

# Emax Clinic

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## 1 Introduction

There were days when I would show up to my office at E-mu Systems and not get any work done. Not because the latest CD was blasting out from a co-worker's office, and not because the service department was celebrating the day with bagels and cream cheese topped off with champagne. It was the damn phone. I knew each ring was a cry for help, and only the most heartless soul could ignore the desperation behind the call. Who were these people that filled my days with uninterrupted interruptions? Emax users, that's who!

Some of the calls were from people who didn't understand the manual; some were from people who had not bothered to read the manual; and some were from people who were really trying to tap the power of the Emax and just needed some help finding the right key to open the door. It is to those brave, creative explorers that this clinic is dedicated. (If you're one of those people who has not read your manual, we suggest that you use it in conjunction with this clinic, which is written with the presumption that you have a basic knowledge of Emax operations and terminology.) We'll explore both basic and advanced functions, utilities and applications. Some of them will probably be brand new to you; others may be in your regular repertoire of Emax tricks. All of them, unless otherwise noted, can be performed using software versions 3.2, 1.1 HD, and SE. The clinic follows a general progression, and often

a given function requires knowledge of other functions described earlier. We suggest that you read this clinic with your Emax in front of you, so that you can gain a greater understanding of the processes you will be performing.

There's no place like the beginning to start.

## 2 Sampling

### Setting The Record Level

Want to take a noisy sample? Of course not, but that's exactly what you'll do if you simply enter the Sample module and start sampling. E-mu uses various criteria when determining the parameter default values on their instruments, and in certain cases they subscribe to the "show-the-highest-value-so the-user-will-know-what-it-is" school of thought. That's appropriate for a number of situations, but not for the sample input gain parameter. If you take a sample without first changing the default value of the input preamp, you'll be out before you get up to the plate; one quick value change and you've got a good chance of getting to first base.

Begin by clearing the memory (Master 4), then create a preset (Preset Management 3). Enter the sample module and select VU Mode/Gain 1. Aha! There's the culprit. The LCD shows +40dB in the upper right-hand corner of the screen, indicating that the signal level to the sampling circuitry is being boosted 40dB by the Emax's

internal preamp. Sorry to say, this preamp isn't the world's most quiet, and when it is set for high gain it adds an enormous amount of noise to the sample. Use the data slider to select +0dB. If the proper input signal level can't be attained with this setting, you should make the necessary adjustments at your mixer. If you don't have a mixer or some other way of boosting the output of the source signal, you will have to use the Emax preamp, but we recommend that you do not set it past +10dB.

Now the question arises, what is the proper signal level? The idea is to record the input signal hot enough to maximize the signal to noise ratio but not so hot as to induce distortion. If you look at the LCD from just the right angle you will notice that it contains 32 boxes, 16 in the top row and 16 in the bottom row. Each box is five dots (pixels) wide and seven pixels high. When a signal appears at the sample input, its level is displayed on the bottom row of boxes by the peak/hold bar meter moving left to right across the LCD. Peak/hold metering is used on the Emax so that attack transients may be accurately monitored. To ensure that an optimum record level is attained when using this type of metering, start with the source's output level very low, so that any high-level transients the sound may contain will not overshoot the top range of the screen. Gradually increase the source's output level until the loudest part of the sound causes the bar meter to fill the bottom of the screen except for

the last row of pixels in box 16. If the last pixel is filled, there is a good chance that the sample will be distorted. Conversely, if the pixels in box 15 and 16 are not filled, the input level is too low and the sample will be unnecessarily noisy.

### 3 Digital Processing

#### Blind Truncation

You may not have the luxury of working with a computer and visual editing software, but that doesn't mean that you can't easily and accurately truncate your sound files. Why truncate? For one reason, there is often dead air at the beginning of a sample, which must be cut so that the sample will trigger without delay. (Samplers are often accused of having slow MIDI response times, when the delay is actually the result of a poorly truncated sample.) The problem is that this dead air is usually only a few milliseconds long. How can you tell if you need to cut 1.75 or 2.14 milliseconds, and just how many samples is that anyway? A wonderful technique suggested by E-mu sound department manager Kevin Monahan will enable you to perform exacting truncations like this by ear.

