E-MU EMULATOR II
DIGITAL SAMPLING KEYBOARD

THE EMULATOR II, as its name implies, is the new hotrod version of E-mu's Emulator digital sampling keyboard (see Keyboard Report, Mar.'82). For those not familiar with sampling, it's a process that allows you to make a digital recording of any sound, store it in computer memory, and then manipulate that sound in a number of useful ways, including imposing pitch changes on the sound by controlling it from a keyboard. As you can imagine, sampling is a very powerful tool.

There is quite a variety of sampling instruments currently available, including the Electro Harmonix Super Replay (see Keyboard Report, Aug. '84), the Decillionix DX-1 sampling software for the Apple computer (Keyboard Report, July '84), the Ensoniq Mirage, 360's Digital Keyboard (Keyboard Report, Jan, '84), the Kurzweil 250, the Fairlight, and the Synclavier. Prices on these systems range from $675 for the Electro-Harmonix to upwards of $30,000 for systems like the Fairlight and Synclavier. The original model Emulator set the pace for all of the sampling machines priced beneath the Fairlight and Synclavier. The E-II goes a long way in out-distancing the first Emulator in features, and it manages to come in at the exact same list price of $7,995.00.

The System: We looked at an instrument with two disk drives. The basic system comes with one drive standard. The extra drive is an additional $650.00, and it is there to facilitate live performance. Floppy disks (5.25") are used to store both system operating software and sound samples. It takes about 25 seconds to load a disk into the machine. That's a long time in a musical context, but once loaded, all sounds located on that particular disk can be accessed instantly by using a 10-digit numeric keypad. The number of sounds held by a disk varies with the amount of memory those sounds require.

Disks are used to store samples in banks and presets. A bank is a set of samples, or voices as E-mu calls them, loaded onto a disk until the memory is full or until you've assembled all the voices you need, whichever comes first. A preset consists of some combination of voices from the bank assigned to specific ranges on the keyboard with specific voice parameters and real-time control settings programmed in. Up to 99 presets can be stored on a disk; of course, this number is a maximum, the limit being defined by how much memory is used up by each voice and its accompanying preset(s). A disk needs to be present in one of the drives at all times, since the machine occasionally needs to refer to the operating software located on the disk.

The front panel is laid out in modular fashion, with sections being graphically set off from each other. "Modules" include master controller, sequencer (which wasn't implemented in the 1.7 software we looked at), filter, VCA, LFO, voice definition, preset definition, real-time control, disk, and sampling sections. Most of the sections feature a single pushbutton accompanied by an LED to tell you it has been activated. When a section is activated - and you can only access one section at a time - the 10-key numeric keypad is used to select the function you want to address within that section. The possible selections are printed as menus within each section's front panel graphics.

A backlit LCD display is used to talk to you during all operations. For example, when the machine is first booted up, the display tells you, "This will take a while." Other messages tell you if a sample was good or bad, what the filter settings are, what the level of an incoming signal to be sampled is, and so on. Again, the display serves a different purpose for every function on the machine.

Besides the 10-key keypad and the LCD display, the master controller section features four sliders labeled A-D, three switches, and a mix output volume control. The four sliders are used to adjust various functions depending on the module selection chosen. For example, in filter module menu selection number 1, slider A adjusts the filter's cutoff frequency, B controls the amount of Q, C selects the amount of envelope controlling the cutoff frequency, and D does nothing. Some of slider functions are bipolar (they go both positive and negative), depending on the selection, but more on that later.

The E-II has eight channels, and can produce up to eight different sounds simultaneously. These sounds can be distributed however you like, across the 5-octave velocity-sensitive keyboard. It's possible to assign each sound to its own separate output jack. For example, say you've sampled a piano. You can assign it to some specific number, let's say three, of the eight channels, and determine what area of the keyboard it will be sounded from. At that point, you can only play up to three notes polyphonically of that particular piano sample. Those notes are distributed one to each of the three channels you assigned them to. The other five channels are available for other sounds. The dynamic allocation switch is used to override that voice assignment scheme, making all channels available to all the voices, a handy function for testing things out. If you don't care to use the eight separate channel outputs, a mix output is provided.

