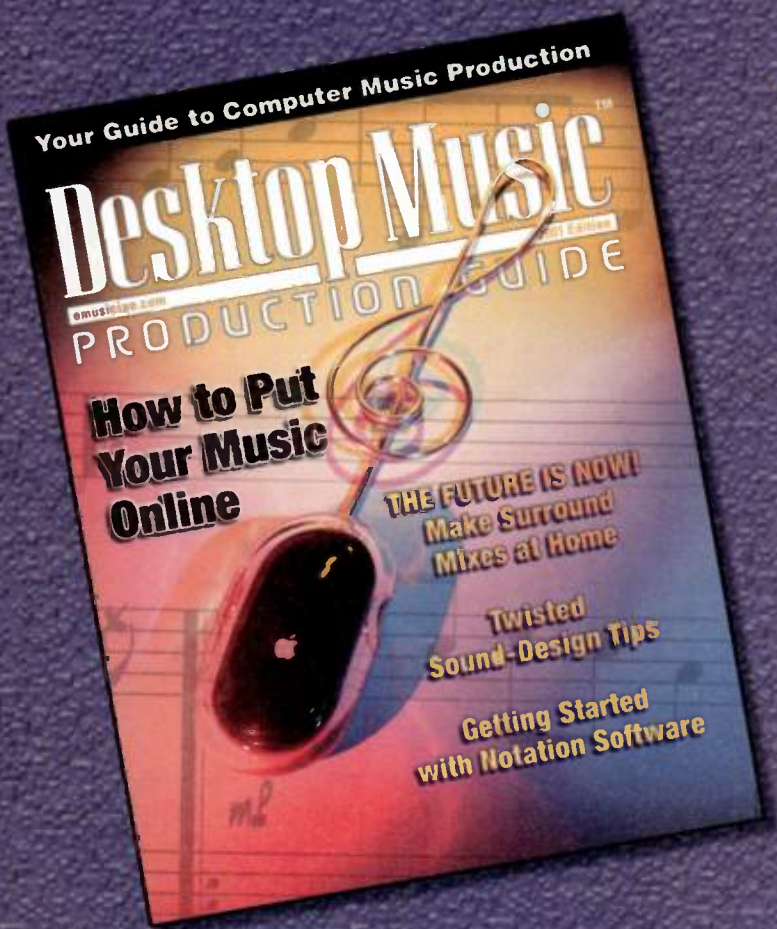
Electronic Musician®

Desktop Music®

2001 Edition

PRODUCTION GUIDE



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SYNTHESIZER & SAMPLER MODULES

MANUFACTURER	PRODUCT	POLYPHONY / MULTITIMBRAL PARTS	WAVEFORM MEMORY ROM / RAM	FILTER TYPES / RESONANCE	SINGLE PROGRAMS ROM / RAM	MULTITIMBRAL PERFORMANCES ROM / RAM	PORTAMENTO	GM / GS / XG COMPATIBLE	BUILT-IN COMPUTER INTERFACE / TYPE	DISK DRIVE TYPE	# AND TYPE OF CONTROLLER INPUTS
Access	Virus B	24/16	N/A	24/12 dB LP; BP; HP; BS	512	128	Yes	No	0	N/A	N/A
Akai	S5000 MIDI Stereo Digital Sampler	64 (exp to 128)/ 32	0/256 MB	26/Yes	0/0	0/No	Yes	No	Yes/SCSI	Floppy	0
Akai	S6000 MIDI Stereo Digital Sampler	64 (exp to 128)/ 32	0/256 MB	26/Yes	RAM dep.	0/128	Yes	No	SCSI-2 x2	Floppy	0
Akai	MPC2000XL MIDI Production Center	32/1	0 (exp to 16 MB)/ 2 (exp to 32)	LP/Yes	0/24	0/1	No	No	Yes/SCSI	Floppy	(1) assignable MIDI footswitch
Alesis	QSR Synth Module	64/16	16/16MB (exp w/flash RAM)	LP/No	512/128	400/100	Yes	Y/N/N	Yes/serial	N/A	0
Alesis	DM5 Drum Module	16/1	4 MB	LP/No	0/20 1/4" inputs	0/0 No	No No	Y/N/N N/A	No No	N/A N/A	12 trigger (4) 1/4"
Alesis	DM Pro	64/16	16/8 MB w/flash cards	LP/No	1,536/128	0/64	Yes	Y/N/N	No	N/A	Pedal; 16 trigger 1/4" inputs
Doepfer	A-100 Modular Synthesizer	Depends on configuration	No	6	None	None	With port. module	Optional	No	N/A	N/A
E-mu Systems	Proteus 2000	128/32	32 MB (exp to 128)	17/Yes	1024/512	64	Yes	No	No	N/A	0
E-mu Systems	E6400 Ultra	64 (exp to 128)/ 16 (exp to 32)	0 MB (exp to 64)/ 16 (exp to 128)	21/Yes	1,000/1,000	1	Yes	No	Yes/SCSI	Yes	0
E-mu Systems	E-Synth Ultra	64 (exp 128)/ 16 (exp to 32)	0 MB (exp to 64)/ 16 (exp to 128)	N/A	1,000/1,000	1	Yes	No	Yes/SCSI	Floppy	0
E-mu Systems	ESI2000	64/16	0/4 MB (ex. to 128)	21/Yes	0/256	1	Yes	No	Yes/SCSI	Floppy	0
E-mu Systems	Xtreme Lead-1	64/16	32 MB (exp to 64)/0	12th order/Yes	512/512	64	Yes	No	No	N/A	N/A
E-mu Systems	B-3	64/16	32 MB (exp to 64)/0	17/Yes	384/512	64	Yes	No	N/A	N/A	N/A
E-mu Systems	Virtuoso 2000	128/32	64 MB (exp to 128)/0	17/Yes	512/512	64	Yes	No	N/A	N/A	N/A
E-mu Systems	Mo'Phatt	64 (exp to 128)/ 16 (exp to 32)	32 MB (exp to 64)/0	12th order/Yes	512/512	64	Yes	No	N/A	N/A	N/A
E-mu Systems	E5000 Ultra	64/16 (exp to 32)	0 MB (exp to 64)/ 4 (exp to 128)	21/Yes	1,000/1,000	1	Yes	No	Yes/SCSI	None	N/A
E-mu Systems	E5000 Ultra Turbo	64/16 (exp to 32)	32 MB (exp to 64)/ 64 (expto 128)	21/Yes	1,000/1,000	1	Yes	No	Yes/SCSI	IDE	N/A
E-mu Systems	Planet Earth	64/16 (exp to 128)	32 MB (ex. to 64/128)	50/Yes	512/512	64	Yes	No	No	N/A	0
E-mu Systems	XL-1 Turbo	128/32	32 (exp to 128)	50/Yes	512/512	64	Yes	No	No	N/A	0
E-mu Systems	Turbo Phatt	128/32	32 (exp to 128)	50/Yes	512/512	64	Yes	No	No	N/A	0
Edirol	ED SC-8850 Sound Canvas	128/64	64 MB	Yes	1,640	N/A	Yes	Y/Y/N	Yes/serial, USB; Mac/PC	N/A	N/A
Edirol	ED SC-8820 Sound Canvas	64/32	64 MB	Yes	1,608	N/A	Yes	Y/Y/N	Yes/serial, USB; Mac/PC	N/A	N/A
Edirol	ED SC-D70	64/32	64 MB	LP/Yes	1,608	0	Yes	Y/Y/N	Y/USB	N/A	0
Ensoniq	ASR-X Pro	32/16	2/ 2 MB (exp to 66)	LP; HP, BP/Yes	0/0	1	Yes	No	Yes/SCSI	None	0
Future Retro	FR-777	1	None	LP; HP/Yes	N/A	NA	Yes	No	N/A	None	0
Korg	EA-1	2/2	None	LP/Yes	0/448	0/256	Yes	No	No	None	N/A
Korg	ES-1	10/N/A	N/A	LP/resonant algorithm	0/100	N/A	No	No	No	SmartMedia	N/A
Korg	N1R	64/32	18 MB	Reso filter FX algorithm/Y	1,471/200	32	Yes	Yes	Yes/PCIF serial PC/Mac	N/A	0
Korg	MS2000R	4/1	DWGS in ROM	LP; BP; HP	0/128	N/A	Yes	No	N/A	None	(2) assignable pedal and switches
Korg	Triton Rack	62/16	32 MB in ROM up to 128 in RAM	HP, LP	Yes	0/640	0/512	Yes	N/A	3.5" HD/DD	N/A