Enter the Digital Processing module, choose the sound you wish to truncate (try one that you suspect has dead air at the beginning), and select Truncate 1. The LCD shows truncate start at 000000 (the beginning of your file), and truncate end shows the number of the last sample in your file. Move the cursor to truncate end and move the data slider all the way to the

bottom so that the end number becomes 000004.

Your sound is now four samples long. But don't worry; truncation on the Emax is always temporary until you answer yes to "Make Truncation Permanent? Y/N". Start playing the sound repeatedly - you shouldn't be hearing anything since you are presumably playing dead air -and at the same time keep your finger on the Yes button. The truncation end numbers will gradually increase as you scroll through the sound sample by sample, until at some point you hear a click at the end of the sound. The click is the sound of a single sample being played. At this point, play the sound again but this time press the No button - only once, don't hold it down. Play the sound over and over, hitting the No button in between each re-keying of the sound. Continue this process until the click goes away. You will eventually reach a point where a given end number plays silently, while the next higher number makes a click. This click is the start of the sound, and all the samples before it are the dead air.

Whatever you do now, don't press Enter. If you do, you will truncate all of the sound and wind up with nothing but a file full of dead air. First, memorize the end number that does not play with a click. Next, move the data slider all the way to the top so that truncation end displays the last sample in the sound file. Now move the cursor to truncate start and, from the keypad, enter the number you memorized. (Remember to include all the zeros e.g. 000123.) Press Enter to make the truncation permanent.

This process can also be used to truncate space at the end of a sample. Simply follow the same steps exactly upside-down, backwards, and inside-out. Er, use truncate start to find the first sample that does not make a click, enter

that number for truncate end, and return truncate start to zero. If your sound is very short, soft, or non-percussive, making it hard to hear the click, you may find it useful to assign the sound temporarily to the whole keyboard (Preset Definition, Edit Assignment 2) with original pitch placed on C6 and the low note set to C1. This will allow you to play notes considerably lower than original pitch when you are listening for the click. Due to the extreme transposition, the sample will play back very slowly and the subtleties of the click(s) will be much more noticeable. Something to watch out for:

What you think is a click might really be the sound of the Emax VCA opening. If you are in doubt, you can eliminate that sound by setting the VCA envelope attack time to 3.

#### Dual Loops

The Emax allows two independent loops to be defined for each sample - sustain, which plays for as long as the key is held, and release, which plays when the key is released. These loops can be used very effectively for creating delay effects on sampled background vocals. (You do use your Emax for vocals, don't you?) Vocal sampling can come in handy both in live performance and in multi timbral sequencing, when outboard effects are at a premium.

Start by sampling the words "ooh baby I love you" into the Emax using the 28K sample rate. (We hate to give away lyrics that have hit-song potential, but what the heck.) Let's presume that you want to create delay effects that result in the vocals playing as "ooh baby, baby, baby, I love you, love you." Cool, huh? To hear the loops you'll be making, enter the Analog Processing module, select VCA 12, and set envelope sustain to 32 and release to 25. Return to the Digital Processing module and select Loop 2. Set the loop length to about 15,000 samples (the length of the word "baby"), and the loop start to approximately 10,000 (the point in the sample where the word "baby" begins). You may need to use different values depending on the length of your sample. You have selected the proper loop length and start point when you hear "ooh" followed by the word "baby", and "baby" repeats for as long as the key is held down. Fine tune the values as necessary to achieve the desired feel. Now press Enter and answer No to "Autoloop? Y/N", and No to "Truncate After Loop? Y/N". (If there is a glitch in your loop, you may want to use autoloop to eliminate it, but there is a risk that the loop length will be altered and consequently the rhythm of the delay.)

To create the second part of the delay, select Release Loop 3, and answer No to "Same As Sustain Loop? Y/N". Set the loop length to about 20,000 samples (the length of the words "love you"), and the loop start to approximately 25,000 (the

point in the sample where the words "love you" start). Again, you may need to use different values depending on the length of your sample. Remember that this second loop will be heard only after the key is released.

Once you have selected the proper loop lengths and the start points, the sample will play "ooh baby, baby, baby" when the key is held, and "I love you, love you" when released. Experiment with different VCA envelope settings to control the number of repeats and their relative volumes.

