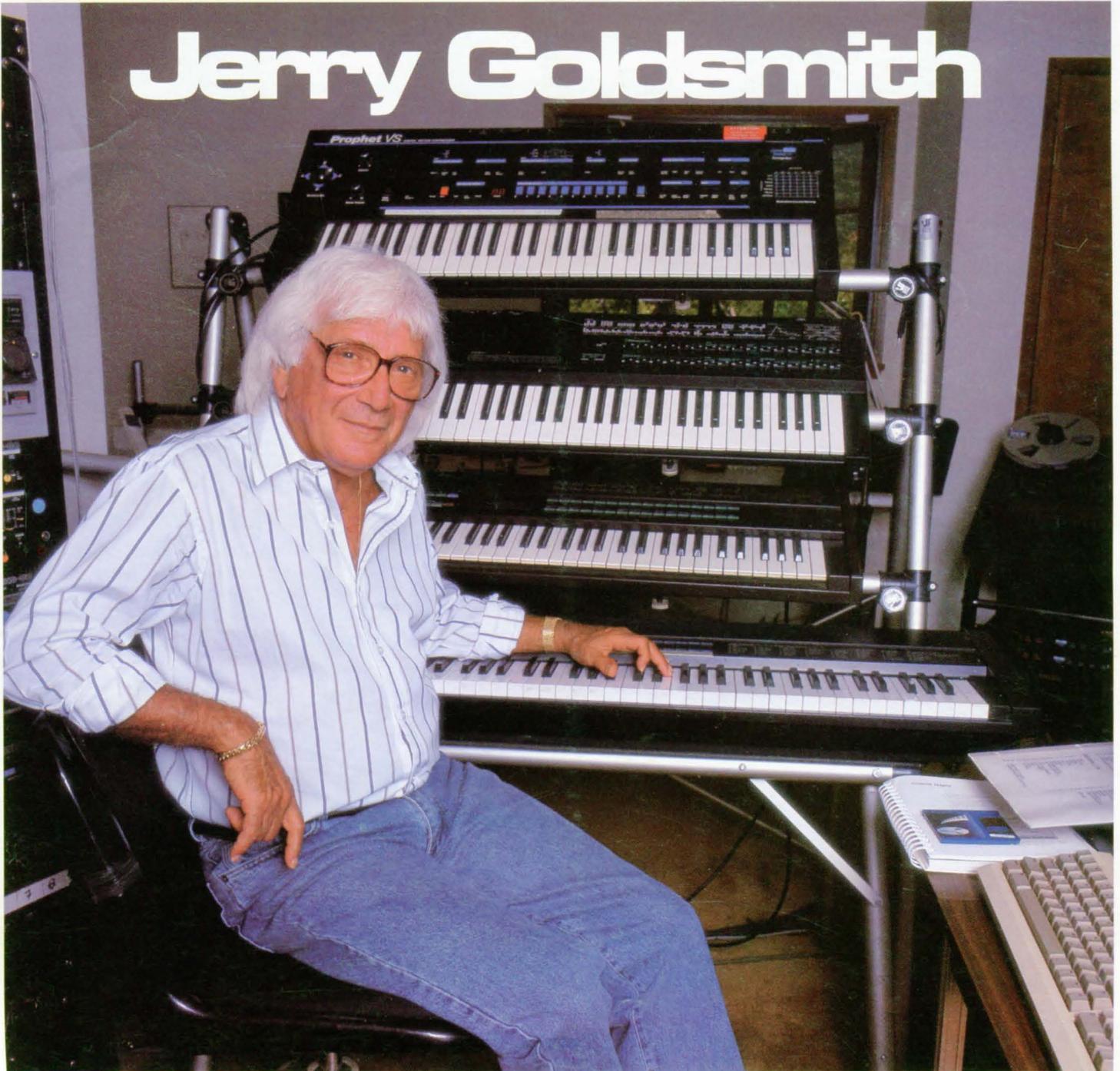


# YAMAHA

THE OFFICIAL PUBLICATION OF THE YAMAHA USERS GROUP

## Jerry Goldsmith



JUNE/JULY 1989

YAMAHA®

# News & Information

## SPECIAL NOTICE!

THE RESPONSIBILITY FOR this publication has been assumed by a new corporation. After this issue, AfterTouch will no longer be published by Yamaha Corporation of America.

An independent corporation, Tesladar Communications, has licensed the "AfterTouch" name and logo, and has been given control over the AfterTouch mailing list. Tesladar, headed by AfterTouch founding editor Tom Darter, took on this responsibility in order to maintain the flow of specific information between musicians that can only be provided by a users group magazine.

AfterTouch provides information that cannot be found in any other publication: material about past Yamaha instruments (including both the original DX7 and the DX7 II, the TX81Z, the SPX90, the FB-01, and many, many more), patches for both past and present Yamaha gear, and solid educational articles covering the basics of today's music industry.

Here are some important things that readers should know:

- If you write to AfterTouch, you are *not* writing to Yamaha. The production of AfterTouch is the responsibility of Tesladar.
- The majority of the contents in all future issues (as in this issue) will come from AfterTouch readers and other interested musicians. The focus of the magazine will be the information requested by and shared by its readers.
- In order to maintain AfterTouch, it is necessary for Tesladar to charge a modest subscription fee of \$12.00 for 12 issues.

If you want to continue receiving AfterTouch, send Tesladar your subscription request as soon as possible. For convenience, use the return envelope bound into this issue. (Make your checks payable to AfterTouch.)

By the way, the bound-in envelope says that paid subscriptions will begin with the July '89 issue—they were printed at a time when Yamaha's plan was to end its publishing involvement with two final issues: the special April/May '89 double issue and the June '89 issue. However, the changeover process took more time than expected, so Yamaha decided to continue its support through one more *double* issue, the June/July '89 issue you are now reading.

### Yamaha Bulletin Board

The Yamaha SGD Division has just opened a new computer Bulletin Board Service (BBS). The system operates at 1200 or 2400 baud, 24 hours a day, 7 days a week. This BBS will initially provide two conferences, a Dealer conference and a C1 conference.

All SGD dealers and C1 customers are welcome to call in from their computer (via modem) and register with the new service. After registration, please allow one week for authorization into the requested conference.

Dealers and C1 users will be able to download the most current C1 SDT (Software Develop-

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Neil Zlozower

*Cover photo: Jerry Goldsmith in his home studio with (bottom to top) Yamaha KX88, Yamaha DX7, Yamaha DX7 II FD, and Sequential Prophet VS.*

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AFTERTOUCH is published monthly. Third class postage paid at Long Prairie, MN and additional points of entry.  
**SUBSCRIPTIONS:** Address subscription correspondence to AFTERTOUCH, P.O. Box 7938, Northridge, CA 91327-7938. **POSTMASTER:** Send form 3579 to P.O. Box 7938, Northridge, CA 91327-7938.

# Questions & Answers

I recently purchased a DX21 second hand without any literature. How can I obtain an owners manual for this instrument?—Reginald R. McNair, Seat Pleasant, MD.

Contact your local authorized Yamaha dealer, and ask them to order a manual for you.

It would be good to get the *internal* piano sounds in the PF2000 so that they could be used with the DX7 II FD/D. Is this possible?—David Bieling, Tallahassee, FL.

No. First of all, even though the sounds in the PF2000 are 6-operator FM digital sounds, no patch charts exist. Secondly, the PF2000 neither sends nor receives MIDI System Exclusive data, so there is no way to send the instrument's internal patches to another unit.

I recently received a set of Demosoft (2.1 version) voices for the TX81Z/WX7 combination. After spending many dollars for third-party data cassette voices, I find that I am using only the Demosoft voices in my 2 TX81Zs. These voices and performances are awesome! Are there any future plans for Yamaha to put out additional sets of Demosoft voices for the TX81Z?—Mike Smiley, Hanford, CA 93230.

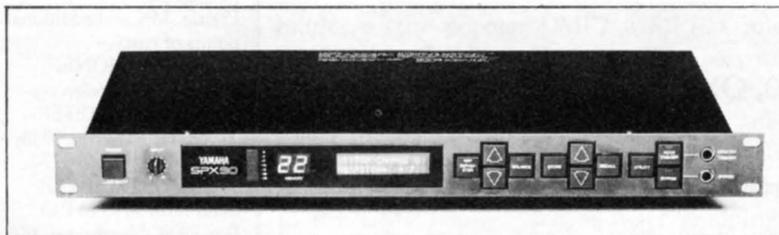
Unfortunately, the TX81Z has been discontinued, so Yamaha has no plans to create more voices for the unit.

I need to know the complete MIDI System Exclusive data for the Yamaha SPX90 (version 1). How can I obtain this information?—Tom Miller, Tempe, AZ.

For the answers to your SPX90 MIDI questions, contact the Yamaha Electronic Service Division's Product Information service: Call 1-800-854-3619, and ask for extension 1.

## Answers To Questions From Readers. By Steve Deming & Tom Darter.

SPX90 multi-effects processor.



Can you explain why the QX3 sequencer will not accept Oberheim System Exclusive information in the MDR mode? Whenever I attempt a complete memory data dump from a DMX drum machine or a Matrix 6, I get a "Buffer Full" message, although the QX3 does accept individual patterns from the DMX. What condition causes this error message in the QX3?—Gene Ryan, Orlando, FL.

Since the QX3 does accept single patterns from the DMX, compatibility does not seem to be the problem. The usual meaning of the "Buffer Full" message is that the material being sent is larger than the buffer of the QX3—all data must be accepted by the buffer before it can be stored to disk. However, the QX3's buffer in MDR mode is 480K, which is quite large. It is still possible that the bulk dumps from each of these two instruments is larger than the QX3's buffer; there is also the possibility that the bulk dump from these two instruments varies from the standard the QX3 looks for in some respects. For answers to these two questions, contact Oberheim to be sure.

I have played classical piano for almost 25 years. After trying out a Yamaha PF85 electronic piano, I decided to sell my acoustic piano and purchase a PF85. My only complaint with it is that when I'm performing complex works by Beethoven or Bach, the limitations of the 16-note polyphony become very apparent. Is it possible for me to double the unit's polyphony by adding a TX1P piano tone generator to my setup and playing them simultaneously via MIDI?—Dameon Likowski, North Royalton, OH.

If you simply buy a TX1P module and connect it to your PF85 via MIDI, both instruments will respond to each note played on the PF85's keyboard—you will have 16-note polyphony.

Yamaha (and other MIDI manufacturers) have created a MIDI function that will solve this problem. Called the Alternate Note function, it allows the user to set one MIDI instrument to respond *only* to odd MIDI note numbers, and to set the other to respond *only* to even MIDI note numbers. This makes it pos-

sible to set two instruments to the same MIDI channel and effectively double the available polyphony.

Unfortunately, the PF85 is not equipped with the Alternate Note feature. In order to double your polyphonic (simultaneous note capability), you need to use a MIDI processor like the Yamaha MEP4 MIDI event processor.

Therefore, it is possible to double the note capability of your PF85-based system, by adding an MEP4 and another tone generator such as the TX1P. Here is the basic procedure (explanation to follow):

- 1) Connect the MIDI OUT of the PF85 to the MIDI IN of the MEP4.
- 2) Connect the MIDI OUT 1 of the MEP4 to the MIDI IN of the PF85.
- 3) Connect the MIDI OUT 2 of the MEP4 to the MIDI IN of the TX1P.
- 4) Make sure that the PF85's MIDI send channel is the same as the MIDI receive channel of both the PF85 and the TX1P.
- 5) On the PF85, turn MIDI Local off.
- 6) Program the MEP4's Processor 1 to send only odd MIDI note numbers back to the PF85 (via its MIDI OUT 1 port).
- 7) Program the MEP4's Processor 2 to send only even MIDI note numbers to the TX1P (via its MIDI OUT 2 port).

In order for the PF85 and TX1P to work together, it is obvious that they have to be set to the same MIDI channel. When MIDI Local is turned off on the PF85, it means that the unit will not respond internally to any notes played on its keyboard; this allows you to send MIDI note data from the PF85 to the MEP4 for processing and back—since MIDI Local is off, the PF85 will respond only to the note data it receives via its MIDI IN port (from the MEP4). The procedures for programming the MEP4's processors to accomplish the tasks mentioned above are outlined in the unit's owners manual.

**I own a number of Yamaha products, and (needless to say) I support and believe in Yamaha products 300%! However, I am in need of a more complete and extensive publication regarding the operation of the QX3. The current owners manual is just too sparse and vague. Are there any books or publications that will help me?—Michael J. Walters, Peoria, IL.**

Shortly after the QX3 was introduced, Yamaha Corporation of America created a detailed Supplemental Booklet on the operation of the QX3



*QX3 digital sequence recorder.*

digital sequence recorder. Designed as a supplement to the QX3 owners manual, this document covers the unit in such detail that it can actually replace the existing manual.

The Booklet is divided into three basic sections. The first section, an owners manual supplement, includes material on the unit's Record/Play mode, a list of Record/Play Job Commands, a list of Edit Mode Job Commands, additional information on the Chain Edit and Other Functions sections in the original manual, a list of errors in the original manual (with corrections), and an appendix. Section two is a detailed applications guide, and Section Three is a useful Operations Map.

For more information on this QX3 Supplemental Booklet, write to: Yamaha Corporation of America, SGD Division, Literature Department, P.O. Box 6600, Buena Park, CA 90622-6600.

**I have a Yamaha DX7 II FD and have recently purchased a Korg M1. Can I dump and save the M1's internal sounds and RAM sounds onto the DX's disk drive?—Alfred Gross, Newton Centre, MA.**

The DX7 II FD has a Disk MDR mode that is set up to record MIDI data from another MIDI instrument (via the MIDI jacks on the back). However, the DX's buffer is only 20K in size, so if the bulk dump from the M1 is larger than 20K, this is not possible.

One way to increase the DX's buffer size is to install an E! kit for the DX7 II FD. The E! disk drive buffer is 128K in its MDR2 mode. For more information on E! for the DX7 II, contact the manufacturer, Grey Matter Response, at 1-408-427-3678.

*Continued on page 6*

# Q & A

Continued from page 5

I have read your Question & Answer column with interest. I have been particularly excited about the possibility of taking voices from other Yamaha 6-operator units and putting them into the memory of the DX7. Would it be possible to put the grand piano voice of the Clavinova (any in the series) into the memory of the DX7? If not, can you recommend a "grand piano" voice that sounds as good as the one found in the Clavinova series?—Marilyn A. Rebhahn, Marietta, OH.

There are many questions buried in this question, and we will do our best to answer as many of them as we can.

First of all, none of the Clavinova units are equipped to perform MIDI System Exclusive transfers, so it is impossible to transfer their grand piano patches to a DX7.

If what you want most is a realistic grand piano sound, you might want to consider adding the Yamaha EMT-10 to your system. The EMT-10 is a portable expander unit that features 10 AWM (sampled) voices, including piano.

Finally, it would be impossible for us to recommend a specific DX7 sound as the best "grand piano" patch. We have offered many such patches in our pages over the last few years, and we consider all of them to be usable. However, the judgement of any particular *sound* is a very subjective thing, and we would never want to ignore the value of our readers' ears by presenting any single patch as *the best* answer to a particular sound problem.

