TS Sequencing Q & A



I've been doing lots of interactive sequencing lately between Vision on my Macintosh and my TS sequencer, which I then sync together with MIDI clocks to a modular digital multi-track recorder. Coincidentally, we've been getting plenty of technical questions about a number of similar topics. With that in mind, this month's column is going to take the form of a Question & Answer forum, where I try to come up with the best solution for the particular user's needs. Here goes:

Question #1: On my TS: sometimes I'll sequence a drum track and find that the bass drum is too loud. How do I edit the sequence so that just that G2 key (bass drum), for example, is quieter throughout the sequence?

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Tony Ferrara

I have tried the Edit Track, Scale, Key Velocity, menu and it doesn't seem to do the trick. Unlike other controllers where there is a plus or minus, on mine, the range is simply from 0 to 9. Selecting different values doesn't seem to change things at all. Am I in the right menu to do what I wish to do?

Answer: Good news! I've got some general suggestions that can help you in regard to quick editing when sequencing drum tracks.

First of all, it's easiest to work with drums by keeping them on separate sequencer tracks. For instance, bass drum on track one, snare on track two, etc. If you designate tracks 1-6 to drums, 7-12 can then be used for your other instruments. That way, you can quickly edit the individual track volume by changing the Mix setting under the Track Parameters sections on your TS front panel. You can also change track programs and transpositions quickly; using this method, you could actually use drums from different kits to create your own unique "virtual" drum kit. Different sequencer tracks (with their own MIDI channel number) will allow you to interface with an external sequencer much more easily, and will enable you to solo tracks for isolation

The Independent News Magazine for Ensoniq Users

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when auditioning instruments or recording to tape.

Question #2: What is the SCALE: KEY VELOCITY menu for? Shouldn't that allow me to control the volume of a single key number in a sequence (for example, the bass drum at G2). I basically want to know if I understood the purpose of this function.

Answer: The Scale command lets you increase or decrease the level of any specific controller information in the selected track by a specified amount. The MIDI controller KEY-VELOCITY pertains to Note-On Velocity, This is a different parameter than track mix (volume). The bottom line is that it's very easy to confuse different types of MIDI controllers.

Question #3: Can you provide any tips on taking an existing TS sequencer track and breaking it into separate tracks? I assume one would do that by simply copying a range of notes to separate tracks. Can one easily "re-combine" tracks back together? In other words, once I have my drum kit on six separate tracks, can I re-combine them into a single track at some point?

Answer: Your assumption is correct — you can take an existing track and break it into separate tracks by copying a specific range of notes to a separate track. You can then easily re-combine multiple tracks by using the merge track function found on the second page under the "edit track" button. You'll also find the copy track command in the same place, at which time you can set the specific key range that you'd like to copy.

Question #4: How come on my TS the KEY VELOCITY scale menu goes from 0-99, not +99 to -99 like some other menus? What does 0 on this menu mean and what does 99 mean?

Answer: in this context, 0-99 are relative; they don't represent dB ranges, nor do they relate in any specific way to dynamic markings found in traditional music notation. They represent a minimum to maximum abstract value, as opposed to system-common information like pan or MIDI controller number values 0-127. Parameters like track pan are generally indicated by a range from +63 to -64 to symbolize hard right to left. Key zone velocity controller values range from 0 through 127; therefore, this particular parameter can be modulated via MIDI.

I hope that these suggestions are helpful to you; they've worked very well for me, and have proven to be the quickest and most accurate way to work with drum track editing on the TS-Series. If I haven't covered a point that relates to your specific interests this month, don't worry! I'll have more along this line next time.

Bio and self-serving plug: Anthony Ferrara has written for Electronic Musician, Recording, and Keyboard magazines. He is still hard at work on some new "snappy ditties," and is playing concerts in the Philadelphia area. Call 215-742-0738 for concert info.

HYPERSONIQ

MIDI Mark Productions announces over 50 brand spanking new disks for the EPS, ASR, and TS-Series samplers/keyboards. Sample disks include drums, synths, effects, rap series and house kits. Nine disks on sale now until December 12, 1995 for \$52.00. For a free catalog, contact: MIDI Mark Productions, P.O. 217, Whittier, CA 90608. Phone: (800) 301-MIDI (6434). Fax: (310) 699-0864. Email: MIDIMARK7@aol.com

*

Waveboy Industries announces the release of the *Transwave Sound Library*. Only Ensoniq instruments can play Transwaves. The five-disk collection showcases this unique feature for the first time.

What are Transwaves? A Transwave is a set of 128 wave-

New Product Announcements

forms of equal size. A Transwave oscillator can switch between different waveforms smoothly, while it is sounding. When the waveforms are properly chosen, the resulting tone is heard as a smooth, continuously changing timbre. To date, Transwaves must be created or at least phase-adjusted on a computer. Transwave Synthesis is unique to Ensoniq, and only the samplers and sample players (TS-10/12, EPS-16 PLUS, and ASR-10) can load new Transwaves.

Waveboy has generated a set of incredibly potent, expressive sounds based on Transwaves. The raw material is drawn from different analog and digital synthesis techniques, including:

- · pulse-width modulation,
- resonant filters,

- frequency modulation,
- hard-sync,
- · formant extraction,
- and additive synthesis.

But these are not just samples. The Transwave gives the player the ability to modulate the filter frequency, the pulse-width, the FM index, and the harmonic content in response to velocity, the wheel, or any modulation source. So these sounds feel alive; they respond to your playing and to parameter tweaks. Transwaves make the sampler into a virtual (if you can stand to hear that word again) synthesizer, where each different Transwave can be a different synthesis technique.

The Waveboy disk set is both a source of raw Transwaves for programmers, and a collection of finished sounds as well. The 30 sounds in the *Transwave Sound Library* run the gamut from aggressive (RUDE REZ) to atmospheric (LIGHTYEARS); from bread-and-butter (BELL BIP FM) to sound-track (HARMONICON); from plain and skinny (303 SAW) to luscious and fat (PWM SYNTH.) There are no pianos.

The Transwave Sound Library works with the Ensoniq ASR-10, the EPS-16 PLUS, and the TS-10/12. (It will not

work with the Kurzweil K2000 or other "Ensoniq compatibles.") It is a five-disk set, includes 30 sounds, and comes with a user's manual for creating new sounds using Transwaves. It is available directly from Waveboy for \$69.00 and at selected dealers across the world.

Bill Mauchly is both "Waveboy" and the creator of Transwaves in the Ensonig product line. Bill played guitar, Mellotron, and guitar synthesizer in art rock bands during the '70s. By 1980 he owned a recording studio, was doing sound programming for the Fairlight, and was getting a degree in Computer Science. When Ensoniq was founded in 1984 he joined to help design the Mirage. As a software engineer for Ensoniq, he created the voice and effects architectures for the ASR-10 and EPS-16 PLUS samplers (as well as for the Mirage, ESQ-1, EPS, VFX, SD-1, DP/4, and TS-10). He is currently heading the Advanced Product Development Group at Ensoniq. Waveboy Industries was started as an outlet for effects algorithms that did not fit into Ensoniq's product plans. As the sole proprietor of Waveboy Industries, Bill gets to develop any hair-brained idea he likes.

For further information, contact: Waveboy Industries, P.O. BOX 233, Paoli, PA 19301. Phone: (610) 251-9562.

Front Panel

RND (JJJ)

Ensoniq Announcements

It's time to introduce you to MR Rack, the latest addition to the Ensoniq family. The MR-Rack is an all-new 64 voice, expandable synth rack. It is based on a new voice engine, new effects chip with new architecture, and new circuitry giving you 44.1 kHz output, and a signal-to-noise spec of 103 dB.

Specifications

Output

- 44.1 kHz sample playback rate
- 18-bit D/A conversion with 108 dB dynamic range
- 103 dB signal-to-noise ratio
- Ground-compensated output circuitry

Inputs/Outputs

- Left/Mono and Right Main outputs (balanced TRS)
- Left/Mono and Right Aux outputs (balanced TRS)
- Stereo headphone output
- MIDI In/Out/Thru

Display/Interface

- 40 (2 x 20) character LCD display
- 2 Data Entry knobs, 10 buttons (6 with LEDs), front panel effects bypass, MIDI indicator LED, Volume knob
- SoundFinderTM interface one knob selects sound type, another knob selects specific sound. Sounds can be searched for alphabetically within musical instrument categories, or numerically within Bank locations (ROM, RAM, card, expansion board, GM, User, Demo)
- Audition button plays musical examples to preview sounds from front panel (your choice of either a musical phrase appropriate to the SoundFinder category, arpeggiated octaves, or a major chord).
- PCMCIA ROM/RAM card slot for Sounds/Performances

Internal Sound/Performance Memory

- 393 ROM sounds (2 Banks of 128, 128 GM sounds, 9 GM/GS drum kits), hundreds of ROM drum elements
- Up to 128 RAM sounds possible (due to the dynamic voice size the final number is still to be determined)
- 65 ROM Performances (16 parts each)
- 32 RAM Performances (16 parts each)

Card Sound/Performance Memory

· ROM cards offer 2 Banks of Sounds (up to 128 per Bank) plus

2 Banks of Performances (32 per Bank)

- RAM cards (PCMCIA format) can be from 512k up to 2 MB. They can hold the following data:
 - 512k 4 Banks of Sounds (up to 128 per Bank) 4 Banks of Performances (32 per Bank)
 - 1 MB 8 Banks of Sounds (up to 128 per Bank) 8 Banks of Performances (32 per Bank)
 - 2 MB 16 Banks of Sounds (up to 128 per Bank) 16 Banks of Performances (32 per Bank)

Wave Memory

- 96 Megabits (12 MB) of 16-bit wave ROM
- Over 5000 variations possible from the over 300 waveform selections, which include:
 - 28 Megabits (3.5 MB) of acoustic piano
 - Multi-sampled acoustic instruments in Keyboard, Synth, String, Brass & Horn, Wind & Reed, Vocal, Bass, Drum, Percussion, and Sound Effect groups
 - 2nd-generation TranswavesTM created through sampling, digital synthesis, and digital resynthesis technologies
 - Sustained synth waveforms with harmonic and inharmonic structures
- · Modulatable start point, forward or backward playback, and multi-sample key shift (4 modes) available on most waves
- · Pitched/non-pitched playback of all waves possible

Expansion Memory

- 3 wave expansion board slots, for a total of up to 672 Megabits (84 MB) of ROM wave data
- A board holds up to 192 Megabits (24 MB) wave data, multiple Banks of Sounds/Performances (up to 128 Sounds and 32 Performances per Bank), demos
- · Expansion boards are user-installable

Voice Architecture (accessible via computer)

- 64 dynamically assigned voices
- · Dynamically sized voice architecture (up to 16 Layers), with the following parameters:
 - Independent waveform selection
 - Programmable layer delay up to 10 secs or by key-up trigger
 - 2 independent multi-mode dynamic digital filters (low-pass, high-pass, variable bandwidth band-pass) with key tracking (33 settings, including many fractional scaling options)
 - 3 five-stage envelope generators (time and level) with level and attack time modulatable by velocity (8 possible curves), key scaling, and response to release velocity
 - 35 ROM Pitch Tables with a resolution of 256 steps per semitone
 - 1 RAM location for receiving a Pitch Table sent from any device that supports the MIDI Tuning Dump Standard (Bulk Tuning Dump and Single Note Tuning Change)
 - 22 modulation sources
 - 8 modulation destinations
 - 1 LFO with 7 waveform choices, sync-able to System clock (definable in BPM) or external MIDI clock, with 12 rhythmic subdivisions
 - Variable rate Noise generator, sync-able to System clock (definable in BPM) or external MIDI clock, with 12 rhythmic subdivisions
 - Volume and Pan

- Semi-tone and fine tuning
- Pitch tracking (33 settings, including many fractional scaling options)
- Key and velocity ranges
- Voice trigger options (key-up, key-down, MIDI controller enable)
- Glide mode, glide time
- Mono/poly playback
- Pitch Bend on/off, Held Pitch Bend mode
- Pitch Bend range is programmable for both positive and negative wheel directions
- Sustain on/off
- Key group assignment
- Drum Kit architecture offers 64 keys (B1 D7): each key points to a fully-developed Sound (up to 16 Layers per key), plus effects bus, volume, pan, and tuning overrides.