### Digital Flange

It's quite easy to create phase and flange effects by assigning the same sound to both the primary and secondary layers of the keyboard, but each time a key is played it uses two voice channels, and the Emax doesn't have enough of those to waste. The sound itself can be flanged digitally, which eliminates the need for an extra voice channel. Flange created this way will be static, rather than dynamic, but try it, you'll like it.

Choose the sound you wish to flange (for this example use a sound sampled at the 28K rate) and copy it to another preset (Preset Definition, Copy Voice 1). Enter the Digital Processing module, select the newly copied voice and answer Yes to "Used Elsewhere Copy First? Y/N". This prompt only appears when the same sound appears in two or more presets. It is important to answer yes to the prompt; otherwise changes made to the copied file will also be made to the original. Now select Truncation 1 and enter 000028 for the start value. Press Enter and make the truncation permanent. Select Combine 7 and enter the number of the preset that contains the original sound. Select the original voice, press Enter, and answer Yes to "Combine Changes Data! OK?" When the process is

complete, your copied sound will be flanged.

Some simple math will allow you to predetermine the amount of flange that occurs as a result of truncating the sample by different amounts. Take the sample rate and divide it by 1000; the result equals one millisecond of time. In the above example, the 28K sample was truncated by 28 samples (one millisecond). If our example used a 42K sample we would have truncated 42 samples. To familiarize yourself with the varying tonalities that are produced by different time values, we suggest that you experiment with a digital delay that is capable of displaying milliseconds. If you like the flange produced by a 4ms delay, you would truncate 112(28 x 4) samples from the sound and combine it with the original.

### Digital Chorus

A technique very similar to the one used for flanging can be used to digitally chorus a sample. (SE software only.) The primary difference between the two techniques is that flange is created by combining two identical sound files that are offset by a small amount of time; chorus is created by combining two identical sound files that are offset by a small amount of pitch. Begin by copying the sample to another preset, enter the Digital Processing module, and answer Yes to the copy prompt. Now select Digital Effects 9, followed by Change Pitch 4. The screen displays 00.00 and the cursor flashes under the zeros to the left of the decimal point.

Semitones are displayed by the zeros on the left, cents by the zeros on the right. (One semitone = 100 cents.) Move the cursor to the zeros on the right side of the decimal and use the data slider to enter a value of 12 cents. Very pleasing chorus effects can be attained by detuning a sound nine to 15 cents.

Detuning less than nine cents is generally too subtle to create chorusing, and over 15 cents takes the sound into honky-tonk land. Press Enter and take a deep breath before you read the next screen, which warns you that it could take five to 15 minutes to compute the pitch change, depending on the length of your file. Be brave, press Yes - the results are worth it. Once the sample's pitch has been changed, combine it with the original, and if the preset contains multiple samples, repeat the process for each of the remaining files.

When this technique is used exactly as described above, the detuned sound file will be 12 cents sharp to the original. If you wish for the detuning to be flat, first enter - 01 in the semitone column, then +88 for the cents value. We can see the LED's flashing in your brain right now. What if I flange *and* chorus a sound? Now you've got the idea. Did you think the E-mu engineers put all of these functions into the Emax just to demonstrate how adept they are at writing code?

### Digital Pitch

The Change Pitch function (SE software) is wonderful for musical applications like creating digital chorus, but it

is also helpful in a number of utilitarian ways such as matching dissimilarly pitched samples for splicing and/or combining. It is extremely useful for extending the tunable range of a sample, especially since the Emax is only capable of limited upward transposition. A sample can, however, be transposed down six octaves. If the original pitch of a sample is too low to be transposed up to a specific note, simply retune the sample digitally to the appropriate pitch, then transpose down as far as is necessary to cover the desired range. Although the retuned sample will be subject to "munchkinization" (play Bruce Springsteen at 45 rpm and you'll swear it's Dolly Parton), the effect is no more pronounced than if the sample were transposed on the keyboard.

### Digital Gain

When a properly recorded sample is examined on a visual editing system, the sound's peaks should show an amplitude value of 100%, indicating that the full dynamic range of the Emax is being utilized. If the peaks are any lower than 100%, the maximum signal-to-noise ratio has not been attained. This can be corrected with the gain normalization function, which analyzes the sample and increases its overall amplitude to the optimum level.

