EMULATOR THREE
16-BIT SAMPLER & PRODUCTION FACILITY

By Dominic Milano

The Story So Far

SIXTEEN BIT LINEAR STEREO SAMPLING combined with lots of internal RAM, dozens of megabytes of hard-drive memory, and a full-function multi-track sequencer with every sync option in the book are the characteristics of your basic high-end digital audio workstation; a.k.a. the all-time god of product coolness, buzzword of supreme awesomeness, sacred cow of instrument dom. Digital audio workstations are the latest in a long line of products from the "do it all in one box school" of design. You know how the pitch goes: "Want to produce records, bucko? Buy this one machine and you'll never need another. No more drum ma chine, no sequencer, no multi-track recorder, no signal processors, no incompatibilities, no problems. All you'll ever need again is a musical idea. You do remember what music is, don't you?" The folly in the one-box-does-it-all approach is that no matter how hip, no matter how good the audio quality, every time a new product falls into the spotlight of must haveness, musicians and producers are going to want to incorporate its sound in their music. Which blows the "own a Synclav and never buy another keyboard again" philosophy out of the water.

Having learned that lesson the hard way, N.E.D., Fairlight, and newcomers Wave-Frame and Mellotron are positioning their "workstations" not as do-everything instruments, but as record-everything, edit-every thing production tools, not so much to replace or supersede instruments but to modernize the studio environment and ultimately supersede multi-track tape recorders by combining sampling (digital recording) and sequencing technology.

The distinction between low- and high-end devices is constantly blurring as manufacturers find ways to provide high-end performance at affordable prices. Aside from competing amongst themselves, the high-end workstations have only had to contend with component MIDI systems (12-bit samplers linked to sequencers, PCs, synthesizers, automated boards, and so on). It's been a battle that the big boys haven't had much trouble winning, at least not when it comes to selling systems to pro studios and AV production houses where price tag jitters take a back seat to audio quality.

Emulator Three

Keyboard: 5-octave, velocity and pressure sensitive.

Voices: 16, expandable to 32.

Memory: 4 megabytes (2 mega words) of RAM, expandable to 8.40-meg internal hard drive, 3.5" internal floppy drive.

Interfacing: SCSI, RS-422, MIDI in, out, thru, SMPTE reader/generator, stereo Outputs, 16 individual ¼" outputs. Footswitch and pedal input.

Features: 16-bit linear 44.1kHz or 33.1kHz stereo sampling with custom 2x oversampling reconstruction filters, individual DAC's and ADC's on all 16 channels,16 VCA's and VCF's, numerous looping modes, digital signal processing including gain change, re-sampling, cut/copy/paste editing, sample calculator. On-board 16-track sequencer, MIDI supermode, analog and digital processing independent for each preset zone, free software updates to registered owners.

Size: Keyboard version: 6.6" high, 18.5" deep, 40.5" wide. Approx. 45 lbs. Rack: 7" high, 16" deep, 19" wide.

List Price: 4-meg with keyboard: $12,995.00; rack, $12,795.00; 8-meg with keyboard: $15,495.00, rack $1 5,295.00. 8-meg upgrade kit: $2,695.00 plus $200.00 installation.

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E-mu was the first company to challenge the high-end of sampling by introducing the Emulator I (see Keyboard Report Mar'82) for just under ten grand. The Fairlight was selling for upwards of $27,000 at the time and offered about the same audio quality. This time around, E-mu's Emulator Three (EIII for short) is the first of a new breed of affordable 16-bit linear machines to hit the market. (Technically, Sequential's Prophet 3000 was first, but only a few of them were shipped before Yamaha bought what was left of Sequential's assets.) Yamaha's P3000 (as in Prophet), Akai's 51000, and Dynacord's ADS are slated for release later this year. All of them offer audio quality once reserved for instruments that cost upwards of $30,000. What sets these units apart from the mega bucks crowd is mostly the amount of internal RAM - it's not uncommon to see a Synclavier with 64 megabytes of internal RAM, while the EIII offers a max of eight meg -and the amount of internal system integration, *i.e.,* on-board SMPTE generators, sequencers, DSP cards, and the like.

