Transoniq Hacker

ASR Tips: Connectivity

Tony Ferrara

Since moving on to some in-depth TS coverage, too many moons have passed since I last wrote about the ASR here in the *Hacker*. I've heard from plenty of people who seem to need some specific suggestions, as well as an overall primer on how everything seems to connect and inter-relate. I'm closing this installment with a general ASR tip about receiving controller #10 pan commands.

So here we go with a little taste of both, for those of you who can't seem to get enough:

DiskTracks and the ASR: Log your work!

I strongly encourage you to keep a paper or computer-based log of important and essential data; this should be mandatory

regarding DiskTracks management. Because of the variable project-specific choices that are possible with parameters like SET SONG ATRK PLAYBACK and SCSI Cartridge Drive ID Number, I suggest that you keep a computerized or hard copy (paper) log of these settings. This would help to prevent much in the way of wasted time on your part later on when trying to reconstruct your particular project configurations.

Using the ASR and DiskTracks in Conjunction with Another Sequencer

If you've created a bunch of sequences using a hardware or computer-based sequencer and tried to link them to Disk-Tracks via MIDI you'll experience a rude awakening: if you attempt to change the ASR's sequencer CLOCK SOURCE to "MIDI" (under Edit-Seq Song) so that it follows the other sequencer, you will get a message saying "NO SCSI ATRK MIDI SYNC." It's important to remember that the ASR must be the master clock when recording and playing back DiskTracks to and from a SCSI hard drive. To get around this so that you can use your outboard sequencer's tracks along with the ASR's DiskTracks function, try my workaround:

Let your computer set up the right length and tempo by recording a new ASR sequence first. Then you can use that as an internal ASR sequence to become the Master, and you now can slave the computer to it, with tempo and length having been determined by the computer.

Press Edit-Seq Song on the ASR; scroll left or right until you get to the screen saying "CLOCK SOURCE= INTERNAL"; change the MIDI clock source to "MIDI" using the up or down arrow button.

Record a song length sequence into the ASR from the other sequencer, then slave the external sequencer to the master clock of the ASR by reversing both clock sources once again. The ASR's sequencer will now correctly drive your external computer, so it can play its files correctly. You are free to record regular or Audio Tracks on your ASR, and the whole shebang will be properly sync'd.

How to Turn Off DiskTracks

There are numerous scenarios in which you would want to turn off the DiskTracks and play back the song and its sequenced tracks only, without any associated Audiosamples. For instance, there are many people who never intend to work with DiskTracks but just want to play sounds and listen to demos without any associated Audiosamples.

To do so, go to the Command-System MIDI button and scroll left or right until you reach the screen saying "CONFIGURE AUDIO TRACKS"; press the Enter/Yes button, at which time the screen will read "ATRK PLAY/REC= ____"; use the up or down arrow buttons to set this to "OFF."

General ASR Tip about Receiving Pan Commands

Here's a scenario in question and answer format that is a helpful workaround to a fairly common situation when using the ASR with an external hardware or software sequencer:

Q: My ASR will not respond to controller #10 pan commands. This is a problem for me; is there a way of solving it?

A: There is a workaround that you can try; I've gotten it to work with no problem. Here are the steps:

1) On the ASR, press Edit-System MIDI; scroll with your left or right arrow buttons until your screen says "MIDI XCTRL NUMBER=____." Set that controller number to 10 for pan.

2) Press Edit-Amp; then scroll left or right until you get the

page that says "Pan Mod =____." Set this to "XCTRL," with a value of +99. On the page that says "WS VOLUME," set the PAN value to -99.

Your ASR will now respond to continuous controller #10 (pan) messages being send from an external sequencer or MIDI controller.

Saving Sys-Ex Data to your ASR-formatted Hard Drive

You may have noticed that one of the ASR's many abilities is to function as a Sys-Ex recorder. If you've been using a SCSI hard drive in conjunction with your ASR, you've probably seen a file type in your default directory entitled "FILE 4 SYSEX FILES." If so, you may have discovered that you cannot load Sys-Ex files directly into your ASR from floppy disk and save them to your hard drive. This is despite the fact that you can do so with various other file types such as ASR sequences, banks, and sampled instruments. If you've already saved a Sys-Ex file to floppy, the following steps will allow you to save the data back to your hard drive:

1. Change the storage device on the ASR to the SCSI device ID number and enter the target folder in your hard drive's directory where you want the Sys-Ex file to end up.

2. Send the file back to your original source synth using the ASR's Sys-Ex "SEND" function.

3. Press Command-System MIDI and scroll left or right until the screen says "MIDI SYS-EX RECORDER"; press enter and the ASR screen will say "WAITING...."

4. Once you send the Sys-Ex data back over to the ASR, the screen will say "RECEIVING..." followed by "CANCEL=QUIT ENTER=SAVE"; if you have followed step number one above, your Sys-Ex file now be saved to your hard drive.

Snappy Closing Statement

I said a taste, and a taste is what you've gotten! In all seriousness, I'll update this series with new installments and observations as they develop. As always, thanks for your interest and your input. ▬

Bio and Self-Serving Plug: Anthony Ferrara is an Ensoniq Corporate Citizen playing and recording original Contemporary Jazz in the Philadelphia area. He uses Ensoniq equipment whenever and wherever possible.

RND (♪♪♪)

Ensoniq News

Ensoniq has released the first in a series of wave expansion boards for the MR-61, MR-76, and MR-Rack instruments. EXP-1 *"The Real World"* is a 24 MB collection of world instruments, containing 442 new waveforms, over 500 new Sounds and Drum Kits, and some stunning demos. After installing EXP-1 in an MR instrument all the sounds are organized by Ensoniq's unique SoundFinder(tm) technology so you can easily locate them by instrument group (strings, percussion, vocals etc.) or by location.

Sounds for this collection have been gathered from eight countries around the world over a two-year period. They are all-new material, never released in an Ensoniq product before. Instruments represented include Conch Trumpet, Udu Drums, Balaphone, Djembe, Banhu, Zheng, Sitar, Tamboura, Kabuki Bells, Taisho Koto, Didjeridu, Crumhorn, Uilleann Pipes, Dobro, Bandonean, Sikus, Charango and a variety of vocal phrases.

EXP-1 has a suggested retail price of $500 and is available now.

Hacker News

Ensoniq has been experiencing some delays in releasing the MR-61 and MR-76. When Ensoniq doesn't ship it makes our new subs coming in temporarily slow down. If you're an advertiser (depending on what you're selling) you may have noticed a certain "softness" the last couple of months — this may be why. In addition, we haven't really been pushing real hard to get MR articles because there are currently so few MR readers. Please hang with us a little while on this — when they do start to ship, things should pick up quite a bit.

*　　　*　　　*

On another note... as of July 1st the U.S. Post Office monopoly is "Reclassifying" a lot of different types of mail. This is Post-Officese for "We're raising your rates about 10%." Once again, we're not going to let this affect your subscription rates, but you may notice some differences in delivery. We're not sure yet, but typically when they raise rates, they like to lower the service to "make up the difference." Meanwhile, any conversions to our e-mail version are *greatly* appreciated.

DP/4 Custom Configs
Part II — Config and Algorithm Parameters

Steve Byhurst

In a previous article we looked at drawing up a config design plan and the preparation needed before programming can begin. Those being the first two stages of creating a custom-built config, let's now move on to the remaining ones.

Stage 3: Config Programming

If you decided in Stage 2 not to use an existing config preset as an editing base because it didn't come close enough to your design plan, you have the alternative of using one of the templates mentioned in part one. Those thoughtful, generous guys at Ensoniq have provided eight ROM config presets (numbers 53-60) which cover all the

input source and mono/stereo variations possible. For example, preset 53 will give you a "1 source: mono input" set-up for 4-unit processing of a single mono source, and preset 60 a "4 source: 4 mono out" set-up for single unit processing of four individual mono signals. Look at your design plan to see which type of source config you require as this will dictate which of the templates to use.

Once you have decided on your template you are ready to select the unit preset(s) it will use. The ROM templates have a dry "no effect" algorithm in each of the units so these need to be replaced with the preset(s) you have (hopefully!) already prepared. So, select your config template, press relevant unit buttons as required, and use

the data entry knob to select your unit preset(s).

If you are not using one of the ROM templates but one of your own previously programmed configs instead, you may want to replace some or all of the existing algorithms. This means you either need to change to a different preset that contains the required effects, or change individual algorithms loaded into the units of the current preset.

In Select mode you can only choose replacement presets that are appropriate to the config type. This is not a problem for 1-source configs where you are only dealing with single units. 2-Source configs can only select 2-unit presets, and with 3-source configs 1- or 2-unit presets are available according to whether a single unit or pair of units is selected. 1-Source configs, being the most complicated, have the most restrictions as they can only select 4-unit presets. However, any individual units in any preset can be easily changed by going into Edit mode, selecting a unit, and then choosing a different algorithm. You will be choosing from the 1-unit presets you already have in memory so make sure that you have an appropriate one available.

One trick to mention here which is tucked away at the back of the manual, is a way to load a 2-unit preset whilst in a 1-source config. In Edit mode press the A and B, or C and D, buttons at the same time and you will have access to the 2 unit presets in memory. You can then select one and place it your chosen pair of units.

Now you can move on to edit the template's config parameters. Make sure the yellow config LED is on (if not, press the Config button) and then press the Edit button.

The first parameter, number 00, doesn't need to be changed because you have already selected an appropriate template, but if you are editing an existing config this is where you can change the input configuration. This governs what the remaining parameters will be and depending on the type there are six, eight, or ten parameters to program. Most of the parameters refer to signal routing between the units.

For 1-source configs you can select routing between units AB to CD, units A to B, and units C to D. There are also two config dependent parameters setting dry path or feedback amounts and a choice of mono or stereo input to the AB units. The last four parameters are the same for all four source types. They decide what happens when a unit bypass button is pressed: bypass lets a dry signal through only, kill lets no signal through.

2-Source configs lose the AB to CD routing, which is not applicable, but gains a mono or stereo input choice for the CD units. Slightly different again are the 3-source config parameters. There is a choice of whether AB unit outputs will be dual mono or mixed stereo, CD unit routing, CD input select, and a config dependent parameter for dry path/feedback amounts.

The 4-source config is the simplest of the four types. Apart from the bypass settings there are only two other parameters: AB and CD units output selection.

Unless you need to have complicated feedback routings, your design plan should make the programming of the config parameters fairly quick and easy.

Stage 4: Algorithm Programming and Testing

This stage is probably the one you will spend most time on as a lot of the algorithms have a large number of para-

meters and you can spend ages tweaking them to get as close as possible to the result you want.

Using your notes from the design plan, start to edit each of the unit's algorithms using the rough parameter values you decided upon in your plan. If the units are serial or feed-back routed you should pay particular attention to the parameters which govern how much signal is passed between them, as these can greatly affect the quality of the output signal.

At this point it may be possible that you need to swap around the order of the loaded algorithms in your config, and there is an easy way of doing this by swapping any two units. Press Edit, Write, and one of the unit buttons you want to swap, and then press and hold that unit button whilst simultaneously pressing the unit button of the second unit. The display will confirm what you are about to do before you finally press Write to do the swap. You can use this method to quickly hear what would happen if the order of units is changed.

Next, put some sample source material through your config and see how close you are to achieving what you want. By using the bypass buttons you can isolate each of the units to check out the way they are processing your sound source. Doing this should very quickly show up any things you might have forgotten in the planning stage. Once any major anomalies have been corrected all you have to do is

make the fine tuning needed to make the config match your exact requirements. Remember that when editing algorithm parameters you can use the Edit button to toggle between the version in the edit buffer and the original to hear what changes your edits have made.

When you are completely happy with your config, make sure that it, and the relevant unit preset(s), are safely written to new presets. It is worth thinking up a good descriptive name for it which you will remember later. Don't forget to use the unit buttons as a shortcut to get to the various groups of alphanumeric characters.

Also, I know it can be a bit of a chore but it really is a good idea to try and keep a note of what each of your custom-built configs does. You never know if you might be able to use one of them as a template for a future config which needs a similar design. It doesn't take that long to jot down a few notes and, after all, why duplicate programming that already exists. You can very quickly build-up a whole library of useful configs in this way.

In Part Three I will provide a real-life example which puts all this theory into practice. See you then. ▄

Bio: Steve Byhurst is a thirty-something British composer of electronic music who is quite adamant that his compositions should not be classed as "New Age" or "New Instrumental."

Some Painless Uses for Ensoniq Sequencers

Frank Fortunato

We continue with another painless entry level article of things the beginner can do to enhance involvement with our Ensoniq synths.

One of the special features of the VFX,SD,TS family of workstations I have appreciated over the years is the large displays that allow six sequences to be accessible on the screen at one time. There are many wonderful ways to take advantage of this, especially in live playing. Let's look at some easy to do things:

1. Simulate Live Drum Playing

In the January 1995 issue I first mentioned taking advantage of the big screen for live playing. Since drum patterns get boring quite quickly if played without variation, by utilizing the six locations of the sequencer we can have quick access to six different patterns. Mixing up those patterns in real time can go a long way in simulating some live playing.

• Find six different drum patterns that are compatible and make one bar sequences out of each.

• Loop all but an ending sequence. Include for example a

couple of basic beat patterns, two or three fills, and the unlooped ending sequence.

• Set the effect the same for each drum pattern. This assures that you can switch back and forth between all six sequences on the screen seamlessly without the momentary halting of sound that occurs when different effects algorithms need to be loaded.

• Load in a favorite patch or preset for all six sequences that you will use to play along with the drum patterns and to play it safe store each sound in the same location for each of the different sequences.

Usually I set up a bass/piano split, a three-way bass/piano/brass patch or some variation of that, or a piano and brass layer. The brass program that I customized for this has all six voices of the patch muted out in its normal setting. Using a dual foot pedal I set up the left pedal to access a left patch select to "open" the brass voices in the program. This allows me to add the brass only when needed. When depressing the left pedal the brass layers with the piano play when I need special brass stabs or for increased sound.

In this way I never have to shift my hands up an octave or two to play the brass sound. (For more on this see *Hacker*, January and February articles on using presets.)

• Set the same tempo for each of the sequences and use a slower tempo for the ending sequence if you want to simulate a ritard ending. I use a crash cymbal for my ending sequence.

You could also experiment with the fills being set one or two beats slower to simulate a slight retard which then leads back into the regular "a tempo" speed with the basic beat patterns/sequences.

• At this point we are now ready to perform by playing live keyboard along with the sequences. With a little practice you can learn to go back and forth between fills and basic patterns by anticipating and selecting the sequence ahead of the downbeat. For very fast songs this will take

some practice to hit the sequence coming up and then get your hand back on the keys to maintain the rhythm.