The Emax doesn't have a gain normalization algorithm, but you can accomplish nearly the same effect using Change! Atten 1, found in the Digital Processing module's Digital Effects 9. Simply select the sample, enter the amount of gain you wish to add (shown on the display in dB), and press Enter.

Start by adding 1dB of gain. If the sample sounds clean, save it to a new disk. (Always make sure a sound has been saved to disk before doing any digital operation,

as the data will be permanently altered and you may need to recover the original.) If the sample is distorted, the peaks were already at maximum amplitude and in creasing them by 1dB caused them to clip. To ensure that the sample reaches optimum level, continue to add 1dB at a time, saving each gain-increased file until there is audible distortion. Virtually any sample can benefit from this technique. Drum samples can be greatly improved, and even boosting them to distortion levels may yield sonically pleasing results. Special note to those of you anxious to create obnoxious sounds: A sample boosted by the maximum setting of +40dB re-defines the word fuzz-tone.

## 4 Preset Definition

### Dual Voice

In these days of multi-timbral sequencing, it is easy to encounter the need for more voice channels than the Emax can provide. Dual mode can be of great help when you need to stretch the Emax's eight-voice limit; two independent samples are routed through each of the Emax's eight voice channels, allowing up to 16 sounds to be played simultaneously. Some clever keyboard mapping, such as assigning velocity crossfade to the dual mode layers, allows one or the other or both sounds to play, yet still only use a single channel. This mode can be extremely useful for drums, which often have multiple sounds occurring on the same beat.

Each Emax voice channel contains a VCA and its envelope generator, a VCF and its envelope generator, an LFO, and a DAC. When two samples are played through the same voice channel they must share whatever parameters may have been assigned to these common ingredients. In other words, both samples will have the same filter setting, amplifier envelope, LFO speed, and stereo image placement. This is not necessarily as limiting as it might seem, because all other analog parameters remain independent. For example, both sounds will have their stereo panning speed controlled by the same LFO, yet one sound could receive pitch modulation from the LFO, while the other has its pitch modulated by velocity.

Dual mode can be used to apply a parameter template to any group of samples. For example, a map could be created wherein every group of five notes occupies a different position in the stereo field. This map would be stored as a preset and loaded into a bank (Preset Management, Load Preset 1) whenever it was needed as a template. (The map should contain extremely short samples due to the memory constraints involved with loading multiple presets into the same bank.) To apply the template to an existing group of samples, simply copy it using Copy Voice(s) 1 to the unoccupied layer (primary or secondary) of the preset you wish to affect. Select Dual Voice 4, and when the prompt asks which set of parameters you wish to use, select the template layer. Once the template has been imposed on the new group of samples, use Erase Voice 3 to remove it from the preset.

There are a number of complex parameters that are ideally suited for storing as templates, including microtonal tunings and output channel assignments. Just remember that all of the template's analog parameters will overwrite the existing setup, so be sure to use this process before you get into any serious tweaking of the preset.

### Stack Mode

Emax SE software lets you stack up to four consecutive presets and play them from a single key. The obvious applications include creating ultra-fat bass and solo sounds, but there are a number of other uses ranging from the subtle to the outrageous.

Using real-time controllers in this mode takes on a whole new dimension, since all presets, and consequently the functions of their controllers, remain independent in the stack. Take a preset (try a solo instrument such as a saxophone) and copy it (Preset Management, Copy Preset 5) to three consecutive preset locations, such as 12,13,14, and 15. Enter the Preset Definition module and select Stack Preset 0. The screen displays two numbers, both of which are the number of the current preset (in this case preset 12). Use the increment buttons to select preset 15 as the last preset in the stack. Next, select Realtime Controls 9, and on the first preset, assign right wheel to LFO to pitch. On the second preset, assign right wheel to Filter (be sure that the preset's filter setting is low enough for the effect of the wheel to be heard). On the third preset, assign right wheel to Panning, and on the fourth assign it to Level. Feel free to get as wild as you wish with the

analog parameters in each preset. (Try different velocity routings and assorted delay settings.)