The EIII is the only one of the new breed to include an on-board sequencer complete with SMPTE generator/reader. At $13,000 for the four-meg model, it's also the most expensive of the pack. But E-mu doesn't want to be thought of as the high end of the low end. Rather, they hope to be thought of as the low end of the high end. Whether they can pull that off depends on how well their audio quality compares to that of the other machines, how quickly third-party support materializes, and whether customers find the built-in sequencer an asset instead of a liability.

The EIII

THE FIRST SIMILARITY BETWEEN high **-** end machines and the EIII is that many of its features are still in the birthing stage. We're happy to report that E-mu's machine is a little further along than some of the high-end units have been at their initial release. Our examinations were run on a production 16-voice EIII with four megabytes of internal RAM, a 40-meg internal hard disk, and a 50-meg external hard disk connected via SCSI. The operating system was version 1.13. We found a couple of glitches that E-mu wasn't aware of and many they were already fixing. We also got a glimpse of a prerelease version of 1.15 software, which fixes 28 known bugs in 113 and adds a number of significantly cool functions, including global MIDI controls, a scrubbing wheel, more automation in the digital processing department, and like that. We're told that when version 1.15 is released in June it will be labeled version 1.2. But our main concern is with what is, not what will be. Suffice to say that 1.15 software is in beta-test at the time of this writing, and E-mu predicts that there will be constant updates (free to registered owners) to the EIII, with the idea that version 2.0 will be the ultimate "stable" system. Which is exactly the kind of thing Fairlight and Synclavier owners have been dealing with for years.

Architecture. Anyone familiar with the Emulator I or II should feel right at home with the EIII's front panel. The layout and nomenclature of the controls are nearly identical. (For you trivia buffs, the M&M-like buttons are new - they came from the aborted LucasFilm SoundDroid project.) Operations are for the most part organized modularly, though many procedures require you to use multiple modules in specific order - a fact woefully unexplained in the reference manual and barely talked about in the preliminary version of the guided tour supplement we looked at. Each module is accessed by a single button, the module's functions are selected via sub menus using a data slider, the ten-key numeric keypad, up/down/ left/right cursor buttons, and increment/ decrement buttons. Individual samples are stored in banks (an empty bank offers up to 47.5 seconds of mono sampling at 44.1kHz, 23.6 in stereo with four meg of internal RAM, twice that with eight meg). Banks can be stored on the internal hard drive, floppy disk (be careful, a bank will fill a lot of floppies), or external hard disks. Banks contain all the information pertaining to each sample (in E-mu-speak, preset information - analog processing, mapping, looping, and digital processing parameters), along with sequences built using the samples in that bank. All save operations involve saving banks, not individual samples or sequences as such.

All visual feedback is supplied by a 4x20 backlit LCD with variable contrast. Parameters and data are entered the keypad, data entry slider, increment/decrement buttons, or the cursor controls. The keyboard is used to select characters when naming banks, samples, and presets. We ran into some minor but potentially confusing inconsistencies in what controls affect what parameters: For example, many modules allow you to input numeric information using the keypad, while some don't - the analog processing module parameters can't be affected at all by the keypad. Truncating samples requires you to choose length in samples, not real-time values which are displayed but they often don't update themselves consistently. Some operations require you to hit 'enter' after you've selected them, though if you use the keypad, the enter key isn't required in most, but not all cases. Once you've armed sampling, you can't get out of the operation in the normal fashion (pressing the sample menu button); instead you have to hit 'enter.' Once you're used to it all, it's no problem. But we think the data entry area could stand a little more consistency for the sake of user sanity.

