tEXtures

a tutorial for the

EX SERIES of Synths

by Joel Silver

t-EX-tures

Forward
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enjoys exploring the outer limits of all forms of music, and has
a deep fascination for anything that is blue with knobs on it.

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Chapter 1. Electronica/The EX frame of mind

Music is, was, and always has been, a relatively direct representation of the popular technologies of the time at which it is created. One of the first musical uses of the personal computer was to record note data, or sequences, as they were called. This, at the time, was quite innovative. As innovative, perhaps, as woodworkers getting excited about their new precision tools, and building intricate violins, or early humans, taking notice of the beauty of the sound that occurs when a piece of rock hits a hollowed-out log. It's all about representing a frame of mind through sound. This is as pertinent now, as it has ever been. Generally speaking, musical styles last as long as technological trends do, changing with current events. It took World War 2 to create rock'n'roll. It's taken a exponential technological growth to give us 'Electronica'.

'Electronica' is a radically broad term used for describing many, many styles of music that rely on technology as their main focus, each one being extremely different from the next. A lot of serious musicians resent the term for that very reason. It tends to put the focus on the machines rather than what's being done with them. It's much like using the term guitar music to describe Led Zeppelin and Liona Boyd. It's not untrue, but it doesn't paint the whole picture. Even worse than that, is the use of the blanket term 'techno' to describe electronic music. 'Techno' is a very specific genre, clearly defined by its own conventions and instruments. Calling 'Drum'n'Bass' music 'Techno' is as incorrect as calling Bach 'Classical', when he's obviously 'Baroque'. Each category of current electronic music has several sub-categories, each sub-category has sub-sub-categories, so clearly, the best thing to do is not concern yourself with genres, but rather experiment with your machines, and define a sound of your own.

For those of you who are new to the scene, there are a few things you might want to know. First of all, many musical conventions that you may hold near and dear are less important. Sometimes, conventional concepts of melody, harmony, and song structure can be your worst enemy. Not to say that there isn't a need for these elements, it's just that a bowel shaking bass-line can be more important than a tuned bass line, an interesting drone is often more effective than stressing about which chords are supposed to resolve to which, and if your concern is about having a 'hit song', it's probably more important to have an interesting vocal loop, or catchy synth riff, than it is to have a good chorus. It's not that these elements have no place in electronic music, it's just that conforming to these elements might prevent you from writing a unique piece of music. It's ALL about timbre, texture, and generally speaking, a good beat.

Enter the EX5/7/R. While some major music corporations have been focusing on building machines that conform to genres, the good folks at Yamaha have been hard at work, building new instruments that ultimately offer nothing but a million options, a ton of expressive controls, a never-ending palate of sounds, and mostly, intriguing new technologies to inspire us. The EX series, aside from having the required presets, have the ability to build deeply fascinating, ever evolving sounds, the likes of which have never been heard before. To maximize the power of your EX, the best thing to do is make as many of your sounds into samples as possible. If you can get into the frame of mind that the EX is a sample-creating-machine-from-the-gods, then you will be

allowing yourself to keep everything as effected as you want, and the sounds that can't be reduced to samples won't have to be compromised. I'd recommend having a bare minimum of 16mb of RAM, although 32mb is much better, and 64mb is fantastic. As soon as you hear something that you like, SAMPLE IT! Making totally unique sounds is the reason that you chose this machine in the first place, isn't it?! Also, if you own the EX7, and not the EX5/R, then you may notice that some of this won't make sense. The EX7 is basically identical, except that it only has a mono audio in (which won't limit you, as you can still LOAD and PLAY stereo samples) and NO VL generator. So, if you are an EX7 user, don't allow yourself to get frustrated by looking for VL voice set-up—it isn't there. Because of this, some of the sounds that are included on the disks may not sound as they should.

A lot of this tutorial assumes that you have a basic grasp of terms and concepts. If you haven't done so already, you should really read The Joy of EX, available from the Yamaha UK web-site, in PDF form. While we're here, I would also like to make the suggestion that you leave your output volume at the halfway point (the 12 o'clock position). Because of the nature of this machine, you'll want to have lots of headroom and foot room. This tutorial conveys actions like this: >> means push these buttons to navigate through your menu to ..., and [......] tells you to perform the specified action. So, for example, to personalize your opening message, do the following:

 UTILITY>>F7(msg)>>[use keypad to enter message].

To select your voice LFO wave, you would do this:

 VOICE>>EDIT>>F6(LFO)>>WAVE[select wave].

By the end of this tutorial, you should have a firm grasp of '3 level' editing. That is, the act of taking a sound, manipulating it, resampling it, manipulating it, and resampling it again. You can repeat this as many times as you want, but I call it '3 Level' editing because no matter what your sources are, 3 tweaks will make any sound unique. Also, keep in mind that these lessons are designed to navigate you through the EX's brain, hopefully finding aspects and techniques that you haven't considered, and in no way should you get the impression that this is the only way to do things. These are only very basic steps and guidelines, and I urge you to personalize at every level, and go off on your own tangents as soon as something interests you. Right then...we've got our AWM's , we got some VL's , a good supply of AN's , a sampler, some frightening filters, some mind-boggling effects...hold the calls, lock the doors, and get your supplies ready...we're going in.

[Insert Disk#1]>> DISK>> F2(LOAD)>> F2(SYN)>> ENTER>> INC/YES

THIS WILL ERASE EVERYTHING—I MEAN EVERYTHING—ON YOUR EX
Back-up any important data.

Chapter 2. The Almighty Breakbeat

Although there is much debate as to the origins of this important aspect of electronica, my sources seem to lead me to these conclusions: The breakbeat originally dates back to the 1960's, when r'n'b and soul were beginning to pave way to a new sound called funk. Funk tended to use aspects of jazz, but in a more rhythmically accessible way. Although soloing was still a part of funk, the soloists were more likely to pick a few cool riffs, and repeat them in a variety of rhythmic offsets. Instead of the stripped-down, all-over-the-place drum solos that jazz offered, r'n'b, soul, and funk tended to offer a drum 'break', where the other instruments (except the bass, perhaps) would drop out, and the drummer would offer a more layered and complex rhythm, perhaps adding a funked up snare line, while the dude that plays trumpet grabs a cowbell and grooves out, and James Brown shouts out "Whoa! Yeah!"

In the eighties, 2 very important things took place...#1 sampling technology became more affordable, and #2, Rap music pioneers hopped on it before anyone else did. Early Rap developed in geographical areas where musical instruments cost more than most people could afford, so jams were usually backed by DJ's who would select their favourite breakbeats, and rapidly switch between them, effectively inventing the drum-loop. If you have ever seen early footage of GrandMasterFlash, or Run DMC, then you've seen this technique...a true art-form, but an extremely frantic one. Once samplers became available, it was only natural that rap artists were the first to see the funk-potential. The most common breakbeats of the 80's: Anything by James Brown, especially the aptly named Funky Drummer, ABC by the Jackson 5, When the Levee Breaks by Led Zepplin, and a hand-full of others that kept popping up.

The next key step in the development of the breakbeat was when American rap music began to merge with UK techno and house, in a more upbeat, electronic manner. Early breakbeat electronica kept the rhythms the same, but sped them up. Eventually, editing samples became easier and more accurate, and people started to displace rhythms and offset the beats. By 1992, it was common to rearrange the sped up breakbeats into almost island-style rhythms, add some dub-style vocals and bass...and Jungle was born. Jungle and its offsprings (Drum'n'Bass, Hardstep, etc.) rely heavily on the process of sample, tweak, resample, tweak, resample. Right about now, you should pat yourself on the back for buying an EX...

***Load BEATZ.S1M off of disk Disk #1.

[Insert Disk#1]>> DISK>> F2(LOAD)>> F4(WAVE)>> ENTER>> INC/YES

First of all, you will have to find a suitable breakbeat. For Drum'n'Bass, you'll want something that can be sped up to at least 165 BPM without losing its effectiveness, while trip-hop may require a beat that can be slowed down to an opiated 65 BPM. For the sake of convenience, this part of the tutorial will be dealing with Drum'n'Bass. If you can tackle Drum'n'Bass editing, you can pretty much take on the world. Goods sources for beats are pretty easy to find. In almost every city there are stores that sell used records for 25 cents to $1.00. Sample CDs are available at any

music store, and Yamaha has been very good about supplying EX users with samples-if you haven't already, go to your local music store and ask for the EX series CDs. In most cases, you will be manipulating beats beyond recognition, so clearance isn't going to be a problem. I'm going to start with a loop # 65 from the Yamaha EX Sound&Utility CD, 'LA Groove' by Neil Conti. You may use whatever loop you like.

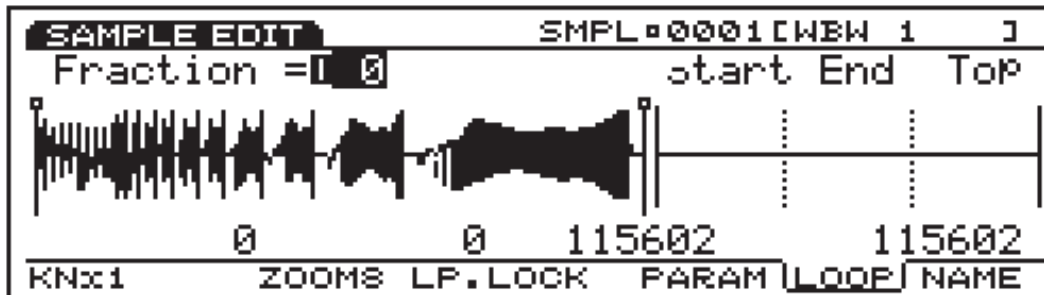Start by sampling the beat of choice:

SAMPLE>>F8(REC)>>[Set Rec Mode, Source, Input, Trigger][Make sure you have a high level on your monitor, but not so high that it clips]>> F8(STANDBY)>> F8(START)>>[let sample play out]>>F8(STOP).

In case you didn't know, Rec Mode allows you to record the left input (L), right input (R), the left and right inputs mixed to one (L+R), or the left and right inputs as a stereo set (stereo). Source tells the EX to get its sample either from the inputs or internally. If you choose 'A/D', which refers to the external inputs, you can also define the gain on the input as line or mic. Because a microphone has a much lower output than a powered source, if you select mic, the input level will be boosted significantly. 'Trigger', as the name implies, is how the EX will know when to start sampling. If this is set to level, then you must set a threshold in the space next to the word level. As you change this value, you'll notice that the little graph beside it will change. This, of course, is to be compared to the graph above, which shows input level. The first time the input level is any higher than the threshold, the EX will start sampling. In most cases, it is better to trigger the sampler manually. This is because you risk losing a few frames of the sample, due to reaction times and subtle settings.