# OF SEQUENCER TRACKS/PPQN	SEQUENCER MEMORY	TYPES OF QUANTIZATION	ARPEGGIATOR	# OF EFFECTS PROCESSORS/ EFFECTS	# OF AUDIO OUTPUTS/TYPE	SPECIAL FEATURES	OPTIONS	PRICE
N/A	N/A	N/A	Yes	1/82	(6) mono; (3) stereo	Stereo filter inputs;, 3 LFOs, 2 env; sine, triangle, saw, pulse, plus 64 waveshape	Virus KB (keyboard version)	$1,795
N/A	N/A	N/A	Yes	0 (exp to 4)/ 0 (exp to 50)	(8 exp to 16) 1/4"	FAT16/FAT32 native discs; word clck; sample format; AK.Sys USB networking; playsheet mode; standard MIDI file player	8 output exp; 64 voice exp; SampleVerb II FX Ak.Sys networking	$2,099–$2999
N/A	N/A	N/A	Yes	0 (exp to 4)	(2) XLR; (16) 1/4"	Removable front control panel; WAV native sample format; AK.Sys USB networking	64 voice exp; SampleVerb II FX; 8/16 ADAT optical I/O	$2,999–$3,799
64/96	300,000 notes	8th; 8th triplet; 16th triplet; 32nd	No	0 (exp to 4)/ 0 (exp to 50)	(2 exp to 10) 1/4"	Reads S1000/S3000/WAV files; velcty & pressure-sensitive pads	8 output exp; SampleVerb FX; SMPTE board; 8 MB flash ROM	$1,649; $1,749
16/480	8 MB/50 seq per flash card	N/A	No	1/8	(4) 1/4"	Add sequences/samples via flash RAM; diigtal outs; CD-ROM		$649
N/A	N/A	N/A	N/A	N/A	N/A			$449
16/480	8 MB/50 seq on PCMCIA card	N/A	No	1/5	(6) 1/4" TRS	16 trigger ins; 2 RCA ins		$899
N/A	N/A	With quantize module	No	Depends on configuration	Depends on configuration	100% analog modular synth system; 50 different modules available		$100 and up
N/A	N/A	N/A	No	2/30+	(6) 1/4"; S/PDIF	32 MIDI channels; 4 real-time controls; plays back custom ROMs	Additional 16 MB and 32 MB sound ROMs	$995
48/480	4 MB	1/4–1/64 w/swing	No	2/60	(8 exp to 16) ADAT	Digital modular synthesis	MIDI x 2 exp; digital I/O; ASCII; resampling; ROM brd; ADAT I/O	$2,295
48/480	4 MB	1/4–1/64 w/swing	No	2/60	(8 exp to 16) ADAT	Digital modular synthesis; resampling	RFX-32 (32-bit FX); flsh ROM brd; ADAT I/O; anlg out exp	$2,895
No	No	N/A	No	Optional	(4 exp to 8)	Trigger mode; SCSI	Turbo option kit adds outputs & effects	$749
N/A	N/A	N/A	Yes	2/30+	(2) 1/4"	4 real-time control knobs; SuperBEATs mod	Expansion sound ROMs	$995
N/A	N/A	N/A	N/A	2/30+	(2) 1/4"	Plays custom ROMs	Expansion sound ROMs; opt. 6 analog outs, S/PDIF	$89
N/A	N/A	N/A	N/A	2/30+	(6) analog; S/PDIF	4 real-time contol knobs; plays back custom sound ROMs	Expansion sound ROMs	$1,395
N/A	N/A	N/A	Yes	2/30+	(2) analog	Control knobs; plays back custom sound ROMs created with E4 Ultra Samplers	Expansion sound ROMs; polyphony/output/MIDI/ROM expansion hardware upgrade	$995
48/480	4 MB	1/4–1/64 w/swing	N/A	2/60	(4) analog; ADAT;,AES/EBU	Digital modular synthesis; burns custom flash ROMs	32 channel RFX-32 FX/Mixer card; MIDI, wordclock, S/PDIF	$1,795
48/480	4 MB	1/4–1/64 w/swing	N/A	2/60	(4 exp to 12 analog); ADAT, AES/EBU, S/PDIF	16 MB Orbit/Phatt sound ROM; int. hard drive; digital modular synthesis	32 channel RFX-32 FX/Mixer; ADAT, AES/EBU,S/PDIF options	$2,395
N/A	N/A	N/A	Yes	2/30+	(2) analog (expand to 6+ S/PDIF)	Expandable sound sets; real-time controls SuperBEATS mode	12 pole filters	$995
N/A	N/A	N/A	Yes	2/30+	(6) analog, (1) S/PDIF	Expandable sound sets; 12 pole filters; SuperBEATS mode	Real-time controls	$1,295
N/A	N/A	N/A	Yes	2/30+	(6) analog;, (1) SPIDF	Expandable sound set; 12 pole filters; SuperBEATS mode	4 real-time controls	$1,295
N/A	N/A	N/A	N/A	1/64	(4) RCA	Includes hybrid CD with GSAE editing software		$1,195
N/A	N/A	N/A	N/A	1/64	(2) RCA	Hybrid CD-ROM; GSAE sound editing software; power supplied by USB bus		$595
N/A	N/A	N/A	No	1/64	(6) RCA	24-bit USB audio/MIDI interface		$798
16/384	70,000 notes (exp to 140,000)	Adjustable	No	2/40	(1) headphone			$1,695
256	256 patterns/ 16 songs	16th note; 3/4; 4/4	No	1/2 (overdrive)	(1) 1/4"	Pure analog synthesis; external audio input; MIDI to CV converter		$777
2 / NA	64 steps	12T; 16; swing 16; 32	No	2/2	(2) 1/4"; (1) heaphone	Audio in for gating audio with patterns; realtime control		$499
10/NA	Up to 64 steps per part	16; 16 swing; 32; triplets	No	2/12	(2) 1/4"	Resample; time motion sequencer; Smartmedia storage		$599
No	N/A	N/A	Yes	2/48	(4) 1/4"	4 assignable real-time control knobs	Serial cable/driver package; 32 parts	$850
3 part motion sequencer	16 steps	N/A	Yes	2/7	(2) 1/4"; RCA	16 band vocoder; 3-part motion sequencer		$800
N/A	NA	N/A	Dual Polyphonic	8/102 inserts; 89 master; 3-band stereo EQ	(6) 1/4" outs; headphone jack	Sampler up to 96 MB; user expandable upgrades	SCSI; physical modeling; sound expansion; MLAN	$2,500