I have a TX81Z, which I've been using exclusively with my Guitar Controller, and it works great. Since I'll be working in a computerized duo, I want to know if I can leave the TX81Z in the Performance Edit mode (using the Mono 8 mode with MIDI channels 1 and 2 for bass and MIDI channels 3-8 for the guitar controller) and send individual patch or volume commands through a merger, so I won't have to write performances (which limits me to 24 possibilities)?—Jeff Guenther, Wayne, NJ.

Yes, this is possible. To set the unit to receive separate program changes for each instrument in a performance, go to the Single Utility mode and set the Program Change parameter to "individual." To set the unit to receive individual volume change messages, go to the Single Utility mode and set Control Change function "normal."

I hope you can help me with a problem I am having with my DX11 concerning the sustain pedal. The pedal works fine with voices such as piano and guitar. However, for voices with a zero sustain rate such as flute or violin, the notes just hold on and overlap as the melody is played. This should not be a problem, since the sustain pedal wouldn't normally be used with these voices.

The problem occurs when I try to set up a split keyboard performance sharing a piano and, say, a violin. I can't find any way to edit the voices or the performance so that the sustain pedal operates on the piano voice and not on the violin. Can you suggest anything?—Marvin Shapiro, Huntington, NY.

To understand one of the problems you are having, check the DX11's owners manual. It is impossible to give the sustain pedal two different assignments in performance mode. To quote the manual, page 14, under the heading Foot Switch Assign: "In performance mode, the first voice in the performance memory will determine the Foot Switch assignment." In other words, the Foot Switch setting in the first voice of any performance memory will establish the Foot Switch assignment for the entire performance.

Your other problem stems from a misunderstanding of the sustain pedal's function: To begin with, there is no "sustain" parameter in the DX11. The sustain pedal on the DX11 has a simpler function than the sustain pedal on a piano: it is the same as holding down a key on the DX11's keyboard. Therefore, if the voice has been created with a D2R of 0, the sound will continue as long as the pedal is held down.

This is a very important understanding: Most piano-like voices use the D2R setting to replicate the slow sound decay heard when notes played on a piano (or piano-like instrument) are held down or sustained using a pedal. Even if the key or the pedal is held down forever, the sound is programmed to decay slowly and finally disappear, because that is the nature of percussive sounds such as those created by a piano.

However, other voice types (such as flute or violin) have a very different D2R setting, because their sounds need to be continuous as long as they are engaged—as long as the key is held down. After all, the sound from a flute would not decay as long as the flautist can maintain his or her airstream, and the sound from a violin would not decay as long as the violinist



DX11 FM digital synthesizer.

continues to move the bow up and down. In other words, the D2R setting must be set to zero for sounds like these, so that the sound can be maintained as long as a key is held down. However, this setting also means that the sound will continue for as long as the sustain pedal is held down (since the sustain pedal has the same effect as holding down a key).

If you do not want the sustain pedal to have this effect on a voice like flute or violin, you must disable the effect of the sustain footswitch in the voice, as follows: 1) Set the FS Assign parameter to "portamento" (so that the footswitch is assigned to portamento, rather than sustain); 2) Set the Portamento Time parameter to 0, which turns portamento off. The footswitch will now have no effect on this voice.

I am using the YMM2 MIDI merge box to merge the MIDI outputs of my DX7 II FD and PF85 (both switched to Local Off) into the MIDI input of the QX5. The MIDI output of

the QX5 is run back to the MIDI input of the DX, and the DX MIDI thru is run to the MIDI input of the PF85. This works fine, except that at least once in every session the darn YMM2 locks up, the "buffer full" LED flashes, and the sound comes to a sudden stop. It doesn't matter how many or how few notes I am playing. Any suggestions?—C. Lloyd Gibson, Mount Hope, WV.

Your problem could be the particular YMM2 unit you have—the opto-isolator components are sometimes a problem with this particular unit. Take your YMM2 to your dealer, set up a similar MIDI system, and try it (to show the dealer the problem). Then switch with another YMM2 and try again. If the problem disappears, you will know that you have a faulty unit, and should look into making a trade.

Another potential problem could be the MIDI aftertouch messages, which the DX7 *always* sends. If you play with a lot of force, you could be sending a lot of unused (but still present) aftertouch data through your YMM2.



QX5 digital sequence recorder.

# TX81Z

## "Ad.Vibe.3." A New TX81Z Voice Larry Mueth.

### Notes:

In case you are wondering about the use of Algorithm #8, Waveform #1, and zero Feedback, I was experimenting with additive synthesis under the guise of FM.

### Notes:

This is a big fat lead voice with a great deal of harmonic change as you vary the pitch. I find it very pleasant all across the keyboard.

### A note on the diagrams:

In these diagrams, the numbers listed for both Frequency Coarse And Frequency Fine do not represent the actual settings—instead, they represent a value level within the range of values available for both parameters. To set each operator's frequency value properly, start with the Frequency Coarse and Frequency Fine both at their lowest possible values; then, go to Frequency Coarse and Increment the value the number of times indicated by the number in the chart; finally, go to Frequency Fine and increment the value the number of times indicated by the number in the chart. The actual frequency values for each operator are shown within dotted boxes at the bottom of the Operator portion of the chart.

OPERATOR	op1	op2	op3	op4
on/off	ON	ON	ON	ON
out level	99	94	84	86
freq. type	RTO	RTO	RTO	RTO
fix range	255	255	255	255
freq. coarse	4	13	34	56
freq. fine	0	1	1	0
detune	+0	-3	+2	+3
	-1-	-2-	-3-	-4-
	1.00	4.06	11.06	20.41

ENVELOPE	op1	op2	op3	op4
attack rate	26	26	25	24
decay 1 rate	5	6	5	6
decay 1 level	12	12	12	12
decay 2 rate	1	2	3	3
release rate	5	5	5	5
eg shift	OFF	OFF	OFF	OFF

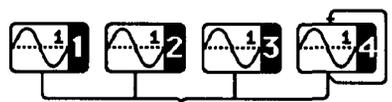
  

SCALING/SENS	op1	op2	op3	op4
rate	2	2	3	3
level	5	8	23	28
ams on/off	ON	ON	ON	OFF
sens eg bias	0	0	0	0
key vel	4	5	6	7

voice name: Ad.Vibe.3

algorithm no. 8  
 feedback 0



**LFO**  
 waveform  sync OFF  
 speed 18 delay 0  
 amp mod depth 0 sens 2  
 pitch mod depth 0 sens 0

**FUNCTION**  
 mode POLY mid C = C 3  
 portamento FULL rev rate 0  
 porta time 0 pb range 1  
 vol 99 pitch 0  
 pitch 0 amp 0  
 amp 0 eg bias 0  
 pitch 0 p bias +0  
 amp 65

OPERATOR	op1	op2	op3	op4
on/off	ON	ON	ON	ON
out level	99	85	78	69
freq. type	RTO	RTO	RTO	RTO
fix range	255	255	255	255
freq. coarse	4	0	8	32
freq. fine	8	4	4	1
detune	+1	-3	+2	-1
	-1-	-2-	-3-	-4-
	1.50	0.75	2.25	10.48

ENVELOPE	op1	op2	op3	op4
attack rate	25	19	10	4
decay 1 rate	3	9	13	12
decay 1 level	12	14	15	14
decay 2 rate	9	0	7	11
release rate	4	3	1	8
eg shift	OFF	OFF	OFF	OFF

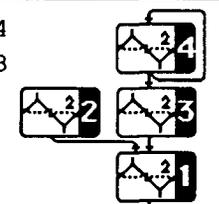
  

SCALING/SENS	op1	op2	op3	op4
rate	1	0	0	0
level	9	4	10	20
ams on/off	OFF	OFF	ON	ON
sens eg bias	1	2	4	3
key vel	3	3	3	4

voice name: Biotoxic

algorithm no. 4  
 feedback 3



**LFO**  
 waveform  sync OFF  
 speed 20 delay 0  
 amp mod depth 20 sens 2  
 pitch mod depth 0 sens 0

**FUNCTION**  
 mode POLY mid C = F 2  
 portamento FULL rev rate 4  
 porta time 0 pb range 0  
 vol 99 pitch 0  
 pitch 0 amp 0  
 amp 0 eg bias 60  
 pitch 0 p bias +0  
 amp 0

# TX81Z

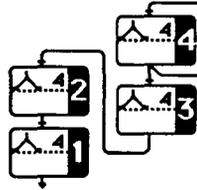
OPERATOR	op1	op2	op3	op4
on/off	ON	ON	ON	ON
out level	99	82	76	69
freq. type	RTO	RTO	RTO	RTO
fix range	255	255	255	255
freq. coarse	4	10	19	36
freq. fine	0	0	0	0
detune	+1	-1	-2	-3
-1-	-2-	-3-	-4-	
1.00	3.00	6.00	12.00	

ENVELOPE	op1	op2	op3	op4
attack rate	22	16	13	0
decay 1 rate	11	20	14	10
decay 1 level	12	15	15	15
decay 2 rate	8	5	4	3
release rate	6	1	2	5
eg shift	OFF	OFF	OFF	OFF

SCALING/SENS	op1	op2	op3	op4
rate	1	1	2	2
level	3	13	15	18
ams on/off	ON	ON	ON	ON
sens eg bias	1	2	3	4
key vel	2	2	2	3

voice name: Doublehi

algorithm no. 1  
feedback 4



### LFO

waveform sync ON  
speed 20 delay 0  
amp mod depth 17 sens 1  
pitch mod depth 0 sens 0

### FUNCTION

mode POLY mid C = C 3  
portamento FULL rev rate 4  
porta time 0 pb range 0  
vol 99 pitch 0  
pitch 0 amp 0  
amp 0 eg bias 60  
pitch 0 p bias +0  
amp 0

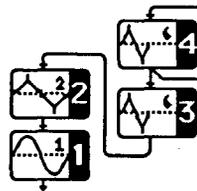
OPERATOR	op1	op2	op3	op4
on/off	ON	ON	ON	ON
out level	99	88	79	77
freq. type	RTO	RTO	RTO	RTO
fix range	255	255	255	255
freq. coarse	0	4	10	19
freq. fine	0	8	0	0
detune	+1	-1	-2	-3
-1-	-2-	-3-	-4-	
0.50	1.50	3.00	6.00	

ENVELOPE	op1	op2	op3	op4
attack rate	22	15	5	0
decay 1 rate	13	15	14	13
decay 1 level	15	15	15	14
decay 2 rate	12	5	4	3
release rate	7	1	2	5
eg shift	OFF	OFF	OFF	OFF

SCALING/SENS	op1	op2	op3	op4
rate	0	1	1	2
level	2	8	13	15
ams on/off	ON	ON	ON	ON
sens eg bias	1	2	3	4
key vel	2	2	2	3

voice name: Doublelo

algorithm no. 1  
feedback 4



### LFO

waveform sync ON  
speed 20 delay 0  
amp mod depth 17 sens 1  
pitch mod depth 0 sens 0

### FUNCTION

mode POLY mid C = C 3  
portamento FULL rev rate 4  
porta time 0 pb range 0  
vol 0 pitch 0  
pitch 0 amp 0  
amp 0 eg bias 60  
pitch 0 p bias +0  
amp 0

## "Doublehi," "Doublelo," & "Biotoxic." Three New TX81Z Voices By Gregory Thomas Meyers.

Notes:

This patch is designed to be used with "doublelo" in a split performance on the same MIDI channel. Together, the two voices sound very much like a stringed keyboard instrument, very delicate and rich, with some sweeping harmonics.

Notes:

This patch is designed to be used with "doublehi" in a split performance on the same MIDI channel. Together, the two voices sound very much like a stringed keyboard instrument, very delicate and rich, with some sweeping harmonics.

# DX7

## "Nightshade" & "felamevibe." Two New DX7 Voices By Gregory Thomas Meyers.

Notes:

The voice most resembles a combination plucked chime/bowed trombone sound.

Notes:

This voice is like a raspy mallet sound, but it allows the use of legato.

**Nightshade** alg 21

transpose C 3  
feedback 3  
sync OFF

pitch range 7  
step 0

key mode POLY  
porta mode RETAIN  
time 0

	range	pitch	amp	egb
mw	8	ON	OFF	OFF
fc	8	OFF	OFF	OFF
bc	8	OFF	OFF	ON
at	0	OFF	OFF	OFF

**LFO**

wave sync OFF pms 0  
speed 0 pmd 0 ams .....  
delay 0 amd 0 **1** 2 3 4 5 6  
mode SINGLE 0 0 0 0 0 0

**FREQUENCY/OUTPUT**

	on	out	freq	vel
	off	lev	crs fin mode	det sens
<b>1</b>	ON	95	0.50 x	- 1 4
<b>2</b>	ON	95	1.00 x	- 1 5
<b>3</b>	ON	88	3.00 x	+ 1 5
<b>4</b>	ON	85	0.50 x	+ 1 4
<b>5</b>	ON	87	1.585 Hz	+ 0 4
<b>6</b>	ON	98	0.50 x	+ 0 5

**SCALING**

	mode	left	break	right
		dep crv	point	crv dep
<b>1</b>	NORM	20	C 3	18
<b>2</b>	NORM	18	C 3	14
<b>3</b>	NORM	14	C 3	16
<b>4</b>	NORM	18	C 3	24
<b>5</b>	NORM	14	C 3	12
<b>6</b>	NORM	12	C 3	16

**ENVELOPE**

	R1	L1	R2	L2	R3	L3	R4	L4	RS
<b>1</b>	85	91	35	57	0	64	25	0	2
<b>2</b>	45	98	24	42	5	37	28	0	2
<b>3</b>	54	95	41	71	28	62	25	0	2
<b>4</b>	43	91	24	94	23	85	28	0	2
<b>5</b>	85	92	10	61	13	53	28	0	0
<b>6</b>	32	99	16	71	40	71	28	0	0
<b>P</b>	67	50	94	50	95	50	60	50	0

**Femalevibe** alg 7

transpose C 3  
feedback 7  
sync ON

pitch range 7  
step 0

key mode POLY  
porta mode RETAIN  
time 0

	range	pitch	amp	egb
mw	8	ON	OFF	OFF
fc	8	OFF	OFF	OFF
bc	15	OFF	OFF	ON
at	8	OFF	OFF	OFF

**LFO**

wave sync OFF pms 4  
speed 35 pmd 0 ams .....  
delay 35 amd 8 **1** 2 3 4 5 6  
mode SINGLE 0 2 0 2 0 1

**FREQUENCY/OUTPUT**

	on	out	freq	vel
	off	lev	crs fin mode	det sens
<b>1</b>	ON	99	1.00 x	- 1 2
<b>2</b>	ON	77	3.00 x	+ 1 3
<b>3</b>	ON	93	1.00 x	- 1 3
<b>4</b>	ON	92	0.50 x	- 1 2
<b>5</b>	ON	65	1.00 x	- 1 4
<b>6</b>	ON	76	5.10 x	+ 1 3

**SCALING**

	mode	left	break	right
		dep crv	point	crv dep
<b>1</b>	NORM	0	A-1	0
<b>2</b>	NORM	0	A-1	0
<b>3</b>	NORM	0	A-1	0
<b>4</b>	NORM	0	A-1	0
<b>5</b>	NORM	0	A-1	0
<b>6</b>	NORM	0	A-1	0

**ENVELOPE**

	R1	L1	R2	L2	R3	L3	R4	L4	RS
<b>1</b>	68	96	30	61	49	14	36	0	2
<b>2</b>	60	99	48	59	47	39	40	0	4
<b>3</b>	47	92	19	95	12	88	39	0	0
<b>4</b>	27	76	21	92	8	84	33	0	2
<b>5</b>	17	89	8	90	0	70	32	0	1
<b>6</b>	38	88	0	84	11	85	23	0	2
<b>P</b>	18	48	60	51	95	50	60	50	0

# DX7

“Dharma” &  
“Pop Bass.”  
Two New DX7  
Voices By Ben  
Moore.