Performances

- A Performance is a 16-part combination of Sounds. Parts can be assigned their own MIDI channel for multi-timbral reception, be set to the same MIDI channel for layer and split options, and be individually muted.
- Parts can be set to a global "Stack" channel, which can then be defined as any channel (1-16). In this way layered or split setups can be quickly matched to a given input channel by simply changing the Stack channel setting.
- Each Performance contains a custom Drum Kit
- Extensive sound "tweaking" (global editing) for each Part in a Performance, including:
- Volume
- Volume Polarity
- Expression
- Pan
- Effects Bus
- Pitch Bend Up
- Pitch Bend Down
- Tuning (Octave,
- Pitch Table
- Glide Mode
- Glide Time
- LFO/Noise Tempo Sync
- LFO Rate Filters

Effects

- New VLSI 24-bit digital signal processor (ESP-2) with 26 MIPS of DSP power, 52-bit accumulation
- New UMTE (Universal Multi-Timbral Effects) architecture with 6 stereo busses:
- 3 stereo busses with independent send amounts into global reverb
- 1 stereo bus with a wet/dry mix into global chorus and a send amount into global reverb
- 1 stereo bus with a wet/dry mix into an insert effect, a wet/dry mix into global chorus, and a send amount into global reverb
- 1 dedicated stereo dry bus

- LFO Depth, Delay - Amp Envelope
- Attack, Decay, Release
- Filter Cutoff
- Filter Envelope
- Attack, Decay, Release
- Amp & Filter
- **Envelope** Velocity
- Key Range Lo/Hi
- Velocity Range Lo/Hi
- Velocity Mode
- Pressure Mode
- Controller Reception
- Coarse, Fine)

· Insert effect choices include:

Hall Reverb Small Room Large Room-Small Plate Large Plate Gated Reverb NonLin Reverb1 NonLin Reverb2 Reverb->Chorus Reverb->Flanger Reverb->Phaser **Rich Chorus** 8-Voice Chorus Chorus->Reverb EQ->Reverb Flanger->Reverb Phaser->Reverb Spinner->Reverb Multi-Tap Delay DDL->Chorus

DDL->EO DDL->Flanger DDL->Phaser Parametric EQ Rotor Speaker Guitar Amp VCF Distort Distortion->Chorus Distortion->Flanger Distortion->Phaser VCF Envelope Loudspeaker Guitar Chain1 Guitar Chain2 Guitar Chain3 EQ->Comp/Gate Pitch Shift Chatter Box Pitch Morph

· Insert effect dynamically modulated by MIDI controllers

· Effects can be routed to Main or Aux outputs

MIDI

- Always able to respond in Multi Mode on up to 16 polyphonic channels
- A single MIDI channel can trigger multiple sounds
- Responds to Poly-Key TM polyphonic pressure
- 4 System controllers (any of 120 MIDI controllers) can be used for voice or effects modulation
- General MIDI compatible

Standard Accessories

- Musician's Manual
- Mark Of The Unicorn UNISYN TM MR-editor/librarian software
 - (redeemed by returning completed warranty card)

Optional Accessories

- MRC Series ROM Sound cards
- EXP Series Wave Expansion boards
- MC-512 SRAM card (PCMCIA format) larger card sizes are available through most computer memory retailers

Physical

- Dimensions: 19" (48.26 cm) wide by 1 3/4" (4.29 cm) high x 9 3/4" (23.6 cm) deep
- 19" rack mount standard, 1U high
- Weight: 9.7 pounds (4.4 kilograms) unboxed, 17.6 pounds (8.0 kilograms) boxed

The MR-Rack will ship in late November/early December and costs \$1695 suggested retail price.

Hacker News

It's always hard to remember in the middle of November that *this* is the issue to convey our holiday greetings. (Well, we've only been doing this for ten years or so...) Anyway, season's greetings to all, thanks to everyone who has anything to do with this, be careful, and enjoy the holidays! — Eric & Jane

Third-Party News

Rubber Chicken has new links — phone: 612-235-9798, and e-mail: chickenEPS@willmar.com.

XLS Food, that oddly-named vendor of samples (reviewed in #124), has a new street address: Total Recording Studio, Parkst 76, CH2000 Neuchatel, Switzerland.

As far as we can tell, **Ramtek** is still in business. However, we do feel obligated to warn readers that they may want to exercise extra caution...

Stupid Sequencer Tricks

Jack Schieffer

Many people think of onboard sequencers as nothing more than fancy tape recorders. However, there is much more to Ensoniq sequencers than merely recording and playing. By use of clever editing and "template" sequences, you can maximize the power of your sequencer as a creative tool. In this article, we are going to focus on the some of the ways to use sequencer templates on the EPS and ASR samplers. However, these same techniques should work on any keyboard with a sequencer and an event edit function. You'll just have to adapt the specific directions to fit the architecture of your machine.

Modulation Templates

One of the primary functions of a sequencer is to record a series of notes and then play them back. However, most modern sequencers record much more than just notes. They record all kinds of events: mod wheel movements, velocity levels, aftertouch amount, and so forth. These events are usually used to modulate or alter the sound produced by the oscillator.

You can use these types of sequenced modulation to produce a variety of effects. By sequencing abrupt changes in volume, you can reproduce the "transforming" effect heard on many techno and dance tracks. By sequencing filter movements, you can create a sampleand-hold type of sound. Pitch modulation can be used to create arpeggiator-like effects.

One of the best things about using the sequencer for these tricks is that you can save your sequence and adapt it for use in many different situations. Simply use the sequencer commands to merge this "template" with the particular sequence you are working on.

Since the modulation is sequencer-based, it will fit perfectly with the rhythm of your song. And you can change the speed of the effect simply by changing the tempo of the sequence. That's much easier than monkeying around with a bunch of LFO and envelope parameters for each instrument.

Using Filter Modulation

Choose a sustaining sound, preferably one with only one sample in it (to make editing easier). If you want, you can use the Create Wavesample command to give you a simple square wave. Go to the Edit Filter page and set the filters to 2 LP and 2 HP. Set both F1 and F2 CUTOFF FREQ to 0. Make sure there is no envelope modulation by setting ENV 2 AMT to 0 on both filters. So far, we've set up a voice with a totally closed band-pass filter which will sound somewhat like a resonant filter.

Next, we want to set the modulation parameters so that we can simulate a sample-and-hold sound. Set the F1 MOD and F2 MOD parameters to WHEEL. Set the MOD AMT parameters to +99 for both filters. Before we forget, let's disable the vibrato effect brought in by the mod wheel. Go to the Edit Pitch page and set LFO AMOUNT to 0. Try playing a few notes while moving the mod wheel back and forth. The timbre of the sound changes pretty dramatically. However, the movement of the filter is continuous. To simulate a sample-and-hold, we need to introduce sudden changes in the filter cutoff. That's where the sequencer template comes in.

Use the sequencer to hold down a single note for one bar. Now, go to the Edit Sequencer page and set RECORD MODE = ADD. Next record an overdub and move the mod wheel once during the bar.

Next, we have to tackle the nuts-and-bolts of the event editor. This can get a little complicated, so you may want to have your manual handy. Go to the Command Track page and use the EVENT EDIT TRACKS command. Scroll left to the Event Type and change it to MW so we can edit the mod wheel movements. Using the insert and delete functions, add a mod wheel event to each sixteenth note (Clocks 1, 25, 49, and 73 of each beat). Now, you want to edit the event value of each sixteenth note. Try to mix high and low values and don't repeat any values. The whole idea is to simulate a wild, pseudo-random jumping-around effect. I used the following values (in order): 30, 70, 110, 55, 125, 90, 45, 70, 25, 95, 65, 35, 80, 50, 120, 75. When you are done, exit the event editor by pressing No/Cancel.

Press Play and let your sequencer run. The filter should be jumping around in response to the sequenced mod wheel movements, creating sudden, rhythmic changes in the sound. After you've admired your handiwork for a while, save this sequence to disk, giving it an appropriate name like SH MODWHL. One of the best things about this template is that it is sequencer-based. You can use it with any instrument that responds to the mod wheel.

Variations On A Theme

This technique works well in the example we just did, but don't limit yourself to just the mod wheel and the filter. You can sequence any of the different modulators: mod wheel, volume pedal, key pressure, pitch wheel, and so on. You can use these modulators to control a variety of parameters: filter cutoff, volume, panning, pitch, effects parameters, loop position, transwaves, or whatever. One of the best features of Ensoniq keyboards is their flexible modulation routings, and with this techniques you can create some wild sounds. Your imagination is the limit. Save your sequences and try them out with different sounds and different modulation routings. You are sure to stumble across some pretty cool combinations.

Test Patterns

When editing patches or tweaking samples, you probably find yourself switching back and forth quite often between the programming buttons and the keyboard to test the new sound. Make things easier on yourself by letting the sequencer handle the keyboard.

Create a simple test pattern; it doesn't need to be a masterpiece, just noodle around a little. Try varying the velocity and duration of the notes. Then let the sequencer loop on this pattern while you adjust parameters. You should find that editing goes much more quickly. You might want to create a separate test pattern for each different type of sound: pianos, strings, drums, etc., so that the pattern can repeat typical performance techniques and phrasings.

Conclusions

As you can see, your sequencer can be a valuable tool in developing your sounds, not just in recording the notes of a song. As always, experimentation is the key. Try out lots of different ideas and see what works. Ensoniq sequencers are so easy to use that you're crazy not to use them for as many tasks as possible.

Bio: Jack Schieffer lives in Dallas, TX which is okay. He plays Ensoniq samplers which is better. He reads the Hacker which is the best of all.

The DP/4 and MIDI System • MIDI Parameters — Part II

Steve Byhurst

As promised last time, we are now going even further into the depths of System.Midi Land. Hold on tight! (I do my best to make this stuff more exciting than it really is you know).

Global Parameters

Right, we've seen what can be done to program the four effect units and the config unit for MIDI, what about parameters that affect the DP/4 as a whole, irrespective of individual settings? Well, these can be found from parameter number 35 onwards. Pressing the System.Midi button repeatedly will get you there quickly and then cycle through groups of related commands. Let's start at the beginning — numbers 35/36.

Control Channel/Enable

This page allows you to choose which channel number from 1-16 you want to function as a controller channel and whether it is enabled or not. As I said in part one, all modulation sources are received on this channel as well as bypass controllers. How the modulation sources change effects in individual units is part of the preset programming. Here we just select which channel the modulation is received on.

The chosen channel can be the same as any or all of the

units, or completely separate. How it is used depends on your normal working method. If you are driving the DP/4 from a sequencer you may wish to have a separate controller track with associated unique channel number, keeping it apart from other data. However, in most cases there are advantages in using the same channel you have allocated for the config unit. With this method, as long as you don't wish to control units A-D separately, all your MIDI data can be sent down one channel.

The Enable parameter can be used to filter out all modulation to the DP/4 by simply setting it to "Disabled." Thus you can eliminate the effect of the modulation without changing the modulation parameters of the preset or the parameters set within your MIDI controller.

DP/4 Controllers 1-8

These parameters are used to select a list of eight DP/4 system controller sources. Any two of these may then be chosen to modulate any two parameters of an algorithm. There are non-MIDI as well as MIDI sources. The MIDI ones are the same as those on offer as bypass controllers (listed in part one).

If you are really into MIDI programming you will probably find a total of only eight system controllers quite restrictive, but the rest of us will probably be happy enough with a few old favorites like pitch bend and mod wheel. In fact continuous controllers like these are the most useful modulators, but also try to include a switch (like the patch buttons on some Ensoniq keyboards) because these can be very useful for certain effects. For instance, if you have programmed a restricted modulator range for an algorithm parameter you can instantly switch from off to a preset depth of modulation.

Program Change Master Switch

This on or off parameter governs whether any program change messages will be recognized irrespective of individual unit setups. It must be set to "On" if you wish any of the units to receive program changes.

One use of this function is to act as a temporary override, stopping the DP/4 from receiving program change commands without having to change the individual unit parameters.

Control 7 Reception

MIDI controller 7 messages are used to send volume data. If this parameter is set to "Yes" the DP/4 will receive this data and direct it to a unit with a matching channel number, otherwise it will be ignored.

If any of the units share a channel number all of those units will be affected by volume changes from a matching channel. For complete MIDI controlled effects mixing give each unit a different channel number and each one can then receive different volume changes on matching channels.

Send Program Change/Controllers

The DP/4, as a multi-effects generator, is geared toward being a MIDI reception unit. By using a computer software program or a workstation-type keyboard, it can be easily incorporated into a complete system and controlled remotely and/or automatically. If this is how it is normally used in your setup it is unlikely that you will want anything coming out of the Midi Out socket, so most of the time this parameter will be set to "No."

Having said this, there may well be MIDI setups where you do want to send out data from the DP/4 and this parameter caters for that possibility. Program changes will be sent out on the preset's primary unit channel and also any controller data generated by a connected foot switch or pedal.

Modulation Response Rate

When using controllers to modulate parameters it is possible that a parameter will be changed either too quickly or too slowly when modulation is applied. This control supplies some help by changing how quickly modulation destination values are altered by incoming data. The slowest response of 1 has the highest resolution with the fastest response of 30 giving a rougher modulation.

This is very much a try-it-and-see parameter. If the sound of your modulation is not what you want, try using this control to see if it can make an improvement. Each effect/ modulator combination can require a different rate but generally a fine resolution is more useful, especially when controlling volume.

System Exclusive Dumps

Outgoing system exclusive dumps are controlled by a facility which is accessed by pressing the Write.Copy button whilst System.Midi mode is selected. A screen with two parameters lets us send system dumps from the DP/4 to an external storage device. Combinations of the two parameters allow individual preset or bank dumps of any 1, 2, 4, or config units, as well as all preset banks, all system/midi parameters, or a complete dump of all data in the DP/4 (except ROM presets). Once a type of dump has been selected another press of the Write.Copy button actually sends the data. The screen indicates that a transfer is being made and then tells you if it was okay or not.

Incoming dumps are automatic with the screen displaying a message to say what kind of dump has been received or if there was an error. However, there are two System.Midi parameters which let us set a system exclusive ID and allow or disallow reception of system exclusive messages. The first only needs changing if you are using multiple DP/4s with something like a computer librarian program, and the second only needs to be set to "Disabled" if you have the unlikely problem of stray MIDI dumps affecting the DP/4.

Maybe these parameters cannot be described as particularly exciting, but given a bit of imagination they can be utilized to give some exciting results. Ta-ta for now.

Bio: Steve Byhurst is a British composer of electronic music. He is still trying to convince people that his work is serious enough to be paid for.

