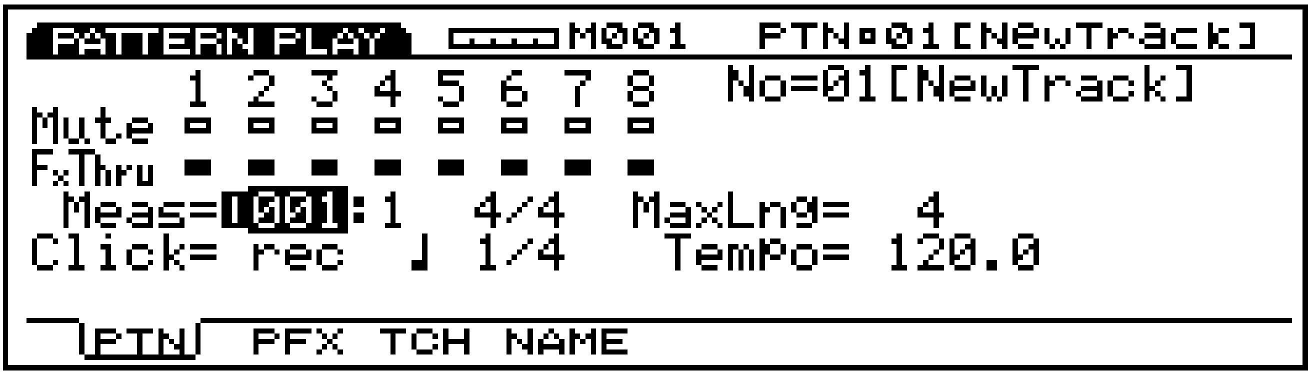
Patterns and Keymaps

Making a Pattern

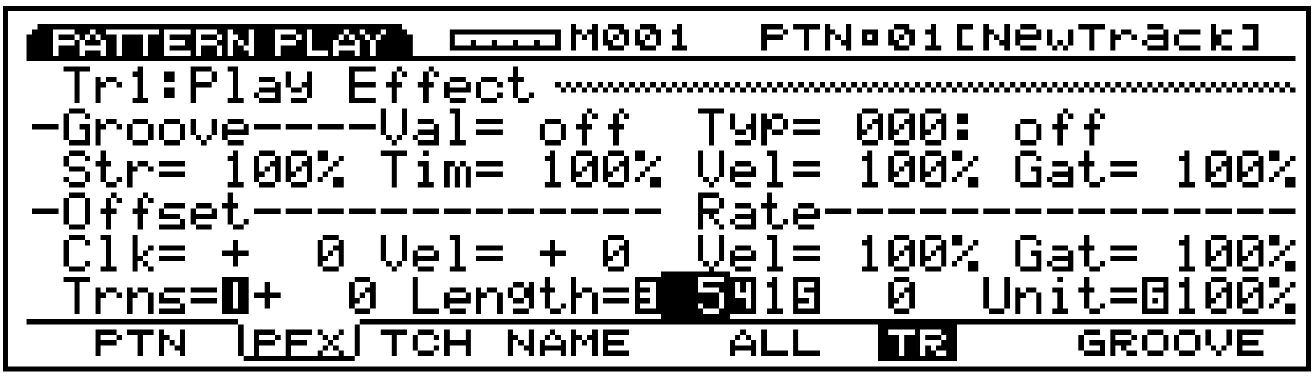
Patterns are actually used in various ways in the EX, so it's useful to learn how to make your own. Basically, you can think of the Pattern as a loop, much as if you'd loop a section in a sequencer. It can have 8 tracks, each of which can be on any MIDI channel and can have different loop lengths. A Pattern cannot be longer than 16 bars, though.

To demonstrate a Pattern that uses different Voices at the same time, let's choose a Performance to work within. Choose the one called "Pattern", Performance #004. This Performance has a Drum Kit on channel 1, then an Rhodes sound on 2, and upright bass on 3.

Hit the Pattern button.



Set a tempo you want to try working at, and then hit the PFX f-key. This takes you to the Performance Effects page, which we looked at briefly in the Arpeggiator chapter.

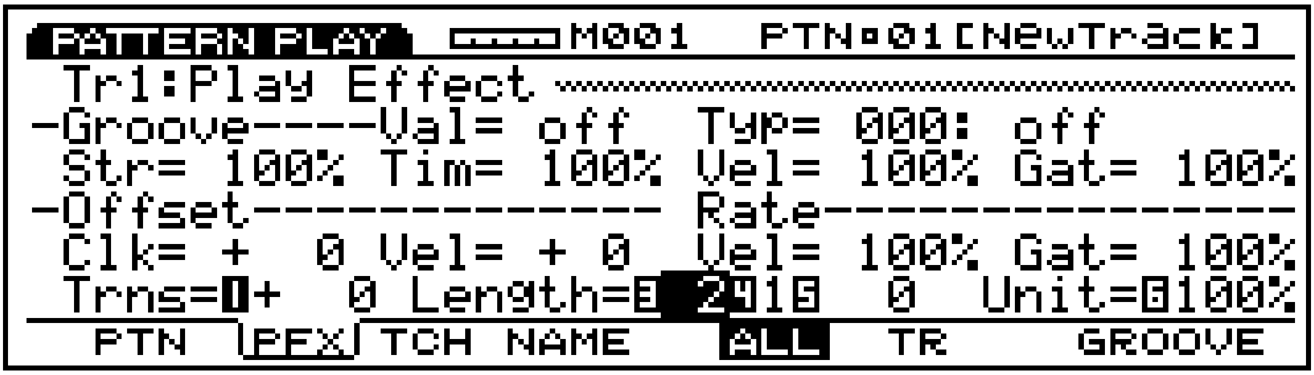


Again, with the patterns, you can offset many "performance" features of each track. You can introduce groove quantisations, transpositions, etc. But for now, all we're interested in is the "Length" parameter, since this sets the length of the track.

Notice that when changing the Length parameter, there are two options. These are shown by the f-key selection below, which can be ALL or TR. If it's set to ALL, then ALL the tracks in that pattern will be set to the Length you choose. If it's set to TR then only the selected Track will be affected. Same goes for the other parameters in the PFX page.