**Dharma** alg 2

transpose C 1  
feedback 7  
sync OFF

pitch range 7  
step 0

key mode POLY  
porta mode RETAIN  
time 0

	range	pitch	amp	egb
mw	0	OFF	OFF	OFF
fc	0	OFF	OFF	OFF
bc	0	OFF	OFF	OFF
at	0	OFF	OFF	OFF

**LFO**

wave sync ON pms 1  
speed 35 pmd 11 ams .....  
delay 0 amd 0 1 2 3 4 5 6  
mode SINGLE 0 0 0 0 0 0

**FREQUENCY/OUTPUT**

	on	out	freq	vel
	off	lev	crs fin mode	det sens
1	ON	99	1.259 Hz	- 1 2
2	ON	76	2.00 x	- 7 0
3	ON	99	2.00 x	- 1 7
4	ON	87	2.00 x	+ 0 0
5	ON	77	8.00 x	+ 3 2
6	ON	44	2042 Hz	+ 5 0

**SCALING**

	mode	left dep	left crv	break point	right crv	right dep
1	NORM	0	\	B-1	\	0
2	NORM	1	\	C 3	\	6
3	NORM	0	\	B-1	\	0
4	NORM	4	\	G 3	\	13
5	NORM	0	\	B 2	\	22
6	NORM	0	\	F#2	\	45

**ENVELOPE**

	R1	L1	R2	L2	R3	L3	R4	L4	RS
1	99	99	25	97	22	86	45	0	4
2	99	99	0	98	0	97	30	0	1
3	99	99	18	95	17	92	55	0	2
4	99	99	30	98	0	90	35	0	3
5	99	99	49	90	55	80	45	0	2
6	99	99	42	99	50	99	60	0	0
P	87	50	94	50	0	50	0	50	0

Notes:

This is the most useful sound I have in my collection. It is a percussive string sound that can be used for just about anything.

**Pop Bass** alg 17

transpose C 2  
feedback 7  
sync ON

pitch range 7  
step 0

key mode POLY  
porta mode RETAIN  
time 0

	range	pitch	amp	egb
mw	0	OFF	OFF	OFF
fc	0	OFF	OFF	OFF
bc	0	OFF	OFF	OFF
at	0	OFF	OFF	OFF

**LFO**

wave sync ON pms 3  
speed 35 pmd 0 ams .....  
delay 0 amd 0 1 2 3 4 5 6  
mode SINGLE 0 0 0 0 0 0

**FREQUENCY/OUTPUT**

	on	out	freq	vel
	off	lev	crs fin mode	det sens
1	ON	99	1.00 x	+ 2 2
2	ON	69	3.00 x	+ 5 5
3	ON	75	0.50 x	+ 0 0
4	ON	63	9.00 x	- 1 7
5	ON	70	9.00 x	+ 0 7
6	ON	94	6.00 x	+ 0 5

**SCALING**

	mode	left dep	left crv	break point	right crv	right dep
1	NORM	0	/	A-1	/	0
2	NORM	0	/	A-1	/	0
3	NORM	0	/	A-1	/	0
4	NORM	0	/	A-1	/	0
5	NORM	0	/	A-1	/	0
6	NORM	0	/	A-1	/	0

**ENVELOPE**

	R1	L1	R2	L2	R3	L3	R4	L4	RS
1	99	99	64	86	33	0	71	0	0
2	59	99	99	86	22	0	71	0	5
3	59	99	99	99	99	99	71	0	5
4	59	99	99	99	41	0	71	0	5
5	99	99	99	99	38	0	99	0	5
6	99	99	99	99	62	0	99	0	4
P	99	50	99	50	99	50	99	50	0

Notes:

This bass sound offers great dynamics for up-beat rock music.

These DX7 voices can also be loaded into all of the other Yamaha 6-operator FM digital synthesizers and tone generators, including the DX7 II FD/D, DX7s, TX802, TX816, TX216, TF1, TX7, DX5, and DX1.

# RX5

## A New RX5 Voice Edit By Douglas Masla.

Notes:

*This is a rich, deep, mallet-struck gong sound. Use hall reverb to taste.*

Voice Name: <b>Large Gong</b>		Origin: <b>Int-China</b>	
<b>JOB #</b>	<b>PARAMETER</b>	<b>RANGE</b>	<b>NEW VALUE</b>
02	<b>Pitch</b>	-3600 ~ 2400	<b>-1700 cent</b>
03/1	<b>Attack Rate</b>	1 ~ 99	<b>92</b>
03/2	<b>Decay 1 Rate</b>	1 ~ 99	<b>49</b>
03/3	<b>Decay 1 Level</b>	1 ~ 60	<b>60</b>
03/4	<b>Decay 2 Rate</b>	1 ~ 99	<b>33</b>
03/5	<b>Release Rate</b>	1 ~ 99	<b>44</b>
03/6	<b>Gate Time</b>	100 ~ 6500	<b>3200 ms</b>
04/1	<b>Bend Rate</b>	1 ~ 60	<b>01</b>
04/2	<b>Bend Range</b>	-60 ~ +60	<b>+01</b>
05	<b>Inst Level</b>	0 ~ 31	<b>30</b>
06	<b>Sound Loop</b>	OFF / ON	<b>ON</b>

# RX5

## A New RX5 Voice Edit By Douglas Masla.

Notes:

*This sound is good for use in Indian or Far-Eastern ethnic moods.*

Voice Name: <b>Chant Cymbal</b>		Origin: <b>Int-Edge</b>	
<b>JOB #</b>	<b>PARAMETER</b>	<b>RANGE</b>	<b>NEW VALUE</b>
02	<b>Pitch</b>	-3600 ~ 2400	<b>+1000 cent</b>
03/1	<b>Attack Rate</b>	1 ~ 99	<b>99</b>
03/2	<b>Decay 1 Rate</b>	1 ~ 99	<b>45</b>
03/3	<b>Decay 1 Level</b>	1 ~ 60	<b>50</b>
03/4	<b>Decay 2 Rate</b>	1 ~ 99	<b>37</b>
03/5	<b>Release Rate</b>	1 ~ 99	<b>60</b>
03/6	<b>Gate Time</b>	100 ~ 6500	<b>1500 ms</b>
04/1	<b>Bend Rate</b>	1 ~ 60	<b>21</b>
04/2	<b>Bend Range</b>	-60 ~ +60	<b>-10</b>
05	<b>Inst Level</b>	0 ~ 31	<b>30</b>
06	<b>Sound Loop</b>	OFF / ON	<b>ON</b>

# RX5

A New RX5  
Voice Edit By  
Douglas Masla.

Notes:

*This is a strong, punchy,  
electronic snare timbre.*

Voice Name: <b>Gliss Snare</b>		Origin: <b>Int-SD 2</b>	
<b>JOB #</b>	<b>PARAMETER</b>	<b>RANGE</b>	<b>NEW VALUE</b>
02	<b>Pitch</b>	-3600 ~ 2400	<b>0 cent</b>
03/1	<b>Attack Rate</b>	1 ~ 99	<b>99</b>
03/2	<b>Decay 1 Rate</b>	1 ~ 99	<b>37</b>
03/3	<b>Decay 1 Level</b>	1 ~ 60	<b>57</b>
03/4	<b>Decay 2 Rate</b>	1 ~ 99	<b>53</b>
03/5	<b>Release Rate</b>	1 ~ 99	<b>59</b>
03/6	<b>Gate Time</b>	100 ~ 6500	<b>750 ms</b>
04/1	<b>Bend Rate</b>	1 ~ 60	<b>02</b>
04/2	<b>Bend Range</b>	-60 ~ +60	<b>+08</b>
05	<b>Inst Level</b>	0 ~ 31	<b>28</b>
06	<b>Sound Loop</b>	OFF / ON	<b>ON</b>

# RX5

A New RX5  
Voice Edit By  
Douglas Masla.

Notes:

*This is a good accent snare  
sound for rock and dance  
grooves.*

Voice Name: <b>Eko Snare</b>		Origin: <b>Int-SD 1</b>	
<b>JOB #</b>	<b>PARAMETER</b>	<b>RANGE</b>	<b>NEW VALUE</b>
02	<b>Pitch</b>	-3600 ~ 2400	<b>-100 cent</b>
03/1	<b>Attack Rate</b>	1 ~ 99	<b>99</b>
03/2	<b>Decay 1 Rate</b>	1 ~ 99	<b>45</b>
03/3	<b>Decay 1 Level</b>	1 ~ 60	<b>39</b>
03/4	<b>Decay 2 Rate</b>	1 ~ 99	<b>75</b>
03/5	<b>Release Rate</b>	1 ~ 99	<b>60</b>
03/6	<b>Gate Time</b>	100 ~ 6500	<b>1100 ms</b>
04/1	<b>Bend Rate</b>	1 ~ 60	<b>00</b>
04/2	<b>Bend Range</b>	-60 ~ +60	<b>00</b>
05	<b>Inst Level</b>	0 ~ 31	<b>28</b>
06	<b>Sound Loop</b>	OFF / ON	<b>ON</b>

# New Yamaha Products

THE PROFESSIONAL AUDIO DIVISION of Yamaha Corporation of America introduced a number of innovative new products at this year's summer NAMM (National Association of Music Merchants) trade show in Chicago, Illinois. Here is a rundown on some of the major new products they introduced. In upcoming issues, we will introduce more of these new products, and provide more detailed information on the products outlined below.

## MV1602 Rack-Mount Audio Mixer

The MV1602 is a compact, rack-mount, 16-input mixer designed for today's high-density, high performance audio applications. This unit, with 16 inputs in one rack-mount package, offers Yamaha quality and performance at a very affordable price.

The MV1602 has two mixing busses and four Aux sends from each input. Inputs one through eight are mic/line selectable with adjustable gain, and have a peak overload LED for optimum level matching. Inputs nine through fourteen are line inputs with a 20 dB pad, switch selectable. Inputs fifteen and sixteen are ganged stereo inputs, making this, for all practical purposes, an 18-input mixer.

Four auxiliary send controls on each input provide for a wide range of effects and/or sub-mixes. The two Aux returns are stereo for compatibility with today's advanced signal processors. The MV1602 has LED meters for precise level setting, and clip indicators for the stereo busses. The main outputs are provided via

## A Quick Look At Some Of Yamaha's Major New Pro Audio Products.

both balanced XLR and unbalanced 1/4-inch phone jacks.

The MV1602 will be available from Yamaha Professional Audio dealers in the fall of 1989, at a suggested retail price of \$1,195.00. For more information, write to: Yamaha Corporation of America, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622-6600.

## AM602 Audio Mixer

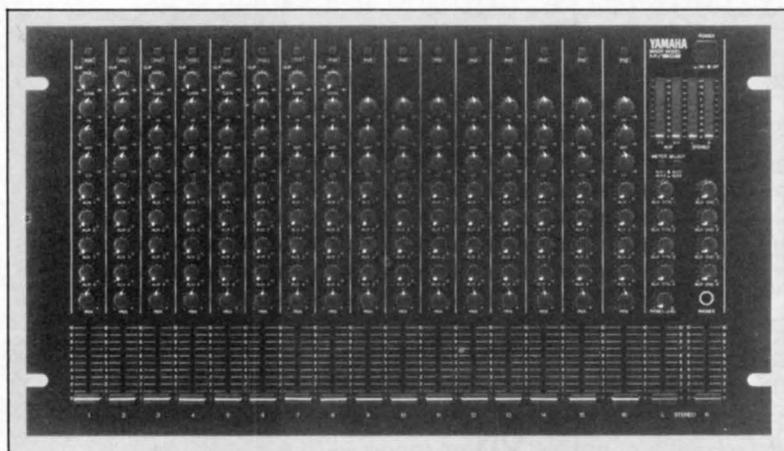
The AM602 is a compact, versatile, six-input mixer designed for many audio applications. The unit has provisions for a separate sub mix in/out for advanced signal processing.

The AM602 has two mixing busses and two Aux sends from each input. Inputs one and two are mic/line selectable with adjustable gain for optimum level matching, and include two-stage + 15 dB boost/cut equalizers.

Two auxiliary send controls on each input provide for a wide range of effects and/or sub mixes. The AM602 features a sub mix in/out and an additional Aux mix in/out. The sub mix output, when selected, is taken from the stereo buss Aux. The additional Aux mix is taken from the Aux send mix. The sub input, when selected, is sent to the Aux return system, post fader, and the additional Aux input is sent to the Aux busses. The AM602 has LED meters for precise level setting and clip indicators for the stereo busses.

The AM602 is now available from Yamaha Professional Audio dealers, at a suggested retail price of \$295.00. For more information, write to: Yamaha Corporation of America, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622-6600.

MV1602 rack-mount audio mixer.



## MR Series Professional Mixing Consoles

Available in three different input configurations, the MR Series of professional mixing consoles is ideal for recording studio use, sound production applications, and small sound reinforcement needs.

Each MR Series console has four mixing busses and a stereo master buss. They are available as the MR842 (with 8 inputs), the MR1242 (with 23 inputs), or the MR1642 (with 16

inputs). Each input features a choice of electronically balanced, low-impedance, XLR-type inputs, or balanced TRS phone jack inputs, with built-in phantom power for condenser microphones. Each input feeds a switchable 20 dB pad to a continuously variable input trim control to a precision long-throw fader.

Each channel includes a three-band equalizer with +15 dB boost/cut and sweepable peaking mid-frequency control. There is a pre-EQ insert jack, and each input has a peak overload indicator for precise level optimization.

The Group assign switches and Pan Pot on each input allow signals to be either independently assigned to one of the program busses or simultaneously panned between busses 1 & 2, 3 & 4, or L & R. Three auxiliary send controls on each input provide for a wide range of effects and/or sub mixes. The Aux 3 send is selectable to the pre-EQ channel or the channel's tape input, facilitating an independent multitrack mix, while leaving channel faders open for input sources. The Aux 3 mix can be selected for monitoring at the Monitor Out jacks.