Start with the wheel fully off and play a few notes. Gradually move the wheel up and listen to the effect. When the wheel is down, the sounds in the first and third presets are heard. As the wheel is moved up, the filter opens in the second preset, adding real-time filter sweep to the blended sound. Additionally, preset number four starts to be heard as its level increases, giving the overall sound a dynamic swell. As if that weren't enough, preset number three is adding animation to the stereo image. One of the more interesting uses of stack mode involves assigning not only different sounds to each preset, but also using the layers to build harmonic relationships. You might, for example, start with a bass sound on the first preset. The secondary layer of the preset could contain a few isolated bass notes in intervals a fifth or a tenth above the root. The second preset could be a synth pad in fifths on two layers. The third preset could have bells randomly assigned to the keys. The fourth preset - use your imagination. Coupled with real-time controller assignments, velocity cross-fades, and assorted analog processing, virtually every new note played will be some sort of timbral or chordal surprise. You could actually plot out very specific pieces of music with full chords, melody, and percussion played from single keystrokes.

## Copy Voice.

One of the most basic functions on the Emax also happens to be among the most powerful. Copy Voice(s) was designed primarily as a preset creation utility for copying sections of the keyboard from one preset to another. This is very handy for creating splits and doubles, but it has a lesser known application that unlocks the heart of the Emax.

When a sample is mapped to the Emax keyboard, the internal processor gives it a name (e.g. Pri. 5, or Sec. 18) so that both the computer and the user can keep track of it. When that particular voice is accessed in Analog Processing, the user need only select a note that contains the voice, and all parameter changes made to the note will automatically affect the entire note range to which the sample has been assigned. This default speeds up the process of changing analog parameters, because there is never a need to hunt for voice boundaries. But what if a voice covers a three-octave range and you want to process the middle octave without affecting the outer two? This can only be accomplished by creating a new voice. Heeeeere's Copy Voice(s)!

Enter the Preset Definition Module and select copy Voice(s) 1. The Emax asks you to enter low and high key boundaries, which define the voice you wish to copy. Select the desired range and press Enter. Now for the really hard part; keep pressing Enter after each of the copy prompts until the menu returns to the module identifier. The voice has now been copied to itself, but the Emax is fooled into thinking that there is a whole new voice. If you now look at the voice

names in the Analog Processing module, you will see that there is a newly numbered voice covering the range specified by the copy process. There is no limit to how many times this process can be performed. In fact, each layer of each key may be an independent voice across an entire 88-key range even if there is only one sample covering the keyboard. Additionally, groups of individual and overlapping voices may be redefined to become new voices. Creative possibilities abound when you take advantage of voice independence, and they are all accomplished without using additional sample memory.

## 5 Analog Processing

### Stereo Imaging

The Emax offers a number of ways to dynamically manipulate the stereo placement of a voice: velocity, LFO, and all MIDI real-time controllers. The LFO is probably the most widely used panning modulator as it has the advantage of being able to set the tempo of the panning. This is useful not only for matching the rhythmic character of the music but - of all things - wonderful for disguising loops. Pan a sound at the tempo of its loop, and *voila!* The once eminently detectable loop blurs into a mesh of swirling stereo images.

String sections benefit immensely from dynamic stereo imaging. Start by using the Copy Voice function outlined above to divide the keyboard into ten sections— half an octave per section. Assign each note group to a different point in the stereo spectrum using Panning 19. Next, assign different amounts of LFO to pan

(LFO 16) to each group. (Be subtle - use small values. The idea is to add life and realism to the string sound, not create stereo hell.) Use assorted LFO speeds (LFO 15) and a small amount of the Variation parameter, which will change the LFO speed slightly with each new keystroke. LFO delay can be used to add a nice touch, especially if the LFO is also being routed to pitch. Stereo doubling works wonders to enlarge a sound, and is especially effective for background vocals. Start by copying your sound to the secondary layer. Do not use Dual Voice mode after copying - this is a four-voice effect. Enter the Analog Processing module and select the entire primary layer. Press Panning 19 and move the pan value to hard left (-7). Now select the secondary layer and pan it hard right (+7). Next, press Voice 11 and move the cursor to the delay parameter. If you are using standard (non-SE) software, set the delay value to 1. If you're using SE software, try values between 3 and 7. You should hear a very wide stereo image, and the sound will appear to be considerably larger than before. If the sound is "left heavy" (apparently louder on the left side than on the right), select Voice 11 and attenuate the primary voice by 1 to 3dB.