I'm going to set the entire pattern to 1 bar for the time being. You can always go in and lengthen another track afterwards. So I set the Length to "2". Why 2? Well, because actually you're setting the Loop position, not the Length. So I want it to Loop at Bar 2, Beat 1. You can actually Loop at a beat within a bar if you like. (The Beat is displayed after the Bar in the Length parameter).

This setting sets all tracks to Loop after 1 bar:

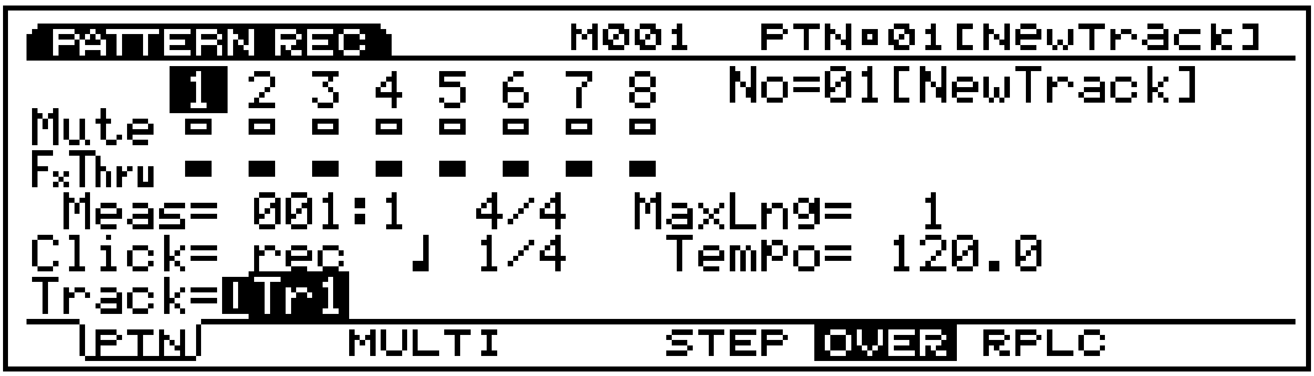


So now, I go back to the PTN page, and I see that the MaxLng display says 1. This displays the length of the longest track in the pattern.

Using the Sequencer transport controls, hit Record then Play. The metronome should be ticking out it's beat, and you can play something into the pattern. It's just looping the one bar around. Play a simple Kick drum or something into the pattern. Hit Stop, and then hit Play to make sure you've got it.

Notice that when you hit the Record button, there are a choice of Record modes, which are equivalent to similar choices you have in your software sequencer.

Here are your options.



Multi This can be on or off. If Multi is on, then all tracks are recording at once. This can be useful if you have several MIDI channels coming into the EX from another sequencer, or from live MIDI instruments.
When it's OFF, you can choose which track to record on the bottom left of the screen.

RPLC Replace mode. Whatever you record will replace the entire contents of that track.

OVER Overdub mode.
Whatever you record will be added to the existing data within that track.

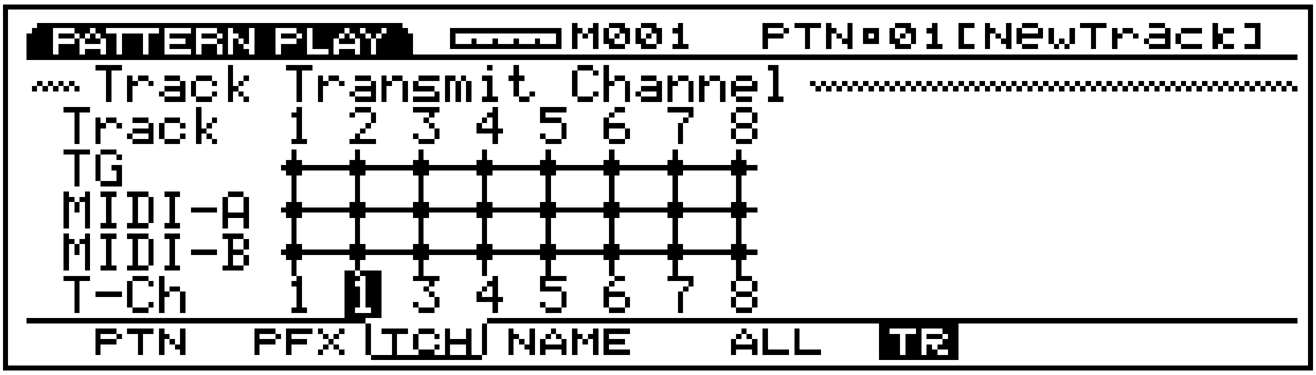
STEP Step entry. Sometimes this is the quickest way to get what you want, just choose the value of each note and hit it. In it goes. Notice two things in the step entry:
1- the note values can be quickly chosen with the ten-key pad
2 - if you set the velocity to EXT it uses the velocity at which you enter it

So we'll record a kick drum, or something simple into Track 1. Make sure MULTI is OFF, and put yourself in RPLC mode. Hit Record, Play. Play a couple of beats. Stop, play it back, make sure you've got it.

Second track. Choose Track 2 with the Program buttons on the right end of the front panel.

It's a Rhodes piano sound. Hang on, we want another drum track, but this Performance has a Rhodes on channel 2. No problem, we change the channel of track 2 to 1.

Hit TCH page:



Change the Transmit Channel of track 2 to 1 (as I've done in the screen above).

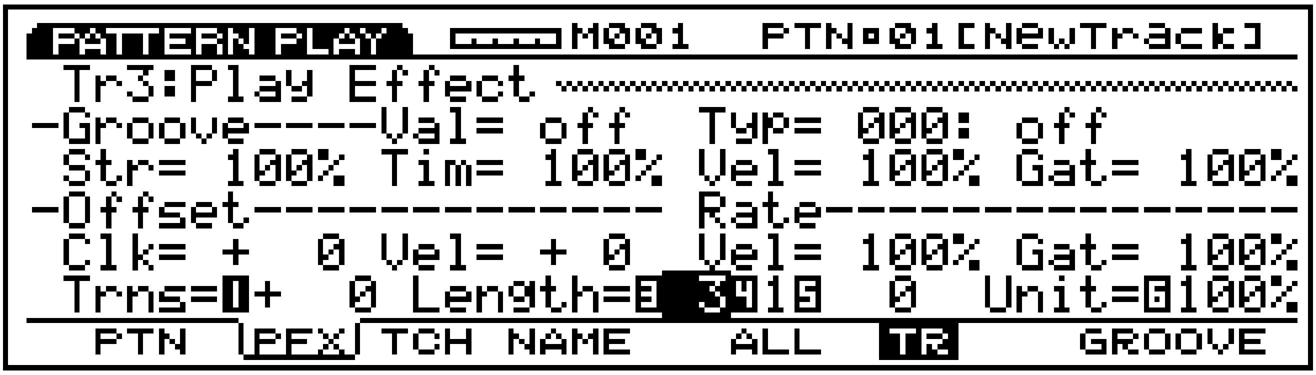
Notice you can also choose which MIDI output to transmit on, and you can disable the internal tone generator (TG) if you want to control only other instruments with the pattern.