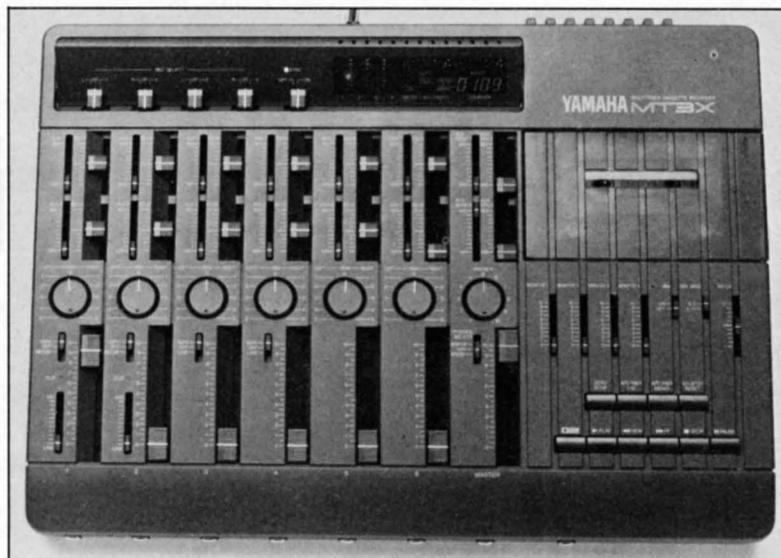
The four group modules include three master Aux Send controls and dual-stereo Aux Return level controls with assignment switches to the four group and master stereo busses. The MR Series consoles have complete monitoring assign and talkback facilities. Illuminated precision VU meters are switchable to provide accurate monitoring of program, Aux Cue, and stereo mixing busses. Each meter includes a peak overload LED indicator for accurate and safe level control.

The MR Series is now available from Yamaha Professional Audio dealers. Suggested retail prices are: MR842—\$1,295.00, MR1242—\$1,595.00, and MR1642—\$1,895.00. For more information, write to: Yamaha Corporation of America, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622-6600.

### MT3X Multitrack Cassette Recorder/Mixer

The MT3X is a flexible six-input mixer designed with studio-like features and a precision quality four-track cassette transport built in. With enhanced monitoring and versatile signal routing capabilities, the MT3X is the most advanced recorder/mixer ever offered by Yamaha.

The MT3X has six input channels, each with two auxiliary sends, L/R pan control, and two-band equalizer. All inputs accept line level, while inputs 1 and 2 are also selectable for mic level. Tape can be selected on inputs 1 through 4. The MT3X features a comprehensive meter



MT3X multitrack cassette recorder/mixer.

section with individual 12-segment LED meters for each of the four recorder tracks, two 19-segments LED meters in an arced VU-like display for the two Master Stereo busses, a digital electronic tape counter display, and additional status indicators for its unique auto punch-in feature.

The 4-track recorder has two speeds, a Permalloy Record/Play head, and a ferrite Erase head. It has a  $\pm 10\%$  pitch control and a switch-selectable dbx noise reduction system. The MT3X features a comprehensive counter system with a programmable Auto Punch In/Out capability. This feature, coupled with the unit's Punch In/Out footswitch jack, allows for hands-free punch in while overdubbing. The MT3X will record up to four tracks simultaneously or individually in any combination. At the faster record speed (9.5 cm/sec), the MT3X delivers high frequency playback response of up to 18 kHz.

There is a comprehensive signal processing routing system in the MT3X. In addition to the two Aux Send controls on each input, there are two master Aux send faders for level matching and level optimization. The return system is stereo with a master level control before the mix buss. Four monitor level controls facilitate a monitor mix of recorder tracks, leaving channel faders free to accommodate input sources. The Phones Select switch selects monitoring of either the tape monitor mix, the stereo buss mix, or both simultaneously. This combined system is available via the Monitor Out L&R jacks. The stereo buss mix is available via separate Stereo L&R mix-down outputs.

A special feature of the MT3X is the sync selector circuit. With this, the user can assign track 4 to the optional YMC10 MIDI converter or to an external MIDI converter. With this

*Continued on page 34*

# V50 Applications

**Understanding  
The V50  
Sequencer,  
Part 2. By  
Chris Cotton.**

**T**HIS IS THE SECOND in a series of articles designed to help you make the best use of the sequencer section of the Yamaha V50 synthesizer. In part 1 (April/May '89, page 15), we provided a quick overview of the V50's sequencer, explained how tracks work, and talked about how to use V50 performances in conjunction with the sequencer. This time, we will cover the use of the V50's Rhythm section in conjunction with the sequencer, and will look at the concept of "tracks" in more detail.

## Using the Rhythm Section

The Rhythm section of the V50 (the "Rhythm Track") is totally independent of the 8 sequencer tracks. When in the Sequencer mode, the Rhythm track will play whatever pattern or song the Rhythm section is currently set at.

To see how this works, enter the Rhythm section by pressing the Rhythm System button, and choose a pattern in the pattern mode. Then enter the Sequencer section by pressing the Sequencer (SEQ) System button, and select a sequenced song to play. The pattern chosen in the Rhythm section will play along as the rhythm track to that song. If you select a song in the Rhythm mode, that song will become the Rhythm track for the sequence. It is possible to save a sequence with a specific Setup, so that whenever that sequence is selected, it will also select the accompanying Rhythm song and Performance voices automatically.

Usually it is a good idea to start by using a simple pattern as a rhythm reference while you are recording your sequencer tracks. For instance, if the Rhythm section is set to a basic 1- or 2-bar pattern that contains a simple kick, snare, and hi-hat part similar to the type needed for the song, this pattern will loop continuously while you record the instrument tracks into the sequencer. This way, you will have a basic reference rhythm track to use. You can then create the actual finished rhythm part after you record the instrument parts. Generally, this is a good method to use, because it is easier to create the rhythm part later in the sequencing process, to make sure it works well with the actual instrument parts in the sequence.

The V50 contains 100 preset rhythm patterns in its internal memory. Each of these patterns is 1 or 2 bars long in either 4/4 or 3/4 time. The preset patterns range in style from basic back beats to more elaborate feels with percussion instruments and fills. In addition to the 100 preset rhythm patterns, the V50 is also capable of storing up to 100 programmable rhythm patterns in its internal memory area. This makes it possible to copy a preset pattern to an internal pattern location and then edit or modify it for your needs. You may also create rhythm patterns in the internal memory from scratch. This means that you can create your own rhythm patterns in blank internal pattern locations.

A blank pattern can be set to be from 1 to 4 bars long, in almost any time signature (1/4 through 8/4, 1/8 through 16/8, and 1/16 through 32/16). This makes it possible to create very specific rhythm parts that can be as elaborate as you like. Once again, even if you will be creating an original Rhythm song to go with your sequence, it is often easiest to begin sequencing using a basic reference pattern, and program the actual rhythm part later in the sequencing process.

To select a rhythm pattern from the V50 rhythm section, follow these steps:

- 1) Enter the V50 Rhythm section by pressing the Rhythm System button.
- 2) Make sure you are in the Pattern (PTN) Play mode by using the 1st LCD Function button.
- 3) Press the 3rd LCD Function button in order to select a pattern.
- 4) Press the Preset or Internal (INT) memory button to select a pattern memory area.
- 5) Select a pattern number (00-99) using the numeric keypad or the +1/-1 buttons.
- 6) Press the Play button to listen to the selected pattern.

It is also possible to select another pattern while the currently selected pattern is playing. This is done simply by selecting another pattern number (using the numeric keypad or the +1/-1 buttons) while the current pattern is playing. When the current pattern reaches its end,



V50 digital synthesizer.

the next selected pattern will pick up right where the last pattern ended—without breaking the time. This is often a quicker way to locate a rhythm pattern to begin sequencing with, since you don't need to start and stop each pattern as you listen to them.

Also, be aware that it is possible to use sequencer tracks for rhythm parts. This can be a very useful sequencing technique when it is utilized properly. The basic idea is to use the Rhythm programmer section of the V50 for simple repetitive parts such as the kick and snare drum, and use a sequencer track for more diversified and sporadic parts such as tom fills and percussion. When a sequencer track is used to record rhythm parts, all 61 of the PCM drum and percussion sounds can be accessed on the V50 keyboard. Parts can be recorded in real-time or step-time, and as many parts as necessary can be overdubbed on a single sequencer track.

### What Is A Track?

As mentioned in part 1 of this series of articles, each sequence or "song" on the V50 can have up to 8 tracks. Each track is generally an individual instrument part such as a bass line, a piano part, a horn section, or a string section.

Each instrument will use one of the V50's single voices. Single voices can be chosen from the 100 preset, 100 internal, or 100 card voices (if you have an alternate voice card). The eight instrument slots in the SEQ8 performance will contain the voice numbers and the settings for each of these individual instruments as you create parts. Remember from the graphic example given last issue that every instrument does not play all of the time during a sequence. Whenever an instrument is not playing it is possible to change the instrument voice on that track temporarily (or permanently) to play another part. For instance, if track 3 is the horn section part

and the horns are only used in the choruses of the song, then the horn voice could be changed to a string section sound for each of the verses and the bridge, and then be switched back to the horn section for the choruses. This way you can consolidate two or more potentially different tracks into one, and open up entire tracks for other instrument parts.

Utilizing patch changes within tracks can consolidate instrument parts that are sparse or sporadic. This will open up track space for more instruments in the song, making it possible to create more elaborate arrangements. It is possible to use this method of track efficiency for very complex instrumentation. A single track can utilize dozens of different voices, with patch changes every couple of measures (or even beats). With up to 300 single voices available in the V50, the possibilities are virtually endless. However, don't try to get too sophisticated too quickly: With 8 instrument tracks, it is possible to create very full arrangements without using any patch changes at all. Tracks can always be mixed and consolidated after they have been recorded, using the V50's Track Edit features. Therefore, unless you anticipate very elaborate instrumentation in your sequence, it is usually easiest to start by recording each instrument part on a separate track.

♦ ♦ ♦ ♦

Next issue, we will continue our series on the use of the V50 sequencer, and will cover the creation of tracks in real time and use of the V50's Track Edit features.

# Jerry Goldsmith

JERRY GOLDSMITH IS one of the most well-known and most respected film score composers in the world. During a career now well into its fourth decade, Jerry has scored more than 130 feature films (plus countless TV and radio episodes).

One hallmark of Goldsmith's approach to film music composition is his constant search for new and unusual sounds. This led him to explore the possibilities of electronic music very early in his career. He continues this involvement in the MIDI age by consistently including two to four MIDI keyboardists as part of his orchestra. Still, as you will see from the interview below, his attitudes have changed quite a bit in the last few years.

Another hallmark of Goldsmith's approach is that he keeps changing—he actively resists imitating himself (or anyone else). As a journalist, I have enjoyed the opportunity to interview Jerry Goldsmith more than once. And, as a musician, I have appreciated the opportunity to work as a keyboardist in his orchestra on a number of film scoring sessions (including *Explorers*, *Poltergeist II*, *Innerspace*, *The 'burbs*, and *Star Trek V*). In all of my encounters with Jerry, he has proven to be a musician whose attitudes and approaches constantly change. The interview below reflects the current state of Jerry's art.

....

*Let's talk about Star Trek V. That's the first time you used the C1 music computer with the Auricle program, isn't it?*

Yes.

*Had you ever used the Auricle program before?*

Oh, yeah; I used the original version, on the Commodore 64. It was amazing that it was all programmed in 64K. The first film I used it on was *Extreme Prejudice*, two and a half years ago. I've used Auricle ever since. It's a real godsend. It's great for working out timings, laying out cues, providing clicks—everything. And on the C1, the whole program is much richer, because it's written for a computer that has much more memory, and is much faster.

*So, the main advantage of using Auricle on the C1 is that you can do a lot more things, because there's more inside the computer?*

## A Film Scoring Legend Talks About His Use Of Synthesizers, His Work With The C1, And His Score For Star Trek V. By Tom Darter.

Well, there's more memory, so the program has more bells and whistles on it. And it is a lot faster. In the original program for the Commodore, it would only go up to 84 bars, and then you had to go to "Page 1," and it would go another 84 bars, so you were left with no more than 168 bars per cue.

*And that was it?*

And that was it. That's fine if you're doing a television show. When you're doing a feature, where a cue could run 200-250 bars, it's a problem. I think the new program has some 300-odd bars per cue. Also, because of the meters that I do—the  $\frac{7}{8}$  bars, the  $\frac{5}{8}$  bars—you couldn't get proper clicks on the original program for the Commodore.

*You couldn't?*

No. You could create a  $\frac{7}{8}$  bar, but if you wanted to use the clicks, they would come out as, say, seven eighth-notes in a  $\frac{7}{8}$  bar, which is impossible. As you well know, I always divide irregular bars up, and reflect that in the clicks: a  $\frac{5}{8}$  bar might be divided 1-2, 1-2-3, and the clicks should reflect that—x-o-x-x-x; or a  $\frac{7}{8}$  might be divided 1-2, 3-4, 5-6-7, and the clicks should reflect that—x-o-x-o-x-x-x. With the program on the C1, you can do that. On the Commodore, to get the same click, you had to break a  $\frac{7}{8}$  bar down to a  $\frac{3}{4}$  bar plus a  $\frac{1}{8}$  bar, which meant that you were using up twice as much space: If I had a cue in  $\frac{7}{8}$ , 42 bars of the actual music would use up 84 bars of the program's space per cue.

*That must have been a real problem sometimes. I remember that Innerspace had a number of cues that involved irregular meters.*

Yeah, that was a real nightmare. Another problem with that setup was, if I wanted to check the timing in a certain bar, the bar number in my music didn't necessarily correspond to the bar number in the program. But now, with the Auricle program running on the C1, that's all been resolved. On *Star Trek V*, I had to deal with a scene that had been re-edited, and I was able, on the soundstage with the aid of the computer, to remake the cue—to make cuts and changes in the music, and figure out where certain events had to land. I would say it took me ten minutes to redo a two-minute cue. If I'd had to do it longhand, it would have been impos-

sible. The computer told me how to redo it, and then I just dictated the changes to the orchestra, and the clicks were already redone.

On *Star Trek V*, you wanted to have a video screen displaying the Auricle program's output, so that you could see it while you were conducting.

Well, that was nice to have, but, to be perfectly honest, it was a luxury on the soundstage—I never looked at it very much.

But, when I'm writing, I'm terrible: I often get lost when I'm writing; I am constantly remetering bars, and then I forget where I put the meter change in, and then I figure it wrong. With the new version of the program for C1, I can have a number of screens up at one time. So, I can always put, say, the meter map in one corner of the screen, so I'll always know the current bar and meter situation. Previously, I would, for instance, forget to change the meter on the score sheet for the cue, and then I'd forget where I was, and would be writing in the wrong meter. With the Auricle program for the C1, I always have a graphic display of the bars and meters.

*It's not the easiest thing in the world to figure all of the film timings out and then think about clicks in terms of frames, when all of your musical training is in terms of metronome markings.*

Yes. It's tough enough to write and just concentrate on the music; if you do enough films, you realize that it's sort of a "split personality" thing—you have to be constantly aware of the time values and the parameters that you're writing in. And everybody's got a different way of doing it: Some composers will lay out the whole cue in advance. A lot of times, I'll change the meter to catch a certain timing; although, meter changes are also important for many of the musical effects I do.