LFO Fundamentals 101 (No Prerequisite) (or) Hello, I'm Joe's LFO

Dan Rohde

One of the mysteries to unravel when learning to program your own sounds is how to use an LFO to your best advantage. Everyone who has acquired a synthesizer has experienced the initial bewilderment of wondering what the heck all those buttons and sliders do. In my case, that occurred when I bought a KS-32. It played great at the store, and I figured any questions I'd have when I took it home would be answered by a perfunctory perusal of the *Musician's Manual*. (Yeah, right.) Actually, learning to program the LFO is not that difficult if you approach it in a step by step manner.

But first of all, what *is* an LFO? A Low Frequency Oscillator generates frequencies too slow to hear but very useful in making other sounds more human-like or interesting. As an analogy, imagine a bobber floating in a lake. As the waves move faster or with greater depth, so does the bobber. An LFO also creates waves, but they are waves of sound energy that are much more precisely generated and controllable than water in a lake. In the LFO's case, the bobber is also where we attach the wave's energy to the signal of yet another oscillator.

In the old days, which in synthesizer time means less than 20 years ago, connecting oscillators was done with electric cords on what looked like old telephone switchboards. The word "patch," in fact, comes from the way these cords seemed to sew the control panels together. Today's buttons and sliders are wonderfully convenient, but ironically they can make the connections harder to visualize.

The KS-32 uses a separate LFO for each of its three Voices within a Sound. Its 32 Voice polyphony means that as many as 32 Voices can be used at once, and therefore so can 32 LFOs. That's a lot of bobbers in the lake — and other Ensoniq keyboards have even higher numbers. Each Voice can use its LFO in up to three different ways at the same time: to modulate Pitch, Filters, and/or Output. Using the LFO as a Modsource for Pitch produces a vibrato effect by varying the Voice's frequency. Using the LFO as a Modsource for Filters produces fluctuations in the Voice's treble and bass qualities. Used as a Modsource for Output, the LFO varies the Voice's volume. Other uses may exist, but they are like eating pizza from the crust side — just too weird for this article.

Now let's look at how an LFO is programmed in a Voice so that we can then reprogram it to better meet our particular needs. Here's how to do it:

1. Choose any Sound, then press the Edit Sound button.

2. Select Wave (00) and pick one of the three Voices to view. If more than one Voice is on, solo one of them to hear your changes more easily.

3. Change the Effect (80) to any of the first four reverbs since the others, such as Chorus or Rotary Speaker, also make changes harder to hear because of their tremolo effects.

4. Select LFO (30) to adjust its parameters.

5. Select Pitch (11), Filters (42) and/or Output (70) to determine whether and/or to what depth LFO is used as a Modsource.

6. Unsolo the Voice if necessary to hear how it sounds with the other voices.

7. Choose the effect you want.

8. Save the reprogrammed Sound.

No sweat, eh?

Let's move from theory to practice now by working with a Sound in the KS-32, (your friend and mine, ladies and gentlemen, let's give it up for...) Multi Sax! Following our procedure above, we select ROM 05 and then press Edit Sound. Under Wave (00) we find that only Voice 2 is on. Let's change the Effect (80) to a reverb, such as Warm Chamber, to eliminate the Chorus.

Next we select LFO (30) to find the parameters Speed and Noise. Speed refers to how many waves the LFO generates

Prog: MULTI SAX (ROM 07)

Notes: Only the LFO-related parameters are shown.

TCH 1	2	3	LFO	1	2	3	OUTPUT	1	2	3
Octave			LFO Speed		36		VOL			
Semitone			Noise Rate		40		Boost			
Fine			Level		02		MODSRC		Wheel	
ENV1	00	-	Delay		40		MODAMT		+05	
LFO	+03		MODSRC		Whl+Pr		KBD Scale			
MODSCR	Noise2		Wave		Sine/Tri		Key Range			
NODAMT	+03		Restart		On		Output Bus			
KBD Ptch Track			····		1		Priority			
Glide			FILTER	4	2	3	Pan			
Glide Time			Filter 1			3	Vel window			
	and the state of the		Filter 2		1 - 1					
			FC1 Cutoff		1.		EFFECTS - C	HORUS	+ REVER	B
			ENV 2		1 million		FX-1			in the last
			FC1 KBD		Y		FX-2			
			MODSCR		LFO		Decay time			
			MODAMT		-09		HF Damping			
			FC2 Cutoff				BY (MODSRC)	Wheel		
			ENV2				MODAMT	+00		
			FC2 KBD							
			FC1MOD-FC2				C.A. uni eristante estas			
			TO INOD-FOZ				the standard in the			

per second. The chart on page 6-22 of the manual lists the frequency for each value, though it's easier to adjust Speed by ear. Multi Sax is set at 36, which the chart shows to be 5.25 waves per second. The Noise parameter on this screen refers to how fast this wave's upward and down directions change. Noise is actually a Modsource but works like a randomized LFO, making it especially handy for producing subtle, irregular shifts in a Voice.

The next LFO screen shows the Level parameter, which works like a master volume control. You must have some value here higher than 0 to enable the LFO to work as a Modsource on the other screens elsewhere in this Voice. Multi Sax's Level is 2. Its Delay is set at 40, which means the LFO kicks in about a half second after each key is depressed. This Delay can be set for up to about two seconds.

At the bottom of this screen is Modsource. Though each of the KS's 16 Modsources has its own unique abilities, Modwheel, Pressure, Modpedal, and Timbre are the most commonly used as real time controllers for the LFO. Multi Sax's LFO Modsource is Wheel/Pressure, so either can initiate the LFO IF it is chosen elsewhere in this Voice as Modsource. Moving the Wheel lets you vary the depth of the LFO's effect, whereas Pressure works as an on-off toggle switch. You can adjust the keyboard's sensitivity to Pressure, by the way, under the System-MIDI button. (I like mine al dente.)

The last LFO screen shows Wave and Restart. The KS's LFO can produce seven different waveshapes, shown in the manual on page 6-24. Like waves on a lake, these waveshapes can be either smooth or choppy, each causing our bobber to bounce in a different vertical pattern. Four waveshapes, Triangle, Sine, Sine/Tri, and Sawtooth, are

similar in that they all undulate both above and below a starting point set by Pitch, Filters, and Output parameters. The other three waveshapes, Pos/Sine, Pos/Tri, and Square, can only move in one direction away from their starting point, but this direction can be programmed by giving the Modsource a positive or negative value. Triangle, Sawtooth, and Square have the sharpest angles so they produce the most abrupt modulations. Only experimentation will help you choose the best waveshape to use in your Voice. Multi Sax uses the Sine/Tri waveshape. That is, only Sine/Tri's pattern will be used whenever LFO is chosen as Modsource throughout this Voice. Restart On means that the LFO begins from its starting point with each key down. Multi Sax Restart is On, so we always get a slight Delay before the LFO's effect can be heard.

The next place we'll look for LFO programming is under Pitch (11), which shows four parameters. Env1 is always routed (hardwired) to Pitch. Clearing this value to 0 helps us hear the LFO's effect more easily. Multi Sax's Env1 is already 0.

To the right of Env1 is... LFO again?!? The LFO is hardwired to Pitch because it's used so often here as a Modsource to create a vibrato effect. This also allows us to choose another Modsource below on the same screen to modulate Pitch. Another advantage of this permanent LFO parameter is that we can give it a plus or minus value not possible on the first LFO screens. A positive value begins the waveshape in an upward direction; a negative value, downward. Multi Sax is set at LFO=+03 so its vibrato will move upward first — but only when we use its Modsources, Wheel/Pressure.

The Modsource below shows Noise 2=+03. On page 6-23

the manual shows us an illustration of Noise 2, which has smoother variations than Noise 1. We've already set its Rate at 40. This screen, however, is where we set to what depth Noise 2 affects Multi Sax's Pitch. Noise 2 is not affected by the LFO's Modsource, by the way, but always creates a constant, random fluctuation. If you want to hear Noise 2's effect more clearly, raise its value to +25. (Just for fun, set Pitch LFO=0 and Noise 2=+25 at Output Modsource to hear the Annoying Horsefly patch.)

The last place Multi Sax uses its LFO is for the Filters Modsource (42). LFO=-09 here means the Sine/Tri waveshape modulates this Voice's treble-bass quality at the same rate of 36. The minus value means the waveshape's first moves downward toward a bassier sound. This LFO = -09 at Filters is happening at the same time that the LFO = +03 at Pitch. Therefore, and *e pluribus unum*, as you activate LFO's Modsource Wheel/Pressure, the higher pitches will sound bassier and the lower pitches will sound treblier. To hear this effect more clearly, slow the LFO's Rate down to 20, then set Pitch LFO=+08 and Filter LFO = -25, which keeps their same ratio. (See Figure 1.)

You may also have noticed during your wanderings through Multi Sax that the Modsource parameter for Output (70) is Wheel=+05. That is, as you boost the LFO's modulation of Pitch and Filters with Wheel, you also increase the overall volume a little. However, though Pressure activates the LFO, only Wheel, not Pressure, will also increase volume. If you were to choose the LFO as the Output Modsource, then Wheel/Pressure would both produce a tremolo. Since we already have plenty of LFO modulation at Pitch and Filter, more at Output might be overkill.

Now that we're all LFO experts, let's reprogram Multi Sax to better suit our needs. Personally, I like a little slower LFO Rate of 31, which the table in the manual shows is four cycles per second. Let's also lessen the severity of this up-down effect. To do this, lower the Pitch LFO to +01, then reduce the Filter's LFO depth to -03.

Suppose that, since we rarely use Multi Sax as a live lead sound, we'd prefer a controller for the LFO that will let us adjust its depth to a fixed rate in a sequence or Preset. Timbre does just that. For the same reason, we could substitute Timbre for Wheel at Output Modsource. As Timbre raises the LFO's depth, it would also raise the Voice's volume.

This ends our surgery on Multi Sax. The last two things to do are to choose the Effect (we began with Chorus + Reverb) and to save our altered Sound to a RAM location or MIDI storage device.

I've found that as I've learned more about LFO programming, I've gotten pickier about the Sounds I use. For instance, though Fretless Bass (RAM 10) is an excellent patch, it bugs me that Wheel is used as the LFO's Modsource. This routing causes the bass guitar to sound wacky when I use it with other sounds that also need Wheel in a Preset. How would you eliminate the Wheel's effect on



11

Fretless Bass. You make the call, then check the answer at the end of this article.

I hope this step-by-step procedure has helped demystify the LFO for you. Once you get the hang of it, you'll be programming those multi-Voice patches in no time.

(Answer: Change Pitch LFO to 00. Another way, assuming you aren't planning to use an external MIDI controller, is to change the LFO's Modsource to XCTRL.) -



Bio: Dan Rohde is a part time musician and high school teacher who lives in Muscatine, Iowa. He hopes someday to balance his classic-motorcycle-repairperson self with his romantic ride-thewind self. He also hopes to win the Powerball and to see the end to all conflicts in the world.

VFX Hackerpatch

Kirk's patch is the runner-up in our fattest-sound-usingone-voice patch contest. His sound, admittedly quite portly for one solitary voice, owes its obesity to two main factors. First of all, Kirk employed a pulse-width-modulation transwave, and swept through the wavetable using an LFO. In other words, the waveform is constantly changing as a note is held, adding a nice animation to the sound. Secondly, it's all pumped through Ensoniq's wonderful 8-voice chorus effect for extra largeness and motion.

The only tweak I added to Kirk's handiwork was to speed up the attack scoop created by Envelope 1 — it was a little sluggish for my tastes. I simply changed the attack from 08 to 04, and that did it.

SD & VFX Prog: LARGE-SOUND

By: Kirk Slinkard

NOTES: This is a Pulse-wave pad sound that I tried to make compatible with the SQ types. (Or at least translatable.).

AVES	1
Wave	Pulse .1-x
Wave Class	Transwave
Delay	000
Start	42
MODSRC	LFO
MODAMT	+42
OD MIXER	1
SRC-1	
SRC-2	-
SRC-2 Scale	
SRC-2 Shape	•
тсн	1
Octave	-1
Semitone	+00
Fine	+00
Pitch Table	System

00	1
0*	1
*0	1
**	1
NV1	1
Initial	99
Peak	00
Break 1	00
Break 2	00
Sustain	00
Attack	08
Decay 1	00
Decay 2	00
Decay 3	00
Release	00
KBD Track	+00
Vel Curve	Linear
Mode	Norma
Vel-Level	99
Vel-Attack	00

PITCH MODS	1
MODSRC	Timbr
MODAMT	+62
Glide	None
ENV1	-44
LFO1	+00
FILTER 1	1
Mode	LoPass/2
Cutoff	127
KBD	+00
MODSRC	Wheel
MODAMT	-44
ENV2	+00
FILTER 2	1
Mode	HiPass/2
Cutoff	127
KBD	+00
MODSRC	Wheel
MODAMT	+50
ENV2	+00
OUTPUT	1
VOL	99
MODSRC	Veloc
MODAMT	+14
KBD Scale	+00
LO/HI Key	100
Dest Bus	FX1
Pan	00
MODSRC	Env2
MODAMT	+99
Pre-Gain	Off
Voice Prior	Med
Vel Thresh	+000
LFO	1
Rate	1
MODSRC	18 Noise
MODAMT	-01
Level	99
MODSRC	*Off*
Delay	00
Waveshape	Triangle
Restart	Off
Noise SRC RT	92
PERFORMANC	= 09(1)71.5
Timbre	00
THINK	
Release	00

Sam Mims

ENV2	1
Initial	00
Peak	99
Break 1	00
Break 2	99
Sustain	00
Attack	45
Decay 1	45
Decay 2	45
Decay 3	45
Release	45
KBD Track	+00
Vel Curve	Linear
Mode	Repeat
Vel-Level	00
Vel-Attack	00
Vel-Allack	00
ENV3	1
Initial	70
Peak	99
Break 1	99
Break 2	99
Sustain	99
Attack	07
Decay 1	00
Decay 2	00
Decay 3	00
Release	51
KBD Track	+00
Vel Curve	Linear
Mode	Normal
Vel-Level	00
Vel-Attack	99
PGM CONTRO Pitch Table Bend Range Delay	Off X1
Restrike	50
Glide Time	00
Canda Little	00
EFFECTS (1)	1. Strate
Effect 8-	Voice Chorus
FX1	67
FX2	50
EFFECTS (2)	
	05
Chorus Rate	99
	99
Chorus Rate	54
Chorus Rate Depth	
Chorus Rate Depth Delay	54

Vocoded Vocals

Tom Shear

Say what you will about your Ensoniq sampler. It only lets me have eight instruments at once. It doesn't have enough polyphony. It doesn't let me have all the memory I need. Yes, say what you will, but there is no denying that the EPS-16+ and ASR-10 have the industry's best on-board effects of any synth or sampler. Hands down. This will probably change, but then again, we *are* due for a new Ensoniq sampler any time now, aren't we?