SYNTHESIZER & SAMPLER MODULES

MANUFACTURER	PRODUCT	POLYPHONY / MULTITIMBRAL PARTS	WAVEFORM MEMORY ROM / RAM	FILTER TYPES / RESONANCE	SINGLE PROGRAMS ROM / RAM	MULTITIMBRAL PERFORMANCES ROM / RAM	PORTAMENTO	GM / GS / XG COMPATIBLE	BUILT-IN COMPUTER INTERFACE / TYPE	DISK DRIVE TYPE	# AND TYPE OF CONTROLLER INPUTS
Kurzweil	Micropiano	32/0	6/0	N/A	32/0	0/0	No	No	No	N/A	0
Kurzweil	K2600R/K2600RS	48/16	8 MB ROM (exp to 128)	LP; HP; BP; notch; AP/Yes	200/1,000	100/1,000	Yes	Y/N/N	Yes/SCSI	Floppy	0
Nord	Nord Rack2	16/4	N/A	24dB BP; HP; LP/Yes	99	100	Yes	N/A	N/A	N/A	(1) pedal switch; exp pedal
Nord	Nord Modular Rack	16/4	N/A	15 types	500	N/A	Yes	N/A	No	N/A	(1) switch; (1) exp pedal
Nord	Nord Micro Modular	4/1	N/A	15 types	99	N/A	Yes	N/A	No	N/A	0
Quasimidi	Polymorph	16/4	N/A	24 dB LP;+ 12 dB HP	0/128	0/64	Yes	No	No	N/A	0
Quasimidi	QM-309 Rave-O-Lution	17/5	N/A	24 dB/12 dB LP	0/384	N/A	Yes	No	No	N/A	(1) footswitch
Roland	GR-33 Guitar Synthesizer	48/1	384 tones ROM	Digital	N/A	256: 128/128	Yes	No	No	N/A	2
Roland	JV-1010	64/16	32 MB (exp to 48)	LP; BP; HP; peaking/Yes	895/128	64 Prest/ 32 User	Yes	Y/N/N	Yes/serial; Mac/PC	N/A	0
Roland	JV-1080	64/16	16 (exp to 80)	LP; BP; HP; peaking/Yes	512/128	64/32	Yes	Y/N/N	No	N/A	0
Roland	JP-8080	10/2	N/A	LP; BP; HP; peaking/Yes	384/128	192/64	Yes	No	No	N/A	0
Roland	SP-303	8	16/internal	16/card	Yes	Yes	No	No	No	SmartMedia	0
Roland	SP-808 Exemix-studio	4/ N/A 1	250 MB RAM cached from disk	HP; LP; BP	1,024 RAM	None	No	No	No	ZIP 250 MB (2x)	(6) assign knobs; D-Beam
Roland	VP-9000	6/6	8 MB (exp to 136)	LP/Yes	1,024 RAM	1 RAM	Yes	No	No	ZIP 250 MB	0
Roland	XV-3080	128/16	64MB ROM (exp to 256)	LP; BP; HP; peaking/Yes	1,024/128	64/64	Yes	Y/N/N	N/A	SmartMedia	0
Roland	XV-5080	128/32	64MB ROM (exp to 512)	LP; BP; HP; peaking/Yes	1,152/128	64/64	Yes	Y/N/N	No	128 MB SmartMedia; SCSI	N/A
Studio Electronics	SE-1	Monophonic	N/A	12 dB LP; BP Moog 24 dB	198 RAM	0	Yes	No	No	None	0
Studio Electronics	Omega 2	2/2	N/A	12 dB LP; BP; HP; BR; Moog 24 dB	256 RAM	128 RAM	Yes	No	No	None	0
Studio Electronics	ATC-1	Monophonic	N/A	Moog 24 dB	512 RAM	0	Yes	No	No	None	CV/gate I/O
Studio Logic/Fatar	Blue Chip OX-7 Virtual Tonewheel Drawbar Module	24/3	N/A	N/A	99/99	N/A	No	No	No	N/A	0
Studio Logic/Fatar	Blue Chip Baby B Virtual Tonewheel Drawbar Module	24/3	N/A	N/A	99/99	N/A	No	No	No	N/A	0
Voce	V5	Full/ 3	N/A	N/A	0/7	0/3	None	No	No	None	0
Voce	Electric Piano	32/3	N/A	BP/Yes	32/0	3/0	No	No	No	None	0
Voce	Micro B II	32/3	N/A	HP; LP/Yes	36/0	3/0	None	No	No	None	0
Wiard Synthesizer	Wiard Modular System	1/1	N/A	Multi-type resonant	N/A	N/A	Patchable		N/A	N/A	N/A
Yamaha	AN200	5/32	N/A	LP; HP; BP; BE/Y	256/128	256/128	Yes	No	No	No	18 knobs
Yamaha	A4000	64/16	0/4 (exp to 128)	16/Yes	0/128	0/128	Yes	No	Yes/SCSI	Floppy	(5) assign knobs
Yamaha	A5000	128/32	0/4 (exp to 128)	16/Yes	0/128	0/128	Yes	No	Yes/SCSI	Floppy	(5) assign knobs
Yamaha	CS6R	64/19	28.8	12/Yes	Same as CS6x	Same as CS6x	Yes	Optional	Yes/serial; Mac/PC	SmartMedia	Breath controller
Yamaha	DX200	16/32	N/A	LP; HP; BP; BE/Y	256/128	256/128	Yes	No	Y/Serial	No	18 knobs
Yamaha	EX5R	128/16	16/1	8/Yes	256/256	0/128	Yes	Optional	N/A	Floppy	Breath controller
Yamaha	FS1R	32/4	N/A	6/Yes	1,408/128	384/128	Yes	No	No	No	N/A
Yamaha	QY700	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A
Yamaha	QY70	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A
Yamaha	SU700	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A	N/A