After you have recorded your sample, press the exit button located beside the F8 button. This tells the EX the you are happy with the sample, and you want it to proceed. It is important that you do this, even if you are recording a stream of samples. If you don't, then the EX wont remember what it has just sampled. Now we want to get rid of any sample information that is unimportant. To do this, go in to your Sample edit loop page:

SAMPLE>>[select sample]>>EDIT>>LOOP.

 In most cases, you'll want to make your sample as tight and as accurate as possible, right away. The display on the EX should now be showing you the typical amplitude wave data read out. Hit the cursor down button, so that your knob movements will now edit locations. F1 allows you to define your knob movements as increments of 1, 10, 100, 1000, or 10000. Using a high setting like x1000 or x10000, get your start and finish point to the approximate area. This is an easy step...use your ears and eyes. Once you have found gotten close, use the F8 button to lower the edit increment to 10 or even 1. To make your sample flawless, you need to find the zero-crossing. The zero-crossing is the point where your sample level hits the zero point (displayed as a straight line that goes horizontally through the centre of the page). Don't get lazy and forget to name your samples.

```
SAMPLE EDIT          SMPL∎0001[WBW 1    ]
Fraction =[ 0              start End   Top


         0          0    115602        115602
KNx1    ZOOM8 LP.LOCK   PARAM [LOOP] NAME
```

Not naming your samples will lead you to much, much grief. It can also be in your best interest to develop some sort of language that you use with yourself that is consistent, so that you can accurately describe your samples to yourself. I've named mine BB Raw, because I always use 'BB' to mean 'breakbeat', and I always use 'Raw' to mean 'as-of-yet-unaffected'. Obviously.

If you put your EX to I1-001 Raw Loop, you'll notice that I have it set up as an FDSP voice. I have done this to take full advantage of the effects processing that the EX offers. At the most basic level, using an FDSP voice gives you the addition of an extra insert/variation style effect. Initializing a patch as an FDSP voice requires two steps:

Then hit EXIT a few times until the display shows you an 'Init Voice' (Initialized voice). The EX automatically defaults to an AWM voice, so, step two is switching it to an FDSP voice:

If you examine this page in (I1-001 Raw Loop), you may be confused as to why the sample that I made before (BB Raw), is now called 'Raw Loop'. This has to do with another one of those powerful-yet-confusing EX features. Instead of simply asking you to put samples into your voices, the EX asks you to put waves in. A wave can, and actually does in this example, consist of just one sample. This can confuse those of us who have previous sampler experience, because it seems to add an unnecessary programming step. But again, there is much power in wave programming—you can have up to 128 samples in just 1 Element of your Voice!

To program a wave, follow the above steps to get to the OSC page, then hit F3(WAVE EDIT). From here, you can create some pretty complex-sounding tricks...but more about that later...

Drum'n'Bass beats rely on sonic minimalism, with rhythmic maximilism (!?), so I've taken

advantage of the extra effect that we have earned by using an FDSP voice and, inserted the 'EG Pick-up' algorithm, removing a lot of the fullness, and to giving the beat that oh-so-popular lo-fi feeling. To make the FDSP affect your sample:

VOICE>> EDIT>> F1(COM)>> F5(FDSP)

Make sure that the Element Switch (ElmSw) that pertains to your wave is turned to 'ON'.

```
╔══════════════════════════════════════════════╗
║ ▐VOICE EDIT▌          I1-001[Raw Loop     ]   ║
║ -COM▪FDSP --┌─FDSP───(02:EG Pickup      >     ║
║  ElmSw│Type │1│Pickup Type =▐humbuck▌         ║
║ ▐1▌: on │ 02 │ │Coarse      =▐4▌  + 0          ║
║ -:---      │ │Picking Pos =▐6▌   36           ║
║ -:---      │ │Picking P Kf=▐6▌  +16           ║
║ -:---      └─┘                                 ║
║ ▐5▌>OSC >              ▐FDSP▌PARAM ARP NAME   ║
╚══════════════════════════════════════════════╝
```

It took me no time at all to edit this FDSP effect to sound the way I liked, and, to be totally honest, I have no clue what many of the effect parameters mean. Remember, with EX, there are so many possibilities, that it can be hard to know when to stop...the answer is: stop the second your hear something you like. If you wish to tweak further, then store the voice by hitting Store (duh....), then ENTER followed by INC/YES. Store a second time, but store it to a different location.

```
╔══════════════════════════════════════════════╗
║ ▐STORE▌                                        ║
║ ⌐⌐ₒTO E:Voice ⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐⌐   ║
║                                                ║
║          I1-001(A01)[ aw Loop    ]            ║
║                       ↓                        ║
║   ₒet&Press ▐I▌1▐B▌▐001▌(A07)[Init Voice  ]   ║
║      [ENTE ]                                   ║
║                                      DIR       ║
╚══════════════════════════════════════════════╝
```

This will erase any data that was previously in those locations.
I chose to keep fiddling with the sound...it's just so fun. Next stop, more effects!!

VOICE>> EDIT>> F8(EFCT)>> F5(INS2)

You'll see that I'm sending the sound through an amp simulator, with a tube setting, and the slightest bit of drive, to further mangle the sound but not so much that it loses the detail in the playing. You'll probably want your sound to be rather gritty, but not too painful. Remember, your focus for this step is to give the snare drum plenty of definition and character, so if necessary, use the EQ effect (INS1 24 or 23) or a filter:

For a grittier sound, lower the volume on the low frequencies, and boost the volume on the mids. Don't worry about the kick drum…we can put it back in later. You'll want a pretty dry sound, so I've also removed any reverb or chorus. As well, I recommend that you get in the habit of turning the AMP release time to 0:

VOICE>> EDIT>>F5(AMP)>>F8(EG)>> [turn knob 6 to edit AMP Release]

This will give you have absolute control over the duration of the sound. The AMP page is where you define all things pertaining to the volume or, if you really want to impress your friends, amplitude(AMP). From this page you can control level (how loud) and time (how long the sound takes to get to the specified level) in a shape (or envelope, if you really want to impress your friends) that you can design. EG, two letters that appear together often, stand for 'Envelope Generator'. You can alter, from left to right, ATTACK (the amount of time it takes from note-on, to get to the level you set), DECAY 1(where the level goes after reaching the attack level), DECAY 2 (you figure it out), SUSTAIN (The release controls the time that it takes for your sound to fade to silence after you release the note, and in this case, you want to have the ability to go from noise to silence in the fastest time possible.

After you are satisfied that you've given the sound a new character, it's time to resample. Remember that the EX purposely lowers the volume when you resample to give you lots of headroom, so go to your SCF:

 VOICE>> EDIT>> F4(FILT)>> F4(SCF)>> [Use dial to select SCF Filter type]

Set it to an appropriate level (usually +12 dB is enough), or boost the Out Level on the fx all the way. REMEMBER TO TURN THE OUTPUT VOLUME DOWN, because your ears are more sensitive than the EX's ears. To resample, simply follow the steps for sampling, but when you choose your Source, select internal, and set 'Trigger' to 'key'. If you happen to be resampling a sequence that you've composed on one of the EX sequencers, then you can also set trigger to 'seq', which starts sampling when you hit PLAY.

After you've successfully resampled, have a comparative listen between the original sample and your new sample (I've called mine BB Step2). If you are satisfied that you've started to personalize the sound, then you are ready to proceed. If not, don't get discouraged, but don't compromise. Take your time and play around with sounds. You want it to be distinctive and characteristic, but you also want it to be usable!

Now comes the painstaking part. You want to separate several components of the beat. If you are lucky enough to have a good computer, you can save yourself some time by exporting your sounds to Tiny Wave Editor, or TWE, which is a visual sample editor available from the Sound and Utility CD/ROM included with your EX, or from the Yamaha UK website. The rest of us can get comfy and role up our sleeves for some serious editing.

We want to turn our loop into several smaller loops, each starting at a different point in the beat. There are 2 basic ways to divide up the beat: #1 pick some different sounds within the beat (i.e.. Kick, Snare 1, Snare 2) or into 4, 8, or 16 mathematically even phrases (assuming your beat is 4, 8,

or 16 beats). Both work well, and both give surprisingly different feels when you get to programming. For this example, I'm going to separate based on sounds. To begin, copy your beat to as many other sample locations as you want fragments. For example, let's pretend that the sample we are using is the phrase I love my EX, and the name of the sample is ILoveMyEX, stored as RAM 0001. Copy your RAM 0001 to RAM 0002:

SAMPLE>> **JOB**>> **F1(COPY)**>> **[Select your source sample and your destination]** >>**ENTER** >>**INC/YES**

Naturally, 'Source' refers to your original, and 'Destination' refers to the location that the source is being copied to. Be sure that you aren't copying your source to locations where other samples that you want already exist...it will erase anything that was at that location. Now, because we want to separate the components of the phrase "I love my EX", we will need to repeat that step five times:(1)I, (2)love, (3)my, (4)E, (5)X. Each time you perform the copy function, make sure that you are sending your source to another location.

What you should end up with is five samples that all say "I love my EX". Not very useful at all, really. In order to make these samples usable, we now want to have them each start on a different word. Follow the previous steps for sample loop editing. On your first sample, only allow the word "I" to be heard. On the second sample, only the word "love", and so on. This concept is much easier to understand than it is to explain! Another option is to only edit the start point of the sample, so that sample 1 remains the same, sample 2 becomes "love my EX", sample 3 becomes "my EX", etc..

In order to save precious memory, we can now extract the parts of the sample we are not using. The extract function acts exactly like the 'copy' function, except that it only copies the information that lies between the Sample Start point and the Sample End point.
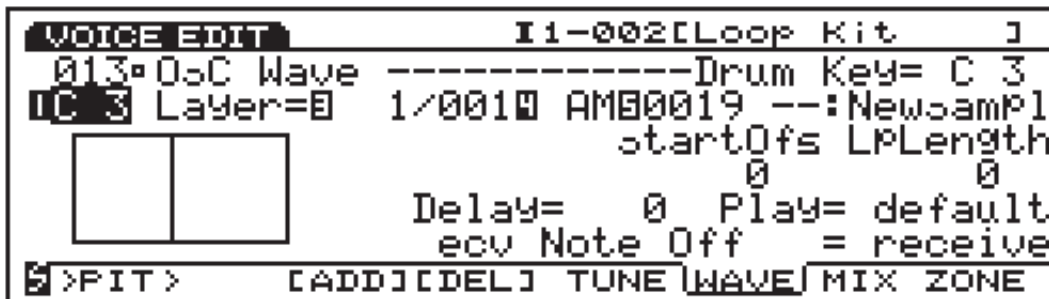
SAMPLE>> **JOB**>> **F5(EXTRACT)**>> **[Set source sample and destination]**>> **ENTER**>> **INC/YES**

In this case, you'll likely be making the destination the same as the source. More advanced EXers may want to eliminate a step: instead of copying your sample five times as before, do all your editing on the source sample, and extract the bits that you want as samples to a different location. Then, edit your Start and End point of the source again, and extract that to a different location, and so on.