The effects go a long way toward making samples sound really professional, but not many people consider using them for vocals. Still fewer consider the vocal processing possibilities other than some reverb and a little chorusing. However, if you have the inclination, you can twist your voice around in some pretty wild ways on your trusty ol' 16+/ASR.

But just how do you get your voice to go through the effects in your sampler? Well, there are two ways. If you happen to get your hands on Waveboy's Audio-In disk (all of Waveboy's products are highly recommended for Hacker types...), you can simply plug a mic into the input jack on the back of your sampler and go crazy. If you don't have the Waveboy disk, you can get the same results by sampling your vocals and feeding them through the on-board effects.

With that out of the way, let's decide how we're going to mangle our voice this month, shall we? With the recent popularity of retro sounds, I thought this month I'd show you how to produce a fairly decent vocoder emulation. Keep in mind this is a very basic imitation with static pitch... this'll basically give you robotic, Kraftwerk-like vocals, and not the eerie "sung" vocals also made popular by vocoders.

Fire up your sampler and either call up the Flanger+Rev algorithm (available on both the EPS-16+ or the ASR-10) or load the corresponding Waveboy effect. Hit EDIT-EFFECT and prepare to dig in.

First, set your BUS 1 REVERB MIX to 99. This is a 100% wet effect, so that the result will be totally effected with none of your voice's "natural" sound coming through. Next, set your BUS2 REVERB MIX to whatever number you want. If you wanted to process your voice AND an

onboard sound, you'd send your onboard sound to BUS 2 and set your reverb mix on bus 2 to taste. Next, give your reverb a relatively short decay by setting REVERB DE-CAY TIME to 15. Next, set your LOW FREQ DECAY to +15. This parameter lets you make low frequencies decay faster or slower than high frequencies. In terms of vocal processing, it can help to eliminate "boominess" in effected vocals, a common problem in home recording situations. After this, set your DIFFUSION to 50. Diffusion is basically a "smoothness" control. Lower values give a very rough reverb in which individual echoes are often discernible, and higher values give smoother results, although on these samplers, there is often a little graininess associated with very high values, so a value of 50 is a good compromise.

Next, we'll want to set HIGH FREQ DAMPING to 0, or no damping of high frequencies, and set the HIGH FREQ BANDWIDTH to 99. The effect we're after is a high frequency effect, so we want all those frequencies to cut through. Continue to scroll right and set your REVERB DETUNE RATE to any number and your REVERB DE-TUNE AMOUNT to 0. With all of the reverb parameters set, keep scrolling right so you can set the flanging parameters.

First, set your FLANGE CENTER to 20. This is one of the most crucial parameters — as I will explain later. Set FLANGE RATE to 20 and FLANGE WIDTH to 0. With no flange width, you get a "static" flange, or one that doesn't sweep back and forth as we are generally accustomed to. Set your FLANGE FEEDBACK to +93. This will give your "vocoder" some tone without totally feeding back and getting nasty. Finally, set your FLANGE DEPTH to 99 and FBACK MODWHEEL to 0. If you're using a Waveboy effect, also make sure to set it up for external input.

Now that your effect is keyed in, make sure to save it. As with many instruments and effects, this one requires special performance techniques to be most effective. Basically, you will want to whisper your vocals through it. This will eliminate any identifiable "human" vocal timbres. So sample yourself whispering a vocal, or whisper into your mic if you're using the Waveboy. This is a relatively high-pitched effect and I like to layer it under a track of vocals processed in a different way. So what if you don't like the pitch of the effect and want a still more robotic sound? This is where the FLANGE CENTER parameter comes into play. This parameter will basically adjust the pitch of your "vocoded" vocals. A higher value will give you a lower pitch, and vice versa. With some fancy fingerwork, you can adjust this parameter in real-time using the data slider to open up a variety of pitches in performance.

So there you have it. No, it's not a real vocoder, and accordingly isn't nearly as flexible, but it's a pretty cool effect, and it's free, so whadayawant?!? Experiment with the parameters to get variations on this effect, and try sending instruments through the effect as well as vocals. Drums sound particularly cool and give effects that were utilized by industrialites like Skinny Puppy and Front Line Assembly in their early days. So enjoy your new robotic friend, and by all means pass on any further tips you discover along the way.

Bio: Tom Shear is a fictional character and any resemblance to persons living or dead is purely coincidence.

Samplin' From Your VCR

J. D. Ryan

My ears are burnin'. I can hear the readership now: "...boy this is a no-brainer...find the audio out from your VCR and run it into your sampler input, duh..." Well, okay. But you may find though that this little treatise may save you some time and perhaps diminish the hassle factor somewhat. Why would anyone want to sample audio off a VCR? How can I sample a tape legally?

To the first question I'll ask you to just think about it... And here I'll ask the legal beagles to please remain calm and shut up! In the interest of liability correctness, we will explore this exercise only in terms of you digital devotees extracting audio elements from your home videos. Truth is, video tape is a rather pristine medium for analog audio. It's often used for mixdown mastering, so why not freeze that special giggle or your child's first "Ma Ma...Da Da"? No problem.

The only technical obstacle is the lack of a volume control between the source (VCR) and the sampler. Often, you may be able to sample without introducing a choke of any kind. But if that "AMP" lamp should flicker even the eensiest, you'll need to run through your mixer. With all the volume pots in the fully closed position, simply run the VCR audio out (which is a line level signal) to an empty channel on your mixer. Now run your sampler cable to the mixer headphone output. Yow! Am I trying to fry your ears in a feedback loop? Trust me, there is no danger until you push the "yes" button after you set up your sample screens, ready to go into record. Have the appropriate operations disk booted when you hit "sample" (should your machine require it) and run through the default screens: sample rate = 44.6, filter cutoff = 20.0, pre-trigger: I usually use 7 MSEC, input level = LINE, sample time = 22.97 sec. (Remember, we decided long ago that it's much better to convert down from the highest possible sample rate than to initially sample at a lower rate, hence the short sample time on my expanded EPS-16 Plus. You will find that you'll make very short takes when extracting from video anyway.) Click on the right arrow button one more time and you are at your digital VU meter — STOP!

Now we're ready to que up the VCR. But first, our sound check. Turn on the tv and rewind your selected tape to at least two minutes before the sound byte you wish to capture. Put on the earphones and turn down the tv's volume completely. Put the VCR in "PLAY" and watch the tv for the video. Now turn or slide up your master on your mixer to whatever level you normally use (usually about 50%). Make sure everything else is turned off or down - especially the channel that receives your samplers regular outs. Now open the channel receiving the VCR audio to about 50%... Now slowly open the headphone volume control on the mixer, which is supplying the audio to the meter and A/D converter on your sampler. That headphone output is pre-amped which means it's hot! Standby at about 20% while you watch the meter on your sampler and listen to the soundtrack off your video cassette. Now turn down the channel receiving the VCR on your mixer and listen ...

Plug a set of earphones into the sampler earphone jack.

You shouldn't hear much of anything. If you hear a hum

or buzz it's because tvs and samplers sometimes don't get along so well. Also it could be 60-cycle hum so see to it that the cables from the VCR don't reside near any power cords. Stop the VCR. Any change? Probably not, so any noise will be IF interference set up between the tv and the sampler. You can rectify this by moving the tv away from the sampler. Also, turn off any amplifiers, *all* other instruments and equipment. All you need is the tv, the sampler, the mixer, and the VCR. The sound should be better than from your tv speaker. Now you have acceptable quality sound in your phones and you see a healthy signal on your meter. If it's clipping, attenuate it at the headphone control.

Don't be afraid to dress up the sound with the EQ on your mixer channel. You are hearing exactly what your sampler will digitize so get it the way you want it. Rolling up to the point you want to sample, bring the level up to the point where the AMP lamp blinks on just a bit, then back off until it remains unlit. Let the VCR roll past the point you wish to sample just to make sure there is no clip. Now you're ready to take your sample. Just rewind the VCR to about 30 sec or so *before* the point you will sample (it's easiest to gauge the upcoming audio by watching the video) As you put the VCR into play (use your remote control) your finger should just hover atop the "YES" button. As the VCR completes its preroll and approaches the audio segment you want, press the button a second before the segment you wish to capture. As soon as you hear the audio portion you wanted, push the "NO" button and you have your sample. Stop the VCR.

Assign the root key to your sample. To listen to your sample and truncate, edit, whatever, just remove the sample cord from the mixer and carry on. Now you can get those phones off and go back to your speakers. You have also discovered how to enjoy home theater audio without buying the home theater. Some camcorders do a pretty basic job of recording audio, and you will be amazed at how much you can clean it up. That, however, is another story.



Bio: J.D. Ryan is a composer/writer/dreamer somebody living back among the chickens and corn plants of his summer residence near the Atlantic ocean resort town of Ocean City.

Selling Your Services on \$4 a Day

Jeffrey Fisher

There are two crucial keys to building and sustaining your successful music business. They are:

- · Generating and contacting new business leads.
- Bringing your specific benefits to current and past customers.

You should always work hard to keep your name alive and deliver your message to those who need what you offer. Unfortunately, most of us have precious little time and often even smaller checkbooks. Here are two tips I use to promote my music products and services.

Moneymaking Promotional Tip Numero One

Rather than concentrate on sending out huge bulk mailings all the time, it's easier to spend a little time each day marketing. One way to do this is to use what I call fourdollars-a-day marketing. The basic strategy is to contact ten clients or prospects each day, five days each week. You contact five by mail and five by phone. This is very easy to manage, especially for the one-person shop, and with a targeted list of prospects, this method is both inexpensive and potentially lucrative.

Each day, send out five sales messages to five prospects or customers on your list. Using a series of sales letters, brochures, newsletters, thank you notes, special offers, and postcards, try to hit the same list 3-4 times each year. Next, follow up each mailing ten days later with a telephone call checking on the receipt of the package and to answer any questions.

So, on any given day you will mail five promotions and make five telephone follow up calls. That's it. Use these opportunities to keep-in-touch with past clients and to hunt down new prospects. With this plan, you'll put your name in front of those who buy many times in a short period. Your marketing will be better — and more profitable — if you keep plugging away, bit by bit, day in, and day out. You'll begin to be recognized and the work should start rolling in.

Your cost is about 52 cents each to print and mail five letters for a total of \$2.70. The telephone calls cost about a quarter each for a total of \$1.25. This is how you can promote your products and services for \$3.95 each day. That's only \$1000 a year that you pay one tiny morsel at a time. Every single business — including yours — will benefit from this simple, effective strategy. It's easily managed, lets you build your prospect and client database steadily, keeps expenses low, and brings your benefits to the attention of your target market several times in a carefully controlled, methodical way. Try it!

Moneymaking Promotional Tip Numero Two

Running a business requires you to handle many diverse functions. Unless you integrate a computer into your business, you'll face a tough battle handling all these details.

Being prepared means having what you need before you need it. That's why I've relied on my computer to pioneer what I call "Just In Time Marketing." I borrowed the phrase from industry where it means to manufacture products as needed. Rather than stock huge inventories of raw materials, you order them when you need them, use them, and then ship the finished products out the door.

I've modified the principle and applied it to routine business tasks. You keep all your basic business information — everything you use to run your daily business — on your computer so that you can deliver what your clients and prospects need just in time. While your computer can't meet clients, it can help you manage other tasks. Automating parts of your business with your computer helps you save time and money, two important considerations for every burgeoning music business.

First, choose software that complements your method of working. The key is to get your work done efficiently. Today's sophisticated word processors are all many businesses need. You can do mail merge, rudimentary spreadsheets, and keep your files together in one place. Using macros, you can automate many tasks such as standard salutations, closings, and more. Other than a financial and desktop publishing program, my word processor has everything I need to run my music, business consulting, and publishing businesses. Next, put everything on your computer — especially general correspondence, promotional materials, proposals, contracts, budgets, invoices, thank you notes, and more. It takes time to gather this material, but in the long run you save time. Once you develop boilerplate copy, you can use it repeatedly. You can personalize each piece through mail merge or by simply slipping in a custom paragraph or two. That's certainly less time consuming than typing something from scratch every time you get an inquiry.

Finally, let your computer be an integral part of your everyday business. Computers excel at routine tasks and most of your work is repetitive. Delegate some of your workload to that box on your desk. All you should do is select a file, update it quickly, customize it to your target, and print. Your computer frees you from those everyday tasks giving you more time to be creative, productive, and successful.