*I've noticed, a number of times in cues that I've worked on with you, that there'll be a 2/4 bar, when everything else has been in 4/4, and the change was set up to catch something on screen. In other cues, though, like on one of the main themes in Rambo, there was a repetitive group of bars...*

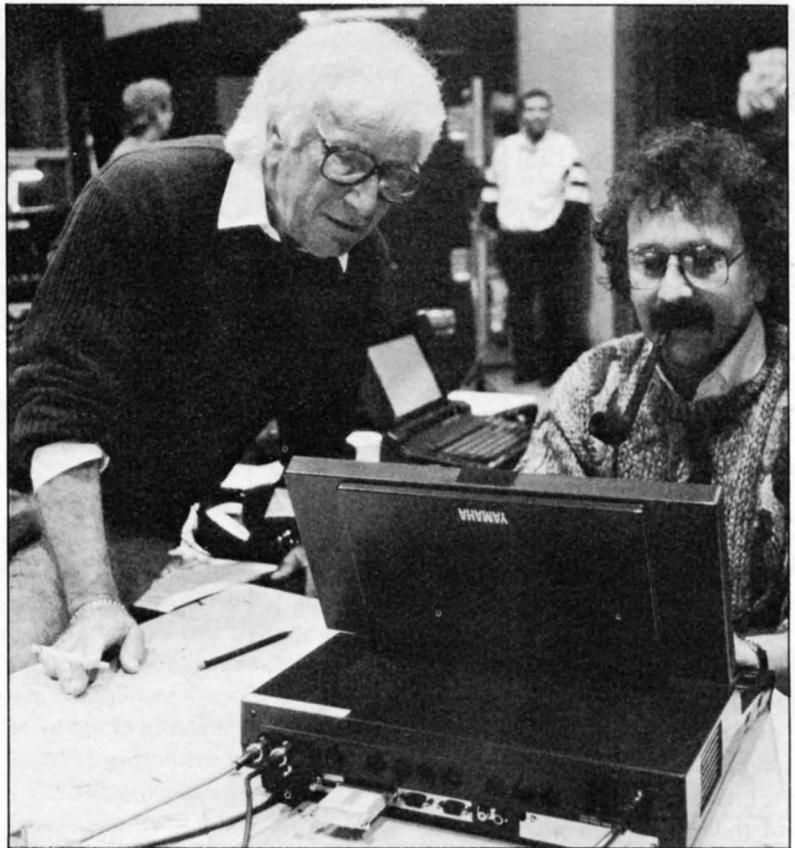
Yes: Three bars of 5/8 followed by one bar of 7/8.

*Obviously, that was the musical idea you started with.*

Yes, that was strictly a musical idea.

*At what point does tying the music to the cue come in? Do you figure out the tempo relative to a rhythm in the cue, or do you get the tempo as part of the musical idea, and then dictate the number of bars based on that, and then figure out what you have to do to fill in the end of the cue?*

Here's an example: Let's use the pattern we've just talked about—three bars of 5/8 followed by one bar of 7/8. Let's say that we want to use this



At one of the *Star Trek V* scoring sessions: Jerry Goldsmith (L) with Richard Grant, principal co-creator of the Auricle Program. (Photo by Rick Oyama.)

repetitive pattern in a cue that lasts one minute, and that there is an event in the picture at fifty seconds that we want to catch. So, we get the tempo we want, and discover that one cycle of the pattern takes 5.3 seconds. Now, what I used to do in a situation like that was divide 5.3 into 50 to see where it would come out. But now, with the C1 and Auricle, it's very easy to set up the computer with an indefinite number of bars following the pattern at the proper tempo, and then ask the program, "Where is 50 seconds? What bar and what beat does it land on?" And, say, maybe it will come in on the fifth beat of a 7/8 bar. So I change that bar—make it a 2/4 bar plus a 3/8 bar—so I can come in on a downbeat at 50 seconds. I like to try and catch things on the downbeat. But, again, you know, the computer works very fast.

For the last four or five years, I've been working mostly with the Mac, so using the C1, which is MS-DOS, took a little readjusting. We haven't begun to appreciate the musical possibilities of the C1. I think having all of the built-in MIDI ports is wonderful. [Note: The C1 has two MIDI INs, one MIDI THRU, and eight MIDI OUTs.] And the programmable sliders are great, although I haven't seen any programs that use them yet. Actually, with the exception of Auricle, I haven't used any of the music programs written specifically for the C1,

*Continued on page 20*

but I have run some IBM MS-DOS programs on it with no problem.

The other thing I like with the C1 is its portability. When I work with the Auricle program, I use a color monitor. I've never stopped to figure out what all the color coding means on the Auricle program, but it does look pretty in color. Frankly, though, I think the graphics on the C1's LCD screen are much sharper—they're pretty good.

*Because of the portability, are you thinking of using the C1 for other music applications?*

Yeah, there have been times when I wanted to be able to carry my sequences around more easily. On *Leviathan*, I had a zillion sequences. I created the sequences on the Mac, but the recording sessions were in Italy, and I didn't want to cart a Mac over there, so I loaded all of the sequences into a small hardware sequencer, and took it instead.

I have often thought about eliminating some of that. It's just a matter of my deciding to change all of my musical applications over from the Mac to the C1 and MS-DOS. My C1 has a 20-megabyte hard disk, which makes it pretty powerful: If you had everything in there, you wouldn't have to be carrying disks around all the time. I just have to ask myself, "Do I want to start a whole new scene again?"

I just wish there was one computer that has it all.

*Yeah, that would be nice, but...*

Forget it.

*That is one of the things that is amazing about MIDI. You know, people who don't work with it think that it cooks dinner and does your taxes, but just the fact that a group of companies that are in competition with each other agreed to do at least one thing the same way is just astounding.*

Well, that is amazing, I suppose. But, only so much works with synthesizers and MIDI—if people only knew that it's not the be-all and end-all. It's a neat part of our vocabulary, but people go overboard: "Oh, we have made MIDI! Everything's musical with MIDI!"

*A lot of people talk about it who don't know much about it.*

I don't know very much about it.

*Well, you know the essential things. On a basic level, it's a lot simpler and a lot more limited than most people think. It's just that what you can do with MIDI expands the possibilities quite a bit.*

Well, as you well know, ninety-nine percent of my musical electronics are used as an adjunct

to the orchestra. And once I got past the glamour of the MIDI thing, my feelings changed. I remember when MIDI first came out, I was layering five synthesizers at a time and playing them against the orchestra. And it became ridiculous! You know, when we had everything going, you couldn't tell if all the synthesizers were coming through on the board! So we spent hours saying, "Let's hear this one first; now let's hear that one; now let's hear this one; okay, mix it, check it—they're all coming through." Dealing with that many synthesizers, you have to get them balanced. And then, when you put it with the orchestra, it doesn't make any difference! I think you've noticed that, on the last few sessions you've been with me, I don't MIDI anything together anymore. I did it on just a few cues on *The 'burbs*: I had two organ sounds MIDI'd together.

*On The 'burbs, I think the most MIDI we ever did on a cue was two instruments. I don't remember ever hooking up to more than one other instrument to the one I was playing.*

On *Star Trek V*, I didn't MIDI anything together, as far as I can remember.

*There weren't any MIDI combinations on the days I worked.*

The only thing involving MIDI at my sessions these days is if you've got a rack-mount unit, and you need a keyboard to play it. Otherwise, nothing's MIDI'd together. The only time that MIDI really plays a big part in what I'm doing now is if I'm running a sequencer.

*You said that it started to feel "ridiculous" to have these six or seven electronic things going all at once, and you really couldn't hear all that supposed depth when you were mixing it with the orchestra.*

Yeah, and even if I'm doing an all-electronic thing, Tom—you know, one patch, one sound on a synthesizer, is a very complex thing. And, we never totally realize all of the overtones in the timbre. Especially when you're doing film, you never understand all of these subtleties, because it is always wiped out by ambient noise. You get some of these wonderful patches, with these subtle sounds and little dancing effects in there, and half of it gets wiped out just with sound effects. And, if you start mixing that with an orchestra, a lot of it gets wiped out by the orchestra. So, I found that when you start MIDI'ing and layering these sounds, it creates a sort of non-distinguished sound.

Now, I look for a patch that has a very unique sound, or is non-acoustic sounding, or suggests

an acoustic sound. And then I let it play by itself. If I have an orchestra of 80 to 85 musicians, I don't need to fatten up the sound with synthesizers.

*So, you don't need a string pad when you have strings.*

No. Oh, there was a time when I thought it was wonderful—I tried to be the Montovani of the 80s. I had some very rich GS1 string sounds, and combined them with violins playing the melody, but all I ended up with was a GS1 sound or an acoustic string sound—it didn't really blend that well. If you've got a properly sized orchestra, you can leave the acoustic things alone, and go for *unique* sounds with the electronics—sounds that you can't get acoustically. And then the secret is to blend them all together.

*Yeah, the blending is the hardest thing. That's one thing I have always felt that you understood very well. I have always admired the fact that you choose all of your electronic sounds before you come to the session.*

Well, yeah, I don't understand how you can't do that. You know, I see Mike Lang and Ralph Grierson come in with these huge rigs of equipment, and these huge libraries of sounds, and I know a lot of composers just write synthesizer parts and give some description of the sound they want. And you know, I don't think I do an amazing job with my approach, but I couldn't compose that way.

To me, that's like composing for an orchestra and not knowing what certain instruments are going to sound like. I mean, if you don't know what the woodwind section is going to sound like, how can you write notes for it? For a lot of my musical ideas, I'll hear a certain patch, and it will trigger a musical response. Most of the time, I'll get the musical idea and try to fit it to some sound that complements the idea. But so many times, I've heard electronic patches that all of a sudden trigger a musical idea inside me. So, to just sort of go at it blindly, that's no way I could work.

*Well, it does seem like a different way to approach things.*

Yeah, everybody wants different things.

*I remember seeing this on a part during a session: "scary analog strings."*

I'm glad I wasn't you trying to figure that one out!

*What I wound up using was the full Yamaha TX816 sound mixed with a little bit of Oberheim Xpander. So, it was actually scary digital/analog strings!*

How would anybody even know what the composer meant by "analog"? Was it a slow



Jerry Goldsmith rehearsing the cello section at one of the scoring sessions for *Star Trek V*. (Photo by Rick Oyama.)

part?

*Yeah, exactly. I think, at that level, the composer is asking the keyboard player to be an arranger or an orchestrator for some of their ideas.*

That's a good way of putting it; I think the composer has to admit that the player, in that situation, is almost being an orchestrator.

*You have always been your own orchestrator. I mean, Arthur Morton does the job for you that's called "orchestration"...*

He does the labor.

*He writes out the parts from your sketch onto a full score. But, your sketch always indicates all of the instrument parts.*

Right; but that's just the way I work. Everybody works differently. I wonder how other people work the way they do, but I'm sure people wonder how I work the way I do. It's just the same way I've done it all my life. I've been interested in the newer electronic things ever since the time when they were cutting and splicing magnetic tape. So, it's not a new-found thing with me: It was always something that I wanted to get involved with.

*You were using tape manipulations back in the early 60s.*

The first film where I really got involved was *Freud*—the dream sequences were all done using these old tape techniques. When you went to school, isn't that what everyone was doing in electronics?

*Oh, yeah. Little pieces of tape hanging off the wall, splicing, changing speeds, reversing sounds, and everything.*

I think it was more interesting then.

*You know, one of the things I thought when I saw samplers and then computer support like the*

Digidesign programs—I said, “Hey! This is great. You can do all of those tape-splicing techniques, right here on the computer with a sampler. You can take this part of the sample, run it backwards, change the speed, and so on.” But, it seems that no one’s doing anything like that with the samplers.

No, it’s becoming a thing where samplers are simply being used to simulate an orchestra. That’s the direction I see it all going. And, at this point in my life, I’m convinced that it can’t do it—I mean, samplers can do very good simulations; I’ve heard some that’ve been pretty scary—but it isn’t the same thing.

*I’ve heard some scary simulations, too, but I think I can still tell the difference—and I’m sure you can.*

Oh, I can tell the difference. But I think what you were saying about using a sampler is a great idea. I had never thought of it that way.

*Yes; here you have something that can do things electronically that used to take incredible amounts of hand labor, and who knows what else it could do? And people don’t seem to be using any of that capability—instead, they’re making little synthetic orchestras.*

I remember an interesting case in a two-part interview you did with Morton Subotnick a few years back. [Note: The interview appeared in the March ’86 and April ’86 issues of *AfterTouch*.] He was talking about a piece he did with the QX1 and the TX816, and he was so happy because he didn’t have to put up with any musicians. I was really fascinated with this; here was a guy who was really happy because he could get his music exactly the way he wanted it. Very interesting. Listen, if that’s what you want, that’s terrific. But I get sort of lonesome without the musicians there. Sloppiness is a part of the human element, which I sort of miss in totally electronic, sequenced music.

*That’s one of the reasons why a simulated string section doesn’t sound like a real string section: it’s too perfect.*

Well, it’s okay for a demo or something. I’ve had composers ask me, “What do you need an orchestra for?” And I say, “I don’t think it works until someone plays it.” And I’ll write here in my studio, and demo the cues with electronics, but the first time I actually hear the orchestra play it, all of a sudden, it just blossoms out. Maybe I didn’t demo it really well with the electronics, but there is just a difference for me. You cannot get that kind of excitement, the air, the breath, and the emotion from the computer. At

least, not yet.

You know, I haven’t really begun to examine what a sequencer can really do, but the exciting thing for me is to use the sequencer to do something that’s “impossible” to do, and then have that effect run with the orchestra.

*Instead of using the sequencer to replace musicians, you’re using the sequencer to do something they can’t do.*

Yeah, and that’s what you should do. Or to get cold, mechanical, “sequence-sounding” effects, it’s better to use a sequencer, even if it’s possible to play the part live.

*In your orchestra, it seems that you treat the electronics as another family: There’s the woodwinds, the brass, the strings, the percussion, and the electronics.*

Yeah, that’s what it should be.

*And, when you pick the orchestra for a score, and decide on the number of brass and the number of woodwinds, you decide the electronic family in the same way, by picking out the specific sounds.*

What I try to do when I sit down to score, I try to have the musical ideas worked out, and I try to say, “This is the library of sounds I’m going to use on this film.” Once I do that, I try to stop, and not keep looking for other sounds that I’ll use “here or there.” Sound is so identifiable, it’s so distinctive, but like anything else, you have to use it several times so the audience will grasp ahold of it.

*I didn’t play on the later sessions for Star Trek V, but I remember one distinctive sampler sound, a sound that would have been hard to get any other way: it was a sackbut sample.*

Oh, yeah, yeah.

*Did you use that a lot, or just at the beginning?*

That idea, that was one of the first cues I wrote, and I thought I would use that idea all through the picture. I didn’t use it constantly, but it appeared several times during the picture. There was an example of a sampled acoustic sound that I do use.