This part of the process is very technical, and can be tedious, but through it all, you get to know your samples quite well. Remember, precision is key. You don't want sloppy loops! If you are finding this difficult, examine these steps in a non-hypothetical situation. Samples 'BB Step2' and 'BBFrag1-13' will show you what the final product sounds like, with the drums that we have been working with. Remember to save often, as nothing is more frustrating than losing your work at this point.

Now that you've done that (and perhaps taken a bit of a break), it's time for the next step: building a basic kit. So, initialize a voice as before, but this time, instead of selecting 'FDSP', select 'Drum'. A 'Drum' voice allows you to place a different wave on every key, originally for the purpose of building kits. To begin, let's assign our samples to some keys.

Follow the previous steps to arrive at the OSC page. Once you get there, you will see that Drum Voice OSC wave edit pages have a slightly different display than the other voice types, simply because within Drum Voices, every key can have a different group of waves layered. This page also allows you to navigate through the different sounds in 2 different ways: by 'Elements' (located in the top left corner), and also by note names (i.e. C2, E#6), located just below the Element number.

```
┌─▐VOICE EDIT▌──────────── I1-002[Loop Kit    ]
│ 013▫OsC Wave ------------Drum Key= C 3
│ ▐C 3▌ Layer=0   1/0010 AM00019 --:NewsamPl
│ ┌────┬────┐            startOfs LPLength
│ │    │    │                  0        0
│ │    │    │         Delay=   0  PlaY= default
│ └────┴────┘          ecv Note Off   = receive
│ ▐5▌>PIT>    [ADD][DEL] TUNE WAVE MIX ZONE
```

To begin, pick a note, any note, as a starting point. Assign the first sample by selecting a note by changing the 'NOTE SELECT', and pressing F3(ADD). This tells the EX that you've added an 'Element' to the set up of this key. The EX will automatically default to an internal (or PRE) wave. If you are building your kit from the internal sounds from the EX ROM, then it is a simple matter of selecting which sounds you want on this key by changing the value of the number that follows PRE. Should you desire a sample that you have created, then you must first change PRE to RAM or, if you have your sample saved to FlashRAM, then select FLS. Now, assign your sample as if you were working with an internal sound; by scrolling through your RAM sounds. Because we are building waves, as before, you can add several layers or samples on every note. Once you've done that, then use the NOTE SELECT to select a different note, and add a new sample or layer of samples, repeating this process until you have keyboard access to all of the samples you wish to use. I'd recommend using a logical structure for sample placement, like consecutive keys in an ascending scale.

At this point, you should decide the relative tempo on your patch. For this tutorial, I'm going to pitch everything up +7, giving the patch a tempo of about 175 BPM. To transpose samples and sounds, you must still be in the OSC VOICE EDIT, but push F5(TUNE) to select the tune edit page. Adjusting the 'Coarse' value will speed up or slow down your sample in semitone increments. Use this to find an approximate tempo, and then use 'Fine' for subtle tuning or tempo adjustments.

In the OSC (F5)MIX page, you might want to send all of your loops to 'InsEF 2', to give yourself more options later. To so this, simply change the value below 'InsEF' in the bottom right hand corner, from off to 2. As well, for waves of the Loop nature, go to the OSC F5(WAVE) page, and set 'Recv Note Off' to 'receive'. When it's set to 'ignore', striking a key simply triggers the sound and allows it to play for a predetermined time as defined in the AMP Edit page, to add realism to AWM drum patches. Unfortunately, in Drum Voice mode, the EX defaults to ignore, so this can become tedious, but you'll soon see the importance. As before, go to the AMP EG page, and set the release to 0. Repeat these steps for all significant loops. To keep the options wide open, you can also put in some other percussive sounds into your Patch. I'd recommend given your self a good solid kick drum sound, some cymbals or hi-hats, and maybe even some more snare drums. On the kick drum, you'll want to pitch it up a bit. The function of the kick drum in Drum'n'Bass music is

different than in other genres; it isn't going to be the deepest, most boomy sound, but it should still have impact. Ultimately, you'll want the cymbal and hats to tingle your scalp, the snares should slap your face, and the kick should, well, kick you, preferably in the chest. You'll also noticed that I've assigned knobs to control the filtering of the different fragments, and MW1 as an IN EFF2 modulator. It's wise to keep the different fragments filter control on as many different knobs (and controller sets) as possible, so you can have more movement and variation. Remember that on the EX, although you have 16 controller sets, when you are in Drum Mode, each set can only affect 1 sample at a time (unless it's a global parameter, like EFF drive or depth), so you have to assign an Element to be affected.

```
┌─────────────────────────────────────────────────┐
│ VOICE EDIT          I1-002[Loop Kit        ]    │
│ -CT ▫Controller set --------Ctrl=set1 -         │
│      Ctrl=█set1                  Elem=0    1     │
│      src: ▫PB                                    │
│            ▫MW1 ▫MW2 ▫AT  ▫FC  ▫BC  ▫  B         │
│            ▪KN1 ▫KN2 ▫KN3 ▫KN4 ▫KN5 ▫KN6         │
│      Dst:  124:D  DCF Freq      Depth= +11       │
│ ▐>EFCT>   [REMAP]               PITCH ⌊SET⌉      │
└─────────────────────────────────────────────────┘
```

If you go to the Controller Set Edit page, you will see the Element number in the top right corner. The Elements correspond to the Element number in the top left hand corner of your OSC edit pages.

After you have assembled your initial palette, it's time to move on to your third level of editing. This is where you will be coming up with several more loops, based on your initial patch, so get ready for some serious sequencing. To help yourself out compositionally, give yourself a few basic motifs, (e.g.. same effects on INS 2, similar rhythms, etc.)and then elaborate on each one. Each sequence only needs to be 1 or 2 bars, but the more you make now, the more options you'll have later. After you have sequenced a beat that you are content with, follow the previous directions for resampling. To save further confusion, sample as you go, always remembering to change the pertinent SCFs to +12, or to boost your effects output (part of the parameter of the actual effect) to slightly beyond your pain threshold (REMEMBER TO LOWER THE OUTPUT VOLUME!!) Also, remember that the EX won't respond to movement on its own knobs while it's resampling, so either sequence your knob movements, or control them from an exterior controller.

While I can't help you out with the specifics of patterns, I can say this...If you are stuck with a dry and uninteresting pattern or the variations just aren't coming to you, try using the Groove Templates to spice things up a bit. Forcing templates on your beats can make them do a total 180˚. Try changing the voice to (I2-123 Dr: Rare Breaks) and going into PATTERN mode by simply pushing the button marked PATTERN.

***Load GROOVES.S1P off of disk Disk #2.

[Insert Disk#2]>> DISK>> F2(LOAD)>> F7(PATTERN)>>>> ENTER>> INC/YES

First of all, go to the PATTERN PLAY page. Make sure that the 'FX thru' box under the desired track is not solid. If it is solid, then 'FX thru' is on, and therefore, you WILL NOT hear the groove.

Turning to pattern 1, we have a pretty basic hip-hop groove...nothing fancy. While in Pattern Mode, push F2 (PFX) and then push F8 (GROOVE). This window shows you a graphic display of the variety of grooves that are available. Hit play on the sequencer, and the beat should start. I have purposely programmed this beat with no dynamics, so that you can hear how the Groove Templates affect the groove. Using Knob 2, try out some different templates. They really do change the entire function of the beat. If you move Knob 1, you will be selecting which sequencer track you are effecting. In this example, I have put all of the beat components onto one track, so you cannot have different templates.

Creating your own groove templates is a great way to personalize your tunes. If you go all the way up to type 101, you'll find an initialized template, which is fully editable. You can set the where the notes fall with reference to the beat (ClkSft), the gate time (GrvTim), and the velocity (VelOft) of every note on a set grid (Quant).

If you record the components of your beat or sequence onto different tracks (i.e. kick and snare on track one, and hi-hat onto track 2), then you can put different templates on the different tracks. If you listen to pattern 2, you'll hear a standard House beat but I've separated the parts, so that the kick is on track 1, and the hi-hats are on track 2. A common technique in Electronica (especially House) is to start with a sturdy, quantized pattern on a significant sound, like a kick. Then, force a heavy swing on sounds like hi-hats, shakers, or a comping part of some kind. This keeps the main beat mechanical, but colours it with a more funky groove. Try changing the groove on Track 2 without changing track 1. You can create some pretty intense rhythms by making a heavily layered beat with different templates on different tracks. As always, once you hear something you like, sample it right away.

If you listen to (I1-003 Phrase Kit), you can hear the examples I have made using the Loop Kit. Notice that I've used several different types of effects, but all of these phrases are built from the same source, and function at the same tempo. The beauty of this process is that when it comes to sequencing your beats for a final mix, you've already created a palette for yourself, making improvisation easy and truly spontaneous, and as well, you get maximum variety and movement without compromising DSP power. The riffs, filter movements, and effects changes have already been performed, so you can focus your energy on resequencing your beats into even more complex grooves. As an aside, if you have less than 32 mb of RAM, or if the SCSI speed irritates you, then you can erase any samples that you aren't using...like your initial loop, or your stage 2 loops.

Using samples isn't the only way to achieve this effect. If you want to try an interesting approach to breakbeat manufacturing, check this out: using a combination of layers and the Delay feature in the OSC Wave page of a Drum voice, it is quite possible to assign mini-sequences that use the internal sounds of the EX.

VOICE>> EDIT>> F2(OSC)>> F6(WAVE)>> [Adjust the number beside Delay to add time between note-on and the sample play-back]

The concept is simple...the Delay function simply adds a set amount of time between the note-on message and delivery of the sound. This is sometimes a desirable way to work, because even though the phrases functions exactly the same as samples would, they use only the internal voices. For those of us who need to perform in a live setting, (and don't feel like springing for

FlashRam) this is a good way to go. If you examine C1-C2 of (I1-004: De-layer) you can get a feel for how this would function. The only truly tricky part of creating patterns this way, is that all timings must be done by ear (!), and again, BPM must be established right away. Another advantage of working in this manner is that it allows for you to add more real-time manipulations by clever use of effects sends. On the same key, you could have every different wave going to a different effects send (follow the instructions for sending your sounds to InsEF 2).