Sampling. Our EIII came packed with sounds loaded at the factory. These included a great piano, E-mu's Arco Strings (which many will remember from the EII days - they don't sound much better), a variety of rock drums and ethnic percussion samples, brass, saxophones, a number of DX7 and Prophet-5 sounds, and so on. There was plenty to get anyone up and running, with more on the way (we've heard Optical Media is already at work on a CD-ROM for the EIII). But what's it like to roll your own samples on an EIII? Once you figure out the correct order to do things in, it's simple as pie. Our one complaint: There's no sample thru, so if you want to listen to what you're sampling, you'll have to use a couple of more mixer channels and a multiple. Bummer and double bummer, because the audio quality of the EIII is so outrageously good that you'll find yourself noticing noise where you thought there was none. In fact, we heard things we thought at first were artifacts of the sampling process of the EIII, but in tracing back through the sampling chain, we found that they were barely audible glitches in the sound effects CDs we used as source material - background noises that we never noticed with our 12-bit machines. The EIII is so clean that routing source material through your mixer may introduce infinitesimal amounts of noise that may be objectionable. It kind of gives new meaning to the old maxim "garbage in, garbage out."

Two sample rates are available (44.1kHz and 33.1kHz), and all the standard sampling stuff is supported: variable trigger threshold, forced sampling, input VU meter display in the LCD, and so on. The VU meters aren't visible during sampling - dumb. But there are also a few nice touches: auto truncation, which trims silence off the front and back of a sample; auto normalization, which increases the peak amplitude of a sample in order to maximize audio quality; and auto sample placement, which maps each new sample to the next highest set of keys, the number of keys being user-programmable. We ran into a bug when building presets that included samples from earlier versions of the operating system. Specifically, when a bank's memory was nearly full, you wouldn't get a memory full message and only some of the samples you mapped to the keyboard would sound. The problem only occurs with corrupted samples. Hopefully, these bugs will all be identified and cleaned up in the not too distant future.

Processing. Once you've got your samples loaded up and placed on the keyboard (yes, the full MIDI note range is supported), you can process them in the analog or digital domains. Analog processing affects primary and/or secondary samples in user-definable keyboard zones. Zones can overlap to any depth, and can be any size, from the length of the keyboard to as small as one note. Parameters affect each zone independently. For example, you could set a VCF to affect the entire keyboard, pan each key differently, have different LFO parameters on each note (handy for sound effects), or use a VCA to affect only certain portions of the keyboard. The possibilities, as they say, are endless. There's enough memory for over a thousand preset zones, far more than any normal human could ever hope to use up. Our wish list includes some zoning automation in the form of templates - a feature that's being added in the next software release.

Each of the 16 VCA's and VCF's (one for each output channel) has its own AHDSR envelope generator (the H is for a hold segment). Values are displayed as real-time numbers, though they didn't check out with our stopwatch - a 19.65-second release timed out to about ten seconds. There's also an auxiliary AHDSR for controlling one of the following per zone: pitch, panning, LFO rate, or amount of LFO control of pitch, loudness, filter, or panning. The filters are 24dB/octave resonant low pass filters, which oscillate fairly easily at high resonance settings. They can be set to track the keyboard - handy for filtering out aliasing or artifacts of pitch-shifting samples with background noise in them. Tracking amount is programmable. The analog processing section includes 16 VCA's and VCF's, each with its own AHDSR envelope plus another set of 16 auxiliary AHDSR's for control of pitch, panning, LFO modulation amounts, and so on.

The LFO's produce triangle, sawtooth, sine, and square wave shapes. The routing in the analog parameters module is extremely flexible. The LFO, velocity, keyboard pressure, a voltage pedal, left and right wheels, and MIDI controllers can be routed to just about every imaginable parameter. Nice.
There are a lot of nifty games you can play with the digital processing module that go beyond the standard looping stuff. For example, portions of samples can be cut, copied, and pasted Macintosh - style, which might sound like no big deal except that the paste operation lets you do things like offset a sample with itself with an adjustable gain. This is great for doing echoes and pseudo-reverb effects. The system will auto-correlate truncations, looping, and even the copy operation, automatically selecting points that will minimize discontinuity. Of course, the process isn't perfect, but the machine will let you do it again and again until you get an acceptable take. All the digital processing operations can be undone, a very nice touch.