Interesting effects can be added to this setup, including detuning either the primary or secondary voices and adding dynamic imaging. If you want to have some real fun, detune one layer by 12 cents, then select both layers and choose a medium to slow LFO speed (15 to 30). Next, assign LFO to pan with a value of 15. The result: Cross-panned stereo chorus echo.

## Dynamic Chorus

We've explained how to create digital chorus effects using the pitch and combine functions. We've discussed layering and detuning sounds to achieve a similar effect. The Emax also has built-in chorus, analog parameter Chorus 20. No clinic would be complete, however, unless it demonstrated at least four ways to create chorus. The next method differs from the first three by the fact that chorus depth may be controlled on a key-by-key basis by velocity.

First, copy the primary layer to the secondary, then place the entire note range in Dual Voice mode. (It does not matter which layer's parameters you choose since the two layers are identical.) Enter the Analog Processing module and select the secondary layer. Select Velocity 17 and move the cursor to Pit (pitch). Enter a value between 1 and 3 depending on the depth of chorus desired. (Note: these values are for SE software. Standard Emax software [3.2 or 1.1 HD] does not offer the pitch resolution necessary for this effect.) Play a few notes with different velocities to determine the best velocity-to-pitch setting. If you are finding it hard to control the amount of chorus due to your particular touch, enter the Master module and select Velocity Curve 3. Try different curves until you find one that gives you the desired response. This curve setting, although accessed through the Master module, is stored as part of the preset.

## Creating Reverb

Run out of reverbs? Try creating the reverb right in the Emax. Start by copying the primary layer to the secondary. Dual

Voice mode should not be used, as each layer will have its own VCA envelope setting and will need to play through its own voice channel. Select the secondary layer and choose VCA Envelope 12. Set the attack time to 6 or 7, and the release time to 12 or 13. Hold, decay, and sustain times should remain at whatever values were programmed for the particular sound. Now select Voice 11 and set the attenuation parameter to 03dB and delay to 05. (Note: This setting is for SE software. Delay should be set to 01 when using non-SE software.) Lastly, the stereo images of the two layers should be slightly offset so that they can be differentiated. For example, if the primary voice is centered, the secondary should be to the left or right. A small amount of LFO to pan can also be added to the secondary voice.

If you think about how these specific functions are being used to simulate reverb, you should be able to easily fine-tune them to create the reverb effect you desire. The slow attack time works like a reverb's early reflection density control, and prevents the secondary voice from having a perceptible attack. The release time controls the length of the reverb's decay. The delay parameter functions as the equivalent of pre-delay, and the attenuation parameter controls the reverb's volume.

## Inverted Envelopes

The Emax is one of the few samplers that allows you to invert the filter envelope. This is essential for creating the classic horn "hit and swell" sound commonly known as sforzando. Start with your favorite horn sound, enter the Analog Processing module, and select

the entire voice range. (If your sound occupies both the primary and secondary layers, be sure to choose both in the voice select prompt.) Select VCA 12 and set attack, hold, and decay to 01, sustain to 32, and release to 06. Next, select Filter 13 and set the filter cutoff to 120. Move the cursor to Env (envelope amount) and set it to -30. Now select Filter 14 and enter the following values for the envelope: attack 22, hold 01, decay 17, sustain 01, and release 20.

Each part of the envelope may be adjusted to suit the particular sforzando effect you wish to create. Attack controls the length of the initial "hit" before the sound dips, and decay controls the length of the swell. A long release time is used to prevent the filter from opening up too quickly after the keys have been released. (Shorter release times can be used to create some interesting double-trigger effects when the keys are released before the envelope completes its cycle.) Sustain functions as a "brightness" control for the portion of the sound that follows the swell. The lower the sustain value, the brighter the sound will be. Envelope amount controls the depth of the overall effect. It is recommended that you set up the shape of the envelope with the depth set fairly high (-30 to -50), then back off the depth to taste.