Now go back to the main PTN page, and Record another drum part into track 2.

Some kind of rhythm happening? Alright, let's add a bassline on track 3.

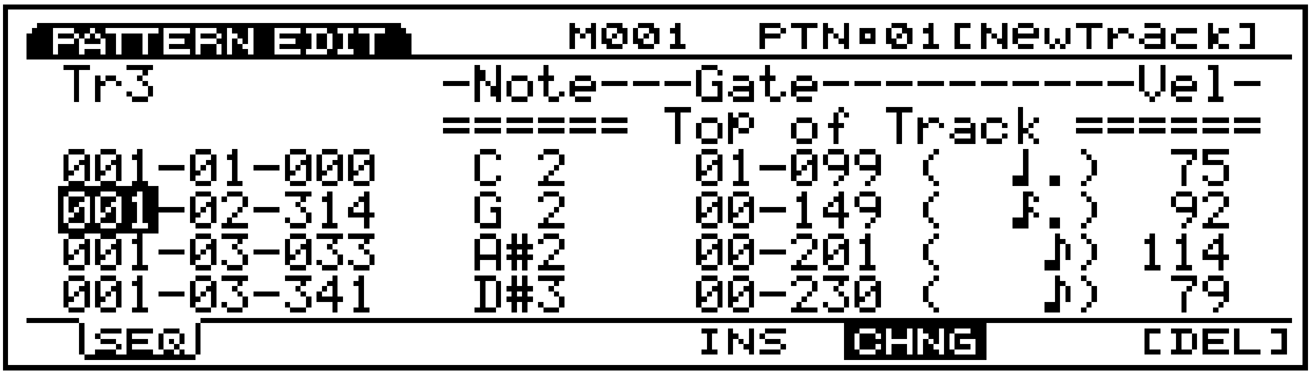
I want to make my bassline two bars long, so now I choose Track 3, and hit the PFX page.

I choose TR, so that what I do only happens to the selected track, and change the Length parameter to 3.1 (2 bars). Like this:



Go back to to the PTN mode and record it in real time.

Damn! I played it perfect, except for one note.
No problem, I just hit EDIT and fix that note.



In EDIT you can see the start times of the events in the left column, written as bars, beats, and ticks (at 480 ticks per quarter note). The next column is the note, then the duration, and finally the velocity. You may see controllers and so on in here also, particularly CAT, which is Yamaha's way of saying Aftertouch. (It's actually Channel AfterTouch, as opposed to Polyphonic AfterTouch). If you see too much CAT in your Patterns, you could always disable the Aftertouch in Utility/SEQ.)

More on this EDIT of MIDI data in the following "Sequencer" section.

For now, let's add some more tracks, and name the pattern something to differentiate it from all the other "New Track"s in there.

But let's see how we can use this pattern.

WARNING

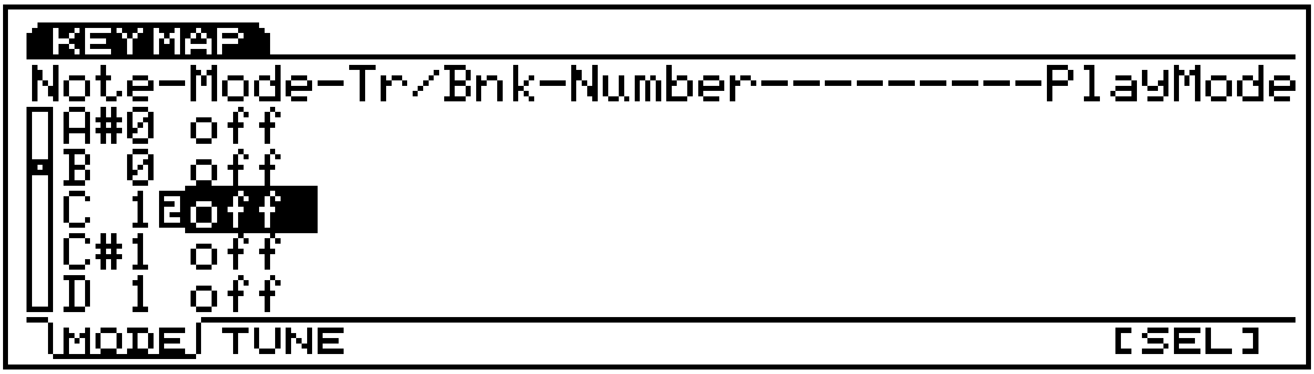
Patterns disappear when you power down.
If you make a good one, save it to disk.

Making a Keymap

Let’s imagine you’ve got a whole load of patterns, a bunch of samples, etc. and you want to jam them out and try various combinations of them. Wouldn’t it be cool if you could have each key on the keyboard trigger a different one of these? Well that’s what the Keymap mode is for.

There is only one KeyMap at a time in the EX. It can be turned ON from either Voice or Performance mode, and this Performance or Voice remains active UNDERNEATH the KeyMap. In the case of our Pattern, we need the Performance called “Pattern” to be active for it to make sense. So let’s stay with “Pattern” and turn on KeyMap (hardware button called “KEYMAP”).