Each sample can be tuned over a +48 to -50 cent range using one of the sliders. A transpose switch is also included. Transposition is possible over a range of an octave up or down, which is very useful, especially when the sample you've input isn't pitch-centered on C. We felt it would have been nice to include an A-440 reference tone to help keep all the samples in tune with one another.

To the left of the 5-octave, C-to-C velocity-sensitive keyboard are two wheels whose function is determined by the real-time control module. The left wheel is spring-loaded and has a center detent. The right wheel is not spring-loaded, and has no detent. It would make sense to think of these as pitch-bend and modulation wheels, except that you can assign either wheel to control functions such as pitch, filter cutoff frequency, level, LEO amount as it affects pitch, LEO amount as it affects the filter, LFO amount as it affects level, and the attack rate. The left wheel's pitch-bending range is ± slightly less than a minor third, with the center position being concert pitch. However, the right wheel's range for pitch-bending is a tritone from lowest (off) setting to full on. Concert pitch is some wherein the middle of its range. Likewise if you assign the spring-loaded left wheel to control any of the LFO functions, its center position leaves them on all the time, while moving the wheel either adds to or subtracts from the effect. You'll probably want to use the left wheel for pitch-bending and the right for modulation control, but you can get some unusual effects by switching their roles.

Each parameter (pitch-bend, LFO amounts, and so on) can only be controlled by one real time controller at a time, and each wheel or other controller can only affect one parameter at a time. Other controllers governed by the real-time section include: an A/D pedal (supplied) or any voltage pedal with a 0-9-volt range, three external MIDI controllers, and two assign able footswitches (also supplied). The foot-switches can be routed to control the sequencer, sustain, release, sustenuto, and preset advance. The sustain function causes sounds to be held in their loops as long as the switch remains pressed. If there is no loop in the sound, it will play its entire recorded length and stop. Sustenuto causes only the notes being held when the switch is pressed to sustain. Release causes all notes to be played for their entire recorded length regardless of how long their keys are depressed. Advance preset does just what its name implies; however, in order to be able to step through all the presets, the switch must be assigned this function in each preset. Note that all real-time controls can be loaded as part of a preset on disk, so that you can tailor their functions individually for each preset. A control enable function (located in the voice definition module) allows you to exempt specific voices on the keyboard from the normally global real-time controls, so you can have all the samples but one responding to pitch-bend or whatever.

Filter & VCA/LFO Sections: Each of the eight channels has two ADSR's, a VCF, a VCA, and an LEO. These can be used to affect a single voice on the keyboard, some combination of voices, or all the voices simultaneously. The machine asks you to choose which voice or voices you want to affect when you select either a VCF or a VCA/LFO parameter.

The filter parameters are: cutoff frequency, Q, envelope amount (a function that's bipolar), LFO amount, and keyboard amount (which can be up to 1.87 volts per octave). The controls for the filter's ADSR are accessed through the filter module as well.

The VCA/LFO module gives you control over an ADSR, which is routed to the VCA, the LFO rate, an initial delay for the LFO, a function called LFO variation, and an LFO amount control. The LEO variation parameter deserves some comment in that it sets the amount of random variation in LFO rate for each key depression over a scale of ito 15. The higher the number, the wider the range of variation. This function is great for creating ensemble effects. It's an idea we're surprised hasn't been picked up by synthesizer manufacturers before now.

The inclusion of analog VCF's and VCA's greatly extends the Emulator's capabilities. With them, you can use sampled sounds as sound sources, and then treat those samples the way you would an analog synthesizer's VCO's by filtering and amplitude-modulating them. Pitch-bent and bowed piano are beautiful effects, as are strings with filter swoops and amplitude modulation.