If your piano style is like mine it means using your left hand to change the patterns keeping the right hand free to maintain the beat at all times.

2. Splitting the Song into Its Component Parts

A variation of the above is to take advantage of the six sequence screen to create real time control of a song by having each section of the song displayed on the screen as a separate sequence. In this way you can choose which section of the song you want to use and repeat as many times as necessary when looped. One more quick tutorial:

After I've created a song or imported a Giebler converted standard MIDI file I do the following:

• Copy the song/sequence four or five times.

• Go to the second copy, listen for the intro, and then delete all bars following the intro so that only the intro remains. Loop the sequence. This way I can have the intro playing while I talk or improvise to my heart's content over the looping intro. Go in and rename the sequence.

• Go to the third copy and delete the intro. Then listen for the verse or whatever the first part may be and then delete all of the following measures. Loop it. Rename it.

• Repeat the process as necessary to work your way through the entire song.

What results is having individual sequences for each section of the song and if you choose to sing three verses in a row or three choruses in a row, or to repeat the bridge or whatever, you have access to each song section at all times on your screen. Playing live on top of the sequences will usually require your pulling back the volume levels of the other parts so that the live accompaniment part will punch through. Further refinement of this process is to have a final chorus that changes key for emphasis. With the transposition feature within the sequencer, this is a breeze. Remember not to transpose the drum part.

Once transposed, if you are too lazy to play in the new key simply transpose the accompaniment voice you are using to the new key and keep playing away in the old key.

And there you have two painless easy ways to use the nice big screen of Ensoniq synths. ■

Plucked Synth à la KS-32
Or — The Attack of Sawtooth!

Dan Rohde

In trying to come up with an interesting lead for this Plucked Synth article, so far I've ruled out the chicken feathers joke (plucked) and the recipe analogy as both being too dumb. *"The Attack of Sawtooth"* had some possibilities as a B grade splatter movie title, but really had nothing to do with the article. And since I can't think of any clever yet inoffensive rhyming puns, I'll just skip the whole lead, if that's all right with you.

There's a great vintage Plucked Synthesizer sound used for a one verse lead in the song *"Breaking Us In Two"* by Joe Jackson on *Night and Day,* 1982. I roamed around the ROM and default RAM programs on the KS-32 but couldn't find one "close enough for horse shoes." In spite of a nice variety of synth programs, none of them approximated the plucked synth with reverb in the song. The other Programs that simulated a plucked sound, such as the guitar, bass, harp, and harpsichord, were also not the right direction.

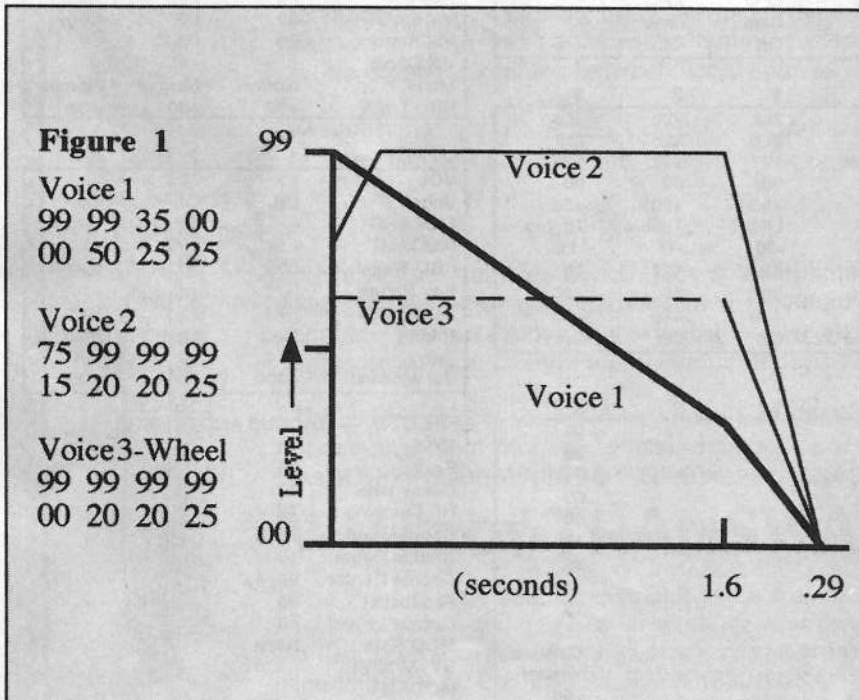Just as I was wondering how I'd use the Envelopes to accomplish that unique pluck attack (about halfway between an elephant snort and the squeak you get when you rub a beach ball) — I stumbled upon Synth Pluck under Tuned Percs Waveforms, the perfect initial sound. It didn't have quite the right sustained sound afterward, but I figured I could drive off that bridge when I came to it.

I set the Pitch screen's Octave = +1 to put Synth Pluck's range more toward the middle of the keyboard and detuned it slightly (-03) to add some interest. The LFO settings of Speed (20) and Level (30) affect the Filters through their Modsource (LFO) to the Modamount of -30 to add additional movement. The Amp parameters form a steep slope (1.89 seconds) downward since all I want to hear is its attack (Figure 1).

For Voice 2, the sustained synth part of the program, nothing else comes close to Sawtooth. Under Pitch, detune it a gnat's eyelash sharp with Fine = +02. Give it a little more action with Modsource = LFO, Amount = +05. Keep the Glide off, unless you want to strike out on your own with Legato. The Joe Jackson synth definitely has no glide. The LFO screen's parameters should give Pitch a subtle movement. I'd rather underdo it than overdo it here.

On the Filter screen I've set Modsource Timbre = -15 because I usually use my programs within sequences; therefore, Timbre enables me to choose and set the bass/treble quality with the Data Entry slider when the Program is used in a sequence or Preset. If you plan to use Plucked Synth as a solo lead, you could choose Wheel or Modpedal as Filters' Modsource instead. I almost never choose Modpedal as a Modsource because I need that left pedal to control my Volume on the sound I'm using in a sequence.

To soften the attack of the Sawtooth wave, Initial = 75 in the Amp envelope. This small


Figure 1

Voice 1
99 99 35 00
00 50 25 25

Voice 2
75 99 99 99
15 20 20 25

Voice 3-Wheel
99 99 99 99
00 20 20 25

"notch" is meant to bring in Sawtooth quickly yet not obscure Synth Pluck; that is, I want Synth Pluck to be most prominent the first 0.14 seconds, then cross fade quickly yet smoothly into Sawtooth so that the program doesn't sound too obviously like two different Voices. (The sherry should enhance — not overpower — the other herbs in a good marinade, know what I mean?) Note that under Output, Timbre = +05 increases the volume slightly as it simultaneously cuts off a few of the higher frequencies under Filter.

Chorus + Reverb is chosen for the Effect for two reasons: One is that the synth sound in the song definitely has some 'Verb behind it. The other reason is that a little Chorus enhances the overall sound. FX1 = 15 because more than that much Chorus seems to produce a strange attack effect and covers up the already good quality of first two Voices. Also, FX2 = 25, which determines the amount of Reverb heard, is about the right level of Reverb for most of the other Programs I use in my sequences.

Voice Three is optional. (Synth Bass-3 is an Expansion

sample, so SQ owners will have to make do without it.) This Voice adds even more punch to the attack and a deeper tone to the sustained Sawtooth wave. As in Voice Two, raising the Timbre setting in a sequence or Preset reduces the passage of the higher frequencies. Also on the Filters screen, Envelope = +20, which moves the 2Hi Cutoff point upwards a little bit, depending on key Velocity (Figure 2). Under Output, Wheel = +25 to give

Figure 2

Voice 3
Filters:
Env 2=+20
99 99 65 65
00 15 15 25

Level

99

00

.15 .15   .25 (seconds)
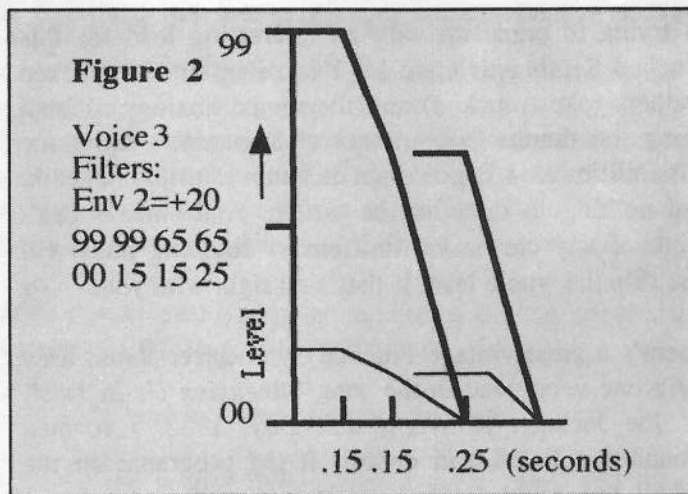
KS Prog: Plucked Synth

By: Dan Rohde

WAVE	1	2	3
Select Voice	On	On	On/Off
Wave Class	TunedPerc	Waveform	Expansion
Wave	SynthPlck	Sawtooth	SynBass3
Delay Time	00	00	00
Wave Direction	For	-	For
Start Index	00	-	00
MODSCR	-	-	-
MODAMT	-	-	-
Restrk Decay	25	25	25

PITCH	1	2	3
Octave	+1	00	+2/+1
Semitone	00	00	00
Fine	-03	+02	-02
ENV1	00	00	00
LFO	00	00	00
MODSCR	-	LFO	-
MODAMT	-	+05	-
KBD Ptch Track	On	On	On
Glide	Off	Off	Off
Glide Time	-	-	-

ENV1	1	2	3
Initial	-	-	-
Peak	-	-	-
Break	-	-	-
Sustain	-	-	-
Attack	-	-	-
Decay 1	-	-	-
Decay 2	-	-	-
Release	-	-	-
Vel-Level	-	-	-
Vel-Attack	-	-	-
Vel Curve	-	-	-
Mode	-	-	-
KBD Track	-	-	-

LFO	1	2	3
LFO Speed	20	25	-
Noise Rate	00	00	-
Level	30	04	00
Delay	00	00	-
MODSRC	LFO	LFO	-
Wave	Sine	Sine	-
Restart	On	On	-

FILTER	1	2	3
Filter 1	2Lo	2Lo	2Lo
Filter 2	2Lo	2Lo	2Hi
FC1 Cutoff	127	127	127
ENV 2	00	00	00
FC1 KBD	+50	+50	+50
MODSCR	LFO	Timbre	Timbre
MODAMT	-30	-15	-15
FC2 Cutoff	127	127	00
ENV2	00	00	+20
FC2 KBD	+50	+50	+50
FC1MOD-FC2	On	On	Off

ENV2	1	2	3
Initial	-	-	99
Peak	-	-	99
Break	-	-	65
Sustain	-	-	65
Attack	-	-	00
Decay 1	-	-	15
Decay 2	-	-	15
Release	-	-	25
Vel-Level	-	-	13
Vel-Attack	-	-	13
Vel Curve	-	-	Quikrise
Mode	-	-	Normal
KBD Track	-	-	+00

AMP	1	2	3
Initial	99	75	99
Peak	99	99	99
Break	35	99	99
Sustain	00	99	99
Attack	00	15	00
Decay 1	50	20	20
Decay 2	25	20	20
Release	25	25	25
Vel-Level	00	00	00
Vel-Attack	00	00	00
Vel Curve	-	-	-
Mode	Normal	Normal	Normal
KBD Track	+00	+00	+00

OUTPUT	1	2	3
VOL	65	70	70
Boost	On	Off	On
MODSRC	-	Timbre	Wheel
MODAMT	-	+05	+25
KBD Scale	00	00	00
Key Range	-	-	-
Output Bus	FX1	FX1	FX1
Priority	Med	Med	Med
Pan	+14	00	-14
Vel window	>000	>000	>000

EFFECTS — CHORUS AND REVERB

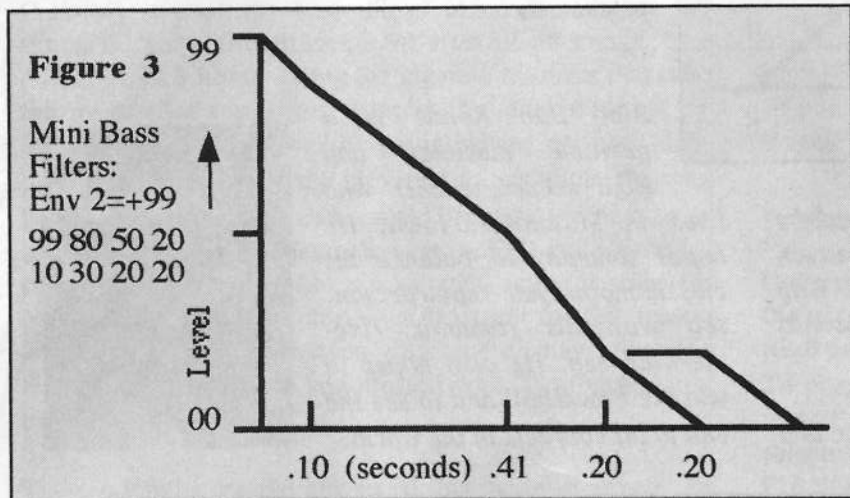|	
FX-1	15
FX-2	25
Decay time	30
HF Damping	72
Chorus Rate	15
Chorus Depth	10
Chorus Center	55
Feedback	00
Chorus Level	46
MOD (Dest)	None
BY (MODSRC)	-
MODAMT	-

this Voice a real-time controller. Timbre or Pressure would also be logical choices for different personal preferences. Under Pitch, Octave = +1 puts this Voice one octave lower than the other two Voices, which you might prefer. I like it at +2.

The synth solo in "Breaking..." uses the Pitch Bend Wheel, probably set to a maximum one whole step, so

Figure 3

Mini Bass
Filters:
Env 2=+99

99 80 50 20
10 30 20 20

Level

99

00

.10 (seconds) .41 .20 .20

you might practice using it before the next big performance.

The other program, Mini Bass (RAM #17) Revisited, modifies a factory RAM program so it sounds more like Joe Jackson's synth bass in *"Stepping Out,"* a piece on the same album/tape/CD. I'm sure the guys at Ensoniq are light years ahead of my humble programming understandings, but this patch used "out of the can" does not meet my needs. To personalize it, I advanced the Start Index on the Wave screen to 10, then set its Modsource (Velocity is the only option) to -13. This gives you the real time ability to vary the point at which the sample starts through your fingers' velocity. That is, a harder key strike will start the sample further toward its beginning.