## 6 The Arpeggiator As Drummer

Very few instruments these days have built-in arpeggiators, and as a result, very few people learn how to use them to their full potential. The Emax sports an "intelligent" arpeggiator, meaning that it is programmable and has a number of sophisticated parameters. It can be programmed to generate a simple repeating pattern or a series of complex rhythms and chords. Using the arpeggiator in conjunction with some of the stack mode setups described earlier can result in music that ranges in style from rigidly controlled to totally random.

It is also possible to use the arpeggiator to enter perfectly quantized drum parts into the sequencer, as they both share the same clock. This is a handy trick for com posers who use the sequencer as a scratch pad. To help you understand how this works, we'll construct a sample drum pattern.

Imagine that the first sixteen notes (C1 to D#2) of the Emax keyboard represent the sixteenth-notes in one bar of

music. Using the Copy Voice function, drums are assigned to the appropriate keys based on where they would play in the bar. Fig. 1 shows how we assigned the drums to create a basic rock beat.

Once you have assigned the drums to the proper keys, enter the Preset Definition module and select Arpeggiator 6. Use the right cursor button to scroll through the menu, and the Yes button to enter the following settings:

table border="0">
Arpeggiator:	On
Tempo:	120.00
Note Value:	1/16
Mode:	Up
Latch:	Auto
Number of Ext:	15
Ext Interval:	m2
Harmony 1:	Off
Harmony 2:	Off
Clock:	Internal
Velocity:	127
Key Repeats:	1
Cruz Control:	Off
Glissando:	Off
Range:	C1 to C1

Play C1 to start the arpeggiator. The pattern will continue playing until a new preset is selected.

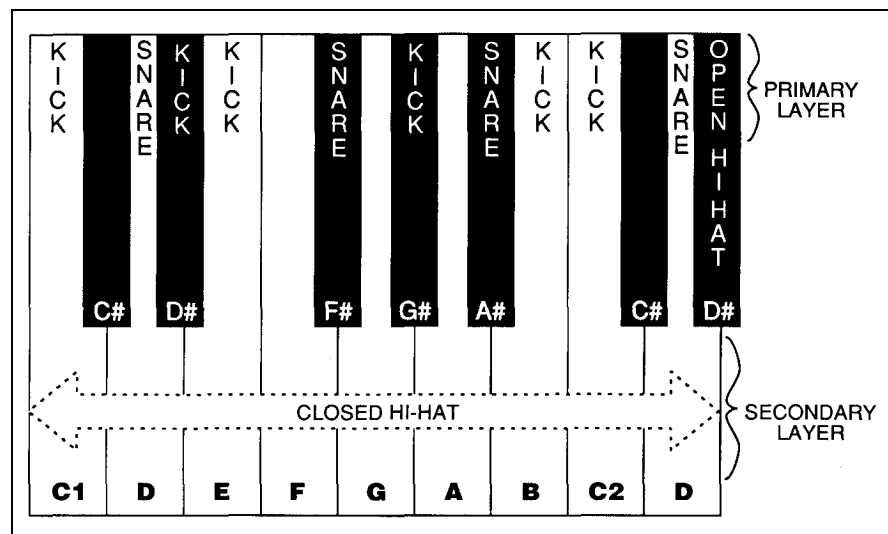
To record the pattern, put the sequencer in Record mode, press Play, then play C1. This will automatically cause the sequencer to start. It is not necessary to hold down the key while recording since the arpeggiator's auto-latch is on. Record the pattern for the full length of your song, and press sequencer stop when the desired length has been reached.

## 7 In Closing

Many people consider an instrument obsolete just because it's old, or because a newer instrument has more features. This idea is often promoted by manufacturers, who would just love to sell you a new keyboard every year. The fact of the matter is:

Your instrument sounded good when you bought it, and guess what? It still sounds good. Major studios and artists swear by their "obsolete" Emulator II's, and have you ever counted the number of hit records that use the Minimoog for bass ?

Happy Sampling !



**Fig. 1.**

Drums are mapped to the Emax keyboard, then arpeggiated to create "quantized" rhythm patterns.

Kick and snare occupy the primary layer, hi-hat the secondary.