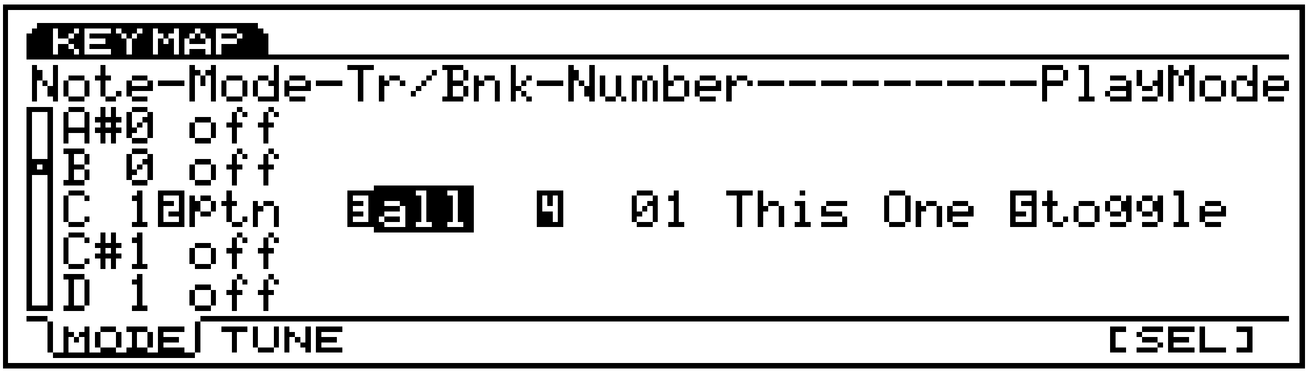
The first thing you might see after turning it on is a screen like this:



If you don’t, hit it again twice. The KEYMAP button is a toggle switch.

Choose C1 either by scrolling or using the SEL key while hitting the MIDI note.

Where it currently says “off” you can choose either a Pattern (ptn) or a Sample (smpl). Choose a pattern first of all. To the right, you’ll see a choice of all the Patterns in memory. In between you’ll also see a column that describes either a Track number or “all”. This is so you can choose just one track from a pattern. Set it to “all “ for the time being:



Choose the Pattern that we made in the previous section, and hit C1.

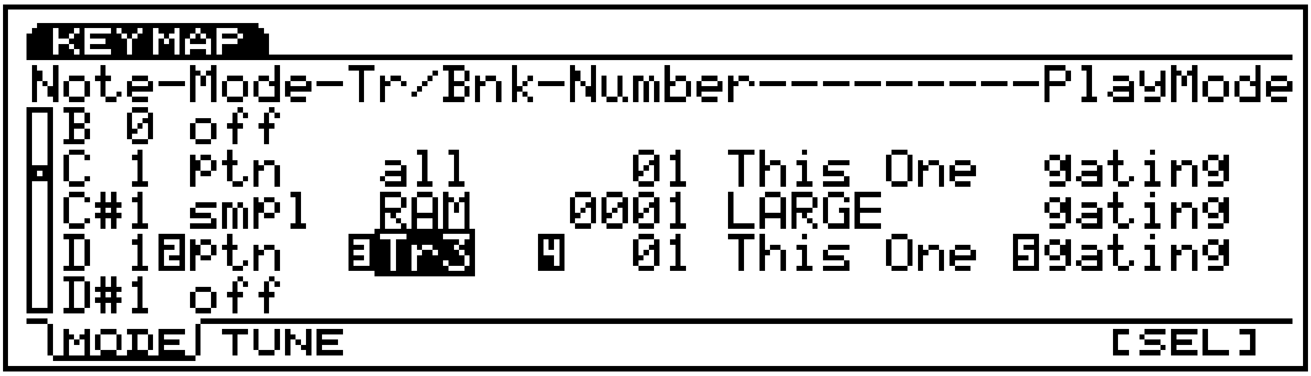
The Pattern starts to play. Hit it again to stop it.

Cursor over to the far right where it says “toggle”. There are three modes of triggering for each Key in the KeyMap. “Toggle” means that you hit the key once to start and again to stop. “Oneshot” means that it will play the entire pattern once, without any looping. “Gating” means that it triggers much like a normal MIDI note, ie. it plays while you hold it down, and stops when you release it. This is my preferred mode, but it’s up to you what’s appropriate for your pattern or jamming style.

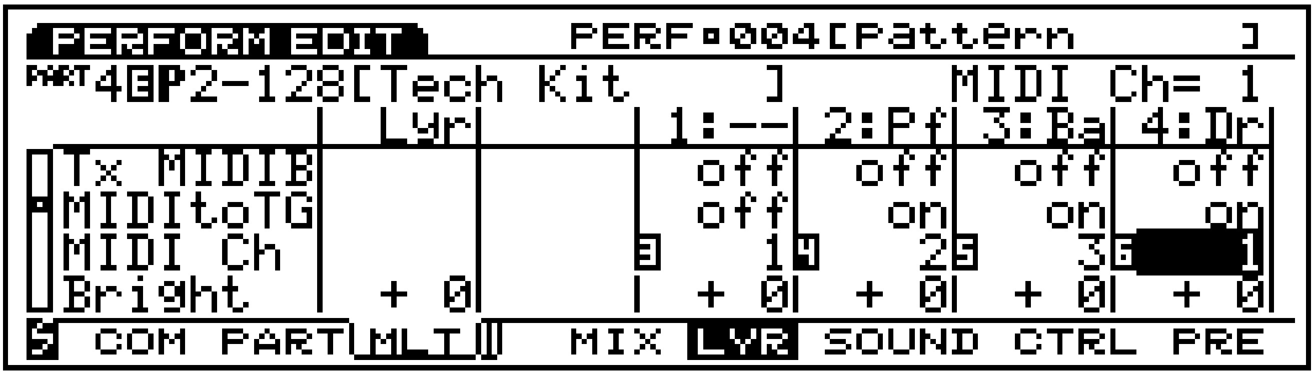
Do you still have the LARGE sample in memory? If not, load it from the floppy.

Choose C#1 and set it to “sample”, you can choose the sample and the type of triggering for the sample also. Additionally, with samples, you can tune them using the TUNE f-key to access the coarse and fine tune parameters. TUNE has no function for Patterns.