The biggest change — and easiest to accomplish — is turning Glide to Off on the Pitch screen. The existing Legato required too much staccato technique to catch the pluck attack. The parameters on the Filter

Figure 4

Mini Bass

Amp Env:
90 99 00 00
05 50 20 20

(Vertical axis: Level, 99 to 00; Horizontal axis: .06 to 1.6 (seconds))

Effect, okay? Here again, though, the factory Mini Bass was set to 8-Voice Chorus, which is fine for a synth sound. But while the Voice was routed to FX1, FX1 = 00, and the Effect's Modsource was not set to increase the Chorus level. Therefore, no Chorus happened (?). Maybe it's my keyboard.

Well, it's time for me to be "step-pin'...step-pin' ou-out...in-to the night...," etc. Hope you like the patches. ■

screen allow Timbre to "subdue" the tone. Envelope 2 (= +99) also puts a little more bite in the tone of the attack (Figure 3). The Amp setting, similar to the Amp Blip Default, limits the sound's duration to 1.32 seconds (Figure 4).

I chose Chorus + Reverb again mostly because I like that

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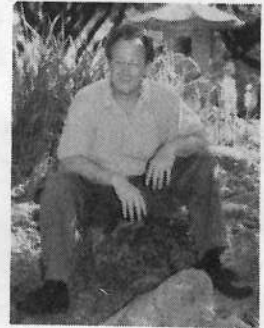
KT Prog: Mini Bass

By: Dan Rohde

WAVE	1	2	3
Select Voice	On		
Wave Class	Expansion		
Wave	SynthBass3		
Delay Time	00		
Wave Direction	Forward		
Start Index	10		
MODSCR	Vel		
MODAMT	-13		
Restrk Decay	20		

PITCH	1	2	3
Octave	+1		
Semitone	00		
Fine	00		
ENV1	00		
LFO	00		
MODSCR	None		
MODAMT	-		
KBD Ptch Track	On		
Glide	Off		
Glide Time	00		

ENV1	1	2	3
Initial	-		
Peak	-		
Break	-		
Sustain	-		
Attack	-		
Decay 1	-		
Decay 2	-		
Release	-		
Vel-Level	-		
Vel-Attack	-		
Vel Curve	-		
Mode	-		
KBD Track	-		

LFO	1	2	3
LFO Speed	-		
Noise Rate	-		
Level	-		
Delay	-		
MODSRC	-		
Wave	-		
Restart	-		

FILTER	1	2	3
Filter 1	2Lo		
Filter 2	2Lo		
FC1 Cutoff	85		
ENV 2	+99		
FC1 KBD	00		
MODSCR	Timbre		
MODAMT	-99		
FC2 Cutoff	85		
ENV2	+99		
FC2 KBD	00		
FC1MOD-FC2	On		

ENV2	1	2	3
Initial	99		
Peak	80		
Break	50		
Sustain	20		
Attack	10		
Decay 1	30		
Decay 2	20		
Release	20		
Vel-Level	13		
Vel-Attack	00		
Vel Curve	Quikrise		
Mode	Normal		
KBD Track	06		

AMP	1	2	3
Initial	90		
Peak	99		
Break	00		
Sustain	00		
Attack	05		
Decay 1	50		
Decay 2	20		
Release	20		
Vel-Level	13		
Vel-Attack	00		
Vel Curve	Quikrise		
Mode	Normal		
KBD Track	00		

OUTPUT	1	2	3
VOL	80		
Boost	Off		
MODSRC	-		
MODAMT	-		
KBD Scale	-10		
Key Range	E1-G7		
Output Bus	FX1		
Priority	Med		
Pan	00		
Vel window	<000		

EFFECTS — CHORUS AND REVERB

FX-1	10
FX-2	15
Decay time	05
HF Damping	53
Chorus Rate	20
Chorus Depth	15
Chorus Center	57
Feedback	+10
Chorus Level	40
MOD (Dest)	-
BY (MODSRC)	-
MODAMT	-

TS Owners: Se Habla ASR?
Editing Sampled Sounds in the TS-10 and TS-12 — Part 1

Robby "Bilingual" Berman

Certainly one of the best things about the mighty TS synths is their ability to load ASR-10/ASR-88 sounds. The ASR/16+/EPS library being the gigantic resource that it is, this means that a galaxy of samples is available for use by intrepid TS owners (boldly going where no one blah-blahblah). And one of the cleverest things about the way the TS imports all of these sampled sounds is that it turns its suite of editing parameters into ASR-style sample-editing parameters whenever a sampled sound is selected. No, you're not going crazy — at least not for this reason — it's true: you have seen different displays revealed from time to time when you pressed your TS programming buttons.

These oddballs are the key to editing sampled sounds on your TS, of course. To those of us with both products, this is nothing but a good thing since these params are old friends. Plus, we've got a big manual devoted to them. To those without an ASR/16+/EPS, however, the whole phenomenon may be a little disorienting: all of these terms that are sorta familiar, but don't quite make sense from a TS point of view. And a bit unsettling as well, I imagine. TS owners who have mastered TS programming may feel a bit ookie having all of these weird little strangers residing in this otherwise totally familiar box. While the TS manual does talk about using these params, why don't we just go ahead and translate them into the native TS tongue? (Thank you for the warm welcome. I am very happy to be in your country.)

This series of articles is intended to de-mystify these foreign parameters once and for all. We're going to assume a certain degree of familiarity with the way a sound is programmed in the TS. The plan is to use that knowledge as a foundation on which to build an understanding of all sampled sound params. If you understand how a TS parameter works on normal TS sounds, it is my fond wish that you'll understand how its counterpart operates in the world of sampled sounds by the time we're done.

It will help to load in a sampled sound so that you can press buttons and check out displays as you follow along.

This series will make most sense if we're both looking at the same things, so git aholt of the SSD-100 sampled sound disk that came with your TS and load in RS PERCS (If you dunno how to load in a sampled sound, see the TS manual).

When Is a Wave a Layer?

Once you've got your sampled sound loaded, ignore it for the nonce. Locate the TS sound BABY-GRAND in Bank R2-0 and select it. Press the Select Voice button. As any TS programmer knows, this is the page on which you can view the TS waves used to create the sound you've got selected. The first wave here, for example is GRND-PIANO.

Perhaps you realize that calling GRND-PIANO a "wave" is actually a bit of an oversimplification. GRND-PIANO is really comprised of a bunch of sound waves, all of which work together to give you a full keyboard's worth of Piano Goodness. If you don't believe me, slowly play up from the bottom of the keyboard — if you listen carefully, you'll hear where one sound wave ends and another begins (one of these transitions occurs between the E two octaves below Middle C and the F just above that E). All synths of the TS's ilk work this way. The art is in matching up these different sound waves so that the transitions aren't noticeable unless you really study them (and Ensoniq's done a good job here, I'm sure you'll agree).

And now it's time to really get to the point: what we're calling a sound wave is actually a sample. Tweaked, poked and optimized, yes, but a sample nonetheless. What a synth like the TS — also called a "sample playback" synth, after all — does for you is that it provides samples trimmed to the nub, pre-looped and mapped out on the keyboard in groups that are accessible as coherent single entities ("waves" in TS-speak). The result is that a whole palette of memory-efficient samples-mapped-as-waves is available to you at all times.

A sampler like the ASR is not that fundamentally different a beastie, and sampled sounds really do operate similarly

to their TS cousins. But in a sampled sound, the various samples that make up a keyboard full of noise are not welded together as they are in TS waves — each sample is still a separate thing. Since you'd still want to construct sounds a keyboard-full at a time, however, there still needs to be a term for such a collection of samples. And there is: a keyboard full of samples is called a "Layer." The fundamental difference between a TS wave and an ASR layer is that in the wave, the samples are joined together into one uber-thing; in the ASR, each sample is still a separate entity. But they're both a TS wave and an ASR layer are merely a keyboard full of stuff.

Layer Down Gently

Select your sampled sound. (If you don't know how to select one of these thangs, as Lord Bowler would say, look for it in sound bank 8 or 9). Press Select Voice again. What you're now seeing is similar to the Select Voice page for TS waves: here's where you turn keyboards full of sound waves on and off, and also where you select what you'll be editing. This page, however, has a bit more complexity to it than the TS Select Voice page, since you've got be able to select layers and turn them on and off — just as you do with waves in TS programming — but you've got to be able to get to individual samples as well.

The bottom line of this display is the location for selecting which layers will be audible, and when. The two main things that make this look different than the TS Select Voice page are that, first, you see the number of each layer, instead of the name of each wave. And, second, you don't see them all at once as you do with standard TS sounds. In the TS world, any waves that don't play in the current Patch Select position are in parentheses; with sampled sounds, they're shown as dashes. Press your Patch Select buttons and watch the display: the layers that are active for each Patch Select position appear on cue — and the sound you hear when you play the keyboard will change — just as with a normal TS sound. Individual layers are turned on or off manually by pressing the middle soft button under the display repeatedly until the position corresponding to the layer you want to switch on or off is underlined (sampled sounds can have up to eight layers). Once it is, you use the up or down arrow buttons to turn the layer on or off.

Navigation Made Sample

The top line of the display is about selecting what you

want to edit. The sampled sound editing process begins with the selection of something to work on, just as in the rest of the TS. However, with a sampled sound, that could be any number of things. For instance, if you wanted to work on all of the samples in a sampled sound at the same time, you'd press the button above the instrument name on the top line of the display. Do it to it. As you can see, the instrument's name becomes underlined. This means that, should you change anything on any of the other editing pages, the entire sampled sound will be affected.

You can also press the button above LAYER=A. When LAYER=A is underlined, any editing you do will apply to all the samples in the selected layer. "Ah," you say, "how do you choose the layer you want to work on?" You can use the up and down arrow buttons to select any of the eight possible layers in the sampled sound. Only layers that actually have samples in 'em can be chosen.

Suppose you want to edit one single lowly sample? Simple. If the layer the sample lives in has been selected in the middle of the top line on the display, you can press the button above WAVESMPL (for "wavesample"). If there's more than one sample in the selected layer, you can play the keyboard and you'll see this number change as the different keys trigger different samples in the layer. You can also access these samples by using the up and down buttons, but playing the keyboard is a more intuitive (not to mention fun) way to be sure the sample you're hearing is the one you're editing. If you play the keyboard and the number doesn't change, or it shows ALL, you've merely got the wrong layer selected.

That's it for this class, class. Next time we'll pick up who knows where, and you'll be messing up sampled sounds like nobody's business in no time. ■

Bio: Robby Berman is a musician finally starting to understand what music's all about (send $5 SASE, no-no-no-no). His most recent gift to humanity is "Rings and Rings" (must've been a gift; nobody bought it).

12

Precambrian Synthesis

The DinosauRemains Series

Pat Finnigan

For: EPS/16+/ASR variants.

Product: *The DinosauRemains Vintage Sounds Disks.*

Price: $9.95 each or $39.95 for five.

From: DinosauRemains, 10702 Sagewillow, Houston, TX 77089, phone: (713)-481-1256 Monday thru Friday 9:00-5:00 CST.

This month we give a listen to the DinosauRemains disks. This series is aimed at the retro crowd looking for those ancient Mellotron, Orchestron, and other Triassic era sounds. Remember the Vako Orchestron with its optical disks and three octave keyboard? Or the two-manual (er, split 2-3 1/2 octave) $8000 Mellotron that would only play notes for 7 seconds before the tape played out? These archaic guys live on as 1's and 0's thanks to Dinosau-Remains.

The set came packaged with plastic disk pages and documentation; this is one of the most attractively packaged sets I've seen in a long time. Even came with a letter for the reviewer (which I don't read until after I've auditioned the sounds), so three stars for appearance. I auditioned these sounds on an ASR-10, EPS 16+ and EPS Classic, and since all sounds are on 800k floppies, they'll run on all Malvern samplers save the Mirage (which isn't exactly Paleolithic just yet).

So let's get paleontologic...

The Roses

These are very accurate captures of these instruments. Since they were taken by a 16+ some granulation is present but not overbearing. The "Mellotron Choir" tapes were pretty used, as a lot of hum is present in the samples. Then again, Mellotron were more inductive beasts than B-3's: any instrument with 40+ playback heads is going to attract any spurious EMF in the area, and I can't remember playing any Mellotron that *didn't* hum. The VP-330 Roland Vocorder samples are just as wimpy as the real thing: I bought a Korg VC-1 one-piece like the Roland for that very reason back in 1979, so again, accurate samples. The loops on the "Mellotron Violins" are very good considering what a PITA it is to loop string samples. Split

points are well placed, albeit some munchkinization is inevitable in the top ranges. Most of these samples don't play in the top octave to minimize interpolation anyway, so hats off again. "Mellotron Brass" is very cool with a built in "disco smile" EQ curve built in: this kind of accuracy is only useful in about an octave and a half range, but reminds me of the original Mirage Brass Disk that pioneered "free-running" loops like the "Tremolo Strings" Mirage disk.

The Thorns

These are very accurate captures of these instruments. The "Mellotron Choir" is a whole step off (Bb plays Ab), but the pitch knob of the representative Mellotron could very well have been off. Most of the sounds have no patch selects or other Ensoniq-unique programming (a glaring oversight). All sounds are *very* granular sounding, which, given, means the samples were taken with filters and envelopes wide open. While this is ideal for grabbing sounds, one usually would massage with ADSR filter and amp envelopes: these sounds use a selection from one of the 18 generic Ensoniq envelopes. Given, the sounds are paleolithic, but I would've liked to have seen a bit more inventive programming here. The "Octave Choir" sounds more like a Wurlitzer Theatre Organ than human voices. The Vako "Orchestral Violin" has an octave and a 4th subharmonic beneath any note you play, and the entire sample is a whole step flat. The Vako "Orchestral Violin 2" at least is in temper but exhibits the same subharmonic. The "Choir 5th" sounds like a B3 attack grafted onto voices. These would be very cool for the TS-variants, but since they have no sample editing capabilities, I can't recommended them for those particular variants until you've washed the limestone outta them...

The Deal

These sounds will take some massaging to work into a musical environment: for just knocking around they'd be fine, but I can't live with the aliasing in the top octave of these sounds. Good stuff for tweaking. The lack of patch selects seems a glaring oversight; thoughtful FX programming would have helped these sounds out, but I don't

remember any FX boxes back in the early '60s except the old Fender Spring Reverb box, so these are pretty raw samples; be advised/caveat emptor. In total, a good package with very usable raw footage with which to build your own retro library.