Voice Definition Module: The voice definition module gives you 15 selections, each dealing with a different aspect of tweaking sampled sounds. The first selection allows you to truncate a sample - that is, cut off the beginning and/or ending portions of a sound. Sliders A, B, C, and Dare used as coarse and fine controls for adjusting where a sound starts and where it ends. You're given the opportunity to make any truncations permanent, in which case the instrument discards the unused portions of your sample, allowing you to make better use of memory space. If you don't make the truncation permanent, it is stored in memory, and the sound can be un-truncated at a later date.

The next selection allows you to set the repeat loop within a sound, again using the four sliders in the master control section. Loops are used to create sustained instrumental voices out of shorter samples. Looping is tricky and takes a lot of patience to get right. The E-II has an auto-looping function that is designed to locate the best loop in the general vicinity of where you have set the loop controls.

If you get frustrated with your attempts at doing regular looping, you can always get experimental and use the next function, which is called back and forth looping mode. In a nor mal looping mode, a loop plays through to its end point, and then returns to its beginning, over and over. In this back-and-forth mode, the loop will play through to its end, and then play itself backwards, returning to its beginning, and then play through to the end, and so on. According to E-mu, some sounds work very well with this type of looping, because there are none of the pops or clicks that might be caused when a loop jumps from the end back to the beginning.

The next function allows you to digitally splice portions of two voices together. The sliders are used to select the points where the two sounds will be joined. An auto-splice function is included. It works like the auto-looping function in that it helps find the best points for splicing located near the area you select with the sliders. The splice feature is great for combining sounds in unusual ways. You can do things like start out with a piano and end with the tail end of a trumpet, or have a car horn end as a flute. Some combinations work better than others. This is obviously one of those potentially very powerful features that you'll have to explore on your own in order to get the most from it.

The keyboard's velocity sensitivity can be routed from a number of selections in the voice definition module. Possible destinations include the VCA level (adjustable over a 0-15 range, higher numbers giving greater dynamic range), VCA attack time (faster key strikes equate with faster attack times; the VCA envelope attack setting determines the fastest attack possible), VCF cutoff frequency (faster strikes increase the cutoff frequency), VCF attack (works just like the VCA attack time function), and VCF Q (this bipolar function can either decrease or increase the amount of filter resonance).

Other functions in this module allow you to display the length of a sound in bytes used, set the initial vibrato depth, attenuate individual voice levels, and tune individual voices over a -50 to +48 cent range. A solo mode is also avail able that lets you assign a voice to a single channel in a single-trigger mode. A loop-in-release mode is provided to hold a sound in its repeat loop after a key is released. Still other modes allow you to play a sample back in reverse (oh no, is the world ready for more subliminal messages from the music plane?), combine two sounds digitally (merges them and replaces the current voice with the newly combined voice), selectively exempt individual voices from real time control assignments, and save voices on disk.

Preset Definition Mode: Here's where the power of this instrument becomes most apparent. Remember that presets on the E-II are combinations of voices assigned to specific ranges on the keyboard with specific real-time and parameter settings. As you might guess, the preset definition module is where you put together presets.

There are a number of utility functions that allow you to move voices around in useful ways. You can load them from disk into a memory bank within the instrument (bank memory has approximately 480,000 bytes in it), copy a voice from one voice slot to another (renaming them if desired), erase voices, erase presets, erase banks, copy, replicate, and/or rename presets, and display lists of voices, sequences, and presets.

A voice assign function allows you to assign a voice to a particular keyboard range in the preset you're building. Voices can be played over a maximum range of one octave above and one below original pitch (that's really all that's usable for most samples anyway). If you want a single sound across the entire 5-octave keyboard, you've got to do multiple samples of that sound at different pitches. Two voices can be assigned so that they butt against each other, overlap each other in part of their ranges, or share exactly the same range. The voice assign function also lets you choose where the original recorded pitch of the sample will appear. You'll want to be careful not to pitch-center different ranges of the keyboard on different notes, unless that's the effect you're after. For example, you sample a piano's G, but select that pitch to appear at Middle C. Then in another part of the keyboard, you assign a flute, sampled at F#, to appear on an E. Things can get confusing if you don't keep track of what pitches you assigned to where.

You can also edit your voice assignments, allowing you to look at where you've assigned a particular voice and change that assignment. Another function lets you de-assign a voice and remove it from the current preset.