Let’s make one more key, on D1. Let’s set this one to pattern, but just trigger Track 3 of our Pattern. This is where I put the bassline in the previous section. And our KeyMap now looks like this:

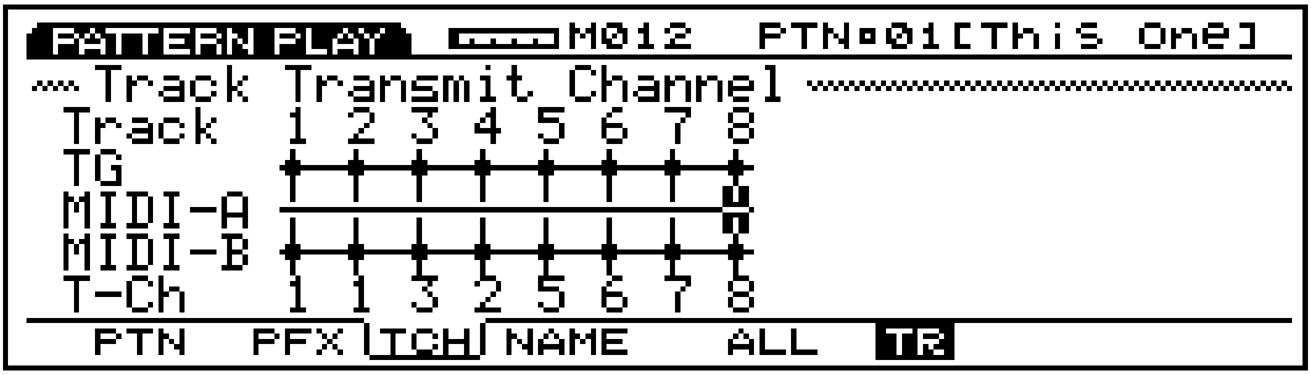


Now, there’s only one problem with this KeyMap so far, and that’s the fact that you’re also triggering the Performance Part 1 (the Tech Kit on Channel 1). There’s no easy way to avoid this, so I suggest choosing the Tech Kit on another Part (let’s use Part 4 for now) and setting that Part to receive MIDI Channel 1.



Then choose a Silent Voice on Part 1, or else turn the volume down.

You may also find that if you’re routing through a sequencer, all the Pattern tracks are triggering the sound on Channel 1 (or wherever your sequencer is re-routing all incoming MIDI). In this case, you should turn OFF the transmit to MIDI on the Pattern, or use a “multi” or “Any” channel track in your sequencer, and turn OFF the TG in the Pattern. For example, here is the Pattern set up so that it doesn’t transmit any MIDI out MIDI port A:



WARNING

Keymaps disappear when you power down.
If you make a nice one, save it to disk.

RECIPE TEN

About the JOBs

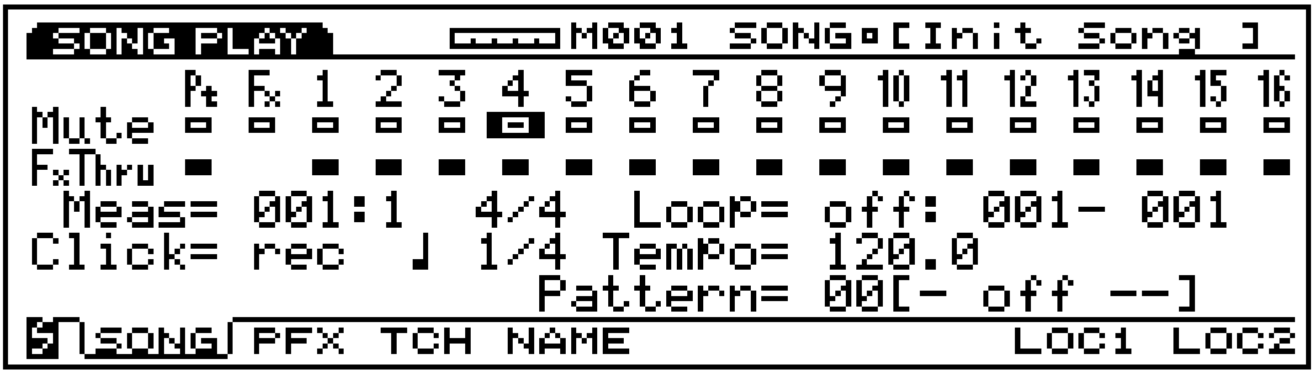
There are many functions in the JOB section with patterns. You can copy a track or a pattern, you can get a phrase from a MIDI file in the Song section and make it a Pattern, you can quantize tracks, etc. Check the manual at page 227 for a full list of JOBs.

9

The Sequencer

If you have a Performance set up for a song, or just want to experiment with our "Pattern" Performance, choose the Performance and hit SONG. (You can also use SONG from Voice mode, but you will have only one Voice at a time in Voice Mode).

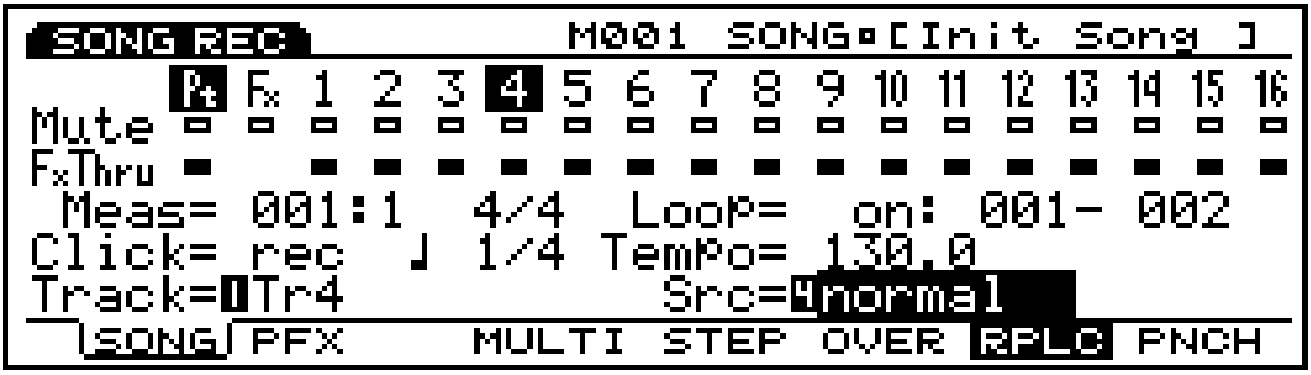
Here you can see the 16 tracks of the sequencer, plus an Fx and a Pt track. The MIDI tracks 1 through 16 can be chosen quickly with the Program/Part/Track buttons.



As with the Pattern mode, each track has "Play Effects" (PFX) and a transmit option (TCH). Similarly, each Play Effect can be assigned to ALL or TR (individual track). You can also record Play Effect changes to happen at various times through the song. This is done on the "Fx" track.