I apologize for not raving about these sounds, but since I had experience playing these keyboards in the late '60s and early '70s, I'd rather let those skeletons hang dusty in the closet. But just because they're coming back in vogue these samples are worth a listen if for no other reason than the historical perspective of the technological hoops we jumped through just to get realistic sounds back in that Precambrian era. If you've ever looked at or seen a Vako acetate you know what I mean. Arp, Solina, Elka, a veritable plethora of Italian electronic nightmares come to mind when researching more cost-effective string mach-

ines from the period. If you've ever changed a 3-voice tape pack on a double-manual Mellotron, you know what I mean and are sure to appreciate the work DinosauRemains has put into this set... ■

Bio: When not answering broadcast NTSC/PAL video-capture questions for TrueVision, Pat Finnigan is a tech support person for RasterOps, a company that makes video boards and monitors for Mac and IBM platforms. He still uses a B-3 for a keyboard stand and watches the alpha channel.

HACKER BASEMENT TAPES

Steve Vincent

Stamina, Jim Reynolds
Thom Johnson, Ed Greenman, Gary Rea, keope & Friends, Russ Stedman

---

Tape: *Affinity of Tribal Love and Perpetual Hate* (c) 1994.
Artist: Stamina (Mike Della Bella, Luisa Capuani, Alessandro Amoroso).
Contact Info: Mike Della Bella, 28 Viale Liegi, 00198 Rome, Italy; Fax: 01139-6-8530-0794.
Equipment: Ensoniq VFX, EPS-16+, ASR-10, DP-4, Kurzweil K2000, guitars.

Affinity of Tribal Love and Perpetual Hate is Mike Della Bella's "very first work" as "a hobbyist musician-producer." I can't wait to hear his second, third, fourth... Here is one talented hacker! Make that one talented trio: Stamina is Della Bella, joined by friend Alessandro Amoroso and singer Luisa Capuani, with very tasteful guitar work shared by Fabio Servilio, Angelo Di Muro and Lorenzo Salvatori (why do I have this sudden craving for lasagna?). Stamina covers similar stylistic territory as Amy Grant and Kathy Troccoli: danceable, upbeat rock and moving power-ballads.

The word "classy" kept coming to mind as I listened to these clean, crisp and captivating tracks. The arrangements take you through always-interesting yet accessible modulations, and the project is replete with very cool intros and outros. Della Bella's lyric writing is top notch: refreshing depictions of the passions and pains of falling in and out of love. He knows how to write about powerful, universal experiences, sans cliches.

I scoured the liner notes for drum credits, but apparently all drum tracks were sequenced. This is amazing; Della Bella and Amoroso obviously poured a lot of energy into making the percussion tracks sound not only human, but at times super-human (can you say Vinnie or Omar?). Capuani's vocals are sweet; she reminds me of Basia Trzetrzelewska (let's just call her "Basia"), substituting Italian for Polish accent.

Highlights from *"Affinity"*:

"Gifts of Life" — The album opens with this upbeat hip-hop dance tune, guaranteed to make you feel almost as good as a Miami Sound Machine opener.

"Under Cover" — This is a power-rocker in the style of Mister Mister, depicting someone daring to think about having hope again.

"If The Darkest Night" — an Amy Grant-ish ballad with tasty guitar soloing and captivating vocal arrangements. I kept waiting for the song to bust loose, but it stayed inside the boundaries.

"Nothing I Wouldn't Do" — keeps up the Stamina tradition of fresh arranging style on this break-up song.

"Affinity" — my favorite; combines all the best of Stamina: riveting lyrics, body-moving rhythm tracks, in-

teresting arrangement, killer guitar solo, Capuani's sexy voice...

"Until A Better Day" — Dann Huff guitar dominates the intro, then anchors the rest of this techno-shuffle. This middle/low-end guitar presence is what was missing on *"If The Darkest Night."*

"Heart And Soul" — Classic Della Bella: paints a landscape of the lessons of love expertly, using the mixed media of lyric, melody, rhythm, harmony and timbre.

"Unchain My Life" — This dark rocker is another Mister Mister-ish guitar symphony, with the best vocals on the album. Bravo!

Nitpicks in the mix: On a couple of tunes, an element or two of the percussion tracks leap too far out of the mix, e.g., a hi-hat pattern or an overly-gated snare. But these are venial sins in an otherwise nearly-faultless production.

Della Bella claims that he is "not at all a gifted musician." C'mon, Mike! You're way too hard on yourself! You can't be serious. I just can't wait to hear Stamina once they get their chops together! In typical self-effacing fashion, Mr. D.B. states in a P.S.: "If the tape proves to be any worth, it is available through me at the price of $10 shipped to the USA (what a deal! Kidding!)." No kidding: it *is* a deal. (See contact address and fax above.)

Tape: *Calenhad* (p) 1994.
Artist: Jim Reynolds.
Contact Info: 3598 Minuteman Way, Boise, ID 83706 (208) 383-0368.
Equipment: Ensoniq SD-1, EPS-16+ Rackmount, Rubber Chicken's Medieval samples, Korg M1 and Wavestation.

If any screenwriter ever gets brave enough to write movie scripts from the works of J.R.R. Tolkien that give his material the attention it deserves (vs. the cartoon versions we've been inflicted with), then Jim Reynolds ought to write the score. His (Reynolds') debut album, *Calenhad,* catches the spirit of Tolkien: a medieval hero's journey with enough tongue in cheek to avoid pretentiousness. Jim calls his musical style "MIDI-eval" or "New Middle Age Music," and it fits to a tee. His soundscapes evoke actual landscapes, and in fact are inspired by the actual Calenhad retreat in Oregon's Willamette Valley. A tour of his five-song tape:

"Getaway" — Jim's notes read, "Your pulse quickens and pounds as you flee the mad maze of the city..." but *"Getaway"* evokes more of a rollicking, joyful dance with its 12/8 time and lilting recorders. The introduction is su-

perb, with one of the best uses of nature sound effects I've heard: crickets and birds fade in, and the sound of horses' galloping hooves segues into the percussion track, all in less than ten seconds. The timing is excellent; most of these types of intros are way overdone.

"Flow Down, Flow Away" — In contrast to *"Getaway,"* which uses wonderful emulations of medieval instruments, this tune uses an English horn or oboe patch which sounds very unconvincing and unexpressive, possibly due to over-quantization. This is too bad, because the intro excellently imitates a babbling, cascading brook with downward arpeggios, but once the melody comes in, the song jerks, not flows. A little elbow grease and humanization could take this great melody and turn it into a great piece. Bring in a real horn player, and this song will rule.

"Leaves" — Another harp-dominated New Age track, evoking images of leaves fluttering in a breeze, or falling. Jim's use of the harp sounds reminds me of Georgia Kelly and Andreas Vollenweider, but a little less organic.

"Rangers" — The intro leads you to expect a rollicking fox hunt, but instead develops into a flowing, meditative song, which then morphs into *Camptown Ladies.* Jim explains that this melody is "usually attributed to Stephen Foster, but is in fact far more ancient." Nevertheless, the association with horse racing is too ingrained in our musical collective unconscious to be evocative of anything medieval, at least to this listener. And the ending is puzzling: the final modulation brings in a honky tonk piano, and the tune ends with what sounds like someone vomiting into the microphone! Don't get me wrong, I love a humorous twist to a song, but this is a rather strange and jarring departure from Jim's otherwise consistent, sincere style. I expected to see the Bros. Ryan credited with the production on this track!

"The Night Is Alive" — This is a twelve and a half minute ambient "nature" soundtrack, with persistent cricket chirps washed with synth pads and simple, soothing melodies. Well done.

My overall criticism would be that Jim's tracks are overly quantized, too robotic, with not enough of a human element. This can be cured! Otherwise, Jim has excellent compositional skills, and his love for the Middle Age comes through loud and clear. If he can get his songs to flow like the beautifully hand-calligraphed writing on his liner notes, he will be right on target. I can see Bilbo Baggins winking through the smoke curling up from his pipe.

Tapes: *Originals; Yin-Yang; Boogie Busters.*
Artist: Thom Johnson.

Contact Info: 2407 So. Lk Michigan Dr, Sturgeon Bay, WI 54235 (414) 743-3655.

Equipment: Ensoniq KT-76, Korg Poly-Six, classical guitar, Ibanez UE45 multieffects, Tascam Porta 1, Sony 690 DAT, Pioneer 703RS cassette.

The material submitted on Thom's three (!) cassettes spans more than a decade, and as many styles. Thom moves from blues noodlings to new age piano to men's movement drumfests to classical Japanese, on into his area of "main income," flamenco and classical guitar. When playing styles outside his comfort zone, these tracks sound like scratch-pads for experimentation. Thom obviously spends a lot of time in front of a red "Record" LED in his spartan studio. Suggestion: seek out collaborations with other composers and MIDI buffs to stretch your songwriting and sequencing chops.

Tape: *Ed Greenman — Composer Demo* (c)1994.

Artist: Ed Greenman.

Contact Info: 8967 David Ave, Los Angeles, CA 90034 Ph/fax: (310) 202-8907.

Equipment: Ensoniq EPS Classic, Yamaha DX-7, Boss DR-550 drum machine, Atari 1040st w/ C-Lab Notator & SMPTE sync, Tascam 38 8-track, Nakamichi cassette, Tascam M224 mixer, Sony DAT, Alesis Monitor Ones, Quadraverb, dbx 266 comp/gate.

Ed Greenman has the perfect demo tape: he presents excerpts from ten tunes, showcasing his compositional chops in a variety of styles. Ed is hawking his wares for scoring gigs for film, TV and computer games, and this tape ought to land him some jobs! He shows great versatility in composing TV show themes, circus/stage act music for a Princess cruise ship, a Processional he wrote for his own wedding, and a pop tune he wrote for singer Juliann Pugh to demo her vocal chops. Ed has some great sequencing here too, with crisp, clean, transparent mixes and slammin' percussion tracks. Overall, he has tasteful selection of voicings, with the exception of the ol' EPS Crunch Guitar sample; better to use a ballsy synth pad as substitute on those power fifths (of course, I'm a biased spanker of the plank). I'll pick one nit on the vocal demo mix: the synth pads compete with Juliann's vocals in the same frequency and pitch ranges; perhaps comping would have been more transparent with something like a classical guitar, to showcase her vocals at center stage. Otherwise, top-notch work! If you want scores for your circus, show, computer game, wedding or vocal demos, call Ed!

Tape: *Gary Rea Demo Tape* (p) 1994.

Artist: Gary S. Rea.

Contact Info: 12012 Camelot Dr, Oklahoma City, OK 73120 (405) 748-6319.

Equipment: Ensoniq ASR-10, Tascam Porta-03 Ministudio.

Gary describes his tape as a sampling of his "electronic" work, intended as film music. It includes thirteen tracks, many of which indeed sound like movie soundtracks, and are named accordingly: *Chase Scene 2, Chase Scene 3, Action Film 3, Suspense 3, Midnight, Darkfall, Glory,* and so on. These titles are helpful, placing each cut in its context. Gary's strengths include his ability to compose appropriate material for specific moods, and his choice of cool drum loops to anchor the music. For the woodshed: work on your synth performance chops on solo lines, specifically your use of the pitch wheel (be careful not to overdo it); and spend a bit more time mixing your sequences so the melody/solo lines don't oscillate between disappearing in the mix or taking the whole tune hostage. The compositions are good; make your demo present them in the best light!

CD: *Worlds Apart* (c) 1994 Keo-Neo Records.

Artist: keope & Friends (Jeff Hall).

Contact Info: P.O. Box 178484, San Diego, CA 92177.

Equipment: Ensoniq Mirage, ESQ-1, Alesis D4, Roland MT-32.

This is a group project knit together by four people who never met in person (keope (aka Jeff Hall), Renee Carter, Sally Daley, Scott Garrigus), but sent sequence files and song ideas to each other through the mail. The simplistic melodies executed with intermediate keyboard/sequencing chops, together with voice-over narrations that sound like pages from personal journey journals, combine to give this CD a "*Sesame Street* for adults" feel. At first I categorized this as a children's project, but it is more like an "adult child's" project, with its references to alcoholism from a recovery perspective and the joys of 96-color Crayon sets. The liner notes claim, "This music was produced without the aid of exciters, compressors, maximizers, noise gates, or other esoteric sound processing equipment." From an engineer's perspective, the CD could have benefited greatly from the use of these "esoteric" devices. On the positive side, this kind of collaboration-by-mail kept a positive spirit flowing that is evident in all the tracks, which can sometimes falter when human musicians actually have to meet each other in the studio.

Tape: *Mormon Cookin'* (c) 1994.

Artist: Russ Stedman and the 11.9% Interest Orchestra.

Contact Info: 4605 W 39th #301, Sioux Falls, SD 57106.

Equipment: Ensoniq ASR-10, Tascam Porta One, Art Proverb, WestOne Concord III, Twisted sense of humor.

Ever been burned by the LDS's? Then Russ Stedman's your man, if you're looking for music to soothe (rather, make that inflame) the burning in your bosom and give vent to your frustrations. Actually, Stedman is pretty spare in his rants over the Mormons. This mostly instrumental tape is filled with avant and Zappa-esque, twisted works that Dr. Demento would be proud of. Song titles include "Burnt Orange Carpet," "Action Potato," "San Diego Summer Seafood Swifty (The Joys of Mormon Cookin')," and one of my favorites, "The Earl Peel Museum." It's a bargain at $2.00 for 17 tracks (or so says the cassette liner). But go dig out your giant magical eyeglasses, 'cuz you're going to need them to read the 4-point font used in the liner notes, which you'll not want to miss, because they include the essay "How to Preserve a Husband." You'll find "Mormon Cookin'" in the frozen food case under "humor," or at Stedman's address printed above. ∎

[TH — For the next several months, Hacker Basement Tapes is going to be in "catch-up mode." We've divided the column into two sections: A main review section, in which we'll try to have at least a couple reviews per month. A "Short Takes" section. This will just be a few sentences on some of the submissions that have been

piling up the longest. If your music falls into this category, and you'd like to have a more thorough going-over, please feel free to re-submit.]

Tapes Recently Received

Synapse Crash — AARD

My Scarlet Life — Preston Klik

If *you* want your tape run through the wringer, err, Hacker, just mail it off to: Basement Tapes, *Transoniq Hacker*, 1402 SW Upland Dr., Portland OR 97221.

Bio: Steve Vincent produces demos and CDs at his home-based Portent Music, and can be reached via email at vincents@harbornet.com, or at his website at http://www.kspace.com/vincent.