You can do a few more tricks with the keyboard velocity sensitivity from the preset definition module, including velocity switches (where playing softly produces one sound and playing hard produces another), velocity cross-fades between two sounds (balance is determined by how hard you strike a key), and positional cross fades. This produces a crossfade whose balance is based on keyboard range. At the bottom of the range, only one voice is heard; as you play up through the range, the first voice slowly fades as the other voice becomes more audible.

The arpeggiator is accessed through the preset module. This is one of the most complete arpeggiators we've seen. Its parameters are set using the sliders in the master controller. Functions include mode (off, up, down, up/down, random, and programmed - the order you play the notes in), range selection (normal, one octave, or two octaves), selectable range (any where on the keyboard, so you can have a preset with only some portion of the keyboard being arpeggiated), note value (can be set in increments from quarter-notes to thirty-second note triplets), and tempo (variable from 40 to 240 bpm).

Another function accessed in the preset module is MIDI. Parameters include omni and poly modes, channel selection (1-16), transmit right and/or left wheel data, and pedal data. You can also determine how the E-II will react to incoming data from up to three MIDI controllers.

Sampling Module: There are eight parameters accessed from the module labeled 'Sample.' These allow you to do things like turn the LCD display into a VU peak-reading display for monitoring the level of the incoming signal to be sampled, adjust the input gain from 0 to +20dB to +40dB (which is useful for getting low-level signals loud enough to produce good samples), and set the record sample threshold. The sample length can be any where from 0.2 seconds to a maximum of 17.5 seconds. Sampling can also be forced - set to start instantly regardless of where the threshold is set.

Disk And Special Module: This module contains disk utilities for accessing disks in drivel or 2 (if your machine is so equipped), listing voices on disk, displaying remaining space, erasing voices, and formatting new disks. The special category is one that E-mu is using as a catch all for new functions as they come up with them - convenient for future software updates.

The Back Panel: This includes MIDI IN and OUT jacks (we're surprised that THRU wasn't included), an RS-232 port, and an array of ¼" jacks. The latter are for an A/D voltage pedal input, two footswitch inputs, a metronome output, SMPTE input and output, a mix output, a sample input, and eight channel outputs. The power cord is detachable. We were surprised that you didn't get a choice of either a ¼" or a balanced XLR connector for the sampling input, although the ¼" jack is probably the more useful of the two.

The Sequencer: This was not implemented on the software we had; however, by the time this review is printed, we're told, the sequencer will be available. Judging from the panel graphics, it is going to allow you to do multi-track oriented recording, with the ability to punch in, bounce tracks, store control settings, reassign presets, sync to SMPTE, and auto-correct your playing.

Conclusions: E-mu is using a proprietary sampling scheme, so they are reluctant to disclose their sampling resolution figures. Suffice it to say that the quality of sampling is more than a cut above that of the original Emulator. How ever, keep in mind when doing your own sampling that sampling quality is only as good as the material being recorded. The disks supplied to us with the instrument we reviewed featured some excellent quality samples. Some were done by E-mu personnel, others by early owners of the E-II. Getting good clean samples of instruments isn't as easy as you might think, so don't be surprised if there's more work to sampling than you expected. Also, truncation and looping are techniques that need to be practiced long and hard before you can expect to feel comfortable with them.

Anyone used to the original Emulator will welcome all the additional features of the E-II. Combining sampling technology with analog synthesizer signal modifier technology (the VCF's, VCA's, and LFO's), makes for some very unusual and beautiful timbres. Add to that functions like digital splicing, back-and-forth looping, MIDI, versatile velocity sensitivity, merging, and a system for setting up very complex keyboard presets, and you've got yourself a reasonably priced (in comparison to the high end sampling instruments) dream machine.

Dimensions: 18 ½" deep, 40 3/8" long, and 9" high. Weight is about 55 lbs. Price is $7,995.00, plus $650.00 for a second disk drive. E-mu Systems. 2815 Chanticleer, Santa Cruz, CA 95065.

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