Here's an example: When someone inquires about my *Musician's Business Building Bookshelf*, I add their name to my mailing list in my word processor and then use mail merge to print a cover letter and envelope. I open another file to print the latest issue and add it to the package. I fold the pieces, stuff them in the envelope, grab a stamp, and the inquiry is filled. What could be easier? You get three advantages to this strategy:

It helps your cash flow. You don't spend money to preprint garden variety brochures or other promotional material. Your materials are always up-to-date. They aren't written in stone; you can change them, customize, and print on demand. Instead of drafting from scratch, you spend your energies on more important duties. You can put together the material you need quickly and easily. Grab an old proposal, update the budget, slip in the prospect's name, and print.

So get started right now and soon you'll have everything you need at your fingertips. And start using your time, resources, and money more effectively so that you are ready for anything that comes your way.



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Hacker Reinitialization

Newbie's Corner

One of our most common requests from new subscribers (new owners) is for more basic tutorial information. We've all been there. Unfortunately, the *Hacker* is usually "there" when a new instrument first makes its appearance — and then we move on. While back issues can answer many questions, not all are still available and they do represent an additional expense for the new reader. Hence, "*Hacker Reinitialization*" — yup, old goods in a new wrapper. We feel a little funny about the whole reprint thing — so we're going to keep it small. Clark's series on the SQs is the most requested, least available, and the most generally applicable (KSs & KTs in particular — and he's checking 'em for freshness), so here we go...

SQ Modulation — The Story

Unfolds

Clark Salisbury

Welcome back to "SQ-1 Programming Made Painfully Obvious," our continuing series on programming the SQ family of synthesizers. Last month we began an in-depth look at modulation. As promised, we'll continue with modulation this time out.

Each SQ-1 sound has up to three voices. The most elemental part of each voice is the wave, which is routed through a filter, an amplifier (also called the "output" section), and the digital effects. Modulators can be applied at many points along this route. They can be applied to various facets of the wave, such as pitch and start index. They can be applied to the filter, to shape the harmonic content of a voice. They can be applied to the amplifier, to shape the way a voice's volume changes over time. And modulators can be applied to the digital effects to allow for real-time effects control, as well as to allow for some sophisticated types of control not always found in outboard effects devices.

All in all, the SQ-1 provides a total of 15 modulation sources, along with a few special purpose modulators (such as the ramps, which are applicable only in certain areas such as the digital effects). These modulators include a Low Frequency Oscillator, or LFO (we've talked about LFO's a bit already); three envelope generators (which we will begin focusing on in this installment); two kinds of noise; velocity and keyboard position; timbre control, pedal controller, and pitchwheel; external controllers; pressure; modwheel; and the "MAX ON" modulator.

Any of these modulators can be connected to the wave, filter, amplifier or output section, and many of them can be used to control a parameter of your choice in the digital effects section. As a matter of fact, it's possible to use two or more modulators simultaneously on a single destination, such as the filter. Because of the complex possibilities presented, we'll want to take our time and be certain that we have a solid handle on modulation before moving on. We begin with envelopes.

An envelope is a generic control signal used to modulate some component within a synthesizer. In the SQ-1, an envelope can run in any of three modes: normal, finish or repeat. In normal mode, an envelope is a one-shot controller, activated by a key depression. This means that once a key is played, the envelope runs once through its predetermined cycle and stops, waiting to be activated again. What the envelope does once it has been activated depends on the settings its settings and what it's routed to control. This will be easier to understand if we approach it through example.

Select ROM program 77, "MALLETS." Press the "EDIT" button, then the "WAVE" button, and then "0." This takes you to the select voice page. >From here, we can see that there are two voices active in this program, (voices one and two). For now, turn off voice two by scrolling to select it, then hitting the down arrow button. This leaves us the basic mallet sound, but with only one voice active. Scroll to select voice one for editing — the word "ON" should start flashing.

Now press "AMP," then "0" to access the amplifier envelope; this is the envelope that generally controls overall dynamic contour. You should find yourself at the first page of the amplifier envelope menu. The display should look something like this:

LEVL =99 99 00 00 TIME =11 50 27 50

The upper row of numbers is showing envelope levels and the bottom row is showing envelope times. In this case, the envelope levels are controlling volume. The times determine how long it will take to go from one level to the next. In this example, the envelope begins at a level of 99, (the maximum level available), then takes a time of 11 to move to the next level, also 99. From there, it takes a time of 50 to move to the next level, which is 00, (the minimum available), and so on. Let's experiment.

Play a note, and listen to how the sound decays as you hold down the key. Now select time 2, the second parameter in the "TIME" row — it's currently set to a value of 50. Change this to 10 and play a note. Notice that the sound decays much more abruptly. That's because it is only taking a time of 10 to get from the second level, 99, to the third level, 00. Try setting the second time to a value of 99 and play a note. Notice that you need to hold the

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Set time 2 back to its original value of 50. Now set level one to 00. The display should be showing:

LEVL =00 99 00 00 TIME =11 50 27 50

Now when you play a note you'll notice that the note fades in, rather than attacking percussively. This is because the SQ-1 envelope is now set to go from level one (00) to level two (99) with a time of 11. Now try setting time one to a value of 00 surprise! The percussive attack is back! You see, even though the envelope is set to start at a level of 00, it takes 00 time to get to the second level, which is 99 (full on). So you can control how long a sound takes to attack by using level one to set the initial level, level two to set the target level, and time one to control how long it takes to get there; it's almost like a game of "connect-the-dots." Level three and time three function in precisely the same way, allowing you to create envelopes that have more complex attack/decay characteristics. Level four and time four, however, function somewhat differently.

Level four is what is known as the sustain level of the envelope. What happens is that the level set at level four will remain constant for as long as a key on the keyboard is held down — that is, once it's been arrived at after the first stages of the envelope have run their course. For example, set level four of the envelope we've been working with to a level of 99, and hold down a single note on the keyboard. Notice that the note attacks percussively, gradually decays to silence, but then swells back to full volume a moment later. If you continue to hold a key down the note will continue to sustain. The envelope we've created should look like this:

LEVL =00 99 00 99 TIME =11 50 27 50

As you can see, this envelope is set to begin at a level of 00, then take a time of 11 to reach a level of 99, then take a time of 50 to reach a level of 00, and then to take a time of 27 to reach level four — the sustain level — set to 99. See? This is easy! "But what is time four for?" you may well ask ask. Well, well, it's called the "release time," because it sets the amount of time it takes for the envelope to get from whatever level it's currently at back to 00 after the key has been released. To hear this effect in action, set time one to a value of 00, level three to 99, and set time four, (the release time), to a value of 00. The display should look like this:

LEVL =00 99 99 99 TIME =00 50 27 00

This envelope goes immediately to a level of 99 (time one is set to 00) and stays there for as long as the key is held down (sustain level — level four — is set to 99). Notice that when the keys are released, the sound decays to zero rather abruptly — that's because time four (release time) is set to 00. Try setting time four to 60 or so; notice how the sound now takes a while to decay after

the keys are released. The higher you set the release time (time four) the longer it takes for the sound to decay once the keys are released.

Let's apply what we've talked about so far to see if we can't make a basic piano sound. First we'll need to select an appropriate wave to use, so press the "WAVE" button, and then "1" to move to the wave page (remember that we still have voice one selected for editing). As you can see, the "MARIMBA" wave from the "Tuned Percussion" group is currently selected. Let's change this to one of the piano wave.

Scroll to select the words "TUNED PERCS" (they should begin flashing). Now use the data slider or the down arrow button to cursor down to the "STRING WAVE" group (the piano waves are found, for the most part, in the "String Wave" group), and cursor over to select the "STRING ENSEMBLE" wave. You can now scroll up to find the piano wave you'd like to use. I'm going to use the "DIGITAL PIANO," but the choice of waves is up to you.

Once the wave has been selected, we'll need to return to the Amp envelope page. Press "AMP," and you should be there. If not, press the "0" button as well.

Now let's think for a moment about what the dynamic contour of a piano sound is like. We know that it's percussive, so we'll probably want to start out at a high level with a short attack time. Setting level one of our amp envelope to a value of 99 should accomplish just this.

We also know that a piano decays as the note is held, so we'll need to set our sustain level (level four) to 00 — we wouldn't want the piano sound to continue to sustain while we hold the keys or sustain pedal down — it might start to sound a bit like an organ. We also know that the piano sound decays fairly rapidly once the keys are released. The sound doesn't die away instantly, though; the notes ring on for a brief moment after the keys are released. This means that we'll need to set our release time (time four) to something appropriate. Let's try a value of 40 for now. Now all that's left is to set how long it will take for the piano sound to decay while a key or the sustain pedal is being held.

The envelope we've designed, so far, should look like this:

LEVL =99 00 00 00 TIME =XX 00 00 40

The wild card here is time one, which controls how long it will take for the piano sound to go from its initial level of 99 to its subsequent level of 00. You can figure out the setting for time one simply by listening to its effect on the sound — select it and adjust it while playing a few notes on the keyboard. I find that a value of 70 or so seems to work OK for the middle notes on the keyboard, but the low notes and high notes are suffering. The high ones sustain too long, giving the upper end of the piano an organ-like quality, while the lowest notes don't seem to sustain quite long enough. What we need is a way to use the keyboard itself to control the envelope times. This would allow us to have shorter envelope times for the high notes and longer envelope

times on the lower notes. Fortunately, just such a parameter exists in the SQ-1, but there's a trick — it only works with time two and time three. This means that if we use time one to set the decay for our piano envelope, the "keyboard tracking" parameter (as it is called) will have no effect. So what we'll need to do is use either time two or time three for our decay parameter. Fortunately, this is easily accomplished.

First, set level two to a value of 99 and time one to a value of 00. This way, the envelope still starts out at full on, but it immediately moves to the second level (since time one is set to 00, the minimum time possible), and from there it can move on to its decay stage. Now let's set the value for time two to the value we originally roughed in for time one: 70. Our envelope:

LEVL =99 99 00 00 TIME =00 70 00 40

Now let's check out the "keyboard tracking" parameter. Press the "2" button, and you'll find yourself at the following display:

Mode=NORMAL KeyboardTrk=+28

We'll talk about the mode parameter in our next installment. Right now we're interested primarily in the keyboard tracking parameter. As you can see, it's set to a value of +28; this is the value originally programmed for the marimba sound that we started with. Try setting this parameter to a value of +98, and play the highest and lowest notes on the keyboard and hold them. The effect obtained is a bit radical for our purposes, but it serves well to illustrate what keyboard tracking does. Try setting the tracking parameter to -98, and play the high and low notes again — the effect is now reversed, with the lower notes decaying quite rapidly, while the high ones hang on forever. With keyboard tracking set to +00, envelope times are equal for all areas of the keyboard; you may want to check this out as well.

At any rate, my tastes tell me that a keyboard tracking value of +28 works pretty well, but you'll may want to play with this one to find out what sounds best to your ear.

That about wraps it up for this installment, but it hardly wraps up the subject of envelopes. We'll be back next time, and by then, we should be fairly well on our way to creating some pretty serious sounds — pushing back the envelope envelope as it were.



Bio: In addition to composing, technical writing and running Virtual Music (a project studio), Clark Salisbury works with Music & Sound Associates, a company that specializes in sound design, sequencing, and consultation.

SQ, KS, KT Family Hackerpatch

By: Mark Haymond

Prog: TUBULAR CHOIR

Notes: This is a light mix of vibes & choir pads. Voice 1, the vibe sound, cuts through with the attack, and Voice 2, the choir, holds on the sustain. The modwheel will increase

choir filter cutoff and raise the pitch one octave. You may want to move this pitch mod to the modpedal.