You can also include Patterns in the song by using the "Pt" track. These can be recorded real time, by entering the Pattern number while recording, or else edited in step time fashion.

Choose a MIDI track first of all, hit Record.

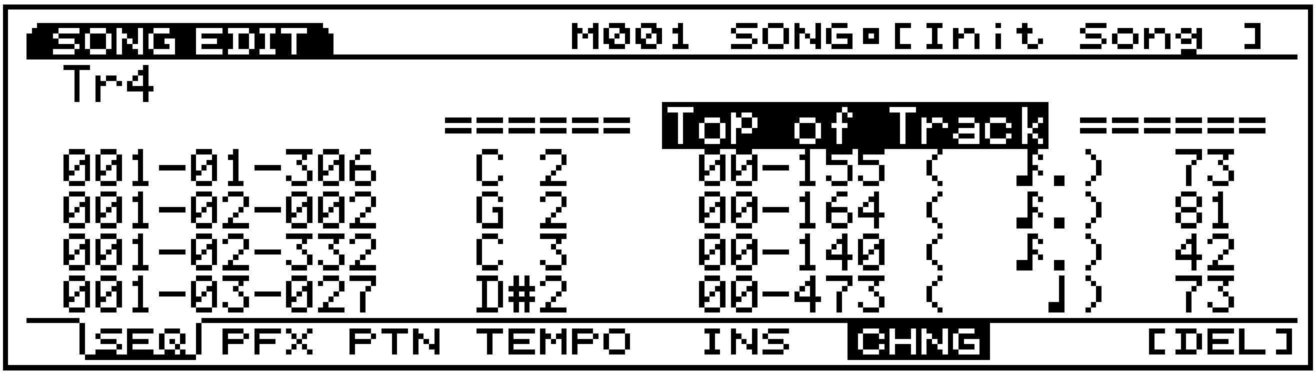


Again, like Pattern, you have a choice of recording MULTI or not, as well as a choice of recording modes. There is one new mode here, not found in Pattern, called PNCH. Punch recording let's you define an in and an out time for recording. This is useful if you just want to record for a short passage within an otherwise perfect take.

The "Src" parameter you see highlighted above can be switched between "normal" and "arpeggio". If it's in "arpeggio" then the output of the arpeggiator is captured as MIDI data, thus freeing the arpeggiator for use elsewhere.

So let's hit record and play any old stuff into a MIDI track. Done? Does it play back?

Okay, hit EDIT.



This is the display of note data, hopefully the notes you've just played.

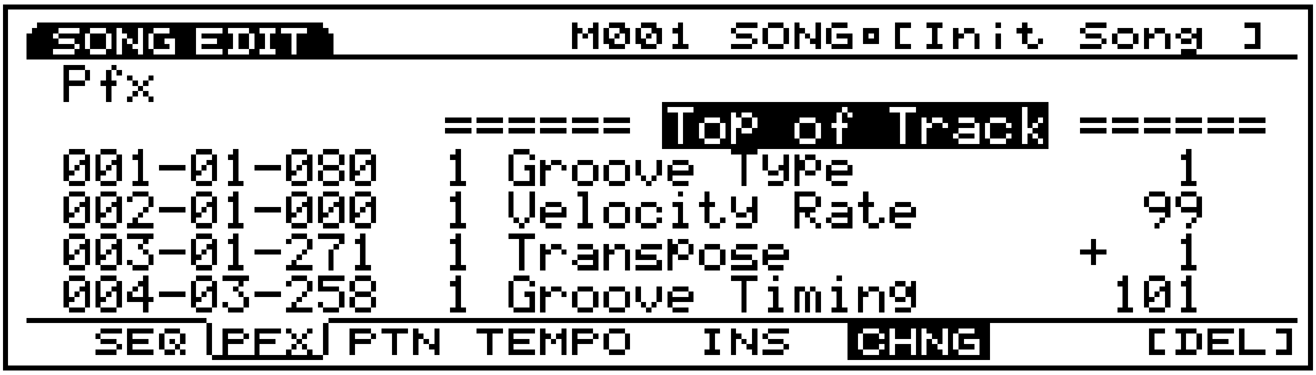
There are two modes to this editor, INS (insert) and CHNG (change). In Change mode, as above, you can alter any of the fields with the data wheel or the ten-key pad. The first field is the time of the event, in bars/beats/ticks (at 480 ticks per quarter note). The next field is the pitch of the note. The third field is the length of the note, again in beats/ticks. (The note-like display to the right of this tries to display the closest musical equivalent of the duration, by the way.) Finally, at the last column, we have the velocity.

If you hit INS, you can insert a note or any other MIDI event. Like this:

Let's have a look at the other types of tracks.

The Tempo track let's you define tempos at any position in the song, so that you can change speed when necessary. It's usually best to do this with INS, since you can't record this real time anyway. If you've imported a MIDI file with tempo changes in it, you can edit them with CHNG.

The Play Effect track (Fx) is a strange one. Here you can record real-time. Try it out. You can adjust the Groove template, change velocity offsets, etc. through the Song. Record some of this and then hit EDIT.



Mad, innit?

Now, no-one's suggesting that this is the way you would like to write music, and it's probably not your preferred way to edit your MIDI data. However, this sequencer can be useful for several things. For example:

- Import a MIDI file to see what the Play Effects can do to your grooves
  There are many Groove templates in here, and you can apply them in varying amounts to the timing, the velocity, etc.

- Use the Sequencer to play sequences live.
  Okay, there's only one Song, but it can hold up to 30,000 MIDI events.
  So you could have an entire set in there, just chained together as one song.

- Use the Sequencer to capture improvised ideas that occur to you when the computer is off.

- Use the Sequencer to capture the Arpeggiator output, copy it to a Pattern, and free up the arpeggiator for something else

- The only way to get your MIDI files into Patterns or Arpeggios is VIA the Sequencer

- Impress your Mum with a MIDI file of Grieg's Piano Concerto while you pretend to play

So you see, it's not a complete waste of space.

WARNING

Sequences disappear when you power down.
If you make a good one, save it to disk.