# OF SEQUENCER TRACKS/PPQN	SEQUENCER MEMORY	TYPES OF QUANTIZATION	ARPEGGIATOR	# OF EFFECTS PROCESSORS/ EFFECTS	# OF AUDIO OUTPUTS/TYPE	SPECIAL FEATURES	OPTIONS	PRICE
N/A	N/A	N/A	No	1/16	(2) 1/4"	Can link two mics for 64-voice polyphony		$550
32/768	30,000 notes	Groove; reference; swing; shift	Yes	1/37	(8) 1/4"; AES/EBU; XLR; optical KDS	Reads smpls frm Akai, Roland, Ensoniq, AIFF, Wave	128 MB smpl RAM; 28 MB smpl ROM	$4,475/$5,250
N/A	N/A	N/A	Yes	N/A	(4) 1/4"	Analog modeling synth	PCMCIA cards w/1200 presets, 400 performances	$1,399
N/A	16 steps	N/A	Yes	4/30 (+)	(4) 1/4"	Mac & PC editor allows you to build a synth	16 voice expander board	$1,899
N/A	N/A	N/A	Yes	1/30 (+)	(2) 1/4"	Mac & PC editor allows you to build a synth	3, 000(+) presets avail from co. website	$749
4	N/A	N/A	N/A	16	4	Analog-style sequencer; AES synthesis; 2 audio inputs		$1,199
5/24	20,000 events	N/A	No	3/17	(4) RCA	Full synth features on drum sounds; 2 audio inputs	Sound expansions for drums and synth sounds	$899
NA	NA	N/A	Yes	Chorus/flange; delay/rev	(2) 1/4"	Guitar synthesizer 40 multi FX + arpeggiator; harmonist	Built-in expression & control pedal; GR carrying bag	$695
NA	NA	N/A	N/A	3/49	(2) RCA; headphones	Built-in SR-JV80 session exp board; sound diver editing software bundled	1 SR-JV80 expansion slot	$595
N/A	N/A	N/A	N/A	3/49	L/R x3 (str); headphone	4 exp slots; LFO sync-to-MIDI	SR-JV80 exp boards; M-512E data card	$1,195
1/96	23,000 notes	Grid	Yes	3/19	L/R (str); headphone	38 knobs/sliders; line/mic ins; 12-bnd frmnt fltr/vcdr/robot; vcl morph	SmartMedia card	$1,595
4	17,000	Grid	No	26	(2) RCA	Memory expander		$395
8	N/A	Grid	No	1/149	(4) RCA	DBeam, COSM effects (incl. guitar amps, microphone modeling)	SP-808-OP1; SP-808-OP2 expansion boards	$1,695
N/A	N/A	N/A	N/A	3/40	(6) 1/4"	Variphrase processsing	DR-10/ DR-20	$3,295
N/A	N/A	N/A	N/A	3/69	(4) 1/4"	SmartMedia; new effects processor; sync exp bal beatloop to MIDI clock	4x SR-JV80 exp& 2x SRX exp (64MB boards)	$1,795
N/A	N/A	N/A	No	5/96	(4) 1/4"	Wordclock; R-BUS; supports Akai, Roland, AIFF, WAV libraries	SmartMedia; SCSI CD-R; SCSI HD or removable media	$2,495
0	0	0	No	0	(1) 1/4"	3 LFOs, 4 envelopes, ring mod;osc sync; 3 voltage-controlled osc. per voice	External input	$1,699
0	0	0	Yes	0	RCA; 1/4"	3 LFOs; 3 envelopes; sub osc; 2 ext ins; 2 voltage-controlled osc. per voice	2 addtional filter slots per voice	$1,995
0	0	0	No	0	(1) 1/4"	2 LFOs; 3 envelopes; ext in; plug in filters ; 2 voltage-controlled osc. per voice	Additional filter cartridges: SEM,303,2600	$1,099
N/A	N/A	N/A	N/A	1/6	(2) 1/4"	Drawbars; rotorsound simulator	VP-26 pedal $39; PS-10 footswitch $18	$1,595
N/A	N/A	N/A	N/A	1/6	(2) 1/4"	Drawbars; rotorsound simulator	VP-26 pedal $39.95/ PS-10 footswitch $18	$895
0/0	0	No	No	1/4	(1) 1/4"	Drawbar organ synthesizer	Spin II rotary speaker simulator	$975
0/0	0	No	No	1/4	(2) 1/4"	Digital model based with analog effects	None	$579
0/0	0	No	No	1/4	(2) 1/4"	Half-rack organ simulator	Spin II rotary speaker simulator	$579
N/A	N/A	N/A	N/A	N/A	N/A	Complete modular system	Built-to-order, several modules available.	$2,999 and up
4/4	2,048	16th swing	No	1/24	(2) 1/4"	2 scene memories/voice	4 track free RG/voice	$630
16/480	100,000+ notes	N/A	N/A	3/96	(4) 1/4"	Loop divide; loop remix, CD burning capable (data and Red Book audio)	Output expander, internal HDD, zip drive	$1,595
16/480	100,000+ notes	N/A	N/A	6/96	(4) 1/4"	Loop divide; loop remix; CD burning	Output expander, internal HDD; zip drive option	$2,295
16/480	N/A	N/A	Yes	2/130	(4) 1/4"	Phrase clip sampling 4 MB; stereo A/D input, scenes	PLG series expansion boards x 2, 6 types available	$1,495
4	2,048	16th/swing	No	1/24	(2) 1/4"	2 scenes/voice	4-track free LG	$630
16/480	30,000 notes	Percentage; destructive; non-destructive; groove	Yes	2/122	(4) 1/4"	2x A/D inputs; 5 types of extended synthesis; sampling	Flash ROM 8/16 MB; SCSI	$2,195
N/A	N/A	N/A	N/A	1/40	(4) 1/4"	Formant sequences; DX7 compatable		$1,000
48/480	N/A	Input; percentage; groove templates; user definable	N/A	1/3	N/A	XG tone generator w/ 480 normal voices; 11 drum kits; 3.5" FDD		$1,495
24/480	N/A	Input; percentage; groove templates	N/A	1/3	N/A	Hand-held seq./tone generator	Battery powered; music database w/4,167 phrases	$600
42/480	N/A	Step; real; pattern; linear	N/A	N/A	N/A	Auto looping and auto sync of audio to MIDI or MTC; 20 songs		$1,495