Computation times in the digital processing module vary according to the length of the samples being manipulated and the type of processing being handled. For example, we pasted a 2.4-second brass section sample onto a .5-second French horn. The French horn had been digitally pitch-shifted (so its sample rate was 43,842Hz rather than44.1kHz). It took 2 minutes, 31 seconds for the machine to convert the sample rates and paste the selections together. We found it more typical that digital processing would take anywhere from a couple of seconds to half a minute - not bad at all. We hit a nasty crash (lots of noises and flashing lights) while copying one half of a stereo sample. It turns out that version 1.13 software has trouble handling various copy and paste operations when the contents of the clipboard are a certain size, a sample has been pitch-shifted, and different sample rates are involved. Note, however, that these problems don't occur constantly - it's only when certain conditions exist in the clipboard. Most of the time copy and paste operations gave us no problems.
The current software will not let you paste to one side of a stereo sample or turn loops on and off independently for each side of a stereo sample, though the next version of software will. All the basic types of loops are supported: forward, backward/forward, cross-fade, and like that. Believe it or not, even with four meg of RAM, it's easy to run out of memory. Of course, looping shorter sounds helps a lot, but if you don't mind losing some of the high-end 44.1kHz sampling provides, you can use the sample rate conversion function to con serve memory. It works quite well, even in torture tests involving sounds with lots of high harmonics, though computation time is pretty long (6 minutes, 45 seconds to convert a 3.7-second 44.1kHz sample to 26,484Hz).

By the way, the minimum rate is 7kHz, and the max is 50kHz, though why you'd want to convert upwards is another question. The idea is to enable you to match sample rates from samples done on other machines, though the sample dump standard hasn't been implemented on the EIII yet.

Miscellany. The EIII's five-octave keyboard is both velocity and pressure sensitive. It feels slightly more weighted than a DX7's keyboard - not bad. (We found that when the keyboard was put into solo [monophonic] mode with the envelopes set to gate, audible pops were present.) For those gasping at the thought of buying yet another keyboard-equipped box, rest easy, the rack version should be out shortly. Though we can't imagine working with the rack version without a remote control - there are just too many keystrokes in some of the procedures.

Remember the Emax arpeggiator? Remember arpeggiators at all? It might seem weird that a $13,000 instrument should include an arpeggiator, but the EIII's is pretty hip. It does more things than the Emax arpeggiator (one of our all-time faves, see Keyboard Report, Jan. '87 for details). Two harmonies can be programmed, the key board can be tracked in all kinds of directions, it will sync to MIDI and TTL clocks, number of repeats and echo decays are programmable, and on and on. Note that the sequencer does record arpeggiation data, not just what notes you play. The arpeggiator goes out over MIDI. A dual footswitch and a volume pedal come with the EIII. Both can be assigned to a plethora of parameters, including starting and stopping sequences, latching the arpeggiator, acting as various MIDI controllers, and soon. These functions are independently assignable in each preset, which is great. Be careful that you don't start assigning your real-time controllers, including the pitch and mod wheels, to so many different functions in different presets that you loose track of what's what.

Other good things include the ability to assign output channels (voices) to the 16 individual 1/4" outputs on the back panel. You can specify the total number of voices available for each zone, or you can use the default dynamic voice allocation. Thankfully, plugging a jack into one of the individual outputs decouples that voice from the stereo outs, so you can still place your snare drums in one output, and get a stereo mix of your other samples. For you technicians out there, one of the master module's sub-menus offers automatic calibration functions and a disable output selection that also displays output channel activity VU-style. And there are a lot of utilities for copying presets, moving samples from one bank to another, initializing disk drives when they've been connected after the machine is already booted, and so on. (If you boot with an external drive connected but not turned on, the EIII locks up.) Some of these operations aren't explained in the manual. For example, to copy a bank you simply save it to a new bank.