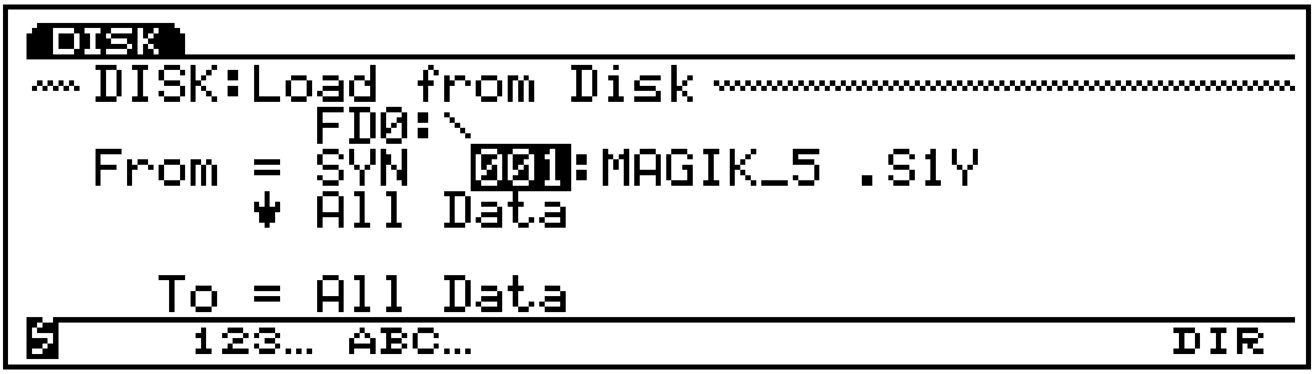
10

Data Management and Utilities

How do I Save/Load one Voice?

You cannot Save one voice. You can Load one voice from either a Voice file or a Synth ALL file. (see next section)

So, if you want to build your own library of Voices, you load individual voices from disks with Synth ALL or Voice files on them. Let's say you want to load a single voice from the floppy that came with this guide. Go to DISK/File Load/Synth ALL



Where it says "All Data", you can change this to decide exactly what you want to load. You can also choose the destination if you are loading a single object. Like this:

This is what we call heirarchical loading.

With this technique, you can fill the user memory of your EX with your preferred voices.

For more detail on the various saved files, check the next section.

What is Saved With What?

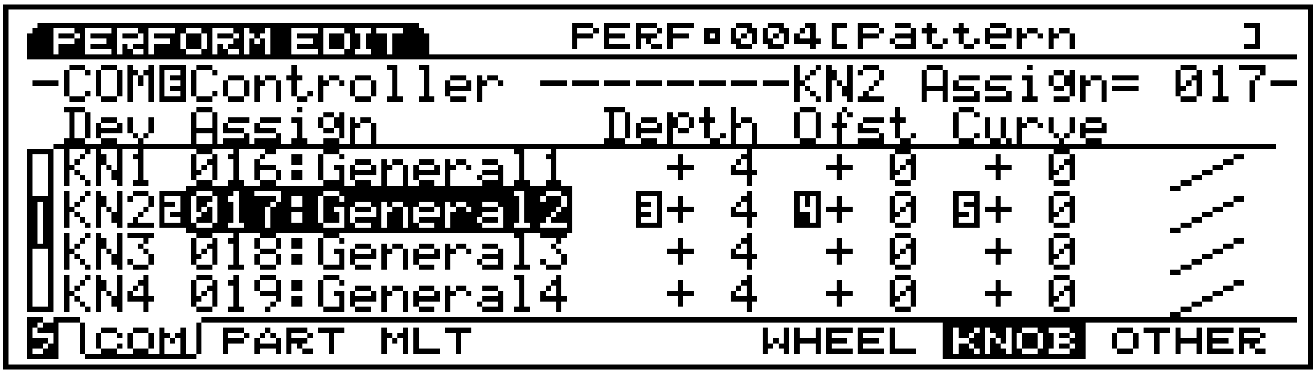
There are many different file types for use with the EX. It's important to know what is saved, and how you can load it again. Here are the Save types, along with a description of what can be loaded from them.

ALL Data	This saves everything; samples, sequences, patterns, etc. If you save this you are sure to be able to get it all back again. The only drawback with Save All is that it's non-hierarchical when loading. In other words, when you have an ALL file on disk, you can only load ALL of it again.
Synth ALL	Synth ALL saves all the voices and performances. Not waves, not patterns, not arpeggiators, not sequences, not keymap. Synth ALL DOES save your system setup, however. Synth ALL is hierarchical, so you can load one single voice or performance from a Synth ALL file on disk.
Voice	This saves only the Voices, not the performances or system. Voice is hierarchical when loading.
Wave	Wave saves all the samples and waves created that use those samples. Wave is hierarchical when loading. You can load a single wave from a Wave file on disk.
SMF	Standard MIDI File. This saves the Song to disk as a MIDI file type 0. Play effects are not included in a MIDI file, obviously. Note: You can play MIDI file type 0 directly from disk if you like.
Song	This saves the current Song in Yamaha's own Song format. All aspects of the Song are retained (play effects, etc.).
PTN	Pattern save. This saves all the patterns in memory to a single file. Loading of patterns is heirarchical, you can load just one.
ARP	Arpeggio save. This saves all the current arpeggio User patterns to disk. Loading of arpeggios is heirachical, you can load just one.

How do I change the MIDI controllers on the Knobs?

The MIDI controllers are assigned to the Knobs at the factory as CC 16, 17, 18, 19, 20, 21.

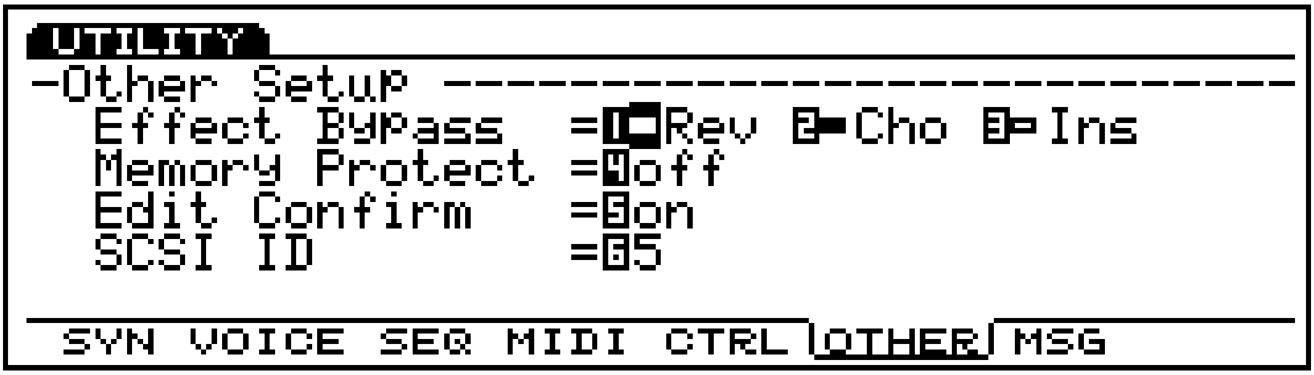
If you want to change this, you can do it for each Performance. EDIT/COM/CTRL/KNOB