SYNCHRONIZERS

MANUFACTURER	PRODUCT	SMPTE	JAM SYNC	FREEWHEELING	MTC/MMC	SPECIAL FEATURES	PRICE
Aardvark	AardSync II-Master Clock Generator	No	N/A	N/A	N/A	WC, 2X WC, 256 superclock; AES/EBU;' video blackburst lock; low-jitter clock; eliminates digital clicks; NTSC/PAL; optional video blackburst generator $495	$1,795
Aardvark	Sync DA-Word Clock Distribution Amp	No	N/A	N/A	No/No	Generate wordclock from AES/EBU; 6 wordclock outputs; supports Digidesign 256 superclock	$845
Aardvark	AES/EBU Distribution Amp	No	N/A	N/A	No/No	1x6 distribution amp	$645
Aardvark	TimeSync II-Universal Time Code Synchronization	Yes	Yes	Yes	Yes/Yes	Derives ultra-low jitter wordclock/Superclock & MTC from LTC/VITC in; locks in 4 forms	$1,295
Alesis	BRC Master Remote	Yes	No	Yes	Yes/Yes	Connects directly to ADAT; video in; wordclock I/O	$1,499
JLCooper	dataSYNC2	Yes	No	No	Yes/Yes	ADAT interface; drives sequencers/DAWs w/o track; MTC/SMPTE/MIDI clock with song position pointer; bi-dir MIDI	$300
JLCooper	MMC/9 Pin	Yes	Yes	Yes	Yes/Yes	9-pin control; records from MMC-sequencers, DAWs	$500
JLCooper	PPS-2	Yes	Yes	Yes	Yes/No	Reads/gens SMPTE tc & converts SMPTE to MTC or DTL	$170
Mark of the Unicorn	Digital Timepiece	Yes (LTC; VITC)	Yes	Yes	Yes/Yes	MTC; Sony 9-pin; video, ADAT, DA-88, S/PDIF sync; 0.1% pull-up/down sample rates (44.1/48 kHz)	$995
Midiman	Syncman	Yes	Yes	No	Yes/No	Syncs sequencers to tape or video	$200
Soundscape	Rosendahl Nanosync	Yes	N/A	N/A	N/A	Low-jitter master clock generator, S/PDIF, AES, (6) word/super clock outputs; (4) video outputs; video sync generator	$1,000
Soundscape	Rosendahl WIF2	Yes (LTC; VITC)	Yes	Yes	Yes/No	Word clock/superclock out from LTC or video onput; film/video pull up/down; MTC out for VITC jog/shuttle	$775
Soundscape	Rosendahl MIF3	Yes (LTC)	Yes	Yes	Yes/No	Reclocks LTC for stable output from MTC input with jitter; LTC stationary frame output for MTC full frame	$399
Soundscape	Rosendahl BIF	Yes	Yes	Yes	Yes/Yes	Positional LTC and MTC full/MMC locate in jog/shuttle; LTC and MTC gen from bi-phase signals	$775
Soundscape	Rosendahl LIF	Yes	Yes	Yes	Yes/Yes	Generates LTC, MTC from LANC-embedded code; positional LTC and MTC full/MMC locate in jog/shuttle	$775
Soundscape	Rosendahl VIF	Yes	Yes	Yes	Yes/No	VITC-LTC/MTC and LTC-MTC conversion; MTC full frame output for VITC jog/shuttle	$649
Steinberg	Steinberg Time Lock Pro	Yes	Yes	Yes	Yes/Yes	Word clock or Digidesign superclock lockup less than 0.4 seconds	$999

REVIEWING THE BASICS: DELAY

From page 169—DELAY

through a single delay set between 30 and 50 ms with no feedback. This setting effectively clones the original signal. Pan the dry signal hard left and the delayed signal hard right, and voilá: faux stereo! If you don't hard-pan the signals, the stereo image collapses, and the original signal sounds as if it was doubled.