*How many sackbut players are there these days?*

There are not too many of them, and most of them are very unreliable. The samples I used were made in London. I know the guys who played them, and I’ve used them on live dates. The intonation on medieval instruments is very unreliable, especially if you want to use the entire chromatic scale—forget it! But, if you get good, clean samples—which these all were—then you can use instruments like the serpent, the sackbut, and the crumhorn.

*Those are all impossible instruments to play.*

They are impossible to play; believe me, I've used them and I've tried. But now, if you have good samples of them, you can do anything you want with them! The intonation is perfect, you've got all the chromatics you want, and you can play chords with them. That's one instance where I think the sampler's terrific.

*What size electronic section do you see in your next few films?*

As you know, I always used to use 3 or 4 musicians playing synthesizers; I've reduced that way down, too. For what I'm doing now, I don't need that many guys. It was too much of a good thing, you know? What sounds good when you're writing it doesn't necessarily sound good when you're performing it.

On one level, I wouldn't mind not using anybody: I love to go back and do it myself—"Now it's my chance to play!" And I have put in a lot of electronic stuff like that, in the mix stage. The thing is, the way Bruce Botnick has been recording, we don't even have to mix down! It's a great live sound. And it is sort of exciting to be able to put it all together at once.

*Do you still add electronic parts later?*

Yes. I really didn't on *Star Trek V*. I think I replaced a part on one cue because, when we checked it over, it just didn't mix down. In the earlier days, long before there was digital, I could just raise and lower levels of certain things, and not destroy the overall orchestral mix. And, actually, on a couple of cues for *Star Trek V*, I think I raised or lowered, a couple of electronics. That was all. And I found one, as I said, that distorted when it came on; so I replaced that. But I have spent hours in there overdubbing. You can get so carried away: "Oh, we'll double the strings on this, it'll sound great!"

*But, in general, you're going more in the direction of having a smaller electronic contingent—more like two players or something.*

Which means I'm not using as many of them, but what I'm using is standing out more. On some of the earlier pictures, it was too much. You don't need it all. I've learned that.

Also, if you use too much electronics, it's going to blot out the orchestra. It throws the orchestra out of proportion. There's too much presence, and it all has to be reduced in size. And that's the thing—I don't really know, until I get it all mixed together. A lot of times I just have to say goodbye to the electronics, because I'm not hearing the orchestra. It's still new!

It's very new—it's a whole different thing, thinking about orchestration with electronics. And it makes it more difficult to learn when



*Jerry Goldsmith conducting one of the scoring sessions for Star Trek V. (Photo by Rick Oyama.)*

you're using so many different patches. Consider them all to be different instruments. I mean, if you had thousands of different woodwind instruments, instead of just the basic four in the section—flute, oboe, clarinet, and bassoon—you could go crazy. But that's the kind of thing you're facing with a synthesizer; each particular patch has different characteristics, which are going to react with the acoustic instruments in different ways. And I haven't figured out how to tell that until I've actually applied them.

*No one's written an orchestration book about electronics; it would be almost impossible, because the instruments change every six months.*

Well, they use the term now, "I'm orchestrating electronically." When they do a pure electronic thing now, there's a lot of talk about orchestrating. I know what they mean, but maybe the word "orchestration" is not the right term for it.

You can't use the principles of orchestration, of acoustic orchestration, with the electronic assembling of sounds. I mean, I've tried it enough times, and I can tell you, it doesn't work. You cannot write contrapuntally for an electronic ensemble the same way you would write for an orchestra.

*When putting together your electronic section, do you work out your own sounds?*

No, I don't make the sounds myself. I'm not very good at that. I'm really terrible at it. So I'm dependent on third-party, you know, the factory stuff, and I'm proud of the fact that that's all I use. I don't care. I have enough trouble writing music without sitting around making sounds. I love the sound of the Yamaha, and the Roland. And the new Korg M1 I've got. The sounds in it are very good. But, there again, you're depen-

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# DX11 & TX81Z

## Adding TX81Z Voices To The DX11—A Tutorial For Non-Programmers. By Steve Ahola.

THE MANY VALUABLE FEATURES of the Yamaha multi-timbral DX11 synthesizer were first presented in the February 1988 issue of *AfterTouch*. If you are working within a limited budget and looking to step up from your non-velocity keyboard, the DX11 is definitely worth looking into. The ROM voices and factory Performance presets sound good enough that you'll be too busy *playing* it to bother with any of the programming parameters for quite a while.

Fortunately for those of you who'd prefer to postpone learning the theory and techniques of programming the DX11, there is also a readily available source of 128 additional voices and 24 additional performances: the TX81Z. This article will step you through the procedures needed to save your DX11 factory RAM presets to tape, and then transfer the internal voice and performance banks from a TX81Z to your DX11.

### Saving DX11 Voices & Performances To Tape

Start by connecting the cassette cable that came with the DX11 to an appropriate tape recorder. The white wire goes to the tape output jack (labelled EAR, EXT SPKR, or TAPE OUT on various machines). The red wire goes to the tape input jack (which may be labelled MIC, AUX, or TAPE IN). The black wire with the small plug would go into the REMOTE jack on small portable recorders, although it is not used in connection with any of the Yamaha products I've worked with.

I have had good results using a Radio Shack CCR-81 recorder designed for recording computer data; one of its features is an LED that lights up when it is actually recording data (the LED goes out when the data transmission ends). A generic \$29 one-speaker portable tape recorder worked fairly well for me before that; the built-in ALC (Automatic Level Control) on these generic cheapies seems to be a positive feature for data recordings. There is probably one of these around your house (or one of your friend's) that you can put to use until you get around to buying a CCR-81. If you try using a high-quality audio stereo or 4-track deck, any built-in signal processors (such as Dolby or DBX) should be switched out, and tape equal-

ization should be set approximately.

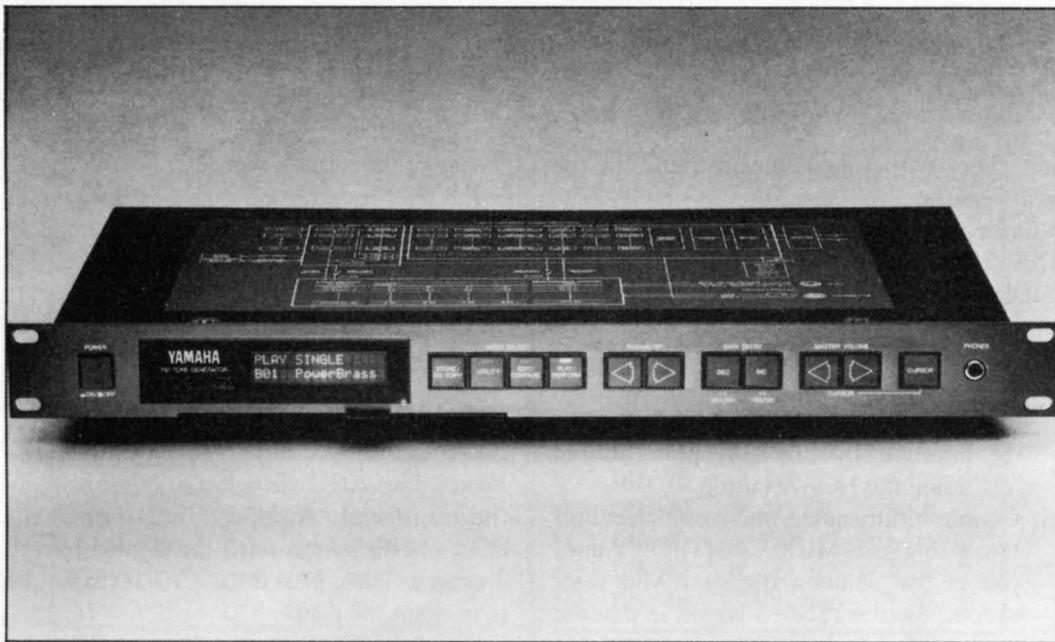
As for the tape itself, I've always used the Radio Shack Computer Cassettes that come in 10 and 20 minutes lengths. You can load approximately 12 banks on a 10 minute tape. One final note on the taping end of this process: Make sure that the head and transport are clean, and that the machine will play audio tapes without noticeable wow or flutter.

Now, to save the DX11's 32 internal voices to tape, follow these steps:

1. Press the Utility mode button to the left of the LCD display.
2. Press Select Button #22 (these are the buttons below the algorithm charts on the front panel). The LCD should read "UT CASS CONTROL" on the top line.
3. Press Select Button #22 repeatedly to cycle through the 11 jobs available in this utility.
4. To save the internal RAM voice bank, the bottom line of the LCD should read "Save32Voice?(VD)." Pressing the NO/-1 button will change the "VD" (for DX11 format, to "TX" or "DX" for alternate formats (not needed if you will be reloading the data back into your DX11). Pressing the YES/+1 button will result in the prompt "Save Ready?"
5. Start recording on your tape recorder, and press the YES/+1 button a second time to execute the save.
6. After all 32 voices have been saved, the prompt changes to "Verify 32 Voice?" Stop your tape recorder, rewind the tape to a spot immediately before the recording began, and press the YES/+1 button once; the prompt will read "Verify Ready?"
7. Start playing back your tape and press the YES/+1 button one more time. After all 32 voices have been verified, the display will read "Verify Completed."

To save the DX11's 32 performances to tape, follow these steps:

1. Press the Utility mode button to left of the LCD display.
2. Press Select Button #22. The LCD should read "UT CASS CONTROL" on the top



TX81Z FM digital tone generator.

line.

3. Press Select Button #22 repeatedly to cycle through the 11 jobs available in this utility.
4. To save the internal RAM performance bank, the bottom line of the LCD should read "Save 32 Perfrm?" Pressing the YES/+1 button will result in the prompt "Save Ready?"
5. Start recording on your tape recorder, and press the YES/+1 button a second time to execute the save.
6. After all 32 performances have been saved, the prompt changes to "Verify 32 Perfrm?" Stop your tape recorder, rewind the tape to a spot immediately before the recording began, and press the YES/+1 button once; the prompt will read "Verify Ready?"
7. Start playing back your tape and press the YES/+1 button one more time. After all 32 performances have been verified, the display will read "Verify Completed."

### Loading Data Banks From Tape

The procedure for reloading voice and performance data from tape back into the DX11 is almost identical to that for verifying saved data. Follow these steps:

1. Press the Utility mode button to the left of the LCD display.
2. Press Select Button #22. The LCD should read "UT CASS CONTROL" on the top line.
3. Press Select Button #22 repeatedly to cycle through the 11 jobs available in this utility.
4. To load data into the internal RAM voice bank, the bottom line of the LCD should

read "Load 32 Voice?" To load data into the internal RAM performance bank, the bottom line of the LCD should read "Load 32 Perfrm?"

5. Once you have chosen the load you want to perform, press the YES/+1 button so that the prompt reads "Load All Ready?"
6. Begin playback on your tape recorder, and press the YES/+1 button a second time. The display will monitor the progress of the load and notify you when it is completed.
7. To load a single voice or performance, press Select Button #22 until the display reads "Load 1 Voice?" or "Load 1 Perfrm?" Then press YES/+1 once, and use the appropriate Select Button to choose the single voice or performance from tape.
8. Press YES/+1 again to initiate the search and load. For single loads like these, the tape doesn't have to start at the beginning of the bank dump.
9. Once the single voice or performance is found, it is loaded temporarily into the DX11's Edit/Play buffer. To save the data to the DX11's internal memory, hold down the Store button and press the numbered Select Button of your choice. (To store to a cartridge instead, press the Cartridge button before pressing the Select Button.)

### Loading TX81Z Voices And Performances Into The DX11

The first step here is to locate a TX81Z to borrow voices from. Hopefully, the dealer you bought the DX11 from has a TX81Z in stock, and is willing to let you make the transfer. If

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# DX11

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not, check with your musician friends in the area.

Once you have located a willing and able TX81Z, connect a MIDI cable from the TX81Z's MIDI OUT to the DX11's MIDI IN. Then, to set the DX11 to receive data, follow these steps:

1. Press the M. PROTECT button to call up the "MEMORY PROTECT" screen, and set the internal (INT. memory protection to "off" using the NO/-1 button.
2. Go into Utility mode, and press Select Button #18 ("UT MIDI CONTROL") until you see the "Midi:on" display. If Midi is set to "off," use the YES/+1 button so that the setting is "on."
3. While still in Utility mode, press Select Button #18 until you see the "Basic Rcv.Ch=xx" display. Use the NO/-1 and YES/+1 buttons to set the basic MIDI receive channel so that it matches the MIDI transmit channel of the TX81Z.
4. While still in Utility mode, press Select Button #20 to access the "Exclusive:on" display. If MIDI Exclusive is set to "off," use the YES/+1 button so that the setting is "on."

To set the TX81Z to transfer its internal voice or performance data to the DX11, follow these steps:

1. Press the Utility mode select button.
2. If the display doesn't read "MIDI Control?," press the parameter button until it does.
3. Press the INC/+1/YES button to cycle through the MIDI Control jobs, and make

sure of the settings indicated below.

4. Job #2 "Transmit Ch=xx." This should be set to match the MIDI receive channel on the DX11.
5. Job #8 "Exclusive=on." This must be set to "on." If it is set to "off," use the INC/+1/YES to switch the setting to "on."

Now, to transmit TX81Z voice data to the DX11, call up Utility Job #9 on the TX81Z—"Voice Trans? (I)." To select another bank for the transfer (I, A, B, C, or D), press the DEC/-1/NO button until the desired bank is displayed. Then, press the INC/+1/YES button to transmit the data.

When the DX11 receives the voice data from the TX81Z, it will display "MIDI Received" for a few seconds, and will then revert to whatever display it had shown prior to the load operation. To verify the operation, press the Single Play and Internal buttons on the DX11, and then use the Select Buttons to see if they call up the voices that were in the TX81Z. If the voice dump was successful, you will probably want to save this new DX11 internal voice bank to tape using the steps outlined above, and then continue with the other TX81Z voice banks.

To transmit TX81Z performance data to the DX11, call up Utility Job #10 on the TX81Z—"Perfrm Trans?." Press the INC/+1/YES button to transmit the data.

Performance data dumps work the same way as voice data dumps, but it should be noted that they call up voice *locations*, not actual voice data. For example, the TX81Z factory performance #1 is called "AcustcGuit." It uses the C01 voice on the TX81Z, which is

DX11 FM digital synthesizer.



"NylonGuit." On the DX11, voice C01 is "Strings 1." In other words, it is necessary to edit the TX81Z performance a bit to be able to recreate them on the DX11. The voices making up a particular TX81Z performance must be loaded into the DX11's internal (or cartridge) bank, and those memory locations must be substituted for the voice locations listed in the original TX81Z performance.

Incidentally, since the TX81Z has only 24 performance slots, your initial performance dump from the TX81Z will leave eight blank slots that you may want to fill in with your favorites from the DX11 factory presets.