VAVE	1	2	3	LFO	1	2	3	AMP	1	2 3	
Select Voice	On	On	Off	LFO Speed		1200 000	112 01010	Initial	99	99	- 44.2
Wave Class	Exp	Exp		Noise Rate		-		Peak	99	95	
Wave	Vibes	Choir		Level	-			Break	00	90	
Delay Time	000	00		Delay				Sustain	00	80	
Wave Direction	Fwd	Fwd		MODSRC	-	-		Attack	16	09	
Start Index	00	00		Wave	-	-		Decay 1	56	00	
MODSCR	-			Restart		-		Decay 2	30	94	
MODAMT	00	00				and the second	1000	Release	46	50	
Restrk Decay	30	30		FILTER	1	2	3	Vel-Level	06	00	
			The second	Filter 1	2Lo	2Lo		Vel-Attack	00	93	
ITCH	1	2	3	Filter 2	2Lo	2Hi		Vel Curve	Lin	Lin	
Octave	+0	-1	and the second	FC1 Cutoff	065	040		Mode	Norm	Norm	
Semitone	00	00		ENV 2	+76	+64		KBD Track	+28	+00	
Fine	00	+04		FC1 KBD	+00	+14					-
ENV1	00	+00		MODSCR	Kybd	WheelP	TOP				
LFO		+00		MODAMT	+00	+07	105	OUTPUT	1	2 3	
MODSCR		Wheel		FC2 Cutoff	096	000		VOL	50	70	11.12
MODAMT		+62		ENV2	+00	+00		Boost	Off	On	
KBD Ptch Track	On	On		FC2 KBD	+00	+00		MODSRC	-	Wheel	
Glide	Off	Off		FC1MOD-FC2	On	On		MODAMT	00	-10	
Glide Time	Uli	UII		1011000-102	UII	Un		KBD Scale	+00	+60	
Calido Tillio			Contraction of the second					Key Range	E1-G7	E1-G7	
								Output Bus	FX2	FX1	
NV1	1	2	3	ENV2	1	2	3	Priority	Med	Med	
Initial	-			Initial	99	36		Pan	+00	+00	
Peak	-	-		Peak	99	60		Vel window	000	000	
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Attack	-			Attack	00	06		FX-1	15	FX-2	1
Decay 1	-			Decay 1	00	24		Decay time	42	HF Damping	3
Decay 2	-			Decay 2	00	86		Chorus Rate	20	Chorus Depth	
Release	12	1 2000		Belease	00	48		Chorus Center	50	Chorus Depth	
Vel-Level	-	4		Vel-Level	00	53		Feedback	+00		
Vel-Attack				Vel-Attack	00	00		Chorus Level	40		
Vel Curve	-			Vel Curve	Conv	Lin		MOD (Dest)	FX1-Mix		
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nee noon			and the second second	INDU HACK	+00	+00	Statistics of	MODAMT	-20		

The Hack: With "Tubular Choir" Mark Hayman's heart is in the right place... but his envelope values... let's just say they're a shade too articulate for my taste. I think both voices fare better with a slower, softer attack and a slightly extended release. The changes in Env 2 values for Voice 1 are as follows: Initial = 00, Attack = 99, and Decay 1 = 99. The changes in Amp values for Voice 1 are: Initial = 54, Peak = 95, Sustain = 16, Attack = 57, Decay 1 = 99, and Release = 65. For Voice 2 the changes in Amp values are: Initial = 00, Peak = 99, Attack = 60, and Vel-Attack = 19. At the Output, Voice 1's Vol balances to 74. Add a little stereo; let the Pan for Voice 1 = +42 and for Voice 2 = -42. Both voices benefit from more chorus within the effect. For simplicity's sake, keep the effect Chorus & Reverb. (Though please note you must change Voice 1's Output Bus to FX1.) At the effect, change FX1 to 75, Chorus Depth to 40, and Feedback to -44.





Bio: Jeffrey Rhoads has been a keyboardist/composer on the Philadelphia Jazz and R + B scene for a period of time resembling forever. He has an interest in cinema and has developed some film courses. Jeff still believes in magic and longs for city lights.

Hackerpatch is intended to be a place where patch vendors can show their wares and musicians can share their goodies and impress their friends. Once something's published here, it's free for all. Please don't submit patches that you know to be minor tweaks of copyrighted commercial patches unless you have permission from the copyright owner. All submitted patches are subject to consideration for mutilation and comments by Sam Mims and Jeffrey Rhoads — our resident patch analysts. If you send in a patch, *please* include your phone number. Requests for particular patches are also very welcome.

Sampling, Sequencing and Live Performance

Tom Shear

We are living in the digital age. It seems that nowadays there's a computer chip in just about everything. While it has undoubtedly made our lives easier, on another level, it complicates them as well. Especially for us, as musicians. Think about it. We can store note-perfect performances of out playing on a floppy disk, correct any timing mistakes, even go in note by note and adjust the velocity until it's perfect.

But with this goes the standard barrage of insults and put-downs from guitarists and other non-electronic musicians who are amused with these machines that "play themselves," never considering that a human being had to put the notes in there in the first place. This is worsened still when another Milli Vanilli comes along and makes fools out of not only themselves, but the record buyers as well. And yet again, electronic music gets a bad name. (Never mind that there are plenty of guitar-based artists from the '60s who never played or sang on their own albums...). So this obviously presents a sticky problem. If you're an electronically-based performer, how can you pull off live performances that all the nay-sayers will be astounded by? Below are a few suggestions to get you started...

1) Drum Pads

If you're a predominantly or totally electronic group, the easiest way to bring the excitement level of your live performance up is to purchase some MIDI drum pads to play live rhythm parts. Whether you just use them for percussive accents, or for playing the entire drum parts of your songs, this is guaranteed to be more interesting to the audience than a bunch of people hunched over their keyboards. There's something very exciting about the physicality of watching a drummer, especially an enthusiastic one, play. And don't assume this is for drum parts only... if you have a product like the Roland Octapad or the DrumKat, why not use them to play melodic parts as well? Those heavily sequenced parts that are difficult to play live on the keyboard might be easier to play with drumsticks, with a little practice.

Sequencing "On the Fly"

Having all your sequenced parts playing back on tape might be convenient, but it is also extremely restrictive. You're stuck with the same arrangement, same song length, and song order night after night. If it is feasible for you to use a sequencer live instead of tape, some new possibilities open up. I, for one, almost never use the SONG function on my EPS-16+. Instead, I just keep the individual sequences by themselves, and switch them as needed as I'm recording. If you've got a free hand during performance, this is a good way to give a more exciting show. As anyone who has played live knows, audience reaction can really affect where the songs goes. If they're going crazy, it'd be nice to be able to extend that song a little bit, and conversely, if they're throwing bottles at you, you'd probably like to cut it short. Leaving your sequences unlinked allows you to put the song together live however you feel like it, adjusting for audience mood.

Not having your sequences on tape also lets you perform different arrangements of your songs if you get the urge. Maybe one night you decide that you're going to change that Hammond organ part to a string section and slow it down by about 20 BPM. If you're on tape, this isn't gonna happen, but with everything in your sequencer, you could practically play different arrangements every time you played, if you so desired.

3) Sampling as an Alternative

Another more risky alternative would be to sample the individual parts of your song and trigger them as "loops" in performance, much as you might use a rap drum loop. While one band member triggers these loops, the others can fill out and add the other parts that change throughout the song. This gives you the same flexibility as #2 above, but allows you to go a little further. You can trigger the loops in a syncopated manner, creating another variation on your riffs. You could even set up patch selects allowing yourself to go into an extended 8 minute backwards jam on the song if you wanted to really freak the audience out. The point is that there are at least 37 ways to make performances more exciting, and the more of these you explore, the more your audience will thank you for it. Of course, with this new-found flexibility comes risks. You'll probably screw up every now and then. But maybe that's what missing from a lot of electronic performances. There's little room for error. There's none of that adrenaline rush that comes with going on stage wondering how your performance is going to turn out. And consequently, little chance that any given performance is going to be sig-



nificantly better than all the rest because you're out to prove yourself to a skeptical audience.

Bio: Tom Shear has been replaced by a computer.

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This is probably one of the most open forums in the music industry. Letter writers are asked to please keep the vitriol to a minimum. Readers are reminded to take everything with a grain of salt. Resident answer-man is Clark Salisbury (CS). Letter publication is subject to space considerations.

Transoniq,

Do you know of any sites on the internet or www that has a list of downloadable ASR samples? If so, could you email me with a response? Any help would be appreciated.

Also, what type of converter would I need to convert the files over to my ASR?

Jason Huffman via Internet

[TH - Probably the easiest way to get this info is for you to check out our web page at http://www.teleport.com/~trnsoniq. From there you can go to our ftp site and get Garth's article covering internet samples and conversion software and stuff. You can also check out our "Other Links" page. It has a link to the Sound Management BBS page and the Oakland EPS Archive - both of which have samples.]

To: interface@transoniq.com Subject: Encoded Files.

Just a note regarding the articles on retrieving files from the Internet and decoding them into a usable form. The writer suggest a few programs for this purpose. I have, however, had much better success with a program (Mac) called Mpack, available on AOL. It seems much more reliable and flexible than the programs mentioned in the article, and is as easy as dragging the encoded file onto the Mpack icon.

Tom Shear Via Internet

[TH - Thanks for the tip!]

To: interface@transoniq.com Subject: Number of sequencer tracks on Ensonia

Ya know, I think it's time to incorporate a multi-channel mode on one INSTRU-MENT of the future Ensonig machines. We have very modest computer systems transmitting more than 16 MIDI channels. It seems reasonable to be able to import a multi-track recording into one INSTRUMENT for transmission to external modules and still be able to add other on-board music.

It's really frustrating to limit the onboard musical contribution because you want to include some discreet MIDI channels. Using just four MIDI channels on an EPS or ASR means there are only four instruments left for samples.

It should be possible to record MIDI tracks on the available "hardware" positions, then "bounce" the whole wad down to a single INSTRUMENT. I can even imagine routines for step editing that would be tedious, but do-able in this scenario.

Compared to sound and effect generation, MIDI is really a "low function" job. It shouldn't take that much more cpu power to do what I've described. I'm still using my "more portable" '16+s but I don't recall this feature being available on any of the newer machines.

Tom Jordan

Wired for Sound! - computer music for kids

[CS - As far as the Ensoniq stuff goes, you get one and only one MIDI channel per track with any of their built-in sequencers, but I guess you knew that. I don't think many built-in sequencers offer the kind of track-merging-with-retentionof-MIDI-channels feature that you're talking about, but my guess is it's only a matter of time before this kind of thing becomes de-rigeur for any self-respecting audio workstation.]

Dear Hacker

I still don't know how to use all 16 MIDI channels with my SO-1 and MasterTrax Pro 5.0.1 sequencer software (Mac). All I have is eight channels. Is it possible to have more than eight channels? I have read the manual but there is very little about using SQ-1 with external sequencers.

Regards, Timo Neronen Finland Via Internet

[CS - I'm pretty sure that eight MIDI channels is the max you can get in your setup. Back when I had one of the original SQ racks, you could get it to do more than eight channels by assigning one MIDI channel to a sequence track, and another to the related song track. My experience with this, though, was that it was a bit flaky, and I'm not sure that Ensonig continued to implement this function. You can try it out, though, and see if it does you any good.

Move to the MIDI Status page [press Edit Sequences/Param (Bank 6)/ Screen 4], and set up the MIDI channel for a track as you normally would. Now, rapidly doubleclick the track button for the current track, and you will find yourself looking at the song track (you can use this doubleclick shortcut to toggle between song and sequence tracks). Set a different MIDI channel for the Song track, then doubleclick again to go back to the Sequence track. If the Sequence track retains its original MIDI channel and the Song track retains its new channel, you're living right (sorry I couldn't try this myself; I no longer own an SQ-Series instrument).]

[Ensonig - No, you are using your SQ-1 correctly. It can only receive on 8 MIDI channels. This is the same for the KS-32, and all of our samplers. The KT Series receive on 8 MIDI channels, expect when in GM mode, when they will receive on all 16. The TS series receive on 12 MIDI channels, except when they are in GM mode. Our newest instrument, the MR-

Rack will always receive on 16 channels, whether it (he?) is in GM mode or not.]

Dear People:

I wanted to respond to Ensoniq's reply to Mark Snyder in the November *Hacker*. They made an implied agreement with Mark's statement that "the new Jaz... will not work with my **EPS-16+!**"

Just because Ensoniq doesn't approve something doesn't mean you can't make it work by using some assistance.

Speaking of both the ZipDrive and the Jaz, Ensoniq does not approve the Zip, mainly because it doesn't supply termination power to the SCSI bus. If you hook a Zip to a 16-Plus directly by itself, it has problems working. Ensoniq has told me that in their approval list they want it to be "Plug and Play" – meaning that they only approve devices that don't need something else to work.

However, we have found by adding either a) another drive to the chain, or b) adding an APS SCSI Sentry (an externally-powered termination power device), the Zip drive definitely works. We have done a significant amount of testing.

The Jaz might be the same way – it is not "out" yet, so we haven't had time to test it, but I am sure the same solutions will probably apply.

Actually, at Rubber Chicken we recommend the EZ-135 Syquest for the 16-Plus at the moment. It is inexpensive, the media is cheap, and it hooks right into the 16-Plus without a problem. It really is the best buy. Hope this helps!

Also, we appreciate all reviews of our products, and the latest one by Steve Vincent ("Morpheus Chicken") is no exception. We have revised our Morpheus set based his suggestions in his review, and invite all previous owners to call us (1-800-877-6377, or 612-235-9798) for a free upgrade.

One more thing! Our e-mail address has changed – it is now chickenEPS@willmar.com. Look for our new Web page soon!

We want to wish all at the *Transoniq Hacker* and its faithful, friendly readership a Happy Christmas season.

Garth Hjelte Rubber Chicken Software Co.

[CS – Thanks for taking the time to write, Garth. I'm sure many will be interested to find out if there's a work-around making it possible to use the Jaz drive.]

[Ensoniq – We weren't trying to mislead anybody by our "implied agreement" that the Jaz Drive didn't work with the EPS-16 PLUS. We simply don't have the resources to try all new SCSI devices with our older products that are out of production and which are known to have incompatibilities with newer SCSI devices. If someone tells us it doesn't work, we assume it to be true, and if it truly doesn't work, we really aren't in a position to determine why. We appreciate people like you who try things, make them work if they can and tell us their findings.

We have worked with both the Zip and Jaz drives with our current products and they work fine. Our tests with an EPS-16 PLUS using the APS SCSI Sentry and other devices told us that they may work, but it is not guaranteed. It is likely that you can get them to work, but we cannot guarantee that, so our position remains the same.]

Dear TH:

My TS-10, whether in sound, preset, or sequencer play mode, is giving an output of a wet-only mix of an unknown delay effect algorithm. Changing effects algorithms has no effect. Obviously, the synth is useless in this state. Re-initialization has had no effect.

Any ideas short of a service call?

Colin Lapsley, Hamilton, Ontario

[CS – Nope. Either you have your TS-10 plugged into the wrong input on your mixer, or it's indeed time for a service call.]

Hi TH.