Conclusions. We didn't get the chance to A/B the EIII with a Synclavier or other high-end unit, but compared to any of the 12-bit machines we've heard, the EIII's audio quality is stunning. We heard none of the typical pitch-shifting artifacts (clock noise, aliasing) in any of our torture tests. Of the bugs and crashes we encountered we're happy to report that E-mu was already aware of most of them (and a whole bunch more that we didn't get into) and had either already fixed them in the 1.15 software or were planning on fixing them. Given the free update policy, we don't think anyone should be too worried. The EIII is farther along than the Fairlight Series III and WaveFrame were on their initial release. Be prepared to want more than 16 voices and 4 meg or RAM. We did

after very little time. Of course, after having lived with 1-meg machines for years, this may seem silly - and to be fair, it is possible to get great results with a little intelligent memory management (looping and the like).
High-end people will also find that even 8 meg of RAM does not a virtual digital recorder make. We wish you could go beyond the 8 meg to perhaps 16 - just enough to record a reasonable length vocal line and piece together live vocals over instrumental parts. You know, the stuff people are doing on Synclavs these days.

We're still a bit concerned with the sequencer crashes, but as we said earlier, these could have been isolated incidents or they could be portents of bigger problems. We just don't know. Regardless, after testing the EIII, it's going to be impossible to live with a 12-bit machine again. Just how well it will stand up to a new wave of low-end 16-bit machines is another question. Only time, and some solid A/Bing when those machines hit the market, will tell. Meanwhile, we have no trouble giving the EIII a hearty thumbs up.

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THE EIII ON-BOARD 16-TRACK SEQUENCER

WHY BOTHER PUTTING A SEQUENCER in a $13,000 sampler when many of its potential buyers more than likely to use more powerful computer-based sequencing software? We're sure E-mu is digging in its heels, bracing itself for the worse on this question. Even the EIII manual includes an entire page of justifications for the inclusion of the sequencer: Using an integrated sequencer doesn't require that you boot your computer. The integrated unit is more accessible. You can download sequences into the hardware unit for use in live performance. it extends the total number of MIDI channels/tracks available to you if you sync to an external sequencer. The EIII will extend your sync options and act as a SMPTE-to MIDI converter. Events can be synced to SMPTE Q-Sheet-style for doing sound effects work. And so on. All of which is true, but we think E-mu is going to get grief anyway.

Organization. The sequencer is the only module in the EIII that has its own dedicated controls; i.e., stop/continue, play, record, erase, and fast forward/ rewind buttons. Sequences are collections of segments (100 maximum per song) that have been compiled into songs. Segments can be organized in the musical domain of measures, beats per minute, and so on, or they can be organized as cue lists (sequences of events timed to SMPTE). You have to be careful that the sequencer is in the proper mode (segment or cue list), since there are mode-dependent differences in the sequence functions.
When building a segment, you may specify its length, set whatever count-in you'd like, determine its time signature (multiple meters are accommodated by joining different segments together), and so on.

There are 16 tracks - one per MIDI channel, though channel assignment is determined in an odd way. Well, it's odd if you're used to other sequencers. MIDI channels are tied to presets. If you want to assign a sequencer track to control some external instrument, you have to build a null preset, *i.e.,* one that has no samples in it but is assigned to the desired MlDl channel, and assign that preset to a track. We recommend building a template of null presets, labeled by MIDI channel, since there is currently no global way to override MIDI channel assignments. There's also no local off. We thought MIDI supermode might do it, but it doesn't. Global MlDl assignments and local-off are coming in the 1.15 software.

Now you're asking what MlDl super-mode is. Supermode is E-mu's way of map ping incoming MIDI data to a specific preset within a bank. Supermode allows up to 16 presets to be addressed and played simultaneously over MIDI. But if you want more than 16-note polyphony out of the EIII, you'll have to wait for the voice expander.

Recording tracks is simple. All feedback is supplied by one screen, although you can look at data pertaining to sequences in a variety of ways from different sequence modules. Some of the alternate windows do affect what you're going to listen to, no matter what you've set in the "main" sequence display. For example, if a track is muted in the track status window, you can record it, but listening back requires you to enter the status display and un mute that track.