Many vocalists enjoy using a slapback delay of 75 ms or so, with no feedback. Slapback delay, or echo, was characteristic of the vocal sounds of early Jerry Lee Lewis and Elvis Presley recordings. Some contemporary artists, such as Bono of U2, use a more distinct echo of anywhere between 150 to 500 ms to produce an ethereal, lingering effect. The feedback control is set to repeat the signal a few times before fading away, and the delay time is often synchronized to the tempo of the song.

To sync delay time to a song tempo, simply divide the tempo in beats per minute (bpm) by 60 to get the number of beats per second. For example, 120 bpm = 2 bps. Then, divide 1,000 by the number of beats per second to calculate the number of milliseconds per beat. In this example, each beat takes up 500 ms, or half a second. If the beats are quarter notes, then eighth notes occupy 250 ms, eighth-note triplets occupy 166.67 ms, and sixteenth notes take 125 ms.

These calculations might seem a bit daunting to mathophobes. Fortunately, there are a several ways to avoid the horrors of relearning simple division. For one thing, you can probably find a small computer program that performs delay-time calculations, such as Studio-Pal for the Macintosh. If you own a delay unit with MIDI sync, it's even easier. These devices can synchronize their delay time to incoming MIDI Clock messages, which keeps the delay synchronized even if the tempo changes. Some delay processors, such as the Lexicon JamMan, offer a Tap Tempo feature; you tap the tempo on a front-panel button or footswitch, and the unit automatically calculates the delay time.

With a multitap delay synchronized to the tempo of your song, you can create some wild rhythmic effects by playing nothing more complicated than whole notes, half notes, or quarter notes. Suppose you have a 3-tap delay, and the song tempo is 120 bpm. Set the first tap to 125 ms, the second to 250 ms, and the third to 375 ms. As you play quarter notes, you'll hear sixteenth notes coming from the delay.

You can also set different delay times for each tap. If you want a dotted eighth-sixteenth pattern after each quarter note, set the first tap to 375 ms, the second tap to 500 ms, and the third tap to 875 ms. You can also set different output levels for each tap, which lets you establish an accent pattern. ■

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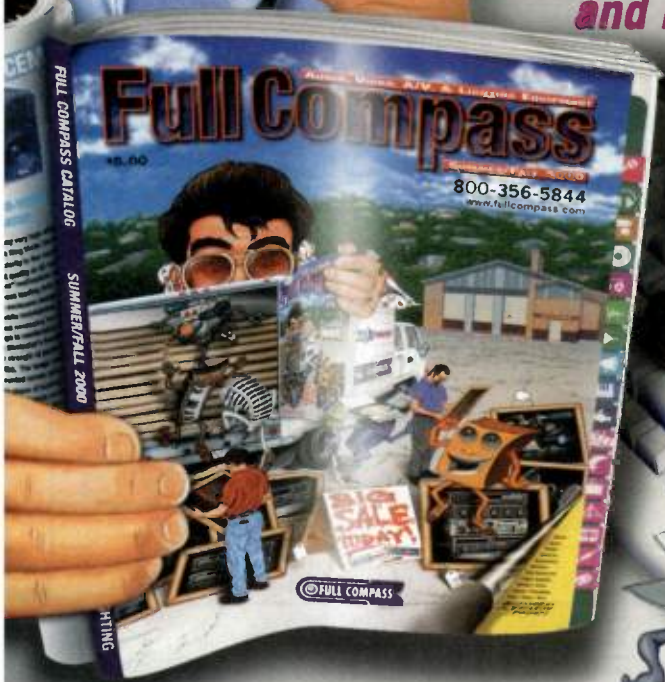

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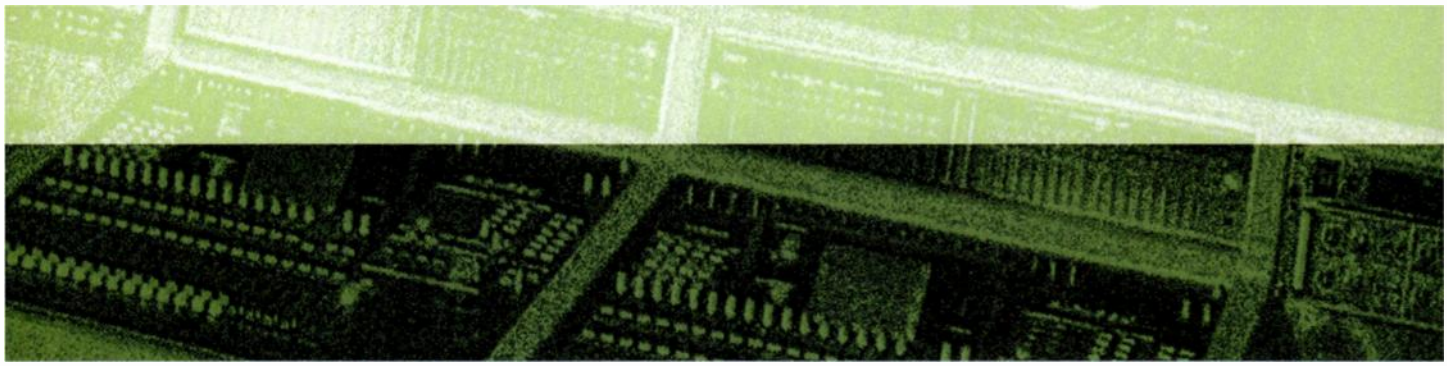