I have decided to get a new computer instead of my old dear Amiga. Which would you consider the best for using with my TS-12? A PC compatible (my guess) or a MAC? I'd like to be able to control ALL (if it's possible) the functions of the keyboard directly from the screen. Am I dreaming or does this software exist for any of the two kinds of computers? Hopefully it does. Anyway, I'd like your opinion on the best software solution for the TS-12 around right now.

Ensoniq and TH fan, Frederik Brons

[CS – Well, I'm partial to the Mac, primarily because it's generally easier to use (particularly in terms of its plug-andplayfulness), and because there's better support for music. This is changing, though, with more professional musicoriented apps appearing for the "Wintel" (Windows running on an Intel-based machine) computers. Also, the music area where the Mac has had its most entrenched stronghold has been in the realm of digital audio recording, an area you may not care much about.

Change of Address

Please let us know at least four weeks in advance to avoid missing any issues. The Post Office really will *NOT* reliably forward this type of mail. (Believe us, not them!) We need to know both your old and your new address. (Issues missed due to late or no change notification are your own dumb fault – we mailed them!)

Missing or Damaged Issues?

Every month we mail out thousands of issues and every month about a dozen get "misplaced" by the Post Office. If you're ever one of the winners of this lottery, just give us a call (503-227-6848, 8 am - 8 pm Pacific Time) and we'll be happy to mail a replacement copy — no prob. (However, if you accuse us of nefarious schemes to "rip you off," you will be offered a refund and given helpful subscription info for other musician magazines.) On the Wintel side, there seem to be more small (and not so small) utility programs designed to help you manage data for your Ensoniq synths. On the Mac side, though, is a longer history with, and more seamless integration of, SCSI peripherals than you'll find in the Wintel wonderland. This may prove significant, since none of the Ensoniq products do IDE (the more common mass storage technology in the Wintel/PC world).

As far as a software package that will remotely control every aspect of your TS-12, I don't think it exists (although I can't swear to this). However, it is possible to design and create your own front-panel emulations in a couple of the more sophisticated sequencing packages. Logic (from Emagic, available on both platforms) springs immediately to mind, and Performer (from Mark of the Unicorn, available on the Mac now and soon on the PC) may allow you to do some of these kinds of things as well.

As far as the "best software solution," it depends on what you want to do. My personal favorites? For sequencing, Vision (from Opcode Systems) or Logic (Vision should be available on the PC pretty soon); for patch editing, Unisyn (from Mark of the Unicorn, available for both platforms) is pretty much the only game in town. And for straight ahead library functions, Galaxy from Opcode (Mac only, I believe) would be my choice.]

Subject: Re: MIDI mapper

**** Announcement ****

For anyone who is interested in using the General MIDI instrument layout with the Ensoniq SQ-1/2 synthesizers, so you can play Internet MIDI files without changes, a solution is available. There were several problems, and compromises had to be made, but a working system is functioning. If you are interested in the details or want a copy of the WINDOWs MIDI mapper file that made it all possible, then visit my internet homepage http://www2.netdoor.com/~rlang for your free copy.

Remember to make a backup of your MIDIMAP.CFG file before replacing it

with mine. If you have comments on the sound matchup or suggested improvements, please let me know via email.

Bob Lang rlang@netdoor.com.

[CS – Way to go, Bob. I guess all us Mac guys will just have to eat our hearts out for now.]

To: Hacker

I'd like any information and tips available on the EPS-16 Plus. I am in the process of putting together a home studio and I need to know as much as I can about my unit. I have been using my EPS for about a year and have made decent progress, but there are a lot of things that I don't know – like how to have each instrument under a different effect yet still under the same bank effect [ie: dist+wah (inst) and dual delays (bank)].

Thank you for your assistance, Darnley Hodge Via Internet

[CS - I hate to be the bearer of bad tidings, but you can't use more than one effect algorithm at a time. The EPS-16 Plus uses a single signal processing chip, called the ESP chip. This chip is sophisticated enough to for multiple effects types within a single algorithm (such as the chorus and reverb algorithms), and the signal routing capabilities found within the programming of the EPS gives you a variety of ways to route sounds - some could go to one of the effect busses, while others are routed through another of the busses, allowing for things like having reverb on a guitar patch, and chorusing and reverb on an electric piano patch, simultaneously. But you can't use two different effect algorithms at the same time. Sorry.

On the other hand, I can bear some reasonably good tidings, to wit; there's been a lot written about the EPS, EPS-16+ and ASR-Series (most of the info is interchangeable, you know), and it's easily obtained. One good source is these very pages, but at an earlier point in time. If time travel isn't your thing, you can probably just get back issues, which should work almost as well.

Also, Ensoniq has some additional instructional stuff not found in the manuals. Contact Ensoniq Customer Service at (610) 647-3930 for more info. Or, you can virtually drop in at their web site: http://www.ensoniq.com/.]

To: Transoniq

I just bought a used EPS. It has the 4x expander and a 40-meg Orchard drive. The first day I used it, it loaded samples with no problems, and played them back with no problems. However, yesterday it started going %%% when I played them back instead of the sampled sound. The pitch control of the samples still works and although it makes a good industrial sound, this is not why I bought it.

Seth Morth Via Internet

[CS – Well, you don't ask for any advice, but I'll give you some anyway. Talk to



Ensoniq Customer Service. To the best of my knowledge, the EPS should not go $^{\%*\%\#@.1}$

Hi,

I want to connect a Syquest drive, PC, K2000, and ASR-10 with SCSI with each other.

Problem: neither the K2000 nor the ASR have SCSI-thru. How can I connect them so that the K2000 can read from PC and Syquest and the ASR can read from PC and Syquest?

I have a "strange" SCSI-cable which allows me to put another cable on the back, but I then have three terminators in the SCSI chain – so I will blow up my whole setup. (Hint: I will not connect them in this way.) And when I disable a terminator (don't know how the K2k or the ASR can do this), I have a Star network – which I think SCSI is not made for.

Who knows what to do?

End of transmission. <click> Angelo van der Weerden Via Internet

[CS – In my experience, the ASR does not like to share SCSI peripherals (such as Syquest drives) with other devices, so your lack of SCSI-thrus may be the least of your worries (although this probably wouldn't affect the peaceful co-existence of your instruments and the PC).

In actuality, the thru on most SCSI devices is simply a parallel bus - in effect, a SCSI "Y" cord - so there's no real voodoo involved in the mechanical hookup. As far as termination, you're not likely to do any damage by having three SCSI devices with termination all on the same buss, but operation may not be as reliable as one might hope. One solution (and a good one, in my opinion) is to invest in an active SCSI terminator, such as the "SCSI Sentry" from APS (you can reach them toll-free at 1-800-677-3294). For around 80 bucks or so, numerous termination headaches can be nipped in the bud. bud.

As far as getting everything physically hooked up, I would think that any supplier of computer accessories and peripherals should have some sort of device that you could use to provide an additional SCSI out. However, your "strange" SCSI cable may be just the ticket for this, but I'm not quite understanding what it is you are saying about having a "star" network. I don't think there'd be any problem using the configuration I think you're describing, but I'm not completely sure I get it. As for termination, you should be able to remove termination from any of the devices you wish. If you want help with the termination in your ASR-10, I'd suggest contacting Ensoniq Customer Service at (610) 647-3930.1

[Ensoniq - Actually, SCSI is a daisy chain network so there is no need for a "SCSI Thru." Typically a SCSI storage device will have two connectors on the back to allow daisy chaining. Neither the PC, the ASR or the K2000 offer the second connector, so daisy chaining must be accomplished with a cable like you have described, or preferably by using a SCSI switcher box. Without knowing whether your PC supplies termination or whether the K2000 supplies termination, it is difficult to tell where each device should go in the chain. The basic idea is to set up multiple "hosts" to the inputs of the switcher (most likely your K2000 and ASR) and then have the output of the switcher going to the Syquest and finally into the PC. You can try the various combinations and see if any work. You are unlikely to "blow up" your rig. The most likely problem is that it simply won't work. It will require some experimentation, but this is the right path to go down. Good luck!1

Bonjour!

I am in love with my KS-32 for more than two years (luckily enough I bought a female one), so we decided to marry and have sounds. Although we have consulted several sound-programming doctors, read books, and even watched videos (with suggestive titles like, "Physical Oscillations and Acoustic Results," or "New Key Positions to Achieve Harmony"), we have not had any happy event to announce yet.

Therefore, we are now considering adopting sounds conceived by others. The problem is that here in France the adoption procedure is long and expensive...

When you go to a reseller (as shocking as it may seem, IT IS LEGAL to sell sounds in this country!), they often don't have sounds in stock (would it be human to keep them in a back room all the time?), so they have to get them from the importer (I don't know exactly, but I understand it is a kind of sound-nursery). But how can you choose a sound when you have never heard the sound? Most of the time all the reseller can tell you is the name of the sound... But how do you make a choice between "Hammond" and "Rhodes"? One would like to see how they behave when they play! Good sounds are alive, you know.

One thing that a reseller with few morals proposed to me was to "try" another keyboard which already had sounds. (From a former programmer, I guess!) As tempting as it was, I declined the offer because I wish to be faithful to my KS. (Or is it because I cannot afford a mistress keyboard?)

So far, I think the best solution would be to meet a sister KS which already has bought some sounds in order to hear them play. It could even be possible to do so with a cousin SQ! But when my KS left Malvern a few years ago, she lost contact with her family members, and although we are convinced that some of them also made the trip to France, we don't know where they settled. We keep looking for them... So, please, Transoniq, could you publish our address so that they can join us?

So, we are waiting.

But I am also thinking to myself: Why don't sound programmers – and Ensoniq – record demo cassettes of their sounds? With banks of sounds on cassettes, anyone could make an advised choice when ordering sounds. And I think it would be easier for resellers to let their clients hear, discover their music-mate (or should I say play-mate?) has much more physical and sonic possibilities than they had imagined, and therefore order new sounds! When today, not having the opportunity to hear the sounds, people like me just don't buy – when they are faithful! The worst would be turning to a mistress-keyboard with fresh young sounds. The most perverse could even consider touching keyboards from another brand!

Isn't it a pity that all these sounds which exist should stay in their bank all their life not being played just because no one has had the opportunity to hear them before buying?

Two remarks to conclude: A rich KS owner could of course order all the sounds available, but this would be a sign of mad love because it would cost more than another keyboard and since he would not use them all some would have to be tossed aside.

Maybe sound conceivers are afraid that sounds on cassette could be stolen by the big bad sampling guys. The risk exists. But, in my opinion, professional sample-hackers who resell their spoils probably use the ROM sounds as the source.

My advice to sound vendors: let us hear them – we will buy them!

Friendly yours, Greg & KS 10 rue Jean Bleuzen 92170, Vanves France

[TH – You might also want to check out the French Ensoniq fanzine mentioned in last month's Front Panel. (Philippe LeFevre, 25 rue des Chenes, 82340 Dunes, France.) And watch how you use that word "hacker."]

[CS – A number of vendors do offer cassette recordings featuring their offerings, in order to help you make an informed, er, adoption (Unfortunately, Ensoniq isn't one of them. Oh well...). Check the ads in the Hacker, as well as some of the other publications.]

[Ensoniq – Clark is right, we don't have audio demos of our sounds. It is a huge task, as we have 11 cards of sounds each containing 160 sounds. So we would have to create over 1700 demos. (Okay, we realize we can gang some together, but you have to agree it is a huge task.) But we have all of our sound names (and Categories) available on our Fax-On-Demand system. Call 800-257-1439 to receive a complete directory, or ask for documents 0840 through 0850. Since this number won't work overseas, you'll have to call us and we can fax you copies. Hope this helps.]

Dear Hacker of Great Renown:

Just finished my latest TH - thanks for all your efforts. I'm thoroughly enamored of my ASR-10, 16-meg RAM, Bernoulli 150 setup. Sometimes I just shake my head and marvel at the incredible technical wonder sitting in front of me. I have two questions. I've been using O.S. 2.51 for the past year, with no complaints (thank God!). It's a trouble-free OS for the most part; I can even delete a wavesample and not have the whole instrument disappear (provided Stereo Link is OFF). But lately, I was in the music store and picked up OS 3.53, brought it home and tried it out. Ouch! First I loaded in a song, then went to Event Edit (a supreme test) and immediately got Error 130 Reboot? (Insert foghorn sound.) So I turned it off, waited, then on again, and then reloaded OS 3.53 and another sequence. In the middle of playing the song, Error 215, Reboot? (Insert raspberry sound.) This has happened each time I tried it. What gives? Admittedly, some of the songs were created using OS 1.61 two years ago. But aren't they compatible? I have no problems on 2.51. So for now I'm sticking with 2.51, since I

Transoniq-Net HELP WITH QUESTIONS

All of the individuals listed below are *volunteers*! Please take that into consideration when calling. If you get a recording and leave a message, let 'em know if it's okay to call back collect (this will greatly increase your chances of getting a return call).

All Ensoniq Gear – Ensoniq Customer Service. 9:30 am to noon, 1:15 pm to 6:30 pm EST Monday to Friday. 610-647-3930. Ensoniq's Fax On Demand line, (1-800-257-1439) can also be used to retrieve specs, OS info, hard-drive info, and the like.

All Ensoniq Gear – Electric Factory (Ensoniq's Australia distributor). E-mail address: elfa@ozemail.com.au; their web site at http://www.ozemail.com.au/~elfa; or e-mail their resident clinician, Michael Allen, at mallen@geko.com.au. Phone calls, Business hours – Victoria. (03) 480-5988.

All Ensoniq Gear - The Electric Factory in New Zealand, phone (64) 9-443-5916, fax (64) 9-443-5893, or e-mail geoffm@elfa.co.nz (Geoff Mason).