Automated Delays

Ray Legnini

Hello again delay fans. In today's lesson we will explore more practical applications using Tempo Delay. The information provided here applies to the DP/4, DP/4+ and the DP/2. I'll note any changes that apply specifically to one unit or the other as we go.

In our hypothetical example, we'll presume that your song has a chorus section where you'd like to have quarter note echoes and a verse section where you'd prefer sixteenth note echoes. Let's presume as well that you want to have this change automated so that your MIDI sequencer sends the modulation commands to the DP.

For simplicity, let's send a Mod Wheel (MIDI Controller #1) on Channel 1 to the DP as the command to change the delay units from quarter notes to sixteenth notes. A Mod Wheel value of "0" will select the minimum value and a value of "127" will select the max. You don't actually need a keyboard from which to send the controller information, you can just add the commands to your sequence at the appropriate spots. Once you understand the basic concept, you can customize this setup to suit your individual needs. You can select any controller and any MIDI channel for the messages to be sent on. If you need more info on this topic, see your manual.

You'll need to hook up a MIDI cable from your sequencer's MIDI Out to the DP MIDI In. Set up your sound system with a single aux send connected to the DP input and patch the stereo outputs to your mixer. Select the "Select 1U Preset" Configuration on your DP.

If you call up the 1-Unit Tempo Delay algorithm into Unit A (1U preset #63 in the DP/4 and DP/4+, #88 in the DP/2) you will find that the default version has some modulation routings already set up for you (they are slightly different between the three models). The first modulation is the one we're interested in now. Check with your particular unit that there is nothing being modulated by the second set of parameters. Do this by hitting the "Edit" button, and then use the left/right arrows to move through the editable parameters. Disable the second modulation by setting the parameter "Mod2 Src" to "Off."

The DPs have eight available controllers that can be used as modulators. The sources of these are selectable by the user. They can be any legal MIDI controller. The settings for these are found in the System pages. Hit the "System" button and scroll to parameter #37 on the DP/4's or #23 on the DP/2. The list of eight DP controllers starts here. As you scroll through you'll notice that they have been set at the factory to some of the most used MIDI controllers. Since we can't presume the state that your unit is in from previous uses, let's select "Controller 1" and set the value of this parameter to "Mod Wheel #001." Remember "Controller 1." This is the info you'll need when setting up the modulation in the Tempo Delay. You'll enter this into the "Mod Src" field. While you are still in the System parameters go to parameter #35 on the DP/4's and #22 on the DP/2 (Control Chan =___). This selects the MIDI channel on which controller commands will be received. Again, for simplicity, set your Control Channel to "01" and check to see that the lower portion of the display reads "MIDI is Enabled." We're done with the System parameters. Hit the "Edit" and then the Unit "A" button to return to editing the Tempo Delay.

Select the "Tempo Control" parameter, #06. Set this to "MIDI clocks" — this will allow your sequencer to control the tempo. Scroll to the "Mod1 Src" parameter. Set this to read "Cntrl-1." The lower portion of the display will now read "Mod Wheel #001" assuming you did everything correctly. Set the "Mod Destination" to read "Parameter 003." This is the parameter that controls which musical unit the delay will produce.

But what values do you need to enter into the modulation Minimum and Maximum fields to achieve the desired effect proposed for our lesson? Read on. The available values for modulation amount go from 00% to 99%. To determine the relationship between percentage and the actual note value, set a mod amount for the "Min" parameter (parameter #12 on the DP/4, parameter #13 on the DP/4+ and the DP/2) and then scroll back to parameter 03 and look at the value as you move the controller to the minimum value or send the value from your sequencer. A handy feature to remember when dealing with any modulated parameters: the value for a modulated parameter will update in real time if you are on that parameter's edit page.

There are twelve musical divisions available in the DP's

Tempo Delay, from 1/32nd notes to 1/2 notes. With a calculator, you can determine that approximately every 8 units of modulation you add will give you the next musical division in the list. I've already done the work for you. Here are the modulation values and the corresponding musical unit that they will select:

00% selects 1/32 note
08% selects 1/16 triplet
17% selects 1/16 note
25% selects 1/16 dotted
34% selects 1/8 triplet
42% selects 1/8 notes
50% selects 1/8 dotted
59% selects 1/4 triplet
67% selects 1/4 note
75% selects 1/4 dotted
84% selects 1/2 triplet
92% selects 1/2 note

For our example, we select modulation amounts of 17%

and 67% to get the two delay units we want. Enter "17%" into the "Mod 1 Param Range Min" field and "67%" into the "Mod 1 Param Range Max" field. Now it's a simple matter of adding a Mod Wheel into your sequence at the measure where you want the changes to happen. Insert a value of "0" in your sequencer and you'll get the 1/16 note delay; insert a mod wheel value of "127" and you'll get the 1/4 note delay. Confirm the changes by watching the display update while you are still in edit mode with parameter 03 showing. Armed with the table above, it's easy to change values to suit any song. Since MIDI is controlling the tempo, this setup can work with any song in your library. As always, save this variation. ▄

Bio: Ray Legnini is the founder of the S.P.C.A. (The Society for the Prevention of Cruelty to Analog). Contributions to this worthy fund are always welcomed.

Classifieds

HARDWARE/SOFTWARE

Fostex 450 8x4x2 mixer $400, Alesis Quadraverb $250 or trade. Contact Mike at 1-541-967-7529.

WANTED: Ensoniq SD-1 32-voice. Call Bob, 214-278-9402.

WANTED: Yamaha YC-45D and Vox Continental Baroque combo organs. Contact Kirk at (303) 979-6837 (new, improved phone number - please try again) or PROTIST@AOL.COM.

WANTED: SD-1 32-Voice. Leon Forrest Productions, phone: 541-996-2575.

SAMPLES/PATCHES/SOUNDS

DinosauRemains — vintage sounds for EPS-16+. New Mellotron multi-sample disks. Two-disk sets of violins, choir, brass.

If you're selling your gear...

Please be sure to pass along how absolutely vital it is to have a subscription to the *Transoniq Hacker*. And — we're always happy to do a sub transfer. No charge, and it's a nice extra to help close the deal.

Sets are $19.90 each. (Includes s/h.) Call for free brochure. 713-481-1256.

Prosonus Disk set. Sounds for Akai S-900/S-950, Oberheim DPX-1, Roland S-550/S-330/W-30, E-Mu Emax, Casio FZ-1/FZ-10M/FZ-20M, Yamaha TX16W. Includes: Orchestral Percussion, Drum Kits, Synth Stacks, Violin Sections, Brass Section, Trumpet & Elec. Bass. Phone: (818) 797-3605.

CUSTOM ASR-10 VOCODER SAMPLES! Yes, indicate what phrase needed (up to 12 syllables) and I will create/sample the robotic voice. These aren't vocoder samples, rather Kraftwerk "music nonstop" voices. $10/phrase, 5/$40 pp. For info write: Wayne Schroeder, 573 Scott St., Studio L, San Francisco, CA 94117.

ASR-10 samples of the TS-10. Hi fidelity, fully programmed. $6 per HD disk, $30 for all six (36 sounds). Check or M.O. Or send SASE for complete documentation. James Samp, 90 N. Bell St., Fond du Lac, WI 54935. Phone: (414) 922-4337.

TH's Jack Tolin presents SYNTH-BITS! Classic HR-16 drums (2 disks; 49 sounds), M1 synth-textures (5 disks; 50 sounds) come in easy to manage samples that turn your EPS-16+ or ASR-10 into a synthesizer! Star-Trec SF blurbs (2 disks; 20 sounds). Only $3 per disk. SF blurbs for your PC: $2

per disk, 12 disks available. Jack Tolin, 9314 Myrtle Ave., # 186, Kansas City, MO 64132.

Your ultimate source for synth samples for your Ensoniq Sampler. Moog, Korg, Oberheim, Yamaha, Sequential! Send SASE for free catalog: Tom Shear, 805 Fifth Avenue, Williamsport, PA, 17701.

OUT-OF-PRINT BACK ISSUES

M.U.G. will provide Out-of-Print issues for cost of materials and postage. Write: G-4 Productions, PO Box 615TH, Yonkers, NY 10703. Attn: TH Back Issues. * * * Folks in the New York City area can get copies of unavailable back issues of the *Hacker* – call Jordan Scott, 718- 983-2400.

FREE CLASSIFIEDS!

Well – within limits. We're offering free classified advertising (up to 40 words) for your sampled sounds or patches. Additional words, or ads for other products or services, are $0.25/ word per issue (**BOLD** type: $0.45/word). Unless renewed, freebie ads are removed after 2 issues. While you're welcome to resell copyrighted sounds and programs that you no longer have any use for, ads for *copies* of copyrighted material will not be accepted. Sorry – we can't (we won't!) take ad dictation over the phone!

Prog: GM VIBRANALOG
<div align="right">By: ???</div>

WAVE	1	2	3
Select Voice	On	On	Off
Wave Class	Waveform	Transwave	
Wave	Sawtooth	Res1-X	
Delay Time	0	0	
Wave Direction	-	Forward	
Start Index	-	0	
MODSCR	-	LFO	
MODAMT	-	3	
Restrk Decay	0	0	

PITCH	1	2	3
Octave	0	0	
Semitone	0	0	
Fine	0	4	
ENV1	0	0	
LFO	8	8	
MODSCR	LFO	Noise2	
MODAMT	-3	7	
KBD Ptch Track	On	On	
Glide	Off	Off	
Glide Time	0	0	

ENV1	1	2	3
Initial	0	0	
Peak	127	99	
Break	127	99	
Sustain	127	99	
Attack	15	15	
Decay 1	15	15	
Decay 2	15	15	
Release	15	15	
Vel-Level	0	0	
Vel-Attack	0	0	
Vel Curve	QuikRise	QuikRise	
Mode	Norm	Repeat	
KBD Track	0	0	

LFO	1	2	3
LFO Speed	30	0	
Noise Rate	46	93	
Level	3	0	
Delay	0	0	
MODSRC	Pedal	Pitch	
Wave	Triangle	Triangle	
Restart	On	On	

FILTER	1	2	3
Filter 1	2LoPass	2LoPass	
Filter 2	2LoPass	2HiPass	
FC1 Cutoff	85	84	
ENV 2	-11	-6	
FC1 KBD	0	-11	
MODSCR	Wheel	Velocity	
MODAMT	+11	0	
FC2 Cutoff	90	41	
ENV2	-4	0	
FC2 KBD	0	0	
FC1MOD-FC2	On	On	

ENV2	1	2	3
Initial	31	29	
Peak	85	88	
Break	50	50	
Sustain	88	89	
Attack	2	2	
Decay 1	5	5	
Decay 2	10	10	
Release	25	26	
Vel-Level	26	19	
Vel-Attack	0	0	
Vel Curve	Linear	Linear	
Mode	Repeat	Repeat	
KBD Track	0	0	

AMP	1	2	3
Initial	0	99	
Peak	99	99	
Break	95	99	
Sustain	88	99	
Attack	2	42	
Decay 1	30	42	
Decay 2	42	42	
Release	25	26	
Vel-Level	0	0	
Vel-Attack	0	0	
Vel Curve	Linear	Linear	
Mode	Normal	Normal	
KBD Track	0	+28	

OUTPUT	1	2	3
VOL	38	84	
Boost	Off	On	
MODSRC	LFO	Env2	
MODAMT	0	28	
KBD Scale	0	-19	
Key Range	C2-C7	A0-C8	
Output Bus	FX1	FX1	
Priority	Medium	Medium	
Pan	0	0	
Vel window	>000	>000	

EFFECTS — CHORUS AND REVERB

FX-1	26	FX-2	75
Decay time	25	HF Damping	33
Chorus Rate	20	Chorus Depth	4
Chorus Center	44		
Feedback	42		
Chorus Level	53		
MOD (Dest)	Rate		
BY (MODSRC)	ModWheel		
MODAMT	13		

The Hack: Well folks, here's a *Hacker* first; we're not sure who sent this one in. It arrived on the portal step last April along with a lot of other SQ/KT patches by the same author. What distinguishes this work from the usual hand-written stuff we get is that all the patch parameters are neatly printed from an editor/librarian program. The problem is that it doesn't print the name of the person who created the patch. So fellow Hackers/Hackettes, if you think this is yours, you might let us know.

"VIBRANALOG" is a warm, reedy dessert that may be at home in a bed or as a solo voice. It's "VIBRA" comes from the LFO applied to pitch in Voice 2. The Sawtooth wave is assigned to Voice 1, while Voice 2 relies upon the Transwave Resonant 1-X Waveform: thus the "ANALOG." Note the ModWheel varies the Filter cutoff for Voice 1.

First, for those who prefer "VIBRANALOG" pretty much as is, just some minor suggestions: 1) The volume levels between the two voices don't seem balanced. Try using Vol=55 for Voice 1 and Vol=70 for Voice 2. As long as we're in the Output section

make use of the SQ/KT's stereo outs — Pan for Voice 1 can = -70 and for Voice 2 - +70. 2) The Filter Mode for Voice 2 allows for quite a bit of noisy clatter. Switch to Filter 1 = LoPass Filter 2 = HiPass to reduce the racket.

Some may like these additional Hacks: 3) Once again (and again), if you're going to go to the trouble of building a pitch envelope you've got to use it somewhere as a modulator - you've gotta turn it on. Apply Pitch to Voice 1 only — let Voice 1's Env1 value be negative at -01. 4) Detune the voices; in the Pitch section for Voice 2 set Fine = +07. Finally, 5) the Chorus could be bolstered a little. In the Effects section, set Chorus Depth to 15.
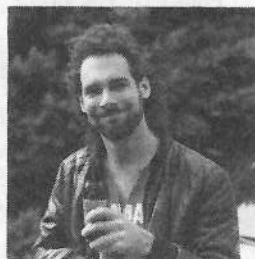
Bio: Jeffrey Rhoads has been a keyboardist/composer on the Philadelphia scene for a period of time resembling forever. Jeff still believes in magic and longs for city lights.

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...

Programming the SQ-1:

Taming Transwaves

Clark Salisbury

Welcome back! Well, I guess it wasn't really you that were gone for the last two months, it was me. Been a busy little beaver out here in the great Pacific Northwest. Busy, busy, busy. That's me, all right.

So. Where were we, now? Ah yes — we've looked at a lot of stuff in the old SQ. We've checked out the waves, LFO's, envelopes, filters, amplifiers, envelopes — covered a lot of ground. So far it's been pretty much your standard synthesizer stuff, though. Now it's about time to start getting into the interesting stuff that makes the SQ unique. We begin this month by looking at the Transwaves.