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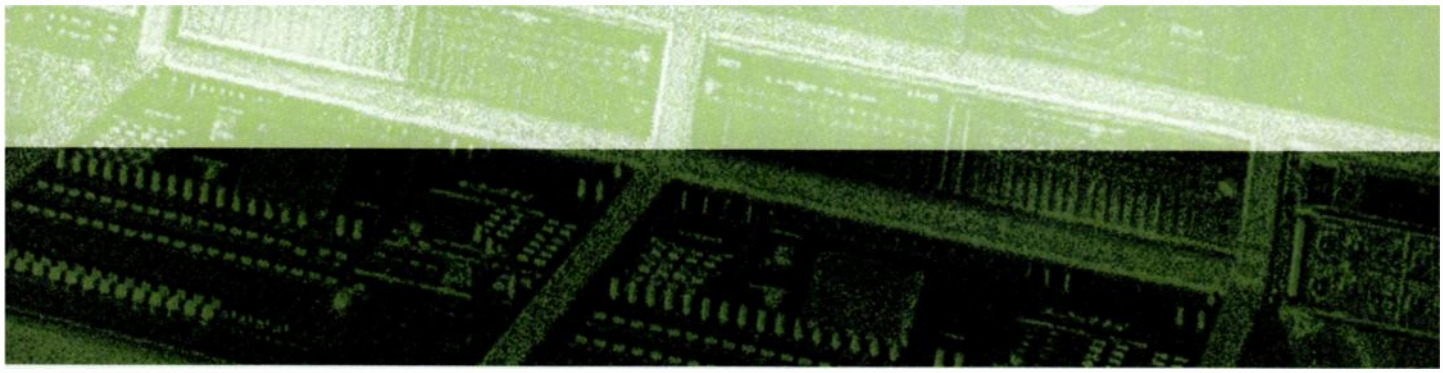

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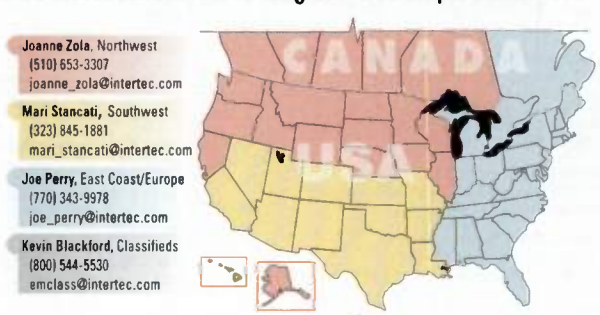

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ZEN AND THE ART OF SURROUND MIXING

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XDR™. The finest microphone preamplifier design ever built into a compact mixer.

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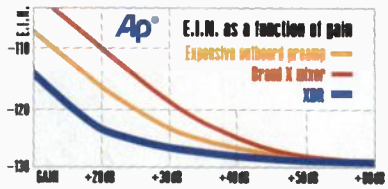
According to numerous cynical recording engineers, magazine reviewers and a lot of satisfied owners, we succeeded.

One reason is the advanced 2068 op-amps that are a foundation of the XDR design. They blow away our competitors' 4580 op-amps in terms of noise and distortion. Consider these real, measurable XDR™ (Extended Dynamic Range) microphone preamp specs:

- **0.0007% Total Harmonic Distortion**
- **0.0008% Intermodulation Distortion**
- **130dB total dynamic range to handle hot digital sources**
- **3Hz to 192kHz bandwidth**
- **Ultra-low mic E.I.N. at practical real-world gain levels**

By "practical" gain levels, we mean the +20 to +30dB normal operating range. The chart below compares E.I.N. for XDR™ ■, an Awesomely Expensive Outboard mic preamp ■ and a compact mixer ■ whose specs *claim* –129dBm E.I.N....but only at +60dB gain.



While XDR™ sonically rivals esoteric designs, it's no creampuff. We also built in protection against damage from shorts and "hot patching"...and the best RFI rejection of any compact mixer.

But don't just take our word for it...

Here are a few comments we gleaned from just *one week* of incoming VLZ™ PRO Series owner registration cards:

"Love the XDR mic preamps...clean and sweet!" C.H., Tampa FL

"A quality replacement for a far more expensive mixer brand." J.C., Arlington TX

"I was contemplating a couple of '——' outboard mic preamps ($2000+). The new XDR preamps let me make my next purchase a CD burner or an outboard processor instead." M.M., Miami FL

"Great product. XDR's are a great innovation and the value's unbeatable." J.K. Boise ID

"I hooked up my brand spankin' new AKG 3000 and all I can say is 'cool'." T.D., Waukesha WI

"I've been though numerous small consoles that were noisy. This one isn't, so it's a keeper!" J.F., Boca Raton FL

"Gorgeous preamps. Nice job, guys." J.C., Provo UT

"Nothing comes close to the quality/price." P.K., Spokane WA

"Excellent frequency response... excellent mic preamp. Worthy of our Neumann mics." C.M., Atascadero CA

"Killer mic preamps!" R.A., New York NY

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than any other competitive mixer line. These are just the awards we remembered to take photos of.

18 Tracks / 24-bit

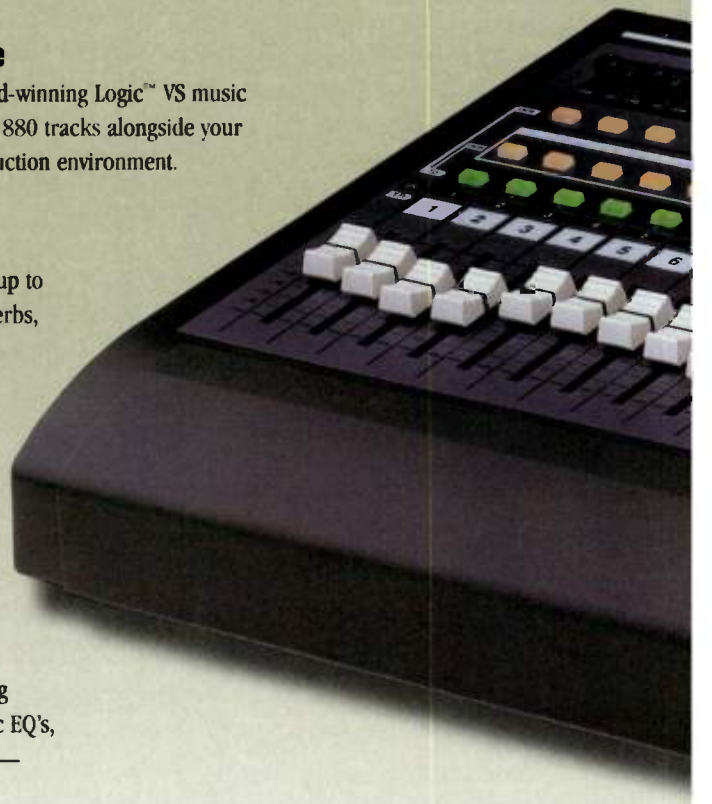
The VS-1880 offers full 18-track simultaneous playback, 288 Virtual Tracks for trying out all your ideas, and 24-bit A/D-D/A converters for superb, award-winning sound.