TS Questions - Pat Esslinger, Internet: pate@execpc.com, Compuserve: 74240,1562, or AOL: ESSLIP.

TS, VFX, and SD-1 Questions - Stuart Hosking, stuh@ozemail.com.au.

MIDI users and ASR-10 Questions – Ariel and Meiri Dvorjetski, Internet: s3761921@techst02.technion.ac.il, or dvorjet@techunix.technion.ac.il. You can also call Sincopated BBS at (Israel country code: 972) 4-776035, 24 hours, 28.8K Modem. Please Login as: ENSONIQ, Password: MIDI.

SD-1 Questions - Philip Magnotta, 401-467-4357, 4 pm - 12:30 EST.

VFX Sound Programming Questions - Dara Jones, Compuserve: 71055,1113 or Internet: ddjones@netcom.com or call 214-361-0829.

SD-1, DP/4, ASR-10 Questions - John Cox, 609-888-5519, (NJ) 5pm - 8 pm EST weekdays. Any time weekends.

SQ-80, VFX Questions - Robert Romano, 607-898-4868. Any ol' time (within reason) EST.

Hard Drives & Drive Systems, Studios, & Computers – Rob Feiner, Cinetunes. 914-963-5818. 11am-3pm EST. Compuserve: 71024,1255.

EPS, EPS-16 PLUS, & ASR-10 Questions – Garth Hjelte. Rubber Chicken Software. Call anytime. If message, 24-hour callback. (612) 235-9798. Email: chickenEPS@willmar.com.

ESQ-1 AND SQ-80 Questions - Tom McCaffrey. ESQUPA. 215-830- 0241, before 11 pm Eastern Time.

EPS/MIRAGE/ESQ/SQ-80 M.U.G. 24-Hour Hotline - 212-465-3430. Leave name, number, address. 24-hr Callback.

MIDI Users – Eric Baragar, Canadian MIDI Users Group, (613) 392- 6296 during business hours, Eastern Time (Toronto, ONT) or call MIDILINE BBS at (613) 966-6823 24 hours.

SQ-1, KS-32, SD-1, SCSI & hard drive Questions – Pat Finnigan, 317-462-8446. 8:00 am to 10:00 pm EST.

ESQ-1, MIDI & Computers - Joe Slater, (404) 925-8881. EST.

don't have a CD-ROM or digital interface yet.

Okay, next I went back east for a two week vacation and, of course, since I'm hopelessly addicted, shipped my ASR with me. Road case, hi-density foam packed, the works. It arrived just fine, but ever since some of the characters on the display do not come on initially. After an hour of warming up, they flicker back in and all is fine. Is this a cold solder joint loosened by the shipping? I'm used to soldering delicate electronic equipment for a living, and since I'm past warranty I could check for this. Or is this an LED breakdown? Have you at Ensoniq had any experience with this? What's more likely?

Thanks again for everything.

Sincerely, Stephen David Hewitt Los Angeles, Calif.

[Ensoniq - Regarding the use of the different Operating Systems, it sounds like



you do have some bad data in your sequences. But we can't be sure.

It also sounds like some of the solder joints may have fractured on either the display tube or the driver. It is possible that the display tube is damaged internally, but this is far less likely. You should definitely call Customer Service at 610-647-3930.

Note to all: it continually perplexes us why you would write a letter to the Hacker, wait for it to be sent to us, replied to, printed, and finally shipped back to you to get service questions answered. It is so much quicker to call us, fax us, e-mail us on CompuServe or the Web to get faster help. It's nothing against the Hacker, but we feel bad that you have to wait so long to get a reply. Especially one that says, "call us."

This interface is a great place for sharing ideas, asking questions, praising, and yes, griping. But it is far from the most effective way to get repair issues dealt with effectively. There, we just had to get that off of our chest(s).]

[TH - Of course we always appreciate the letters and we're sure that readers who have similar problems (and even those who don't) learn a lot from seeing the answers. Any letter that makes it here by the first Wednesday of the month will make it through the cycle and have answers going out the door two and a half weeks later (e-mail subs will get it that day). And when we see a "rush" type situation, we usually try to provide a quickie answer – even if it's just to pass on Ensoniq's (or someone else who may be able to help) number or e-mail address.]

Dear Transoniq Hacker,

Before I even think of continuing my subscription I'd like some very important questions answered.

In the past, I've always bought Ensoniq equipment. I bought a brand new DP/4 when it first came out. I made long distance calls at my expense before buying it. Wanting a processor that has a good, awesome vocoder, I asked that most important question. I wanted a processor that will get my voice from a microphone and change it to a clean electronic robot talking voice – like in groups such as Kraftwerk. I talked with a Bob Brady (?) at Ensoniq. He said the DP/4 would do the electronic voice. After many conversations with him I went ahead and bought the DP/4. But after experimenting with the DP/4 vocoder and other presets to get a robot voice, it did not sound right – not electronic, very noisy, weak.

If I can ever find the company that makes a device that does this that will be my wish come true. Everyone tells me that Roland use to put one out in the late '70s. I was also told that Korg had one at one time. I've had the worst luck searching for this processor.

I've written letters to Ensoniq and haven't gotten any explanation. I've tried everything with my DP/4. Maybe you can help. I've run out of formulas. I've been very frustrated at Ensoniq after their word meant nothing. In the past I've had an SQ-80, an EPS, a Mirage, a TS, and the DP/4. HELP!

Johnny Guillen Gallup, NM

[CS – In my experience, the success (or lack thereof) of most vocoder effects depends greatly on several factors, none of which are under direct control of the vocoder itself. The two main factors are carrier signal and modulator signal.

The modulator is, of course, your voice as interpreted through your microphone. Different combinations of mic and voice will produce different results. If you're using an el-cheapo mic, you may not get as crisp an effect as you would using the same megabucks units used by recording stars. Also, output level for the mic you're using can be fairly critical. If you're not getting enough signal out of your mic to drive the vocoder easily, you may need to boost your mic's output by running it through a spare mixer channel, or using a dedicated mic preamp.

Perhaps even more important is the carrier signal you choose. If your carrier is lacking in harmonics (sine waves, flutes, mellow organ sounds, and the like come to mind) the vocoder won't have enough sonic material to shape into a speech-like sound. It would sort of be like asking a painter to do a photo-realistic painting using only one or two colors – difficult at best.

Finally, it is my experience that all vocoders have their own unique sound. It may be that the DP/4 just won't do exactly the sound you want. Still, I think it's much more likely that you just haven't yet hit on the right combination of factors to produce the sound you want, but eventually you will. In the meantime, you might want to contact Ensoniq Customer Service (610-647-3930) and have them provide you with some more detailed guidance than I can give you here. I'm sure they'd be more than happy to...]

[Ensoniq – Our word meant nothing? Them's fighting words...

Seriously, it is entirely possible to get the type of effect you are talking about with the DP/4. Whether in the final judgment it is one that you will like is not the type of question that anyone but you can answer. Over the phone, how can we guarantee that your taste and ours (meaning the person on the other end of the phone) match up?

Clark offers some good insight. Here's another tip – many people get a robotic voice sound by running a microphone into a Flanger with high regeneration and a very slow sweep. Try experimenting with that as well. And call us – we're glad to continue to work with you. Yes, even Bob.]

Hello,

Problem: MIDI. Situation: I own a TS-10, VFX-sd II, Boss DR-5, a MIDI controller keyboard, and an old Yamaha keyboard. I also own a 486-SX33 PC with a Pro Audio Spectrum sound card, with a Turtle Beach "Maui" daughter board. I also have several pieces of software for music production. For this letter, I'll use "The Jammer" as my reference.

Now I've got "The Jammer" fired up and everything works fine through the Maui board, and when I change the setup "MIDI Output Device" to "Spectrum MIDI Out" and plug into the old Yamaha it also runs fine – using the Yamaha's cheezy sounds. BUT when I plug into the VFX-sdII the computer locks up and has to be rebooted. So I play around with a few things on the computer and on the VFX, and BOOM – same thing. So I plug into the TS-10, enter General MIDI Mode and once again I lose.

Now I've tried everything, changing wires, changing the MIDI Mapper, changing the setups on the individual software programs, and still nothing. Now to confuse me even more, going MIDI in to the computer with the TS-10 works beautifully, same with the VFX. But as soon as I try the MIDI controller keyboard I get stuck notes or the computer locks up.

So before I break out the old sledgehammer and do some damage, do you have any suggestions?

P.S. My TS-10 seems not to recognize the disk name given to Giebler's EDM formatted ASR-10 disks – like when it asks for "Disk 605" I put in "Disk 605" and, SHAZAAM, it asks for it again. Figure that one out please.

Thanks, Eric Beaudry Holyoke, Mass.

[CS – That's a lot of stuff to troubleshoot with just a word processor. I'd recommend you try contacting Ensoniq Customer Service, or perhaps one of our illustrious Transoniq-Net afficianados, and seeing if you can't narrow it down a bit.]

[Ensoniq – It is difficult to imagine how the VFX-SD or TS-10, when being used as a target for MIDI sequences, could cause your computer to crash unless there is a MIDI loop. You don't mention whether the MIDI Out of the VFX or TS is connected to the computer. If that is the case, then the VFX/TS must be transmitting something that your computer software is unable to handle. Try unplugging MIDI In to the computer and see what happens.

You also don't mention what MIDI con-

troller you are using. It is possible that it is sending bad data into the computer. We are also suspicious that since the Pro Audio Spectrum card is an integral link in the chain that it may be causing the problem. But your letter doesn't give enough detail to really tell. This is the type of problem that has to be detailed in full – what's connected to what, what do you press, what happens when you press it, etc. etc. I don't think you'll get it solved in a quick letter response. Call us.]

Hi, Transoniq Hacker,

Concerning "From Cyberspace to Your Ear, Part II." In this article written by Garth Hjelte there was some new information which helped me a little. Not only the EDM program is mentioned, also a new thing for the Mac – the EPSm written by Terje Finstad of Norway in a format called a B-10 and a program called scEPSi.

I bought the EDM program from Rubber Chicken and tried to run it on a DOS-compatible MacPerforma 5200 with a Soft Windows program. It wouldn't run – "Diskette Controller Error."

So where can I get/find/buy these two programs for the Mac?

Finally, two nice gimmicks I found on my TS-12 (OS 3.05). Ensoniq secrets maybe.

1) When you change the click volume to the maximum of 14 dB, the click behaves very individualistically - it counts only the last beats in measure, never the first. In a 4/4 the metronome counts only the last 3/4, in a 3/4 only the two last 3/4 will be audible. It's nice and brings some inspiration while practicing.

2) In the manual all kinds of voice classes are listed and you have a perfect survey. Until now. Because I found a new category of sound – an underground voice which survived all the changes of the operating system. On the Ensoniq TSD 300 diskette you can find a lot of organs. One of which contains the miracle. It's 70s Organ. When you push Select Voice, the fifth voice is called "*unknown*." When you press the wave button, there is the information: "*unknown*=unclassified," and it is not possible to copy that voice. What a pity. Now I can sleep better – even for such a great instrument there still remains unknown sounds.

Thank you very much.

Greetings from Vienna, the city of music.

Sincerely, Herbert Lacina, Austria

[TH – We forwarded your questions regarding the EPSm program to Garth with the following results.]

[Garth - I e-mailed Steve Berkley, the author of scEPSi, and Terje Finstad, the author of EPSm, to give you details on how to obtain/purchase the program. The shareware thing is a little tricky, and I'd rather they handle it.

I also e-mailed Giebler to respond to the EDM-Mac compatibility question.

Thanks for the opportunity – hope I could help.]

[Terje – EPSm is a Mac program for use with EPS/ASR/TS samplers. I am the author of that program. The program is US\$20 shareware.

I don't know what details you would want. For a short reply one might say that one can get the necessary information by sending email to noice@ fys.uio.no.

I don't know if this helps, but I hope so.]

[Steve Berkley, Berkley Integrated Audio Systems –

Dear Transoniq Hacker and Mr. Lacina:

scEPSi, a utility for Macintosh users with EPS, EPS-16+, or ASR-10/88 samplers is shareware. You can download scEPSi from the following ftp site using an anonymous connection:

ftp.crl.com

using the directory path:

lftp/users/ro/sberkley/Utilities/scEP-SiV144.sea.hqx

scEPSi allows Ensonig sampler owners to view the contents of the SCSI storage device they use for storing Ensonig format files. scEPSi also allows users to rename files, effortlessly move files to other directories, scan drives for problems, delete files, savelload files to the Macintosh, update the OS on their drive, save ASR-10 Audio Tracks to your Macintosh for editing, exchange sequences and instruments via modem, search for files, backup and restore Ensoniq formatted volumes, and many other features. scEPSi also works with COMPUTER formatted floppy diskettes (available only on the ASR-10) and CD-ROM drives. I hope

you will find scEPSi a useful utility for your Ensoniq sampler!

Good luck!]

Hacker:

I own a TS-12. We just had an Ensoniq Clinic here at West Music. The rep (forgot his name) said I could add a pedal board to play pipe organ music by setting up MIDI port to accept a sequence. Have not tried it yet.

Also, I am looking for pipe organ sounds – different registers rather than whole organ & ambience.

Thanks, Timothy J. Fees Via Internet

[CS – You can, indeed, use a MIDI-pedalboard with your TS-12. Assign the sound you want your pedalboard to play to a track in the TS sequencer, and then make sure to set that track to respond to whatever MIDI channel your pedalboard is transmitting on, and away you go.]



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