Transwaves are an Ensoniq innovation, and are a response to a problem that has surfaced with the advent of digital synthesizers that rely primarily on wavetables for their raw data — namely that sounds created by playing back digital wavetables can seem rather static. One of the things that makes acoustic instruments so pleasing is that the waveform produced by acoustic instruments is evolutionary — generally, acoustic waves are constantly changing — as the sound moves through its stages, its harmonics interact in complex ways, producing an ever- changing timbre.

The transwaves differ from the other waves contained within the SQ in that they are not really just a single looped wave, but rather a series of different, but closely related, waves through which the loop can be freely moved. The effect is that the Transwave sounds different depending on where the loop is placed; move the loop through the transwave, and the various waves it contains blend one into the next in much the same way that individual frames in a movie blend together to form a moving image. For example, if we were to digitally sample the word "dayo" we could loop the entire word to produce "dayodayodayo..." If we use brackets to indicate the loop, we could show the same thing in this way: [dayo].

Suppose then we loop just the a: d[a]yo. The resulting sound would be "daaaaaa..." Or if we looped just the o (day[o]), we'd get "dayooooo..."

Now let's say that we can control the start point of the sample. We could indicate the start point using the symbol >; in this case, setting the start point to the beginning of our sample could be shown in this way: >dayo. We could then produce sounds such as "ayoooo...," (d>ay[o]), "ayyyy...," (d>a[y]o), "yoooo...," (da>y[o]), and so on. And finally, if we could move the loop while the sound was playing, we could produce things like "dayooyadayooyad...," (>[d]ayo), and "ayyaayya-ayyaay...," (d>[a]yo) by moving the loop back and forth, or things like "dddaaayyyooo,"(>[d]ayo), and "yyyyyaaaaa...," (d>[a]yo), by moving the loop slowly in one direction or the other.

What controls loop position in the transwave? First, the loop can be manually set to any point in the transwave using the start index parameter. This not only sets the transwave to start playing from the point selected, but also places the loop at that point. In other words, setting the start index for the word "dayo" to a, (d[a]yo), will produce the sound "aaaa..." But to actually move the loop through the transwave, we'll need to apply a modulator, such as an LFO or envelope. Let's try it out.

Start by selecting ROM program 04, Today's Organ. First, let's set the effect to something more-or-less generic — perhaps the Concert Reverb would be a good choice. Hit Edit, then Effect, then the 0 button, and use the data slider or Yes/No buttons to change the effect from Phaser+Reverb to Concert REVERB. Now we need to select voice1 for editing. Hit the Wave and then the 0 button to move to the SelectVOICE page. Solo voice1 by selecting it and hitting the Up/Yes button — the display should be showing something like this:

SelectVOICE:ONE
SOLO OFF OFF.

Now hit button 1 to move to the Wave page. Select WAVE-FORM and hit Up/Yes twice to select the TRANSWAVE waveclass. The display should now be showing:

Wave=TRANSWAVE
FORMANT-X.

Hit the 3 button — this takes us to the page where we can set the start index and mod source and depth for this particular transwave. Select the Start Index parameter, and move the data slider while playing something on the keyboard. As you move

the data slider up, you'll hear the sound get thinner. Note that you can leave this parameter set to any value you like, and use the resulting wave in whatever sound you're trying to create. The really interesting stuff, though, occurs when you dynamically modulate the start index (actually, you'll be modulating both the start point and the loop location) with an LFO, envelope, or other modulator. Let's try using an LFO to hear an example of this.

First, set the Start Index back to 00. Now set Mod= to LFO * +99 — you should already be hearing the effect of LFO modulation on the start index, but let's tweak it up a bit.

Head over to the LFO menu (hit LFO). First, let's slow the LFO rate down a bit. Hit the 0 button, and change LFO rate from its current value of 29 to something like 20. Hit the 1 button to move to the next page. From here let's set LFO level to, say, 65, and Dlay (delay) to 00. It should become obvious by now that the LFO is modulating not only the transwave start index, but the pitch of the wave as well. Let's do away with the pitch mod for now — hit the Pitch button, then button 1. Set the LFO= parameter to +00 (it should currently be showing a value of +04). This will eliminate the rather seasick-sounding vibrato.

Now head back to the LFO menu (hit the LFO button), and let's try some different LFO waveshapes. Press the 2 button to move to the LFO wave page — the wave used happens to be the trusty Triangle wave.

As it so happens, the triangle wave is probably not really the best choice for a modulator in this application, since it has both positive and negative movement. You see, we've set the start index of the transwave to 00, the lowest possible number. When the triangle wave moves in a positive direction, of course, it moves the start index of the transwave higher, resulting in the sweeping effect you should be hearing. When it moves in a negative direction, however, the transwave seems to stop sweeping. This is because the transwave can't be modulated any lower than 00, even though the triangle wave produced by the LFO is moving in that direction. So a better choice for the LFO wave would probably be one of the positive-only waves, the POS-SINE or the POS-TRI — either of these waves will modulate the transwave start index with positive numbers only. Give each of them a listen — they

produce subtly different effects. To my ear, the POS-SINE produces the smoothest sweep, so let's use this wave for now.

At this point, you might want to listen to some of the other transwaves to get a feel for how each of them sounds when being modulated in this way. Head back over to the Wave menu and hit the 1 button. From here you can easily select the wave (in this case the Formant-X wave), and use the up/down buttons to step through the other available transwaves. Once you've finished, set the wave to Pulse 1-X.

We're going to use this wave to re-create a popular analog synthesizer string sound based on pulse width modulation. Back in the good old days, any analog synthesizer worth its salt included at least one pulse wave with a modulatable duty-cycle. A pulse wave with a 50% duty cycle (high 50% of the time, and low the other 50%) could also be called a square wave — that's because it looked square when viewed on an oscilloscope (diagram). This wave sounds much like the Pulse 1-X wave with no modulation and its start index set to 00. To hear this sound, set LFO modulation and start index both to 00 — your display should look like this:

Start Index = 00
Mod=LFO * +00.

A 10% duty cycle, on the other hand, means that the pulse wave is high 10% of the time, and low the other 90%. If you now set the start index for the Pulse 1-X wave to 99, you should hear something very close to this sound.

To create more animated sounds, a modulator would often be routed to the pulse wave, much as we have done here with the application of LFO modulation. The resulting sounds were used in many different kinds of patches, one of which is a string-ensemble simulation that has come to be associated with this technique of pulse width modulation.

Now to get back on track, set the start index once again to 00, and LFO mod amount to +99. Next, let's tweak the filter a bit. Hit the Filter button, and then button 1. From here we can adjust the cutoff point of filter1. Let's set it for something like 80. Since filter1 is set up for 3-pole low-pass operation, this should mellow the sound out nicely. Notice, though, that the high notes seem a bit too mellowed out; let's check the status of the keyboard tracking parameter. Hit button 2 and you'll find that, sure enough, keyboard tracking is set to +00 — the filter is not tracking the keyboard at all. Let's try setting the FC1 Keyboard parameter to something like +26. This sounds pleasing to my ear.

Now let's move to the Amp menu, and apply a string-type default envelope. Hit the Amp button, then button 3. Hit the Enter button, scroll up until you reach the SLOW STRING default envelope, and hit the Enter button again to apply the envelope. For the finishing touch, hit the Effect button, then

the 0 button, and scroll up until you reach the CHORUS+ REVERB effect.

There you have it: pulse width modulation strings. For a bigger sound, you might want to try using two voices, with each using the Pulse 1-X transwave, but with slightly different settings for start index and LFO rate on each, and each voice slightly detuned from the other. Or you might want use the voice we've just created, and add another voice based on the STRING ENSEMBLE wave, to create a hybrid analog synth/sampled string section sound. At any rate, if you want to keep this sound, save it now before we move on.

All saved? Good. Let's get back to the transwaves — there are several other things you can do with them that I'd like to point out.

We'll use our new string sound as the starting point for are next foray into transwave territory, a simulation of another classic analog-type sound, a resonant filter sweep.

Head back to the Wave menu, and press button 1 to move to the Wave page. For this sound, we can use any of the resonant transwaves, RESONANT 1-X, RESONANT 2-X, RESONANT 3-X, or RESONANT 4-X. Each of these waves has a sound similar to that of an analog wave being routed through an analog filter with the resonance (or Q as it is sometimes called) set fairly high. The higher numbered resonant waves simulate higher resonance settings — as well as being somewhat different from one another timbrally. For this patch let's use the RESONANT 3-X wave.

First we'll simulate a filter sweep using this wave. As you might guess, this can be accomplished by applying an envelope to modulate the start index of the wave. Hit the 3 button and select the mod parameter — it should currently be set to LFO. Change this to ENV2, and leave the amount of modulation set to +99. The display should be showing this:

Start Index = 00
Mod=ENV2*+99.

Now let's use ENV2 to create our "filter sweep." Head over to the ENV2 defaults page, and hit Enter. Let's try the Ramp Down envelope — scroll until you see it in the display, and hit Enter. Play a few notes to get a feel for the effect — a slow, resonant downward "filter sweep." Let's try speeding it up a touch — hit the 0 button to move to the envelope programming page. You should find the envelope set to begin at a level of 99, move at a time of 10 to the next level, which is also 99, then move at a time of 62 to a level of 00, the level at which it stays. We can speed up this envelope simply by reducing time2, the time it takes to go from level2 (99) to level3 (00). Let's change the value of time2 then from 62 to something like 32 — this speeds up our "filter sweep" rather significantly. Now let's adjust the Amp envelope to something a bit more appropriate for

the sound we're after here. First, press the Amp, then the 3 button, and hit Enter. Let's start with the PIANO DECAY envelope — scroll to it, and press Enter once again. Now hit the 0 button to move to the envelope programming page — I want to make a couple of minor adjustments to the this envelope.

First, I'd like this sound to be a bit more clavinet-like. One thing we can do to give a bit more of this kind of character is to shorten up the envelope's release time — otherwise known as time4. Currently it's set to a value of 30 — let's change this to a value of 06. This, as you will recall, will cause the sound to decay more quickly once the key has been released. Next, let's reduce the effect that velocity has on the volume of the sound. Press button 1, and take the value for LevV down from 19 to 13.

Now let's brighten the filter up a bit. Hit the Filter button, then the 1 button, and increase the FC1 Cutoff (filter 1 cutoff) parameter from 80 to 127 — full open. By now we should have a pretty respectable analog synth simulation. It might be interesting, though, to check back in with the transwave and see what other adjustments might be made there. Hit the Wave and then the 3 button to move back to the Start Index page. From here, we can adjust the depth of the "filter sweep" simply by adjusting the ENV2 mod amount. Try setting ENV2 mod to something like +36 to get a feel for this — now the envelope doesn't sweep quite as high in the transwave. While you're at it, you might try setting the start index to something higher than 00 — try 48, for example. Now the "filter sweep" doesn't end up quite as dark as it did originally.That's because the transwave doesn't come back to a start index of 00 anymore, but to the brighter-sounding start index at 48. As long as you're at it, you can reverse the "filter sweep" direction entirely by setting the transwave start index to 99, and the Env2 mod amount to -99. Or you might try other modulators all together — you could set the start index back to 00, and use WHEEL (the modwheel) to modulate it. Try starting with a mod amount of +99 — this will allow you to control the "filter sweep" from the modwheel to produce those classic wah-wah type effects.

Be sure to check out some of these techniques with the other transwaves, too. If you find a sound you like, be sure to check it out with some of the other effects available in the digital processing section — the addition of a bit of chorusing, flanging or phase-shifting might be a nice touch. I'll see you back here next time for more programming and other digital hijinks. ∎

Bio: No bio, okay?.

[TH — Sure!]

Hi

Do you know a converter for Macintosh from ASR disk images to ASR disk? If yes, please post me where I can download it.

Thanks,
Pierre Alexandre Tremblay

[CS – You can use scEPSi, created by Steve Berkley, and available for downloading from his ftp site, (ftp://ftp.crl.com/ftp/users/ro/sberkley/Utilities/).]

TH –

Is it true that the EPS has been discontinued? A sales associate at Sam Ash gave me that information today. But I have found someone who is willing to sell me a used one that's fully maxed-out for $500. Is that about the right price for an older model?

RGreen5933@aol.com

[CS – The EPS was discontinued several years ago. Still, if the unit you're looking at includes all the accessories (4X memory expander, SCSI, and 8 output expander), $500 seems like a fair price to me. You might want to poke around the 'net a bit, though, and see what you find out. Check out the Used Gear Pricelist, (http://www.synthcom.com/ cgi-bin/ gear) for example, or nose around at the Oakland University EPS archives (http:// www.acs.oakland.edu/oak/eps/eps.html) for more information on what used Ensoniq gear is going for.]

Greetings,

I was vacuuming the carpet when I noticed two items (DP/4 and SQ-1+) went black. Now when the SQ-1+ tries to boot up it goes through part of its routine and then displays *Unexpected Event ID #33*. Sometimes it will go through its complete cycle but if I try to change a sound the *ID #33* appears. I've gotten so far as to reinitialize the RAM but the same problem persists.

Any possible hints as to the cause?

Any response is greatly appreciated.

dru...
dru@dru.seanet.com

[CS – I haven't a guess to hazard, but the problem doesn't sound trivial to me. I'd suggest getting in touch with Ensoniq Customer Support (http://www.ensoniq.com/contact.htm) directly.]

Hi.

I purchased an SD-1 last month, and I'm having a few minor problems interfacing it with my PC (Pentium 75, 16MB Ram, etc.). Hopefully, I'm not the first to experience these problems. I'd like to be able to move patches back and forth between my PC and SD-1. So far, I've tried using two pieces of software, but to no avail. When I try SYXUTILS from the Ensoniq site, it tells me that it cannot access my MIDI Port @ 330, IRQ 2. This is definitely where my MIDI port is set up. Other programs (DOS, WFW 3.11) have no trouble sending MIDI signals to this port, and the SD-1 will respond appropriately.

Next, I've tried a program called SoundLib, which is supposed to be compatible with the SD-1. Yet, when I try to get data from the SD-1 to the PC, any data after 55/60 is not recognized. I'm left with a "Bank Can Not Be Loaded" error message. I am able to send patches to the Ensoniq with (apparently) no problem. Any help would be greatly appreciated.

Robert Campbell
briteboy@betty.bigeasy.com

[The programmer of Ensoniq's SYXUTILS responds – The DOS MPU-401 Sysex utilities that Ensoniq gives away for free (http:// www.ensoniq.com/midi/binary.htm) are DOS utilities and may not work correctly if run from a DOS Shell under Windows. I have seen this happen before with certain MIDI cards and drivers. I suggest you exit Windows entirely, and try running the Sysex utilities from a clean DOS prompt. Also, if your MIDI card is not 100% MPU-401 compatible, then the utils might not recognize your hardware, which might explain the error message as well. Good Luck!