If you adjust MW1 in this patch, you'll notice that I've set it up to adjust the input level of INS EFF 2, which, in this case, is the D. Scratch Algorithm (InsEF2 66). The D.Scratch effect is an adjustable LFO that affects the pitch of a wave in a definable manner. Because it isn't synced to the note-ons, it can be unpredictable, so tweaking is important. To maximize your control over the D. Scratch, I'd suggest that you set it up so that it only gives you the effected and not the dry signal, by changing the dry to wet ratio to D<W. This effect gives you the option to set an initial delay time, but for now let\ s set it to the minimum, 0.1 ms. You can play around with the Scratch Speed (frequency), and the Scratch Depth (amplitude).

To truly emulate the moves of a stylin' DJ, I wouldn't set either of these too extreme. Keeping this effect subtle adds funk-flava in subliminal ways. For further detail on the usage of the D. Scratch, check out patch (I1-006 Br:Braskratch). Here we have a reconstruction of the classic scratch-to-brass-hit, as made popular by rap music from 1985-1989. This effect is obtained on the EX by again, using your layers creatively. Take a look at the OSC wave page...it takes 4 layers to build this sound. The first sound is being sent directly to the D.Scratch, which I have carefully adjusted to be in a good relative tempo with the rest of the action. To keep it realistic, I've set the AMP EG very carefully, so that the scratching ends at about the same time that the other 3 sounds come in. I've staggered the delay times of the other sounds, just to give it a more human feel. On your OSC Mix page, if you want a key to trigger the scratch, but you don't want to hear the initial wave, assign one layer to go directly to InsEF 2, with an appropriately set delay. This layer will go directly to the Effect, and unless you play with the dry to effected ratio on the effects page, you will not hear anything but scratching.

Going back to (I1-004 Dr:De-layer), and look at the OSC Wave page. Notice that I've used FX:Melow.1 as the wave that gets scratched. In most cases, the actual percussion sounds don't last long enough to scratch effectively. Using a non-percussive wave gives the illusion that the beat and scratch are 2 different records on 2 different turntables. Other effects also work quite well; Jump Wave can be inserted at various points to add a mild jungle flavour (I1-005: Roodbwah). Again, this patch uses the MW1 to adjust the balance between beat and effect, so play around with it. Keep in mind that the Jump Wave effect also isn't really synced to tempo, so be clever about it. And again, notice that I've dirtied the sounds up, giving them a more sample-esque feel. It's always effective to degrade quality on drum sounds via DPCM, Tech Mod, LoResolution, Noisy, Attack LoFi, or any of the various distortions and overdrives.

No matter which approach you take, the outcome should be the same; when it comes time to actually start working on a specific song, you want to be working with phrases, reserving individual hits for support only. Sample, edit, effect, and repeat as many times as necessary to ensure functionality and originality.

Chapter 3. AN and VL

In the old days, as we all know by now, there weren't many options when it came to sound synthesis. Keyboards (if you can call them that) were large, unstable beasts that could easily fluctuate out of time and tune. Then came the 80's, and the good people at Yamaha introduced the DX7, with clean, crisp tones that don't ever, ever go out of tune. Hence 80's pop, with its total lack of dissonance. And now, in the true style of late 90's irony, it seems that many people, myself included, crave the instabilities to add warmth and character to their digital sounds.

The good news is that it can happen. Easily. Darn it, I love this machine. Let's start by going way back in time, to the days when synthesizers were just big patch bays, and your only interfaces were cables and knobs.

For this portion, you're gonna want to initialize a voice as an AN(poly)+AWM. Before we get to much further, go right to the OSC-ALG page.

VOICE>> EDIT>> F2(OSC)>> F3(ALG)

This page is where we decide the basic algorithm that the EX will use to emulate analogue synthesis. For this example, let's set it to FM/both, which will attach a sync-pitch to the original wave. A sync pitch, as the name suggests, is a second frequency that is synced to the initial oscillator, or more precisely, a pitch that is generated by the intermingling of the harmonic contents of two pitches that are in sync. It can be set to follow the pitch of the original wave, follow an envelope, an LFO, or controller data. Because the ears are lazy, we generally hear the sync pitch as part of the initial pitch, much in the same way that unless you really focus, we cannot really distinguish between fundamental and harmonic. The sync pitch can be used to fatten up a sound or thin it out, make a sound more mellow, or more biting, depending on how you set it up. Before you start playing around with different pitches and depths, let's go to the VCO page (F4).

The VCO page is where we further define the initial wave.

For those who don't know, VCO stands for Voltage Controlled Oscillator, which, in days of yore, was where the first steps of electronic sound took place. It used to be simple; the oscillator would make a sound, and then you could filter and modulate certain aspects of the sound to your specifications. This is called subtractive-synthesis. Because this is a digital machine, and there isn't really a physical oscillator, but rather a digital simulation, we need to edit our initial oscillator before we go too far.

I've set the VCO1 frequency (FREQ) down all the way to -64, because the kind of synthesizer that we want to emulate doesn't really play pitches in a manner that we would be accustomed to. By now you should be hearing a rather unusable click-click-click. (I1-010 Se:UnusableClik). The clicking you hear is actually single oscillations; if you sped the sound up so that there were 440 of those per second, it would be a perfect A. Don't bother with anything in the VCO2 column, as we're not using it for this example anyway. When you are building your own AN voices, you can use VCO2 as a second oscillator to make more complex sounds and layers.

Go to the EFF page, and turn the reverb and chorus so far down that they're off. Now you should be hearing a DRY unusable click. Let's go to the CTRL set and assign Knob 1 to AN VCF Freq (110).

VOICE>> EDIT>> F7(CTRL)>> F8(SET)

I'd recommend setting it to a depth of about +20. To do this, you must take the following steps:

1. Assign an unused controller set. Controller sets are the set-up parameters for the specified controllers. You have 16 controller sets, and therefore, you can assign 16 events to controllers.

2. Turn the Element on. The leftmost column (Elem) is where you pick which wave you want to be affected by the controller set that you are now on. The elements correspond directly to the items on your OSC wave page. For this example, there is only one wave, but you could have all 4 waves turned on in a controller set. In a drum voice, however, you can only have 1 element on per controller set.

3. Assign a destination. Destination (Dst), which is found at the very bottom of the page, is where you select what you want to affect about the wave that we specified above. Each controller set can only have one Dst.

4. Assign a Source (Src). This allows you to assign which controller(s) will cause the changes to the above destination. You can have as many controllers as you want in a set.

5. Assign a Depth. This refers to how much of a change will occur. The greater the value, the more drastic the change. For example, if you assigned MW1 (Modulation Wheel One) to control Filter Cutoff, and you have assigned a Depth of +32, then moving MW1 from the lowest position to the highest position, will raise the filter cutoff by 32. If the Depth value is set to a -32, then the same movement will lower the cutoff by -32.

We might as well make another controller set, assigning knob 2 to AN VCF Reso, perhaps at a depth of about +23, so that at full twist, it just crosses the self-modulation threshold. It's very important that you, the musician, assign your own custom controller sets in a way that feels

intuitive and natural to you. Since we're here anyway, let's set knob 3 to affect AN VCO1 PMD (096) at a depth of -64. PMD, or pulse modulation depth, defines how much the pulse is affected by the LFO. Although you may not hear how it affects the sound immediately, you will if you go back to the OSC VCO Mod page, and set the Mod to +127.

On this page, you can set your whether your oscillator will follow LFO1 or 2 (Src), at a specified intensity (Mod). By playing around with this, you should actually be able to find several settings where the LFO shape is quite apparent. To demonstrate this further, go to the LFO editing page:

The LFO, or Low Frequency Oscillator generates values (not tones) that can be used to affect the sound in various ways and shapes. On this page, you will see FMD (filter modulation depth) which allows you to have your filter follow the LFO shape. There is also AMD (amplitude modulation depth), which makes your volume follow the LFO shape. Setting either of these to 0 makes them not react to the LFO at all. Because we have set our OSC VCO to +127, the pitch of the initial oscillation will follow the shape of the LFO quite drastically. You are also given the opportunity to sync the LFO to note-ons. When sync is off, the LFO is constantly generating values, and when you hit a note, then the first LFO value that you hear will be whatever value the LFO happens to be at. When sync is on, then every time there is a note-on, the LFO resets to 0, and then proceeds to do what it does. For this kind of sound, I'm gonna put sync on. Since we're here anyway, play around with the different LFO waves. You should be able to hear exactly how the sounds relate to the different LFO's.

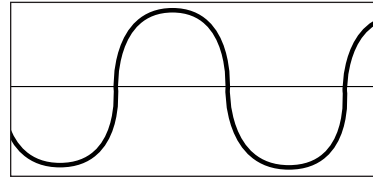[Check out the LFO Waves in your Owner's Manual- pages 100-102]

By now, you will have noticed that my personal tendency is to get the controller up and running as soon as possible. My general rule is if I go from -64 to +64 on any parameter more than three times, grinning foolishly at how it evolves the sounds, it likely deserves a controller set. Heck, you've got 16 of them...use 'em up! Having said that, let's go back to the controller page, and make Knob 4 affect AN LFO1 Speed (087) at a depth of +10. Play with that for awhile. And some more. Go really high...now really low...now see if you can swoop it mid pulse...by now, we have something that sounds like a real synthesizer, perhaps a Synthi. (I1-11) Oooo, remember those? You edited them by sticking pins into a matrix. How about it, Yamaha? How about a Matrix Controller?

What you do from here is pretty much your own decision. Personally, I'm going to add Dst: (090) AN Sync Pitch Depth +18 to a new controller set, which will let us increase the amount that the sync pitch follows the LFO. To make this more apparent, set the initial sync pitch to +63 on the OSC VCO Alg page. As well, I'm going to throw in Dst: (093) AN VCO Edge onto Knob 6 at a depth of +63. Edge affects brightness at a very base level, by controlling the sharpness (or EDGE) of the actual initial wave. Play around to establish some idea of what each knob is doing. In the mean time, I'm going to see what I can do to increase the authenticity of our simulation.
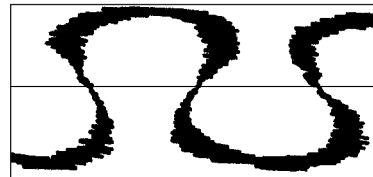
To do this, I'm going to run the entire sound through the Amp Simulator (Insert Effect 2-40). The Amp Simulator physically models a tube (or valve, if you're in the Europe) amplifier, which treats sound entirely differently than the diodes, transistors, and microchips in a digital machine. As

sound travels through a tube, it is a much more physical electricity, so it reacts in a more physically understandable way...for example, if you push the sounds amplitude (or volume) past the limit of the tube, it squishes it down, and spreads it out, creating a pleasant distortion, and adding overtones that weren't as present in the initial sound. When sound is traveling through a digital machine, it's moving as numbers, so it reacts in a more mathematically understandable way. When a digital machine receives an amplitude that is to large for it to handle, it truncates it (ignores values that are greater than its capacity), creating a unique, harsh , and cold distortion.