We wished that there were an easier way to reassign presets to tracks after a track has been recorded. As it is, you can call up new presets for recording blank tracks, but changing an assignment requires you to go into the setup menu, an operation that stops play back. By all appearances, recording is one track/one channel at a time, but that turns out not to be the case. The EIII will record on multiple MIDI channels simultaneously when receiving incoming data over MIDI. Channel assignments in this case are determined by the supermode assignments. Recording in both step and real time is possible, though the manual is short on details and sequencer applications tips, and there's no tutorial at all. Ugh.

When recording in step time, the quanti-zation value determines the default duration. Other values can be input with the numeric keypad. Note values are displayed in dock pulses - 96 per quarter unless you select a lower value in the sync menu.

All kinds of editing utilities are provided, including cutting, pasting, track bouncing, and so on. Cutting a segment enables you to shorten it from any point, down to the clock pulse. If you use the erase button to delete data in step-edit mode, changing the current measure, beat, clock location and then double-clicking pastes the erased data to the new location, which is great for offsetting tracks or segments (a function undocumented in the manual).

Quantize can be set for auto correction on recording or as an edit function. Quantize resolution is variable; as is the on-record tolerance and percentage or distance a note will be moved toward the target value. There's also a skew feature, which will reach out and grab notes that are some percentage away from the target value, even if they're almost on top of the next beat. What this is useful for is capturing data sent in by alternate MlDl controllers -guitars and drum pads - which are known to have a little delay in their response time. Events in a segment can be anything from note data and velocity, to MIDI continuous controller data, to patch changes. A small glitch is that we found that the unit didn't send patch changes reliably. Sometimes our slaves received them as expected, sometimes not. A much more serious problem occurs when you send MlDl data to an external sequencer and route the sequencer's thru back to the Elli's MlDl in. The EIII crashes the instant it gets a note-on. (Both of these bugs are being fixed in 1.15.)

When you build a song out of segments, you can insert track status changes, call sub songs, and set tempo changes. Changing track status is wonderful for conserving memory, especially if you're repeating a segment a dozen times and simply bringing in more instruments as time progresses; however, this is on a segment-by-segment level, so if you want a lot of activity in this area, you'll have to break segments into smaller units. Track status affects external instruments too. We noticed that tempo changes tended to rush over the last couple beats (another 1.15 fix).

All four SMPTE formats can be read and generated by the EIII. We found the unit very slow in the chase-locking department. We clocked it at about 33 seconds. Other forms of external sync include MIDI and 24,48, and 96 PPQ TTL clocks. Song position pointer isn't implemented yet, though the manual says it is. We also couldn't get the sequencer to start from SMPTE without hitting the play key, which can become a real pain to deal with if you're doing a lot of rewinding.

After we spent an entire weekend locked up with the EIII in our studio (it was on the entire time), the sequencer crashed twice in a ten minute period. Both crashes occurred when we moved from one module to another. Why the crashes happened is any body's guess. It could be that the crashes were instrument - dependent - that our particular machine didn't like being on that long in a hot room. Or it could be that no EIII likes being on for long periods in a hot studio. Who knows? We've heard no similar complaints yet.
As for wish list kinds of things: Aside from those points already mentioned, we'd like to see MIDI file support, and track status change events stored on a measure/beat/ clock level instead of as part of a song.

Sequencer Conclusions. After getting past the manual's lack of help with the actual sequencing process, we found ourselves getting around on it quite nicely. But would we forsake our software sequencer? No way, Jose. Having the internal sequencer might be nice for on-stage playback, and it could prove very useful for guys doing cue-list work in video post-production who don't already have a computer. But for in-studio use, we'll take our favorite software sequencer any day. The amount of visual feedback and the sheer number of tracks available on software sequencers have spoiled us. We suspect the sequencer doesn't add that much cost to the EIII. If it is expensive, we suggest it be removed and sold as an option. If it isn't, having it is better than not.

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