[John L. Whipple – I am trying to interface my SD-1 with my IBM. I read your post and wonder if you could let me know if you have found software that works. Thanks, John, mdg@texas.net]

[CS – I'm afraid my experience in this area is pretty limited; hopefully, Ensoniq (or one of our readers) can shed some light on the subject.]

TH –

I have an Ensoniq EPS-16 Plus fully loaded minus Flashbank. I've recently purchased a IOmega Zip Drive and wanted to use it to store samples from Alchemy. I would like to transmit my samples that I have in Alchemy to the EPS-16 Plus by SCSI faster. At present I'm using a Mac Performa 450 with 8meg RAM. I tried to go direct from the sampler to the Mac's hard drive. That didn't work. Then I tried using my Zip Drive. I connected the Mac to the Zip Drive, then the Zip drive to the sampler. This didn't work either.

I've read that from the Alchemy manual that the SCSI ID for the EPS/EPS-16 Plus is fixed at 3. And the Zip Drive only has a switch on the back that reads SCSI ID 5 or 6.

How do I get the EPS-16 Plus to recognize this data transmission from Alchemy? Is there a solution to this without buying an external hard drive?

Thanks,
Larry
lbnet@earthlink.net

[CS – There are several things to keep in mind when attempting to connect your EPS-16 Plus with SCSI devices:

1) Be sure to terminate properly. The first and last devices in a SCSI chain should be terminated. Your EPS and Mac internal hard drive are terminated (unless they've been modified from factory specs), so they should appear at either end of your SCSI chain. For complex systems some sort of active termination (such as the SCSI Sentry from APS) may prove helpful.

2) Use the shortest SCSI cabling possible.

SCSI chains should not exceed 20 feet in length, and can become troublesome even at shorter distances.

3) Make sure your SCSI cables are functioning properly. SCSI cables seem to have an inordinately high failure rate.

4) It may seem obvious, but power up your SCSI devices before your Ensoniq instrument. The EPS will check for SCSI devices as part of its boot routine. If it finds none, it will assume none are attached. The EPS cannot mount SCSI devices powered up after it.

5) Parity must be disabled on SCSI devices to communicate properly. The EPS and EPS-M do not support SCSI devices which perform Unit Attention. EPS and EPS-M users must have at least EPROM 2.40 to support Fujitsu, Conner Peripherals, or Quantum Pro SCSI devices.

Other issues to keep in mind: You must have bi-directional MIDI communication between the MAC and EPS for SCSI transfers to work; be sure you have Sysex enabled on the EPS; make sure you are correctly grounded – I've had some difficulty getting transfers to work predictably in improperly grounded systems.

If you're still having trouble covering all these bases, try getting in touch directly with Ensoniq Customer Service, and go through the process with them.]

[Ensoniq – We just want to be sure something is clear here – the EPS-16 PLUS cannot read Alchemy files directly from a drive. Alchemy stores samples as AIFF files, which it can transmit to the EPS-16 PLUS, but then where do you intend to store them?

If you are trying to get the EPS-16 PLUS to see your Zip drive (or your Mac hard drive) that will not work if you format them as MAC devices. You could use your ZIP drive as storage for the sampler, by formatting the cartridges with the EPS-16 PLUS.

Beyond that, Clark's points about getting the SCSI chain to work are the right things to look into. Still confused/not working? Call our Customer Service department.]

_____

TH –

I am very experienced with the EPS-16+ but even used it's out of my budget. Does anyone have any suggestions? Also, if anyone has the specs for either the Mirage or the original EPS and could send them to me I'd appreciate it.

Thanx,
Dan Caffee
rockstar@erols.com

[CS – The EPS is a lot closer to the EPS 16-Plus, and a vastly more capable and better sounding instrument then the Mirage. I'll bet you could get specs on either instrument if you e-mailed Ensoniq Customer Support (http://www.ensoniq.com/contact.htm) directly, and asked nicely.]

_____

TH –

Three messages left at Ensoniq have failed to get me a reply, so let me try you: I have an EPS-M sampler. I bought it used and I am

unsure of any mods that were made to it.

1. What do I need to do to use a Syquest drive? There is a SCSI port w/ 15 or 19 or however many pins, so it LOOKS like you can just plug a cable right in. But I am aware that these machines require modification if one wishes to use external drive. How can I tell if mine's been modded or not?

2. Likewise, can I increase my sample memory? Currently I have 4096 free blocks in memory. Can I increase this? I recall that there's some memory expansion module, do I have any hope of still finding one?

3. I have a strange sort of interference problem with using the 8 individual outs. If I use neighboring individual outputs, very noticeable static is introduced into one of the signals. However, through experimentation I discovered that if I use every other output (1, 3, 5, and 7) then there's no static, everything sounds clean. My cables work fine with my other gear. I'm using a grounded, surge-protected power strip. Does this sound familiar to anyone? What other remediation/protective measures should I be taking?

Any response is greatly appreciated.

Thanks a lot,
Martin Glaubitz

[CS – 1) If your EPS-M has a SCSI connector (25 pin, by the way), then it's already been modified for use with SCSI devices. Theoretically, you should be able to just plug in your Syquest drive (using a standard 25-pin to 50-pin SCSI cable), and go.

2) Your EPS-M already has the maximum memory it can accommodate.

3) This problem rings no bells with me. I'd suggest getting in touch with Ensoniq Customer Support (http://www.ensoniq.com/contact.htm) directly.]

———————

Gentlemen and/or ladies,

The reason I am writing is I am an owner of a TS-12 and a Mac, and while perusing the latest *Hacker* I came across a couple of Internet sites for finding out how to transfer samples from Mac to ASR. My first question is: does this apply to downloading to a TS or am I just whistling in the wind?

Secondly, my attempts to access Berkely's ftp site and TH's ftp site have been horribly thwarted by either my own computer or serv-

er. Does anyone have the address for either of these, and does anybody know if scEPSi will do what I want?

Thank you,
Alcasio@aol.com

[CS – You can't transfer samples from your computer to the TS-12 via SCSI. Pretty much the only way to do this is to first save samples to ASR/EPS formatted diskettes using something like scEPSi. For more information, I'd suggest checking out Garth Hjelte's "From Cyberspace to your Ear," available at the Hacker's ftp site (ftp://ftp.teleport.com/vendors/trnsoniq/) – if you can access it. If not, the article originally appeared in TH Issues #122 (August '95) and 123 (September '95).]

———————

Hey TH,

I own an ASR-10 and I need to buy more sounds for it. I am not eager to mail order disks because all I have to go by is the names of the disk sets. Is there anywhere on the www to download a little sample of disks you mail order?

Thanx,
Eric Inman
judge@STIC.NET

[TH – Well gee, that's what Hacker reviews are for.]

[CS – You can find audio demos at Ensoniq's Web Site (http://www.ensoniq.com/). Other than that, I don't know of anything in particular, but if you feel like exploring you might start by pointing your browser at Michael Hyman's most excellent Ensoniq Resources on the Internet Guide (http://www.netaxs.com/~mikeh/ensoniq.html).]

[Potaczek (potaczek@misslink.net) – Rubber Chicken Software will send you two disks that have some really cool sounds on them to give you a little taste of what to expect from them. In my mind, Rubber Chicken is one of the best companies to service Ensoniq out there and I would certainly give them a call if I were this guy.]

[Bernie Fab (Bernie_Fab@wow.com) – I second the raves for Rubber Chicken. Their freebie disk has more usable cool stuff on it than most complete sets from other vendors. And the service can't be beat. Garth really has his act together.]

———————

Oh joy, oh bliss!

I just got the update to Giebler's EDM which supposedly formats and reads/writes Ensoniq SCSI drives. So hopefully now I can now transfer my zillions of **TS-12** floppies to a hard drive and take my TS-12 on gigs!

Next step: Can you recommend an external SCSI drive which I can hook to my IBM PC *and* will work with my TS-12? I had heard that RamTek could do this but I also heard they were going out of business.

I need specifics! I have heard too many horror stories of drives that work on the PC but not the keyboard, or vice versa. Before I plunk down the money for the Ensoniq SCSI adapter and the drive I would like some reassurance that I'm not going to be a guinea pig.

Thanks so much,
jc Harris
70242.1520@CompuServe.COM

[TH – Judging by the way RamTek doesn't pay their bills, we would really hesitate before sending any money in their direction.]

[CS – You might surf over to Ensoniq's approved drive list (http://www.ensoniq.com/mid/scsi.htm) to see what's been tested and approved for use with your TS-12. Most of these drives should work with your PC, but when you settle on one, drop the manufacturer a line to find out for sure.]

———————

Hello,

I am writing on behalf of Midiking Technology in South Africa. We are unable to get Simultaneous Record And Playback working under Windows 95 for the SS Elite cards. It seems okay, but then when we play what was recorded, it comes out at about twice the speed.

We are trying to sell the cards with either SAW plus, Cakewalk or Cubase.

Ensoniq hasn't been able to help us yet. If you could help, it would be greatly appreciated.

Thank you,
Anthony Spirou
StreetPrice Music (Showroom for Midiking Technology – Authorized importers for Ensoniq in Africa)

[CS – You might try pointing your browser at Ensoniq's web site (http://www.ensoniq.com/). From here, you can e-mail Ensoniq Multimedia Customer Support

directly (mailto:multimedia@ensoniq.com). And while you're at it, you might want to check out Ensoniq's online FAQ on using the Soundscape with Windows, (http://www.ensoniq.com/mmd/fax/8000a.htm) or any of a number of other helpful documents relating to the Soundscape.]

[Ensoniq – From what we can tell, the rapid playback problem has nothing to do with the sequencer software. Until we know more about the cause of this problem, try using the following workaround:

When you are recording in full duplex, the card needs two DMA channels (generally "0" and "1"). The rapid playback condition will occur when you are recording at 44.1kHz/16-bit. We have found in some cases that if you change the DMAs (to "1" and "3" for example) and record at something other than 44.1, the problem will go away. But this is not guaranteed to work in every case.

If you need additional information e-mail us at multimedia@ensoniq.com.]

––––––

TH –

My name is Shinya Saito. I'm a Japanese student.

The other day, I bought an Ensoniq EPS-16+ which was used, but there is no Operating System Disk.

So I want you to tell me the way to get the OS on the Internet.

Please send me e-mail: j4496041@ed.kagu.sut.ac.jp

Keiichi Saito

[CS – Download EPS-16 PLUS OS Version 1.30 (http://www.ensoniq.com/mid/binary/eps16os.zip), available at Ensoniq's web site.]

––––––

Hey,

I've got an ESQ-1 and would like to get a RAM cartridge for the cart A/B banks. Are there other models whose cartridges would work? VFX or SQ-80 or others?

Thanks,
Dave Fink

[TH – Syntaur Productions would be the ones to ask (they may even have the ESQ cartridges). They can be reached at 1-

713-682-1960.]

[CS – I'd stick to ESQ-1 carts, if I were you. Although the ESQ will recognize SQ-80 carts, some of the waves used in SQ-80 sounds don't exist in the ESQ. This means that some percentage of the programs from an SQ-80 cart won't sound right (and likely won't even sound musical) on an ESQ.

I'd guess that most of the companies that offered ESQ-1 cartridges in the past still do (at least, those companies that are still in business). A couple of other good bets might be Voice Crystal (http://www.voicecrystal.com/) (800-726-7664), or Sound Source Unlimited (they don't advertise with us, so I don't have contact information on file – check recent issues of other industry publications for an address or phone number).]

––––––

Dear Sir/Madam

Would like to know if anyone out there in Ensoniq land knows how to use the full extent of the EPS classic sequencer. I don't know much and the manual is terribly hopeless.

Regards,
Very Confused

[TH – I think if you could narrow your question down a little bit, it might help generate more useful answers. You use the phrase "full extent" so it sounds like you've got the basics down. Is there some specific area that's causing problems?

If it's a more general thing, there are some videos out there that may help. You're welcome to put an ad in our Classifieds (free, if it's just a couple sentences) to see if anyone has one they no longer need.]

[CS – You should also be able to get answers to specific questions (plus some good general info), by contacting Ensoniq Customer Service directly (610-6478-3930). You might also want to check out the listings in Transoniq-Net. These are a group of altruistic individuals who, for no more than the price of a phone call, will try to help you get the most out of your Ensoniq gear. Find them listed elsewhere in this issue. (Or any issue.)]

––––––

TH –

I have purchased an old VFX-synth. (I don't know how old it is but I would guess 10 years.) I use it with a SB-16 Soundcard and Musicator software. My problem is that the

drums sounds on the VFX are very limited – bass-drum, snare-drum, closed hi-hat, crash cymbals – and when I download a MIDI-file the drums don't "fit" the sounds from the VFX. What can I do to get more drum sounds into my VFX and make them compatible with the SB soundcard?

Fredrik Eide Nilsen

[CS – I'm afraid there's not that much that you can do. At the time the VFX was in production, most synths didn't have drum sounds, much less a standardized mapping, such as that used in your SB-16 Soundcard. The VFX can't accommodate more than five sounds in a single drumkit. Even if it could, there aren't many drum sounds in ROM, so trying to create a more robust kit is pretty much hopeless. And yeah, "10 years old" sounds about right to me (what a difference a decade makes...).]

[Ensoniq – Actually the VFX was released in 1989, so the oldest yours could be is seven years. Clark is correct – there are very few drum waves in the VFX so it isn't going to serve your drum kit needs very well. When we released the full-sequencer version of the VFX (the VFX-SD) we added a lot more drum waves to the ROM.]

––––––

TH –

I know it's been discussed for a while, but has there been any progress finding someone with a downloadable collection of the past Hackerpatches? (I'm personally interested in the SQ series patches.)

Bernie Fab

[TH – Well, so far, we've had zip luck in finding anyone who has these or can do this. Steve Vincent did this for all the ESQ/SQ-80 patches. We wish we could clone him.]

––––––

Hello guys,

I'm a reader of Transoniq and I enjoy it – your tips are very cool and useful. Thanks a lot for your support.

In the last edition I saw a guy looking for a sample of a CP-80 piano, and, of course, with his memory configuration it's very difficult. But I have a short version of this sound and it's very cool for memory usage.

Please publish this to let him know.

Bye!!!

Jo borges
joborges@mandic.com.br

———————

Greetings all!

I'm trying to find a patch editor/librarian for the Mac for an ESQ-R+. The ones I've found for ESQ-1's either don't communicate to my ESQ-R+ or don't work with OMS (only a Mac MidiManager).