Regular Sine Wave

Overdriving a Sine Wave Through a Tube or Amp Simulator

Overdriving a Digital Signal (Truncates)

The amp simulator receives sound at an acceptable level, and then simulates overdriving by spreading the sound out as would happen in a more physical reality.

The EX's Amp Simulator is a quite well thought-out physical model. For this example, let's use the Tube amp type, put the drive to 0, put the LPF to 20.0 kHz, and the Edge setting to 127. Being the control freak I am, I'd like to make a motion that we hit the controller page (again!), and assign a controller set to affect Dst:(020)EF2 Drive at a depth of +17 to MW1. We now have a reasonable emulation of one extremely early synth. (I1-012 and I1-013).

***Load WBW.S1M off of disk Disk #2.

[Insert Disk#2]>> DISK>> F2(LOAD)>> F4(WAVE)>> [Select the appropriate file]>> ENTER>> INC/YES

If you go to the 'Sample' page, you can hear these samples. They are all variations of the same voice—the only adjustments have been made by tweaking the knobs and in some cases, changing the Insert Effects.

True to its model, this style of sound can provide hours of fun while creating useful samples, but it's not much good as a pitched instrument. To pull off a good leads or melodic basses, you're gonna need something that can at least be played as a series of pitches, perhaps in series of

twelve tones. For our next tutorial, let's use a VL voice instead of an AN voice. Although each have their advantages and disadvantages, for this example, let's aim for pure warmth and subtle randomness of sound, more easily achieved with VL. Follow the previous instructions for initializing voices, but this time, make it a VL+AWM voice. While you are doing this, you might as well switch the voice into a monophonic voice.

VOICE>> EDIT>> FI(COM)>> [Adjust Mono/Poly]

Now, adjust Mono/Poly so that it says mono. This forces the EX to only play one note at a time, just like in the good ol' days. In this case, it's a good idea, as the VL voice generator is monophonic anyway, so when we start filling it out with AWM voices, it will make more sense to the ear. (I1-015 —:Skware) Because of the nature of VL synthesis, it is quite possible to warm the sound up by overdriving at different stages, without relying on a separate effect. In this example, I've given it some slight adjustments in the EQ area of the VL filter.

VOICE>> EDIT>> F4(FILT)>> F7(EQ)

Keeping the resonance low (thereby affecting the widest band of frequencies), I've boosted the extreme lows, adding more punch. On a higher resonance setting, I've also boosted some of the higher frequencies, giving the sound more of a whiny buzz. Remember that it's OK to overdrive the sound at this stage- in small amounts, it will sound very natural and warm. Now, remembering that our purpose right now is emulation, let's go to the controller set and assign some knobs.

In keeping with the mono leads of the time, I've used controller sets to assign Filter Frequency and Resonance to the usual knobs 1 and 2. Knobs 3 and 4 control VL attack and decay, respectively. One of the advantages of VL synthesis on the EX is that it allows you to add instabilities, which, if you are into psycho-acoustics (the sounds that the ears miss, but the brain and body still perceive), you'll know can help to create warmth. To pull this off, I've used controller sets to tell the EX to use velocity (available only in the VL destinations #126-146) to calculate subtle detuning, Dst:128 VL Detune. I've only given it a depth of 5, because anymore than that and it starts to be too apparent. I've done a similar trick with VL Scream (Dst: 132 VL Scream), only this time I've also added AT (aftertouch) as a controller too. The Scream parameter, which is available for edit only through a controller set, sets a level for the random noise that would be generated by overdriving or overblowing your chosen VL instrument. In this case, it happens to be very subtle, and I like to think of it as evoking images of unstable and dusty circuitry, from the days when it was okay for musicians to smoke cigarettes in the studio. By assigning it to aftertouch, it can also be used as an expressive tool, increasing the hum of the virtual electricity thats flowing through your virtual synthesizer.

There is a definite difference between Scream and Growl, best explained through a saxophone analogy: on a sax, Growl is that growl sound you get from overblowing, Scream is that screaming sound you get from biting the reed and blowing hard. All of these parameters have been assigned to velocity so that there is little thought involved—the EX will create subtle nuances based on how fast the key is hit, and you, as the player have little or no control over specifics. If you have doubts as to the effectiveness of these techniques, compare (I1-015 —:Skware) to (I1-016 — :Skwarer). I think that you'll agree that although the differences are subtle, they are there, and

they do add a certain liveliness and warmth to the sound. Now that we have built a pretty convincing analogue model, there are several directions we could proceed in, depending on what outcome you desire. You could leave well enough alone...this sound will hold its own as a lead or bass...but for the sake of getting the most for our EX dollar, let's push forwards. Let's begin by further developing the sound as a lead.

Many analogue (or analogue modeling) machines give you the option for a sync pitch, which, as we found out in a previous lesson, is a frequency that is attached to our main oscillator. If we were using AN synthesis, we would have the option of adding an actual sync pitch to our physical model. Because of the massive complexities of VL synthesis, Yamaha has opted to give us a vast amount of VL presets, instead of vast amounts of editing options. It's a pretty good thing, too. If you wish to delve into the world of VL editing, check the appendix at the back for the Yamaha UK Website, where you can download several different VL editors. Anyway, the point remains that we want a sync pitch, but have no way of editing our VL voice to have one. That is where AWM comes into play.

Let's go back to our OSC Wave page. You should see that even though we've used up one voice for our VL voice, we still have 3 AWM voices at our disposal. Never underestimate the power of AWM layers. You should become familiar with all of these waves, as they quite often become the icing on our musical cakes. For this example, as I said before, I want to add a sync pitch, to create that piercing lead sound. To do this, I've thrown AWM PRE 0224 Wv:SyncLd in as our second sonic element. Because this particular wave is a sample of a sync LEAD (a lead sound that already has a sync pitch in it) and not just the sync PITCH, we've got to isolate the sync pitch. Are you still with me? To do this, it's a simple matter of filtering.

It is important to understand the differences between the 2 types of filters that the EX offers per voice. The DCF, or Dynamic Controlled Filter, is the type of filter that most machines offer. It is sweepable via knobs, it can follow an envelope or LFO, and it can squeal in a pig-like fashion. If you ever need to toughen a sound up, try raising the gain and resonance and lowering the cutoff. Nasty stuff. The SCF, or Static Control Filter is the same concept, but it is designed to affect the overall timbre of a sound in an unchanging, or static manner. its cutoff and resonance cannot be assigned to any controllers. Now, for our purposes in this example, we want to isolate the sync part of our chosen wave, and, because at no point do we really want it to vary, we should do it with the SCF.

VOICE>> EDIT>> F2(OSC)>> [Use your cursor buttons to select the wave you wish to edit]>> EXIT>> F4(FILTER)>>F4(SCF)

If you examine the SCF on (I1-SkwareSync), you can see that I've used a HPF (a High Pass Filter allows only the frequencies that fall above the cutoff frequency to be heard) to remove all but the sync-pitch. Because the sync-pitch is just about the highest frequency in this wave, I have set the cut-off quite high, so that all the frequencies below the cut-off are filtered out.

In order to make our new layer function as if it were synced to the VL voice, I've also adjusted the DCF, AWM attack, and AWM decay to respond to the same controllers as the VL FREQ, VL Attack, and VL Decay. This takes some fine tuning and tweaking of the initial settings and also in controller depths. Remember, we want to create the illusion that this is one homogenous sound, and not two separate layers. After you are satisfied that you have given the right twists and turn

to make this function correctly, take a few minutes to groove out on the new sound and decide what you'd like to do next. If you try over-lapping notes on the example, you'll see that there is a slide as the EX shifts pitches. This is called 'Portamento', and it can be turned On or Off in the CTRL PITCH menu.

VOICE>> EDIT>> F7(CTRL)>> F7(PITCH)>> [Adjust Portemento Sw to be On or Off]>> [Adjust Time]>> [Adjust Mode between fngr (fingered) and full]

 You can set it to be full (every note slides to the next), or fingered (portamento only occurs when a second note overlaps the first). TIME refers to the time that it takes to get from point A to point B. Personally, I'm satisfied that we've made a convincing sound, and it's time to add more personality and general trippyness.

To further personalize this sound, I'd like to give it more depth and movement, which are always nice additions to any dry sound like this. In any situation, there are no right or wrong ways to approach this, but there are several conventional methods that can be quite effective. If you turn to (I1-018 Ld:KitchenSync) you will be able to hear the subtle differences that I have added to give it a less sterile feel. You may notice the presence of some new waves that weren't there before.

If you turn to the OSC Wave page, you'll see that I have added wave #0227 Wv: 3o3Squ twice. The reason I added it in the first place, is that it is basically a sample of the same type of wave that our VL generator is emulating. The presence of the AWM version serves to thicken our sound, as if there were more than one oscillator per voice. To make the sound even thicker still, I've used the same wave twice, panning one hard left (so that it only comes out of the left channel), and the other hard right (so that it only comes out the right channel). By detuning these layers slightly, the sound becomes thicker, with a slight phase shift.

Because these two waves aren't exactly the same pitch, there are harmonic conflicts that increases the perceived 'depth' of the sound. Again, I've done my best to edit these sounds to function as if they were not different layers, but part of the original. To pull this off, I've changed the release time on their respective AMP EG pages to have a more sudden release. Remember, the lower the value, the faster the release. I've also filtered them to have more in common with the VL wave. Because of the nature of Subtractive Synthesis, each sound should be looked at as more than you'll need, and it is up to you to remove the parts of the sound that don't have any place in your creation. In this case, I wanted the filter to be sweepable, so I've done my editing in the DCF, and left the SCF as thru. I have also assigned the filter cut-off to react to Knob 1 in the controller set, and the Resonance to react to knob 2. Again, I have played with the filter levels to keep all of the sounds together. I have also done the same with Attack and Decay times. If you examine the controller set page, you'll see that instead of assigning these to new sets, I've simply turned elements 3 and 4 on in sets 8, 9, 10, and 11.