### Storing Data To RAM Cartridges

Once I got the TX81Z voices home to try out on my DX11, my only complaint was that the internal voice bank of the DX11 could hold only 32 of the TX81Z's 128 voices. I had assumed that there would be a lot of duplication between the voices in the two instruments, but I found them all to be quite unique and distinctive. I gave up trying to narrow my choices down to 32 voices, and went down to the store to buy a RAM4 cartridge, which gave me immediate access to an additional 32 voices and 32 performances. Here are some things to keep in mind if you decide to use a RAM4 cartridge to expand your memory possibilities with the DX11:

1. RAM cartridges must be formatted before you can use them. First, plug the cartridge into your DX11. Then, make sure Memory Protect is off. Next, enter Utility mode and press Select Button #21 until the display reads "Format? BK 1A." Finally, press the YES/+1 button twice to format the cartridge. (Formatting always erases everything on the cartridge, so the display will ask "Are you sure?" after you press the YES/+1 button the first time.)
2. Individual voices can be stored to the RAM4 cartridge instead of the internal memory by pressing the Cartridge button before selecting a numbered memory location (during the Store operation).
3. The DX11's internal voice and performance banks can be saved to the RAM4 cartridge using the Utility mode. Press Select Button #21 until the display reads "UT CRT CONT. 1A" on the top line and reads "Save? (all)" on the second line. You may want to press the NO/-1 button to select "(v&pfm)" instead of "(all)" to save only the voice and performance memories. The

other data (Program Change Table data, System data, Effect data, and Microtune Scale data) can be added to the cartridge later, once you get all of it programmed to your taste. Press the YES/+1 button twice to save the selected memory to the RAM4 cartridge.

4. Data saved to cartridge can be reloaded back into the DX11's internal memory using the Utility mode. Press Select Button #21 until the display reads "UT CRT CONT. 1A" on the top line and reads "Load? (all)" on the second line. Press the NO/-1 button to cycle through the load choices, and then press the YES/+1 button twice to execute the load. This will overwrite the DX11's internal memories as selected, so be sure you have saved to tape any new voices you have been working on.
5. An additional benefit of having a RAM4 cartridge is to use it to help build your own custom banks from the ROM voices of the DX11 and TX81Z. Rather than go into the step-by-step details of this procedure, I will just outline the basics. You can load one bank at a time from cassette into the DX11's internal memory. Arrange the voices you want to keep, and then save your internal memory to cartridge. Load the next bank of voices from tape, and then store the voices you have selected to cartridge. When you have come up with a customized bank (in the cartridge) you want to keep, load it back into the internal memory, and then save it to tape. (Remember that cartridge memories cannot be saved directly to tape.) This kind of editing process could be done without a RAM4 cartridge, but it would take *considerably* longer—only the first voice bank could be loaded into the DX11 *en masse*; subsequent voices would have been loaded from tape one at a time into the Play/Edit buffer, and then be stored to the desired location.

\* \* \* \*

Following the steps laid out in this article will at least double your library of Yamaha factory preset voices for the DX11, and add several new performances. It will also allow you to use the various TX81Z performance patches listed in AfterTouch that are based on the TX81Z's factory preset voices. For those of you who haven't yet delved into the procedures for editing voices, performances, or global settings, these procedures will introduce you to the basic logic behind the DX11 menus.

# Hot Tips

## Readers Tips For The SPX90, QX3, And More.



QX3 digital sequence recorder.

### Using The QX3 Digital Sequence Recorder In Live Performance

By Paul J. Kurtz

The Yamaha QX3 sequencer is a versatile instrument that lends itself quite nicely to live performance applications because of its design and features. I take advantage of the programmable function keys (F1 and F2), which allow the user to assign a sequence of up to 128 keystrokes to each key, thus speeding up operations that are performed frequently. When using my QX3 in live performance, I have programmed one of these function keys to clear existing song data and load a new song into memory. This helps to reduce the amount of dead time between songs.

To program this operation, perform the following steps:

- 1) While in Record/Play mode, press the Load key and use the front-panel dial to select the song that is to be loaded into memory.
- 2) Hold down the Shift key and the F1 key at the same time to start recording key-

strokes for the F1 key. The LCD display will flash the following:

\*\*\*Log/On (F1)\*\*\*

- 3) Press the Enter key; the LCD display will indicate that song data already exists, and will ask you if you want to clear the data.
- 4) Hold down the Character (CHAR) and Y keys at the same time to clear the data and load the new song.
- 5) Now, hold down the Shift and F1 keys at the same time. The LCD display will stop flashing, indicating that recording of keystrokes for the F1 key has ended, and that the new song has been loaded into memory. (It should be noted that either one of the function keys (F1 or F2) can be used in this way.)

Now, whenever you want to load a new song after an earlier song has been played, simply press the Load key, use the front-panel dial to select your new song, and then press the F1 key. This operation saves keystrokes, saves time, and only requires the use of one hand—all of these

factors are greatly appreciated during the stress of a live performance. Once a function key has been programmed, the data can be saved as part of a Setup file (Record/Play Job #7), and then be recalled (Record/Play Job #8) for future live performances.

I hope that other people using the QX3 sequencer in live performance can take advantage of this feature as I have.

. . . .

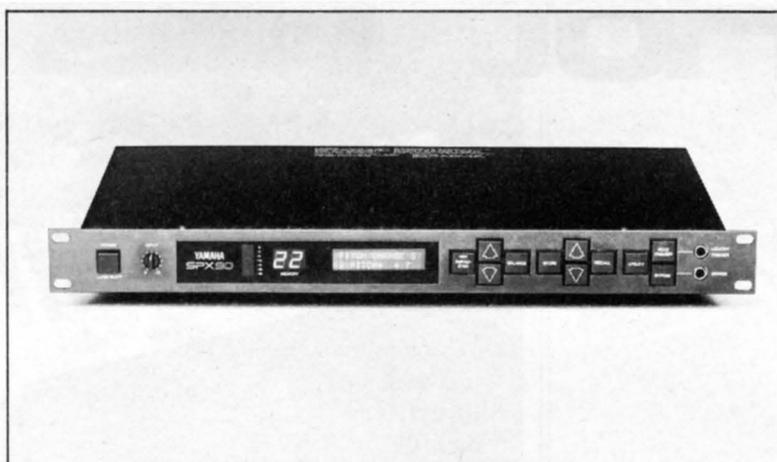
## Four New SPX90 Patches For Use With Guitar, Plus Some General Advice On The Use Of Effects Devices

By Mathew Gurman

I remember some wise old studio sage once saying, "The most important feature of any effects device is its off switch." While I do agree that subtlety is the better part of effects mixing, that approach may not always apply to guitar! Here are a few very subtle guitar effects for the SPX90. They are so transparent and usable that they don't even need an off switch. (Well, to be completely honest, even the most usable sound can be overused....) Anyway, these sounds work great, and they won't burn you out. They are classic, timeless, and so on.

Before giving you the data for these patches, there is one other thing I should mention as general information: The way you patch in your effects and effects loop can have a huge effect on the effect's effect. That is, an effects loop that is not level-matched can mess up your gain stage on high gain cascading preamps and tremendously alter the output of the power amp. Matching the input and output levels on your effects unit with your amp should be done slowly and perfectly. Once this is accomplished, you may then adjust the balance of effect to dry signal.

If your amp has effects level control, set the balance on the effects unit to 100% effect, and then adjust from your amp. If your amp has no effects level control or no effects loop at all, adjust the balance on the effects unit. When using a compressor, it is best to patch into the front of the amp, straight in from the effects unit. A compressor is constantly boosting and squashing. If no signal is coming in to boost or cut, the compressor will boost whatever it can find, and there is always plenty of noise from the preamp for it to play with in an effects loop. Because the SPX90 compressor program is more



SPX90 multi-effects processor.

of a limiter than a compressor, the noise boosting aspect is not a problem, but it can still take the punch out of a high-gain preamp.

I have given each new SPX90 patch below a descriptive name, followed by the preset number from which it was constructed. Following the parameter and front panel values for each new effect, I offer a short description of the effect created by the patch.

### Gentle Whammy (from preset #29)

Trg Level: 58  
Vib Dly: 5 ms  
Vib Rise: 175 ms  
Vib Frq: 3.8 Hz  
Vib Depth: 18%  
Balance: 100%  
Output Level: 100%

This effect gives a gentle shimmering quality to sustained chords and "voice shapes," the way a whammy bar does.

### Double Whammy (from present #29)

Trg Level: 85  
Vib Dly: 149 ms  
Vib Rise: 150 ms  
Vib Frq: 4.2 Hz  
Vib Depth: 9%  
Balance: 100%  
Output Level: 100%

This effect is similar to "Gentle Whammy," but is deeper—it will hold up to an overdrive sound. It is also better suited to lead playing.

### Viberolux (from preset #27)

Pan Speed: 5.8 Hz  
Direction: L to R  
Depth: 38%  
Balance: 100%  
Output Level: 100%

Continued on page 30

# Hot Tips

Continued from page 29

This is the only patch I've ever heard that really sounds like an amp vibrato. It does! The trick is to use this panning effect in mono—use the SPX90's left channel output. Stereo is nice too, but the effect has more “classic appeal” when used in mono.

## Dyna-Comp (from preset #19)

Trg Level: 1  
 Trg Dly: 0 ms  
 Trg Msk: 5 ms  
 Attack: 5 ms  
 Hold: 30,000 ms  
 Hold Level: 30%  
 Release: 32,000 ms  
 Balance: 100%  
 Output Level: 100%

This is a compressor/limiter patch. Adjusting the “Trg Dly” will make the effect sound more or less “squashy.” This patch will work through an effects loop, but probably sounds best when patched through the front of the amp.

\* \* \* \*



FB-01 FM digital tone module.

## A Basic Setup For The FB-01's User Configurations

By Lewis A. Songer

Having spent some time getting acquainted with the DX7s and how it works with Jim Miller's Personal Composer software for the Mac, I have been feeling fairly confident and proud of the fact that at my advanced age I could learn something new and high-tech in the world of music.

I recently acquired a pair of FB-01s from my local MIDI market. I was immediately struck by the power and versatility that was now, literally, at my fingertips. One is often bewildered by the multitude of choices when trying to master the edit voice features of the DX7 series. I have found that the FB-01 presents an orchestrational tool that eliminates the need to

play around with rates and levels and carriers and modulators.

I would like to share one bit of information I discovered, which I hope will be useful for those who have at least one FB-01. I discovered this while trying to answer the following question for myself: What good are all of those user-definable configurations?

I simply figured out all of the possible combinations for distributing the FB-01's eight notes over the eight voice slots in the configuration, and then stored each possibility to its own user configuration. (The voice selection can wait until later.) I have also found it fun and instructive to try different voice combinations while listening to playbacks of my recorded tracks.

Before beginning to set up your user configurations, don't forget to turn the FB-01's Memory Protect off. When you have the configurations set up the way you want them, turn the Memory Protect back on. The chart below shows how I used the FB-01's Config/Store feature to save 16 unique configurations.

voices	notes per voice	config#
8	1-1-1-1-1-1-1-1	ROM 18
6	2-2-1-1-1-1 x x	user 1
5	4-1-1-1-1 x x x	user 2
5	3-2-1-1-1 x x x	user 3
5	2-2-2-1-1 x x x	user 4
4	5-1-1-1 x x x x	user 5
4	4-2-1-1 x x x x	user 6
4	3-2-2-1 x x x x	user 7
4	2-2-2-2 x x x x	user 8
3	6-1-1 x x x x x	user 9
3	5-2-1 x x x x x	user 10
3	4-3-1 x x x x x	user 11
3	4-2-2 x x x x x	user 12
3	3-3-2 x x x x x	user 13
2	7-1 x x x x x x	user 14
2	6-2 x x x x x x	user 15
2	5-3 x x x x x x	user 16
2	4-4 x x x x x x	ROM 19
1	8 x x x x x x x	ROM 17

The settings in this chart fill up the FB-01's user-definable configuration memory. However, the chart does leave out one possibility, the seven voice configuration (2-1-1-1-1-1 x), which I find to be the least useful. Still, I have stored this configuration in my other FB-01, and still have 15 user configurations there for other voicings.

# Goldsmith

Continued from page 23

dent on other people to make the sounds.

*Well, looking at it the way you were explaining before, it's like you're looking for instruments in your orchestra. And you're just going through the sounds trying to find the ones you want.*

So, keep making 'em. I'll take 'em all! Sometimes I'll go through them and try to sort out selectively what I think is a great sound; but to me, the sound doesn't mean a damn thing until there's some music attached to it. You can get some wonderful sound, and try to find a musical application for it, and none of it works! So you get something that you think of as an awful sound, and find a musical application for it, and it works wonderfully.

*For the record, let's run down the synthesizer gear you have in your studio.*

Well, there's an Emulator; there's a Super Jupiter JX10; there's the Korg M1; there's a Prophet VS; a DX7; the DX7 II FD; the D-50; the KX88; the GS1; the RX5. In the rack, I've got the TX16W; the Roland S-550 sampler; the Oberheim DPX-1; the Roland MKS-20 Digital Piano; the Roland MKS-80 Super Jupiter; the TX81Z; the TX816; then the two Cooper MXB 16/20 MIDI Patch Bays. For enhancing the monitor sound, I've got the REV7, the REV5, and the SPX90. Then I have the C1 and the Macintosh computers.

*How do you feel about trying to keep up with the new instruments that keep coming out? I know that there are certain instruments that you have worked with for quite a while, like the original DX7.*

Which I still use more than the DX7 II. That's not the fault of the new instrument, it's just that the old one is familiar; I don't have to learn any new things.

*Right. But do you find yourself continually checking out the new things, and then, picking the ones you think you might use?*

Yeah, but I don't think I've found too many things, you know. For me, the two major steps in the last number of years were the Yamaha DX7 and the Roland D-50. To me, those were the real giants, from digital. Nothing much has been going on with analog that I know of. And what's the next step after this? Right now there are a lot of instruments that are trying to do the D-50 one better, but all they're doing is the same principles. I have not seen anybody come out with anything that's radically new and startling.

I think the reason is that everybody's into sampling now. They're doing all the research

and development in designing a new and better sampler. A 32-bit sampler, two-minute samples, and five-minute samples, and pretty soon you'll record the whole piece into a sampler, with more and more memory. It seems to me that all the development seems to be going in the direction of samplers. Whereas, the Roland D-50 and instruments like the Korg M1 are combining PCM sampled sounds with digital synthesis sounds. It's an interesting concept, and makes for some very interesting effects. But I find most of my time in learning new stuff is with software—either the software for sampling, or for voice editors or librarians, or just keeping up with software updates.

*Is your main keyboard controller the KX88?*

No, the GS1. I just have to get in the habit of using the KX88, you know. It's just more convenient to write on top of the GS1! I've had it for 8 or 9 years now.

*Do you use it anymore on scoring sessions?*

I haven't been using it on sessions too much, no.

*Too hard to move?*

No, I just don't use the internal sounds anymore. I use the keyboard, because I like the way it feels. And, it's convenient to lay music on. You can also lay all sorts of stuff on top of it—I put the computer on top of it, put an ashtray on top of it, and spread my music all over the place. And it has the MIDI, so I can MIDI it to the whole room.

The KX88 is a little more practical as a master controller, because you can change patches with it, which you can't do on with the GS1. The GS1 is only good as a rudimentary keyboard controller, but it's fine for that. It's also good for programming my computer sequencer.