I'd love to find anything that'll work, ideally with a "randomize" function for generating new banks.

Thanks!
Laird Sheldahl, Thine Eyes
laird@technical.powells.com

[CS – The two premiere Editor/librarian packages for the Mac would be Galaxy Plus Editors from Opcode, (http://www.op-code.com/) phone: (415) 856-3333, and Unisyn, from MOTU (Mark of the Unicorn), (http://www.motu.com/MOTUHome.html) phone: (617) 576-3066. I don't know if either company has an editor that'll work with your ESQ-R, but you might give it a shot (MOTU, in particular, seems committed to supporting Ensoniq products).]

———————

Hi there,

What is the best way transferring .WAV files to my ASR-10 with DOS/Windows PC? I prefer a MIDI connection instead of a dis-kette interchanging.

Thanks,
Christian Pesch

[CS – Keep in mind that MIDI transfers of sampled sounds are very slow; using a dis-kette is generally much faster.

You should be able to gather most of the info you need by checking out Garth Hjelte's ar-ticle "From Cyberspace to your Ear" (ftp://ftp.teleport.com/vendors/trnsoniq/), which originally appeared in TH Issues #122 (August '95) and #123 (September '95).]

———————

TH –

Does anyone have info regarding trustworthy stores dealing in used ASR-10 key or rack versions?

Winston Walker

[CS – I recommend getting that sort of gear

from a local dealer, if you can find one you're comfortable with. The service (access to sounds, troubleshooting, operating tips, etc.) you can get from a good dealer can be expensive to come by otherwise.]*

[TH – If you contact Ensoniq, they'll let you know who the Ensoniq dealers are in your area. That would be a start.]

———————

TH –

My EPS is going into obsolescence. I need to upgrade it, but Ensoniq seems to have stopped selling upgrades. Are there any 3rd party vendors who still do and does it make monetary sense?

Bryan Tan
Singapore

[CS – Ensoniq still supports the EPS. Upgrades still available include memory ex-panders, SCSI adapter kits, and the OEX-8 output expander.

Does it make monetary sense to upgrade? Well, if you're asking if upgrading your EPS will increase its resale value, sure it will. But the increase in resale value won't match the expense of the upgrade, so it would be unwise to upgrade for purely economic reasons. Still, if upgrading helps you get more use from your instrument, or more enjoyment, that may be reason enough to spend the money – but only you can make that decision.]

———————

TH –

I am having difficulty in understanding just what I need to do to establish the proper MIDI map for my KS-32 when operating with Band in the Box or Jazz Pianist pro-grams. The Ensoniq manual does not explain the procedure and the manuals for the B in the Box are somewhat vague. The NON-standard nature of the '32 is driving me crazy since I can't get the correct sounds to work whenever the programs are enabled. (The Bass follows the drum patterns and the selected sounds do not respond on the '32.) It's a messed up bunch of noise...

Any suggestions or possibly other books or videos that may clearly explain the way around this non-standard beast?

dibona@mail.sdsu.edu

[CS – In the version of Band-In-A-Box that I'm familiar with, all you need to do is specify the channel and program number for

each instrument you want to use in Band-In-A-Box's MIDI menu. Under the MIDI menu, select Midi Settings; the result-ing dialog box allows you to specify which MIDI channel is used for each sound, along with what program change number should be sent (to select the correct sound on your Key-board) and what octave range the sound should play in. As for the KS-32, you'll need to make sure that its MIDI Mode (MIDI In Mode) is set to MULTI (for multitimbral; find this in the System parameters), and that you've selected (or created) a sequence/preset in which each track is set to its own MIDI channel. The MIDI channel that you assign to a particular track will determine which Band-In-A-Box sound that track will play. If you assign consecutive channels to consecutive tracks (e.g., track one plays MIDI channel one, track two plays channel two, and so on), then you can easily assign Band-In-A-Box sounds to specific tracks just by specifying MIDI channel. For instance, if you set the Bass sound in Band-In-A-Box to play on MIDI channel four, then the bass part will be played on track four of the KS-32 sequence/preset.*

Finally, you'll need to tell Band-In-A-Box which MIDI note numbers play which drum sounds in your KS-32. First, determine which drum kit you want to use most of the time, and use Band-In-A-Box's Midi Settings page to assign the correct patch select number to select this drum kit via MIDI. Next, select Edit Drum Kit (Note Values)... from Band-In-A-Box's MIDI menu. This will produce a dialog box which lists the drum sounds that Band-In-A-Box uses, along with which MIDI note numbers those sounds cor-respond to. You'll need to change these num-bers to correspond to the note numbers used by the KS-32. For example, sending note number 38 from Band-In-A-Box to trigger a snare drum will probably won't trigger a snare drum sound. You'll probably need to change this number to something more like 45, perhaps.

Simpler yet, of course, would be to use the GM drum kit already programmed on the KS-32. The sounds in this kit should match those in Band-In-A-Box, although not all the Band-In-A-Box sounds might be represented.

You can get more specific information by contacting Ensoniq Customer Service (610-647-3930) directly. Also, other pertinent in-formation may be available, quickly and free of charge, via Ensoniq's fax-back system. Dial 1-800-257-1439 and follow the instruc-tions to receive a fax detailing what's avail-able.

And as for the Jazz Pianist program, I've never even heard of it...]

Hi,

Anyone know of a mixer map file for Cubase (PC) for the **DP/4+**? If you do please mail us c/o idlemedia.

Thanks,
Rob Endersby
idlemedia@idlemedia.co.uk
KILLER PRODUCTIONS, LONDON

[CS – Readers?...]

Hi there!

I am an owner of an ASR-10, TS-12 and a Mac with a CD-Recorder connected. So yesterday I tried to backup my Syquest Disks to a CD-R but all the tries failed.

I used scEPSi and GEAR to make it but it does not work! Is there any possibility of this working?

How can I mount an Ensoniq CD-R on a Macintosh??

Thanks for your attention!
Luigi

[CS – Ensoniq uses a Yamaha CDR-100 (4x CD-ROM burner) and Optical Media's Quick Topix software and I use a Ricoh-based burner with Toast software, but both systems function much the same way – and probably pretty much the way your system works, too. Without more detailed info, it's hard to tell where you might be going wrong, but here's the process in a nutshell:

First, create the disk you want to record to CD. For you, this is probably one of your Syquest cartridges. Or, you may want to combine information from two or more syquest carts onto a single, larger hard drive (you can store up to 650 megs of data on a 74 minute CD).

Next, connect your ASR disk (Syquest, hard drive, or whatever) to your Mac, and power up the Mac (the Mac will not be able to mount this disk, although it may try to; if it tries to auto-mount your Syquest cart, you may need to power up the Syquest after the Mac is up and running). Open Gear, and tell it you are going to burn a "device image" (not having used Gear, I can't tell you how this is done; in Toast, you specify "Generic CD-ROM" as the type of disk to make). From

here, you should be prompted for the source file, and the rest of the process is pretty trivial.

If this isn't enough information to get you going, either write again with more specific information on what you're doing, try getting in touch with Ensoniq Customer Support (http://www.ensoniq.com/contact.htm) directly, or give Ensoniq Customer Service (610-647-3930) a call, and go through the process with them.]

TH –

I just bought an **ASR-10** with a CD ROM drive a couple of weeks ago, with the intent to make dance music. A friend of mine recommended I buy an ASR-10 compatible dance CD-ROM as opposed to disks. Unfortunately, I haven't seen a great deal of CD-ROMS out there.

Does anyone know where I might be able to find some?

Raoul Fischer

[Eric N. Michaels – Raoul, you want to call Rob at East West Sounds. They have tons of ROMs that you can use for dance production. I recommend X-static Goldmine Issue 1. It has 4000+ sounds on one disc for $100.00!! That's half of what you will normally pay. Ph. @ 1-800-858-8797. Also check out my letter in this interface. traswave@dream-scape.com]

[CS – Ensoniq provides a catalog of all their CD-ROMs online at their web site (http://www.ensoniq.com/). Also, you might want to check with the big distributors – East-West Communications (800-833-8339), Voice Crystal (http://www.voicecrystal.com/) phone: (800-726-7664), Big Fish Audio (800-717-3474), Ilio Entertainments (http://www.ilio.com) phone: (818) 883-4546, and so on. And don't forget – not only can you use Ensoniq-formatted CDs, but also Roland and Akai CD-ROMs.]

[Ensoniq – For dance music we could recommend our CDR-5 and 15 L.A. Riot Vols. 1 and 2 – they're great collections of rap/hip-hop sounds. For synth sounds our CDR-14 Analog Digital Synth Resource is great. These volumes list at $199.95 each. A great bargain is our CDR-4 AS Archives, which lists for only $99.95. We're close to releasing the DJ Jazzy Jeff Signature Series CD-ROM (SCD-2), which will list for $249.95. We are currently working on a couple more dance-related titles, so stay

tuned to the Hacker for our release announcements!]

Hi there!

We just bought a Zipdrive to use with our **EPS-16+** sampler modified with a SCSI-port. Right now the SCSI-id is set to 5. Formatting a disk and saving the operating system works fine, but when we try to save sounds we get the error message: "Drive not responding." What could be wrong?

Is it possible to perform a so called "low-level format" from the EPS? (They talk about it in the Zip-drive manual.)

Help!!!
Thomas and Magnus,
Sweden

[CS – Garth Hjelte is becoming the recognized expert on Ensoniq products and Zip drives. Here's an excerpt from his article "Zip-ity How-To-Do Da," which appeared in Issue #132 (June, 1996) of TH:

Ensoniq does not have the ZipDrive "approved" for the 16-Plus. But that doesn't mean you can't make it work – with a little work.

The SCSI buss is a "powered" buss in the sense that it needs the data flowing in the correct voltages for the devices to recognize them. The termination buss line is one of these, and usually the initiator (in our case, the EPS/ASR/TS) supplies the correct power to the termination buss. Neither the 16-Plus nor the ASR nor the TS supply that power, unfortunately.

However, the ZipDrive is recognized and will interchange data (read and write) with the ASR and the TS perfectly, when it is hooked up as the only SCSI device in the chain, but the 16-Plus can recognize but not interchange data with a ZipDrive. That's a problem. But according to our experience, once you supply power to the termination buss from an external source (this can be another drive on the system, or an externally powered terminator such as the SCSI Sentry), the 16-Plus could conceivably MARRY the ZipDrive, and be recognized and interchange data. (That's a joke...) In more serious words, it works. Even though the ASR is does not supply term power, it seems to work with the ZipDrive because of its opto-isolation features (the apparent reason why it costs $150 more).]

Dear TH:

In your March issue Guy Miata, aka "The Maestro," was puzzled about program names like "CART-SND-18." If you're still wondering, Maestro, this happens when a cartridge patch is assigned to a track but on playback no storage cartridge is plugged in. What you will hear on that particular track is the patch in the corresponding location in internal RAM. Patch positions are numbered 0 to 59 so you will find the sound "behind" CART-SND-18 in the 19th location (upper left, patch bank 3) in internal RAM.

I've had my SD-1 for five years and expect to turn it on someday soon and find that the RAM battery needs to be replaced. It seems like a simple thing, but then, perhaps the order in which wires are disconnected and reconnected, etc., is very important. Can someone walk me through the procedure? Are the batteries hard to find?

I'm wondering if RAM sequence files can be fragmented. Can editing sequences by deleting and redoing tracks, appending, moving, etc., cause this to happen? Or is fragmenting only something that happens on disks?

This question is for Eric and Jane: has the story been told in some earlier issue about why and how you came to create the Transoniq *Hacker*? Are you performers or keyboard wizards? I'll bet there is a great story behind this unusual adventure.

I am puzzled about some changes on the new MR workstation. Don't get me wrong, I really like some of the new features. What I don't understand is why Ensoniq would decrease the size of the display screen by half. The functions of the MR predecessors were made easily accessible and a breeze to use because of the larger 6-item display, and

I can't for the world understand why that has been taken away from us. Too much wasted time (and inspiration) scrolling through endless subpages and trying to remember context, etc. Good-bye user-friendliness. Good-bye world famous sequencer.

I will also miss the patch select buttons enormously. I use them all the time for a variety of reasons. They provide a great deal of flexibility and this wonderful two-dollar piece of hardware should be elevated to an industry standard right up there with the pitch and mod wheels. I hope this function has been assigned to another easily accessible location or I am going to be very disappointed.

Can someone at Ensoniq explain why they have thrown away these unique features that, up until now, guaranteed my complete loyalty? Any chance we can get these back by popular demand? Just the loss of the larger display is enough to make me finally take a serious look at what the competition is doing.. C'mon Ensoniq, just what were you thinking?

Becky D
Lancaster, Texas

[Jane – The Hacker *started ...um.. about 12 years ago when Eric bullied me into getting a Mirage to use as a teaching tool with my piano students. His observation of what I went through made it quite clear that there was a cool-beans market for an independent source of information. That and his bloodlust for big bucks set the stage.]*

[Eric – Now, after 12 years, I'm starting to seriously question the "big bucks" aspect of this little project...]

[CS – Ensoniq recommends having your battery changed by an authorized technician. The battery is (if I'm not mistaken) soldered

to the main board, so changing it isn't as easy as one might assume. Whether or not Ensoniq would be willing to detail the procedure for you is unknown; guess you'll just have to ask them when the time comes. By the way – I've got keyboards with batteries that are over 10 years old – so you may not need to worry about this for some time yet.

Yes, I believe that sequencer RAM can become fragmented, but as far as I know, sequence RAM is handled efficiently, and will be "shuffled" occasionally (and automatically) to de-fragment as needed.]

[Ensoniq – The actual amount of information in our new display is not markedly less. We just don't have the six soft-buttons anymore. Truth be told, the plant that made that custom display was closed and we can no longer get them. Life goes on...

We believe that the new design of the MRs will be a step forward in spurring your creativity – give them a chance to come out and try them for a while. We're sure that we'll hear from you in these pages, either way :-)]

TH –

I've noticed that some computer peripherals are starting to appear that can be configured via a web browser. Any chance Ensoniq is going to produce a "Java Smart" synth/sampler that can be programmed via a browser? Those Java chips look mighty tasty...

Eric

[CS – Seems like a cool (though futuristic) idea. But what's the appropriate wine to go with Java chips?]

TRANSONIQ HACKER

1402 SW UPLAND DR., PORTLAND, OR 97221

ADDRESS CORRECTION REQUESTED

SUBSCRIPTION MATERIAL
DATED MATERIAL - TIME VALUE

Postmaster: Please return Form 3547 as soon as possible so
we can change our records. This is a monthly publication.