Previously, these controller sets effected only our sync pitch. In the case of any voice except Drum voices, each controller set allows you to have any or all of the controllers (Knobs, MW, Ribbon, etc.) , control any 1 pertinent parameter of any of the selected voices, at a common depth. If you should want the same controller to edit different sounds at different depths, then you will have to assign it to a different set. It's confusing at first, but once you've wrapped your brain around it, it becomes where a lot of the power lies.

Speaking of power, let's take a closer look at a very powerful aspect of this machine: Randomness. The biggest problem with synthesizers, at least until very recently, is that they trigger a note, play it out, and that is that. But many of us like to leave something to chance. The example above shows how to use VL synthesis to achieve minor random variations that tickle our psycho-acoustic register...let's examine something more obvious. The majority of the VL models that are included in the EX use several series of complex mathematical equations to create models of real instruments. We all know how cool it is that a pitchbend on the Trumpet presets don't just bend the pitch; it tightens the virtual embouchure to achieve a rise or fall in pitch that correspond to the actual harmonic series. The editing of VL parameter such as embouchure, growl, and scream can also be used to make wild and unpredictable sounds.

Let's initialize a new voice as a VL+AWM voice. If you switch to the OSC page, it should have defaulted to 201 Alto sax. For this segment, you're going to want to find sound that isn't clean and melodic, but rather, dissonant and inconsistent. Some recommendations: 003 SpaceZoo, 064 WaterPhone, or the one that I have used, good ol' 001 Mad Tube (I1-050 —:Tubes). If you aren't lucky enough to own a wind controller of some kind, then make sure that you've gone into your Utility Controller window, and switch Breathe Controller to Velocity:

UTILITY>> F5(CTRL)>>[Adjust Breath Controller Assign to say Velocity]

This tells the EX get the breath control information, which is vital for VL expression, from the velocity value (how fast the note is struck).

Because our goal for this section of the tutorial is to make a sound that is as unpredictable as possible, let's go to our controller page and assign some controller sets. For this sound, I have assigned the following VL parameters to be adjusted by aftertouch:

Pressure: (how 'hard' you strike, bow, or blow the virtual instrument)
Embouchure: (how tight your mouth is against the virtual mouthpiece)
Scream: (amount of random noise, or as the EX manual describes it, chaotic oscillation)
Damping: (how much the air friction affects the oscillator)
Growl: (the reaction of a sudden extreme pressure on the oscillator)
Absorption: (how much the shape of the virtual instrument affects the sound)

Aftertouch works extremely well for these purposes because it is very hard to keep stable, unlike knobs and wheels. Also, because we are going to be building a very heavily affected sound, using aftertouch to affect these major timbral changes leaves one hand free for knob movements. To give even more control, I have also used MW1 to affect some of these parameters.
(I1-051 —:Twoobs)

For this section, as I said before, we are aiming for drastic and unpredictable changes in the sound. Another fantastic and unpredictable effect that should do the trick: Auto Synth. The Auto Synth effect (InsEF2:67 Auto Synth) takes your wave and runs it through a series of delays, wave modulations and tremolos (a rapid wavering of volume) and spits out a sound that has rhythms and pitches that weren't there in your original.

In order to get the Auto Synth to act as I want it to, I've made a few adjustments to the patch. The first thing that I did was change 'AM R Inverse' to 'reverse'. Translation: AM (amplitude

modulation, or tremolo) R (right channel) Inverse, so that as the Auto Synth effect performs its tremolo, it is staggering the phase of the effect, so that when the left channel is at maximum volume, the right channel is at its minimum, and vice-versa. This produces a trippy, disorienting effect when played through any stereo sound system. But, because too much of this will eventually make you quite ill (not literally), I have assigned controller set #8 to control AM Depth (Dst: 034) from knob 6. AM Depth allows you to alter the difference in the tremolo, between the lowest and highest point. So, for example, a setting of 0 would leave the sound pretty much untouched, but a setting of 127 would cause the amplitude modulation to go from 0-127. (I1-052 —:Threebs). OK! Now play around for awhile, customize, and RESAMPLE! My sample RAM 0041 FX:Threebee was built from this sound. You can hear this sample when you have loaded the samples that go with the demo-song.

Chapter 5. Creating Sounds From Strange Sources

By now, you should have a fairly decent grasp of '3 Level' editing as pertains to break-beats and internal sound sources, but how about using it for other purposes? '3 Level' editing can be your absolute best friend when it comes to building unique sounds. As long as you have no preconceived notions of specifics (i.e.. melody or rhythm), the sample will lead you by the ears to where it wants to be.

Let's start with a sample, any sample of anything, although if you are striving for maximum expression, using a sample with a variety of sounds within it works best. Because tunings are pretty irrelevant, interesting rhythms become more important. Sometimes the best rhythms aren't planned, they just make themselves apparent during the sound creation process. For this example, I will use a phone message from my friend Chris, calling me from beautiful Vancouver, Canada (Sample 0021 Vo:Hey Buddy).

***Load HEYBUDDY.S1M off of disk Disk #2.

[Insert Disk#2]>> DISK>> F2(LOAD)>> F4(WAVE)>> ENTER>> INC/YES

You can hear that it's not a very good recording. As a matter of fact, it's recorded by putting a microphone directly up to the phone and sampling. In this case, good sample quality is irrelevant...by the time we're done with it, it will be unrecognizable.

Again, let's start by placing our sample, as before, into an initialized FDSP voice (I1-020 Vo:Phone Call). For this example, I've used FDSP algorithm 10: Seismic, which acts as a massive compressor, with overdrive. I've done this to bring out all of the little breaths, clicks, pops, and general noise in the sample. Within the DCF, I've set the gain (level boost) quite high, the cut-off frequency quite low, and the resonance is sort of mid-range. This is to help achieve that sought-after massive bass rumble. Sometimes setting the resonance to high or the cut-off too low, to early in the process, eliminates frequencies that you may want later. For this reason, you'll notice that I haven't filtered all of the treble out yet. Using controller set 1, I've assigned FDSP Overdrive to react to aftertouch to add a little bit more randomness, right away. Remember, for bass-lines like the one we're building, it's all about experimenting with sound, so the more opportunity you give for sonic surprises, the more unique your sound will be.
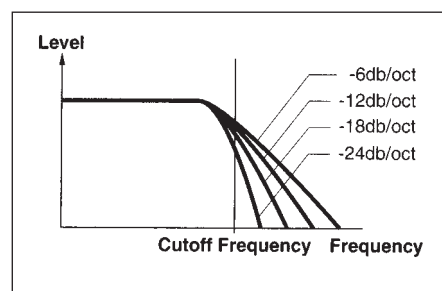
Because we are attempting to mangle the sound into something new and unpredictable, let's use some effects. Waves that are being processed by FDSP function as one wave, SO remember that you cannot send 2 sounds that are going through the FDSP to two different effects. To mess about with this sound, I've used that insane 'Beat Change' Effect (InEF #65). The 'Beat Change' Effect is yet another engineering marvel from the geniuses at Yamaha. Every other machine I have ever worked on requires at least a few seconds to calculate time-stretching. How do they make it happen in real-time? My guess is black-magic. 'Beat Change' can be used for two main purposes: changing tempo without affecting pitch, and changing pitch without affecting tempo. In this case,

I've used it to speed up the sample by adjusting the actual Beat Change parameter (within the Beat Change effect) to +32, and to adjust Beat Range (the maximum value, in semitone increments, that the Beat Change parameter will reach). So, because Beat Change is set to +32 (half of the maximum, which is +64), and Beat Range is set to 12, the entire phrase is sped up by 6 semitones (half of the Range setting of 12). Beat Range allows you to set the maximum change value (100%) that the Beat Change will reach. At this point, you must be saying "Hey...if you wanted the beat sped up by 6 semitones, why didn't you just set the Change to +64 (maximum), and set the Range to 6?" Well, my inquisitive friend, that is because although initially it would sound similar, I wanted to have the ability to manipulate the sound with a controller to a specific value, in a specific way. If you go to the controller set page, and look at set #2, you'll see that I have assigned MW1 to affect Beat Change at a depth of -12. Translation: moving the Modulation wheel (MW1) up all the way will lower the tempo by -12. The -12 is sort of a percentage of the Beat Range. I say sort of because it's not actually -12%, it's really a Beat Change value, which is relevant only to the number that you have set in the Beat Range. If you want to speed your sample up to a specific tempo, use Beat Range to set an approximate speed, and Beat Change to fine tune it. And, as with any voice you intend to resample, set your SCF to boost the level 12dB.

After you have resampled the sound, try putting it into another FDSP voice. If you are ever dealing with a sampled source, including AWM sounds, and you feel a need for randomness, or just that little extra edge, FDSP is always a great way to go. Let's examine (I1-021 Voice Bass 1). Now, what I've done here, is taken my re-sampled phrase, (RAM 0040 Vo:Hey Budd2) and once again, placed it into a Seismic FDSP voice. Again, my reasoning for using this algorithm is that I want the bass sound to be massive, and I want the speech like quality to be apparent but unearthly sounding. Because we've resampled the phrase from the last step, our source is already has the effects from the last step, so we are free to use up our FDSP processor, as well as two insert effects. I've opted to not add any more overdrive to the basic sound, as I want to keep the sound bassy and rumbly, and adding overdrive increases harmonic content, especially in the treble regions. I have, however, made a controller set that uses aftertouch information to increase FDSP drive, NOT overdrive, very subtly, just to add more dynamic control. Drive describes how intensely the FDSP processor performs its over-all function. And, to make sure it doesn't get to out of hand, I've thrown a LPF18 in as a DCF. By using the LPF18 as opposed to, say, the LPF24, I've made it a more gradual filter, that affects a greater range, without REMOVING as many frequencies.
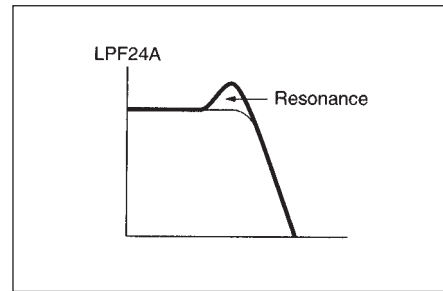
LPF (Low Pass Filter)
The LPF passes a signal below the specified cutoff frequency and cuts the signals above it. There are five different types of Low Pass Filters for the DCF.
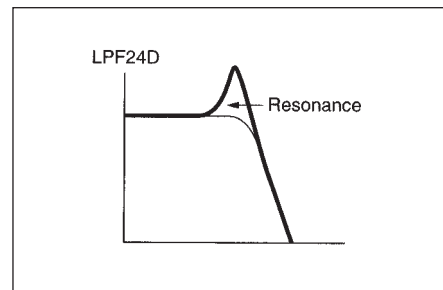
LPF 24A (Low Pass Filter 24A)
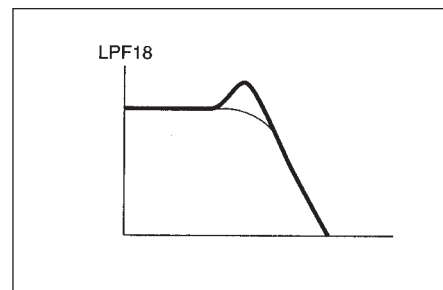Digital dynamic filter similar to a 24dB/oct LPF on a
analogue synth.