*Do you use the KX88 also?*

Yeah, I use it a lot. I have it MIDIed to the room. Personally, I also use Bank B on the KX88 to switch programs on my MIDI patch bay. I really haven't taken the time to find out the zillion different functions the KX88 does. I mean, there can be much more to it than just sitting there and playing, if I really took the time to learn the thing. It outputs MIDI clocks, I understand; I just found that out. You can start and stop sequencers with it, and change parameters with it.

It's like so many electronic instruments: You get these things, and then, you don't take the time to fully exploit what one unit can do. I

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# C1 Users Central

## Windows/A Graphic Inter- face For DOS. By Scott Plunkett.

**E**VEN THOUGH GETTING the hang of MS-DOS isn't too much trouble, you may still find yourself wishing you had the kind of visual assistance and mouse capabilities that graphics-oriented computers such as the Apple Macintosh and Atari ST offer. In response to those who feel MS-DOS isn't always the most convenient or intuitive operating system around, Microsoft developed Windows—a graphic interface for MS-DOS.

### Starting Windows

In order to use Windows efficiently, you'll probably want to buy a mouse. You'll need a serial mouse to hook up to one of the two RS232C ports in the back of the C1. Virtually any serial mouse will do, but, of course, you'll most likely have the best luck with a Microsoft or Microsoft-compatible mouse (funny how that works). Unfortunately, it's not quite as simple as hooking up the mouse and saying goodbye to the cursor keys. You'll need to follow the instructions that come with the mouse to install a mouse driver and make the necessary changes to your CONFIG.SYS file. For those of you who are easily intimidated, don't worry—the whole procedure only takes a few minutes and is easy to perform.

Before you can use Windows, you must first run the Setup routine to install all of the necessary files on your hard drive. (Sorry, this program really doesn't work on the dual floppy version of the C1; if you're really interested, you could replace one of those floppy drives with a hard drive....) Once everything is installed, change to your windows directory, type WIN at the DOS prompt, and you're ready to go.

### MS-DOS Executive

When you first enter Windows you'll find yourself in MS-DOS Executive, which is the Windows version of the Mac Finder. If you're used to using a graphics interface, the first thing you'll recognize is the set of disk drive icons in the upper left portion of the screen. The disk drive that you are currently using is darkened to

indicate that it is selected for use. If you want to use the other drive, you simply move the mouse to its icon and click. The file names of the newly selected drive immediately show up on the screen.

Since you opened Windows from your windows directory, you'll be looking at the names of all of the files and subdirectories that are contained in your windows directory. Subdirectory names are differentiated from file names by a bolder font style. If you want to go to a different directory or start a program, you just point at its name with the mouse and double click. Sound familiar?

Along the top of the screen are the names of three pull-down menus: File, View, and Special. The File menu contains file management commands such as copy, delete and rename. The View menu has a number of selections to let you determine what files you will see and how they will be displayed while you are working in MS-DOS Executive. The Special menu contains directory and disk management commands such as change directory, create directory, and format disk commands. Using the commands is fairly simple and intuitive. To copy a file, for instance, you click on the desired file name, go to the File menu and select Copy. A dialog box pops up with the name of the file you selected, and prompts you to enter the copy destination. After you type in the destination and press Enter, the file is copied.

### Windows Applications

Although Windows acts as an admirable substitute for MS-DOS, it is really designed as an environment where multiple applications can run simultaneously and share data. For example, you could map out your MIDI setup by running your paint and word processing programs together. Use your paint program to make a diagram of your MIDI setup, and copy it to your word processor to write a description of all of the MIDI cable routings. If you prefer to keep your diagram and text separate, you can still edit one of them while viewing the other. Because each application shows up in a separate window,



*C1 music computer.*

that can be individually sized and moved, you can set the applications side-by-side, so that you can work on your diagram while reading and editing your text. In this respect, Windows is very similar to running applications under Multifinder on the Macintosh.

Although you can run most standard programs while you're in Windows, you must use programs that are specifically written for Windows in order to take full advantage of the Windows environment. There are a number of fine programs, such as Microsoft Excel, already available and quite a few more being written, including a number of MIDI editor/librarian and sequencer programs.

To get you started, when you purchase Windows you also receive a paint program, a word processor, and a number of accessory programs. The first thing you'll notice about these programs is that they behave much more like Mac and Atari programs than like typical DOS programs. The mouse is used extensively in the Mac fashion—click on an item to select it, click and drag to move or extend a selection—when editing in a Windows applications. And, unlike some DOS word processors, you can use multiple fonts in a document and see them as they will appear in the finished document.

#### **Windows Or Not?**

Do you need Windows? If you're coming to the C1 from a Mac or ST and want to get up and

running in a familiar setting, then you should check out Windows. Also, if you are interested in using any Windows applications, you may need to purchase Windows. Some Windows applications are sold along with a limited version of Windows so that you don't actually have to purchase the Windows environment separately, but make sure you check on this point before you buy the application. You may still find that it's more convenient to actually buy Windows, especially if you're interested in the extra software that's included with the package.

On the other hand, if you aren't using any Windows applications and you don't need a graphic interface, Windows probably isn't for you. You should also be aware that since Windows needs a certain amount of memory to run, there can be problems if you attempt to run large standard DOS applications while Windows is active. If you tend to use big sequencer or editor/librarian programs, for instance, you'll be better off if you don't run them under Windows. If you don't feel that you need Windows, but you're still interested in something that will make working with DOS a little easier, there are a number of programs—such as Norton Commander—that will do the trick.

If you want to see or purchase Windows, you can find it at practically any computer store that sells IBM/MS-DOS software. The current version for use on the C1 is Microsoft Windows 286, version 2.1. Be sure to ask for the version that comes on 3.5" disks.

# News

Continued from page 2

ment Team) demonstration software, C1 software drivers and utilities (e.g.—a C1 windows driver), assorted 4-operator and 6-operator FM voices, and SGD product listings.

The BBS supports five phone lines simultaneously, allowing for real time “chat mode” between on-line callers. During this initial setup stage, Yamaha will not be able to reply to messages, unless they pertain directly to BBS problems. However, please feel free to leave messages for any and all other BBS users.

The Yamaha SGD BBS can be accessed at 714-522-9464. There are no on-line charges, so the only cost will be the cost of the phone call. If this BBS is successful, Yamaha will continue to expand its features. One feature we’ll be looking into is the possibility of an 800 number!

## G10 Software Update (Version 1.2)

Yamaha’s SGD Division announces the availability of a software update for the G10 MIDI guitar system. This Version 1.2 software consists of a two-chip set, and is available free of charge to any G10 owner. Installation is also available free of charge for all G10s currently in warranty.

The Version 1.2 software update makes the G10 more playable and musical—it responds more like a standard guitar (in situations like, for example, muting of strings). It also eliminates most of the false Note-On events that occurred with the original software.

To acquire this update, contact the Yamaha Electronic Service Division Parts Department (1-800-443-3548) and ask for the G10 Version 1.2 software update.

# Products

Continued from page 15

accessory, the MT3X will supply MIDI clock to a MIDI sequencer or drum machine.

The unit is less than 18” by 13”, just over 4” high, and weighs slightly more than 8 pounds. The unique flush mounted knobs, recessed faders, and recessed switches make the MT3X an attractive high tech unit, while organized and accessible input and output jacks facilitate neat interface of cables.

The MT3X is now available from Yamaha Professional Audio dealers, at a suggested retail price of \$995.00. For more information, write to: Yamaha Corporation of America, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622-6600.

## RTC1 Remote Control Unit

The RTC1 is a powerful MIDI-based control unit that adds new features and advanced capabilities to the popular DMP7, DMP7/D, and DMP11 digital mixing processors. This easy-to-use remote control unit allows centralized control of up to four DMP mixers, and adds “analog-like” control of EQ, pan, and special effect settings. The RTC1 also provides new features to the DMP family, such as fader grouping, single-control cross fade, and linked stereo

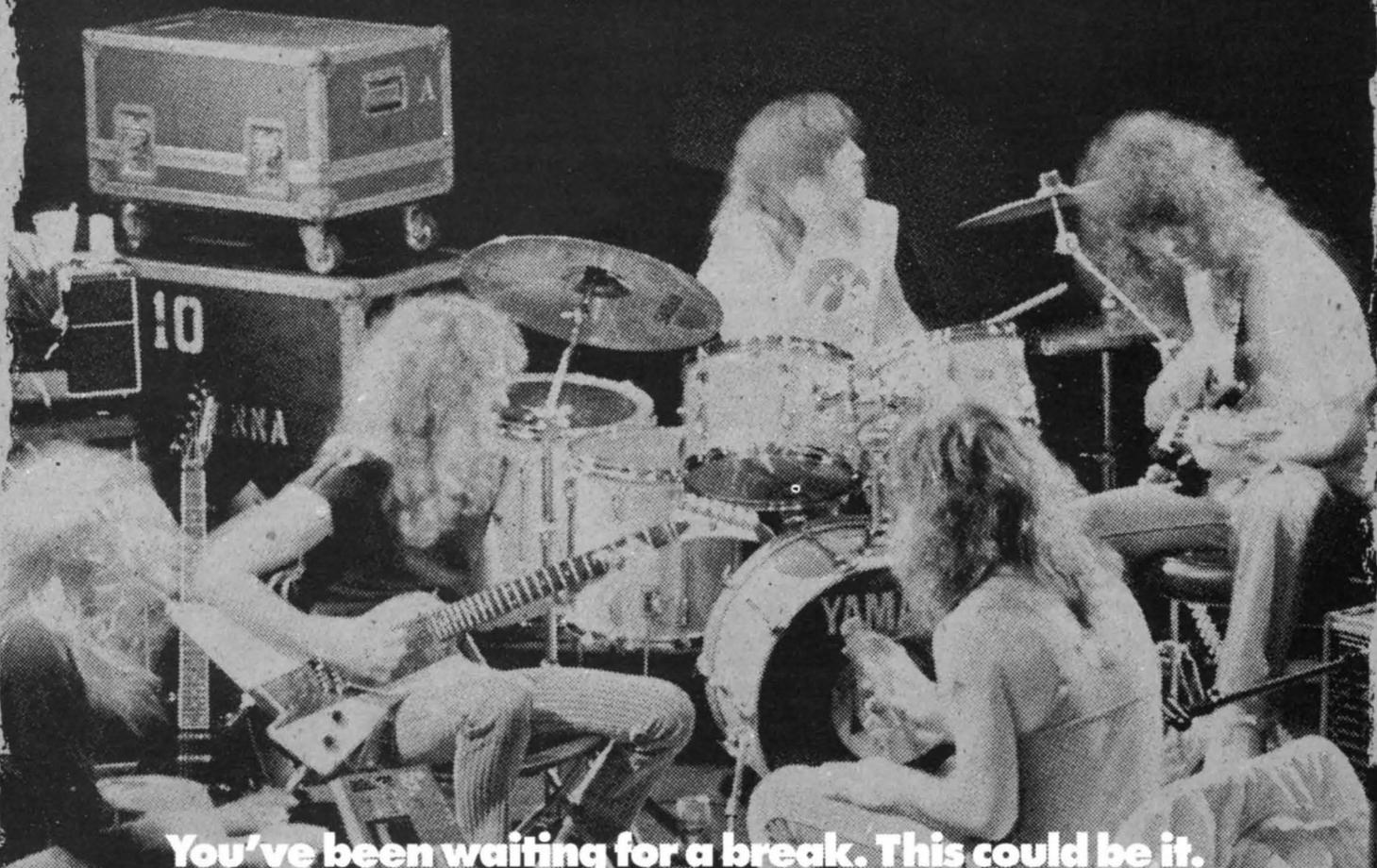
input channel operation.

Four individual push-button switches assign the RTC1 panel controls to the selected mixer or mixers. The large four-line LCD display clearly shows the user several effects parameters and settings simultaneously. Four additional rotary controls may be assigned to any parameter in the mix. On-board memory stores up to 50 sets of control assignments.

The RTC1’s rotary encoders give the user “analog-like” control, just like the rotary pots in conventional mixing consoles. Separate Q, FREQ, and GAIN controls are provided for the High, Mid, and Low EQ bands, and settings are displayed in the large LCD readout. The RTC1 thus provides the user with the ease of EQ setting found in analog consoles, yet retains the precision, the reset capability, and the pristine sound of digital equalization. The assignable rotary controls extend this innovation to every mix and processing parameter in the DMP mixers.

The RTC1 is now available from Yamaha Professional Audio dealers, at a suggested retail price of \$1,095.00. For more information, write to: Yamaha Corporation of America, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622-6600.

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**SOUNDCHECK**  
THE YAMAHA INTERNATIONAL  
ROCK MUSIC COMPETITION

1989 Yamaha Corporation Of America

# Goldsmith

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don't think anyone knows fully what any one of these synthesizers can do. By the time you get somewhat familiar with it, something new has come along, and we have to go to that one.

Like the DX7 II FD. I mean, I haven't begun to really find out what one does with that because I'm more familiar with the old DX7. I don't have as much to learn on the original DX7.

*And the new features haven't pulled you away from the old DX7?*

No. The stereo doesn't really excite me, and I haven't heard that much difference in sounds. I guess the old DX7 sounds recorded onto the new DX7 sound better. I haven't noticed all of the fine differences.

*Well, in order to discover all that, you've got to make a conscious effort to sit down and test it all out.*

I haven't gotten to the fractional scaling, the performance mode, and all that. I've gained a little bit more knowledge of it now that the OpCope program for the DX7 family programs the DX7 II as well. I've gotten to the point where I need the computer to run most of these things.

The RX5 is a good machine. I used the RX5 snare drum on *Extreme Prejudice*. I used Simmons Drums on the score, and for some reason, I didn't like the snare drum sound, so I ran it into the RX5 and I had the drummer trigger the RX5 drum sound, and then I got it adjusted a little bit, and that's what I used. But I love being able to change the parameters and being able to get those effects on it; it's really great.

*What do you think of the TX16W?*

On paper, it looks like the top-of-the-line end-all sampler. On the other hand, I don't make my own samples anymore, so it could be that the samples I've been given aren't very good samples. The sound quality is good; it's just that I don't like many of the samples I have. I wouldn't write the unit off, though. Frankly, I think that the main problem with the unit is that so many people are like me: A lot of people go into a store and buy a synthesizer not because of what's inside but because of what they hear. They push a button, and say, "Oh, I like that patch; this is a great synthesizer." It sounds sort of simplistic, but I think that's the way a lot of people buy synthesizers.

*It's hard to become familiar with an instrument in just a short time at a store.*

That's true. And the older analog units, I think anybody will agree, were a hell of a lot easier to program than the all-digital stuff of today. For one thing, you had all of those knobs in front of you, and you didn't have to go through screens and sub-screens. You may not have known exactly what a knob was going to do, but you could grab this one and get one change, and grab that one and get another change—you used to be able to go by your ear.

But that's why the computer is making it easier again. When I want to tweak a sound or something, it's so much easier to do it with the computer. You can see the DX7 envelopes. It took me a long time to figure out the levels, the rates and all of that on the DX7, but now I think I understand it because of the program.