Why is it in Performance? Well, this could be really useful for setting up controls for an entire keyboard rig in a live situation. In one track you may want to control the EX itself, but in another track you might want the knobs to control another synth with a different controller set. Or, in the studio, you could make a Performance in the EX that will control your Bassstation for example, so you don't have to get out of your chair so much.

Effect Bypass

You can choose the effect of Effect Bypass in Utility / Other. You can choose which effects are affected by the Effect Bypass, effectively.



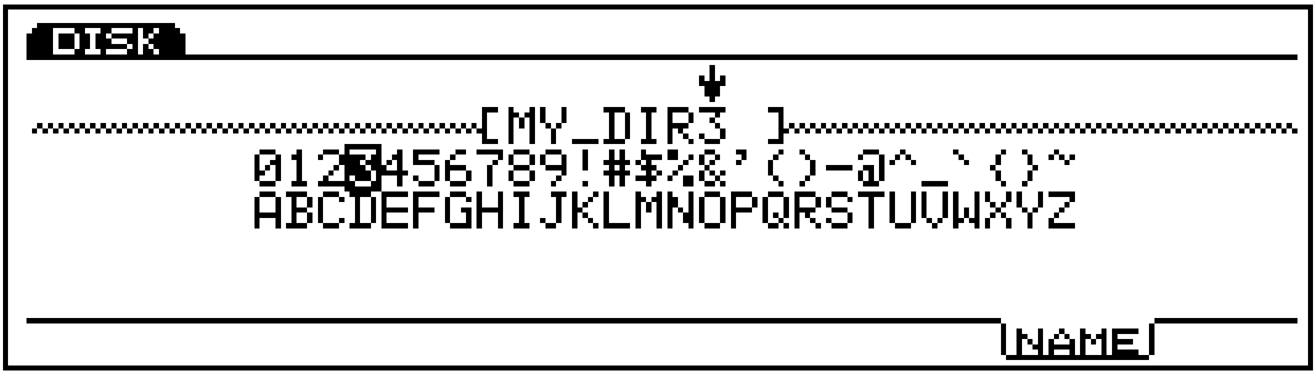
Working with SCSI Drives

To format a SCSI drive, whether it be removable type or a fixed drive, you'll probably have to boot the EX while the SCSI drive is ON and connected. If you turn on the drive after the EX has booted, there's a good chance it won't be able to find the drive.

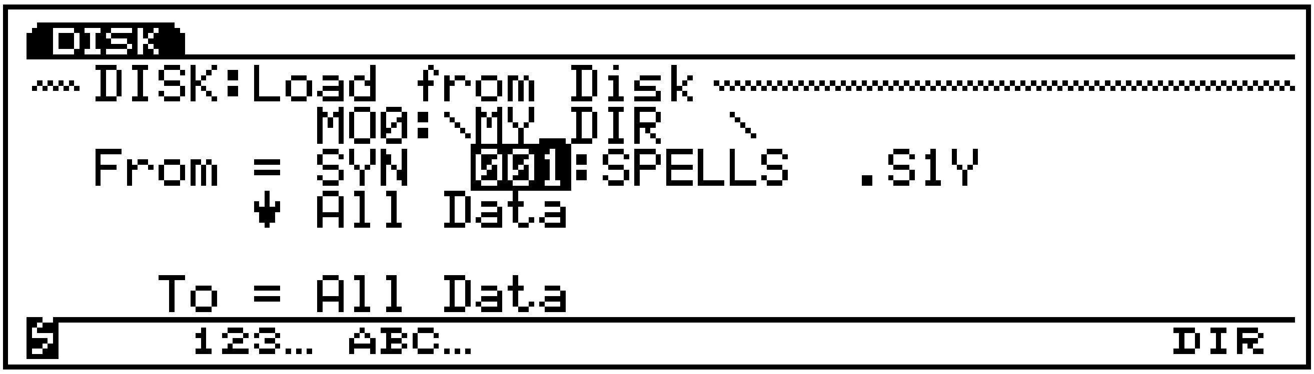
The next thing you'll need to do is format the drive. This is done in the disk page. Select the correct device first of all. Then you can format.

The thing to remember about larger drives is to use directories. Otherwise you may end up with so many files on the root directory that you can't find anything. There's a function for this called "Make Directory".

Try to give your directories names that mean something to you.



Once you've made a couple of these you can navigate during the saving and loading process.



SCSI drives tend to be much faster than floppies also.

And if you're working with samples, you'll need some space. I'd recommend using a SCSI drive of some sort.

Epilogue

Where do we go from here?

If you've followed through most of this book, you should have an idea of the huge potential of this instrument. Not only is the EX providing the raw elements of many synthesis techniques, it's also giving you some compositional tools to help generate ideas.

Once you understand the basic architecture of the machine, it's often useful to forget about the designed purpose of the various sections, and to strike out on your own. For example, the Electric Piano Pickup FDSP algorithm might sound great with a vocal sample going through it. Using the "resampling" feature, you can sample the results of one DSP, and then apply another to the result. You could import your drum grooves, turn them into arpeggio patterns and manipulate them with the Play Effects, transpose it realtime, etc. Put your breakbeats through the Seismic FDSP and turn them to mush. You could import each section of your track as a seperate pattern, and jam the tune live from a KeyMap. Or make a sixteen part keyboard split as a Performance, set them all to the same MIDI channel, turn the arpeggiator on and stand back.

A lot of interesting sounds are creating by merely trying stuff out, experimenting without a fixed purpose. Mistakes can be very cool.

Miles Davis said "There are no wrong notes."

Brian Eno said "Make the machinery fail."

In other words, go for it, break the rules, and make it squeal like a pig.

And now, my son, you must eat this book...