LPF 24D (Low Pass Filter 24D)
-24dB/oct low filter-type digital dynamic filter with
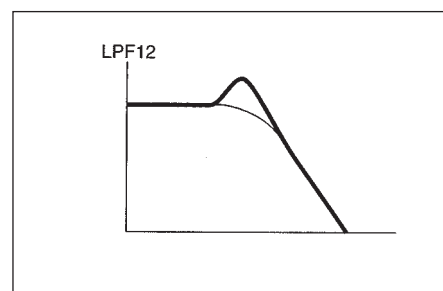pronounced resonance.

LPF 18 (Low Pass Filter 18)
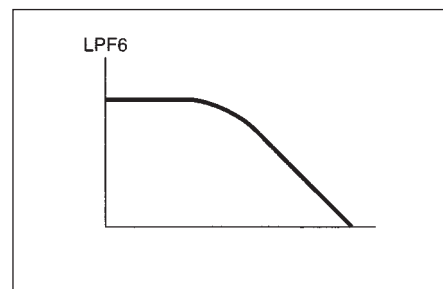-18dB/oct low pass filter-type digital dynamic filter

LPF 12 (Low Pass Filter 12)
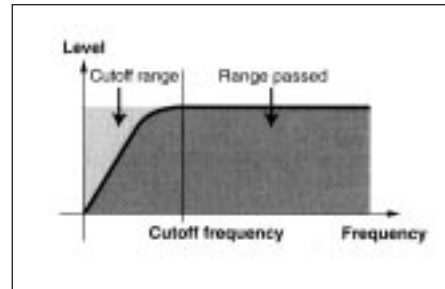-12dB/oct low pass filter-type digital dynamic filter.

LPF 6 (Low Pass Filter 6)
-6dB/oct low pass filter-type digital dynamic filter
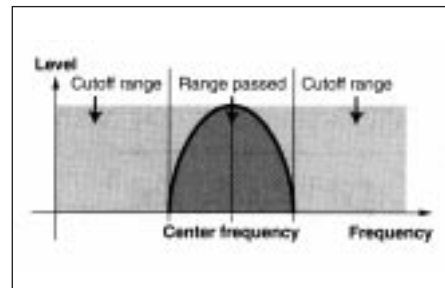with no resonance.

HPF (High Pass Filter)
The HPF passes the signals above the specified cutoff frequency and cuts the signal below it.
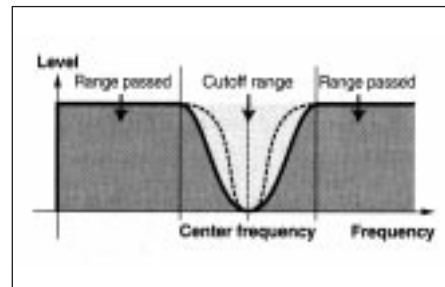

BPF (Band Pass Filter)
The BPF passes the signals around the specified cutoff frequency and cuts other signals.

BEF (Band Elimination Filter)
The BEF filter cuts the signals around the specified cutoff frequency and passes other signals.

Using a controller set, I've also assigned the MW1 to the reverb send, so that upon initial playing of the sound, you wont hear any reverb, but as you push the MW1 up to its maximum, reverb increases. In the case of bass sounds like these, a constant reverb can make the sound too muddy. But in certain situations, it can be quite useful to emphasize notes, in the true dub-style. And speaking of dub-style, don't be afraid of maxing out the parameters of your effects. Just make sure to set up your controllers so that you have options and randomness. There's that word again...

Now, let's take a look at (I1-022 Ba:Voice Bass 2). Take a second to reflect and compare the differences between the 2. One of the first things you'll notice is the presence of a new wave...PRE0229 Wv: MogBs2. I've put this in to add the presence of a true sub-bass, that is, a simple pitched wave that hit frequencies that are so low that they sometimes cant be heard, as much as felt. Generally, a simple low pitched sine-wave is a sure-fire way to make the club-kids freak out. I've filtered this wave quite heavily, so that you hear just the low end. Now we have the unique, bizarre bass phrasings of the sample to groove on, with the kick-in-the-arse sub-bass made from a simple wave, just to reinforce the point.

Because the Moog sample has quite a percussive attack, I've used the filter envelope to quickly but gently ease the sound in.

Again, the greater the number, the slower the Attack. The filter envelope attack acts like any envelope, by allowing you to set up a series of values that the filter will follow. By setting the attack time, you cause the filter to start at 0 (no sound is heard, because it's all filtered out), and then gradually or rapidly raise the level of the filter until the sound can be heard.

To give these sounds more consistency and presence, I've sent them both to a Compressor, using InsEF 1. (I1-023 Ba:Voice Bass 3)

The two waves that we've built in the OSC Edit pages, plus the compression that we've added, give the sound a slow, rumbling release, that stops the sound from being so dry. Take a look at the settings for the compressor.

I've given the compressor a 6 ms attack time, meaning that after a sound passes the threshold (which is also editable), the compressor will wait 6 milliseconds before it lowers the volume on the sound. By leaving it for 6 ms, we end up forcing a more percussive envelope onto the sound. The release time, of course, refers to the amount of time before it stops compressing, and Ratio is the value (in decibels) that the compressor will change into 1 decibel, for every decibel that goes over the threshold, which sounds much more confusing than it really is. For this example, the Ratio is set to 3.0, meaning that if a sound passes the threshold by 3.0 decibels, the compressor effect will turn that sound into being 1.0 decibel over the threshold. This can also be expressed by saying that any sound that crosses the threshold, does so as 1/3 of its volume. If a sound crosses the threshold by 6.0 decibels, then the compressor turns it into 2.0 decibels, regulating the general amplitude of a signal. If the ratio was set to 10, then the compressor turns 10 decibels into 1, making each threshold-crossing sound 1/10 of its level. This can take awhile to understand, but compression can be used in many, many important ways. If you are striving to reach that huge trip-hop snaredrum sound, set your threshold low, your output high, and use a short attack and long release. To keep your bass lines pumping, use a slightly longer attack, like we've done here, to keep the sound percussive. I promise that if you play around with it a lot, you can learn to make it do what you want it to do. The sustain that you hear at the end of each note has been achieved by boosting the output level of the compressor to the maximum, but keeping the threshold low.

Generally, if I find that I need a more common effect, like a compressor, I'll use InsEF 1, saving InsEF 2 for some of those more off-the-wall effects. And speaking of which, try turning the MW1 up on this sound. What the...?! Ahhh, Jump Wave, will you ever cease to amaze me? Now, turn to the CTRL set-up, and you'll notice that the MW1 is doing several things at once. Set 1 assigns it to Dst: 028:EF2 Dry:Wet. In the Effect 2 set-up, the Dry:Wet balance is set to D63>W, that is, you are hearing only the Dry sound, and no Wet (affected) sound. By moving the MW1, you are changing the balance between completely dry and completely effected. Set 2 still uses the MW to control

reverb send, so now with a simple wheel movement, you get a dubby note flutter/plate reverb combo that would make Lee Perry jealous. And finally, as the Mod Wheel raises the Reverb send and Jump Wave balance, Controller Set 3, is slightly lowering the overall volume, to make the effected sounds more distant and less obtrusive in the mix.

'Granvillians (I1-024)' is a further development using the same basic setup. There are a few differences, the first being the addition of a new wave: PRE 0371 Dr:T8Bd. This sound replaces the Moog sample in the lower end with a punchier, nastier, bouncier hit. Also, because it is a percussive sound, it has a fast initial attack, and is gone before you know it, so it is easier to pick out the movement of the initial sample of Chris. The Moog sound has been assigned to a specific zone:

VOICE>> EDIT>> F2(OSC)>> F8(ZONE)

Assigning a Zone simply means that we have limited the notes that trigger the specific wave to a certain range. In this case, we have set it up so that PRE 037 Dr:T8Bd is triggered by C-2 up to B1, and PRE 0229 Wv:MogBs2 starts at C2, and can be played all the way up to G8. Although it is quite possible to overlap sounds, in this example, we have set it up so that T8Bd and MogBs2 don't ever sound at the same time. There is also a pitch envelope on MogBs2:

VOICE>> EDIT>> F3(PITCH)>> F8(EG)

The pitch envelope on MogBs2 has been set up so that once the sound has been triggered, it begins to slide it downwards...very, very effective when heard through big speakers. As well, a few of the parameters in the Jump Wave have been changed. The Depth has been increased, so that the effect will change pitch more, and the Speed has been lowered, so that the Jump Wave takes longer to go through these pitches.

The Jump Wave itself is worth discussing, because it represents some serious ingenuity on behalf of the engineers. As you can probably tell, the Jump Wave effect examines the wave, and, based on a few parameters that you can set, will access the frames of the wave in a non-linear manner, much like a compact disc skipping. The effect allows you to set several parameters, most of which are self-explanatory, some of which are not. The only parameters that aren't really obvious are 'Type', 'Jump Wave', and 'Resolution'.

To make this effect easier to understand, imagine that it is forcing a sequence (which it is) on to the frames of your sample. In a nut shell: Type defines which notes or destinations the sequence will use, Jump Wave tells the EX how to travel from point A to point B (i.e.. step, slide, reverse) , and Resolution controls the base length of the sequence.

If you take a listen to (I1-030)-(I1-041), you can hear all the variations of Type and Jump Wave, as per their names. I have set both the depth and speed low enough for you to hear exactly what's going on and what the differences are. Although the Jump Wave effect isn't syncable to MIDI, there are ways to make it rhythmic. The most successful way to do this is to sequence a part, for example, a bass-line using (I1-024 Ba:Granvillians). Record a sequence of the note-pattern, and then, using the MW1, sequence the point where you want the jump to occur. If you're anything like me, then you might want to mess around with your sequence a bit, quantizing or tweaking as the case may be. Once you have a groovy little bass riff, set your sequencer up to loop that phrase.

While the phrase is looping, tweak the parameters of the Jump Wave to be as close to your desired result as possible. Then, using the EX's internal sampler, record several passes of the loop. Remember...turn the SCF up high and the output volume low. Have a good listen to your samples. There may be several of them, and it may take several tries before you get what you are looking for, but there will definitely be something unique and usable. Once you have found your phrase, trim it down and extract it as before, getting rid of all the unnecessary sample data.

Chapter 4: Song Preparations

***Load TEXTURES.S1S off of disk Disk #2.**

[Insert Disk#2]>> DISK>> F2(LOAD)>> F6(SONG)>> ENTER>> INC/YES

***Load SONGSAMP.S1M off of disk Disk #3.**
(You will need more sample RAM memory than standard – wave file size is just under 1.4MB)

[Insert Disk#2]>> DISK>> F2(LOAD)>> F4(WAVE)>> ENTER>> INC/YES

By now, we have collected many samples and interesting sounds. In my humble opinion, now would be a good time to start organizing sounds in a more usuable manner. To organize your sounds in preparation for sequencing, follow the steps to set up a Drum Voice, but instead of using just drum samples as your waves, use the various samples that you have built from your '3 Level' Editing sessions. If you take a look at (I1-128 Sq:t-EX-tures), you can see an example of one way of organizing; by song. I have placed all the non-drum samples that I want to use in my song into one Drum Voice, with no effects at all. No effects are necessary, because we have built these samples by resampling other samples that already have effects on them. Another way of organizing samples is into Drum Voices that contain similar samples. I have assigned all my drum-loops to their own Drum Voice: (I1-127 Dr:Loop Kit+12). This patch is identical to (I1-002 Dr:Loop Kit), except that I have placed a +12dB boost into the SCF, to make my levels more consistent with the other sounds in my song. Again, there are no effects being used here, because these sounds have already been effected before they were sampled. Because we've successfully created 2 patches that don't need effects at all, we have given ourselves more options when we switch over to Performance Mode, which you can now do, simply by pushing the PERFORMANCE button.

Performance mode is the multi-timbral mode of the EX series. Even though Performance mode allows you to play more than one Voice at once, it is important to understand that you still have the same number of signal processors as you do in Voice Mode. Translation: you are limited to one complex (AN, FDSP, or VL) voice type at a time, while you can layer many simpler voices, like AWM or Drum voices. You still only get 2 Insert effects*, so you have to be very strategic about what voices you choose for your Performance set-up. Now, apt students aren't really going to experience any problems here at all, because we have been sampling as much as possible, and the EX can deal with samples better than Madonna deals with costume changes. Problems and frustrations only arise when you are attempting to get the EX to use more effects or complex voice types than it can handle.

'Performance 001 —:Textures' is the performance set-up for the cute little demo-song that has been included. Although this may not be an award winning tune, it has been very carefully set up to show you an example of how you might approach the task of multi-timbrality on the EX. Go into Performance edit mode:

*The EX7 allows for 1 part (voice) of the performance to have its full insert effects if all parts are AWM or sample based. The EX5 offers up 4 full insert effects in this manner.
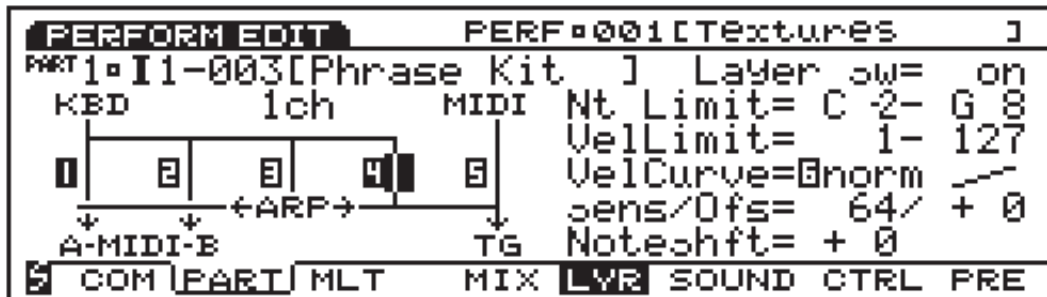
You can see that voices 1, 2, and 3 all are Drum Voices (although only voices 1 and 2 actually use drum sounds). Although this isn't apparent in the Performance Edit screen, you can confirm this by going back into Voice mode and examining the corresponding voices: (I1-003 Dr:Phrase Kit, I1-127 Dr:Loop Kit+12, I1-128 Sq:t-EX-tures) Voice 3 is simply the Drum Voice, t-EX-tures containing all of the samples that we have built from resampling complex voices. Remember, once you are in Peformance mode, you cant really edit the sounds with the same detail as you could in Voice mode, so, if you wish to edit specific elements of a voice, go back into Voice Mode, edit the Voice you wish, and return to Performance mode. The EX will remember the changes you have made to the voice once you have returned to performance mode.

Because none of the voices so far are complex voices, and none of them use Insert effects, we are still free to use both, which I have, in Voice 4 of the performance set-up. Voice 4 is the voice that I have designated as the complex voice. In other words, I have decided to use this voice for any part that doesn't lend itself to sampling. If you press PLAY on the sequencer panel, and listen to the first 16 bars of the song, you can hear that the sounds appears to be quite abstract. You are actually listening to only two main patches. The first is (I2-019 Fx:Luddite Rain), a very complex VL voice with two Insert Effects, assigned to Part 4 of the Performance Set-up. The second sound is actually a sampled loop that is a built from (I1-052 —:Threebs), resampled, and placed into (I1-128 Sq:t-EX-tures). Because we have used up our 1 complex voice and 2 insert effects, the rest of the sounds, should you still need them, can only be Drum Voices or AWM voices. The individual sounds will still react to controller data without affecting each other, so if you feel that your un-complex sounds are dry, or lack movement, then add some extra dynamics using controllers, or edit the voices themselves, using the available groovy maneuvers (LFO, Filter EG, etc.). If you do edit your voices, be sure to save them. If the voice you are editing is a Preset EX voice, then simply perform the functions to Store to a user location (I1 or I2). ***This will erase any data that was previously in this location***.

In Performance mode, the EX assumes that you want each different performance layer to be controlled by the MIDI channel that corresponds to the layer number. For example Voice 1 receives MIDI information that comes in on Channel 1, Voice 2-Channel 2, and so on. Because this may not always be the set up you desire, you can change this quite easily:

This screen shows you a flow chart that describes the path of the MIDI information as pertains to the selected layer. To change the layer that you are editing, move the cursor all the way up to the top left field, and use the dial to change the Part number. To select the MIDI channel that the part will receive, move your cursor to the field that is just below the Voice Name, to the space that says 'ch' with a number before it. The number that comes before the 'ch' is the MIDI channel that will control the selected layer. You can have all the layers receive the same channel, but that means that all channels will be playing the same MIDI information, including controller data.

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┌────────────────────────────────────────────────────────┐
│ ▆PERFORM EDIT▆        PERF▪001[Textures        ]       │
│ ᴾᴬᴿᵀ1▪I1-003[Phrase Kit   ]  Layer sw=    on          │
│  KBD        1ch      MIDI  Nt Limit= C -2- G 8          │
│                           VelLimit=    1- 127           │
│ ┌─┐  ┌─┐  ┌─┐  ┌─┐┃ ┌─┐   VelCurve=▆norm  ̄⁄ ̄           │
│ │1│  │2│  │3│  │4│┃ │5│   sens/Ofs=    64/ + 0          │
│  └   └  ←ARP→  ←┘   └     Noteshft= + 0                 │
│  ↓    ↓           ↓                                     │
│ A-MIDI-B            TG                                  │
│ ▆5 COM│PART│ MLT    MIX ▆LYR▆ SOUND CTRL  PRE          │
└────────────────────────────────────────────────────────┘
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This move is quite effective for building large pads, or combo voices, like that cheesy piano-strings thing, or a heavy Floydian lead of some kind. And don't ever, ever forget that you can resample Performances, no matter what your Performance consists of. If you can reduce your entire song to 4 or 5 groovy loops, then you are well on your way to becoming a hip-hop artist.

Well, here we are...the end of the journey. Because the EX is so very versatile, of course you understand that all of these techniques are here as guides and basic tutorials only. Throughout this process, hopefully you have discovered not only new tricks and techniques, but also tangents that you will be inspired to explore. If you should discover something cool, and you wish to share it, by all means, hop on the Internet, trade patches, share techniques, and help the EX community grow!

Happy EX-ing!

Appendix A: Internet Resources

The Internet has become an invaluable tool for those of us with a taste for tiny machines that make huge noises. There is a crowd for every interest that you may have. Updates are easily made and frequent, so as a disclaimer, firstly, I'm sorry if these sights aren't there when you look, and secondly, I would like to say that neither myself nor Yamaha Corporation are responsible for the content of these sights. With the exception of the Yamaha Web sites, of course. Then you can blame them.

http://www.audiophile.com/alephx/
A personal website that contains links to many EX sites, patches, and FAQs.

http://www.boldersounds.com/
A pro site with news, links, samples, and MP3s.

http://www.harmony-central.com/
A music industry site, with a little something for everyone, industry links, product and event reviews.

http://www.jansz.com/
A personal website that includes EX links, and a searchable EX clinic for all your EX questions.

http://www.ontology.com/
A personal website. Contains Real-Audio, samples, links, and generally inspirational nuttiness.

http://www.prorec.com/
As the name suggests, a pro-recording site with current info on products and techniques.

http://www.samplenet.co.uk/
An excellent site for samples. They are always updating, and they probably have exactly what you are looking for.

http://www.sonicstate.com/
Contains industry information and listings, and loads of tips about tons of different machines

http://www.staccato.com/

http://www.synthzone.com/
A great site with detailed descriptions and well catagorized information about virtually every synth ever made. Tons of links to samples and patches.

http://www.teklab.com/
A cool site made by cool people who know how to do cool things with cool machines. Home of the legendary A3K Central.

http://www.vrsound.com/
Aside from having a nifty title page, they offer great EX native CDroms and sounds.

http://www.wbmedia.com/ex5/
The quintessential site to discuss all things EX.

http://www.yamaha.ca/
Home of Yamaha Canada. Polite and informative, just like all true Canadians.

http://www.yamaha.com/
Yamaha's American Website.

http://www.yamaha.co.uk/
The European Division's Home. Easily the funkiest web page owned by any major corporation.