CD Burning

With the optional CD Recording System*, you can burn your own audio CD's and back up your projects to CD-R at blazing speeds.

FREE Emagic® Music Production Software

Every VS-1880 includes a FREE copy of Emagic's® award-winning Logic™ VS music production software, so you can edit and mix your VS-1880 tracks alongside your MIDI sequences in an integrated computer-based production environment.



Unrivaled Effects

By installing optional VS8F-2 Effect Expansion Boards*, you get up to 4 stereo/8 mono effects processors including studio-caliber reverbs, delays and chorusing, vintage effects simulations, and innovative COSM Guitar Amp, Microphone and Speaker Modeling.

Automated Digital Mixing

Perfect your mixes with the VS-1880's built-in 28-channel digital mixer, which includes time-saving dynamic automation capabilities and routing options galore.

Mastering Tools

The Mastering Tool Kit* gives you professional CD mastering tools including split-frequency compression, full parametric EQ's, extensive dynamics control and preset mastering templates—cleanly integrated into the V-Studio recording environment.

Industry-Leading Support

Answers to any operational questions you may have—from day one through day one million—are never far away, thanks to Roland's extensive product support team.

Proven Performance

When you buy a VS-1880, you're buying into a proven digital studio technology used by nearly 200,000 musicians, engineers and producers worldwide.

**Optional Effects Expansion Boards and CD Recording System sold separately. CD Mastering and "Mastering Tool Kit" require these optional purchases.*

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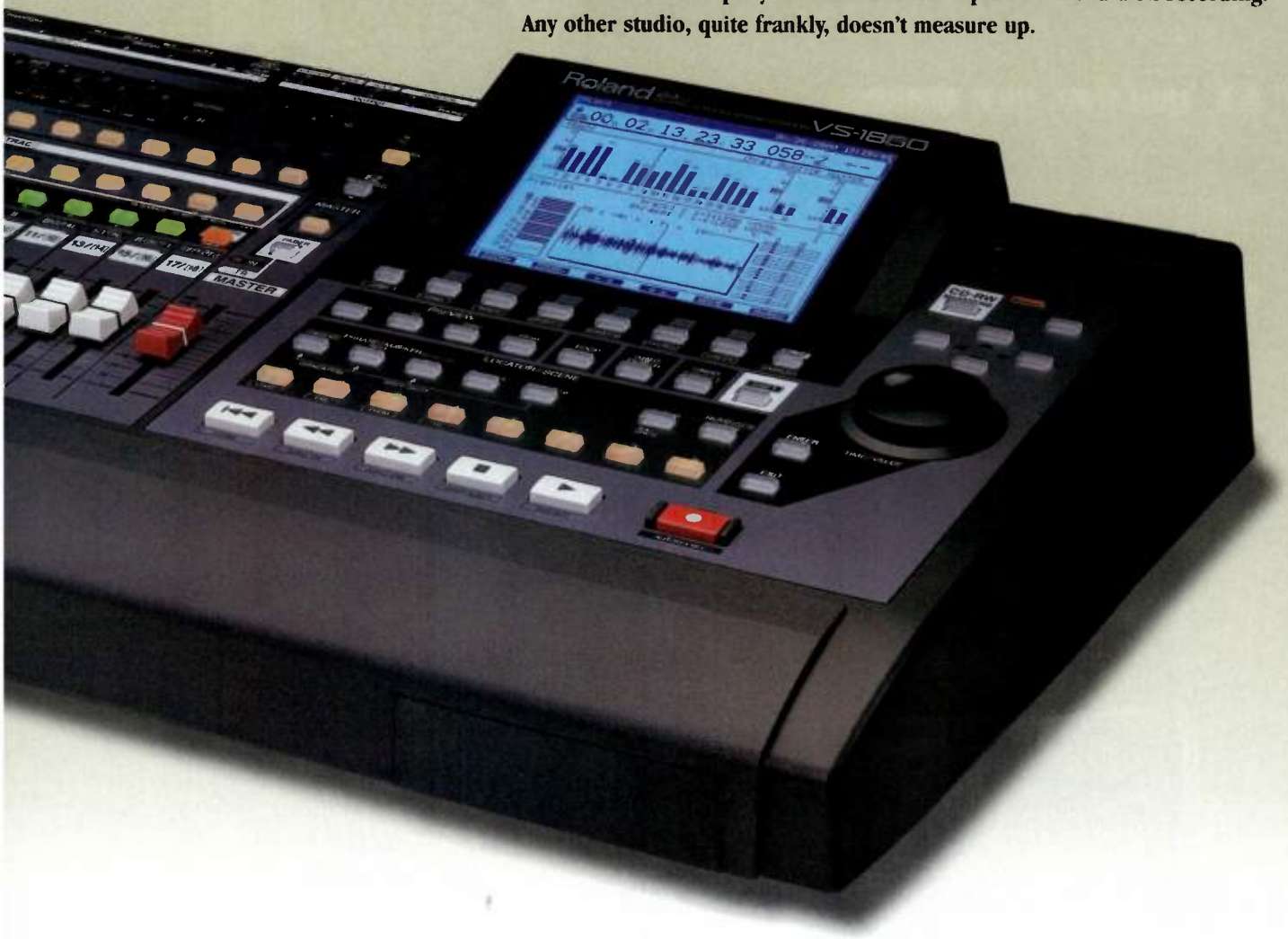

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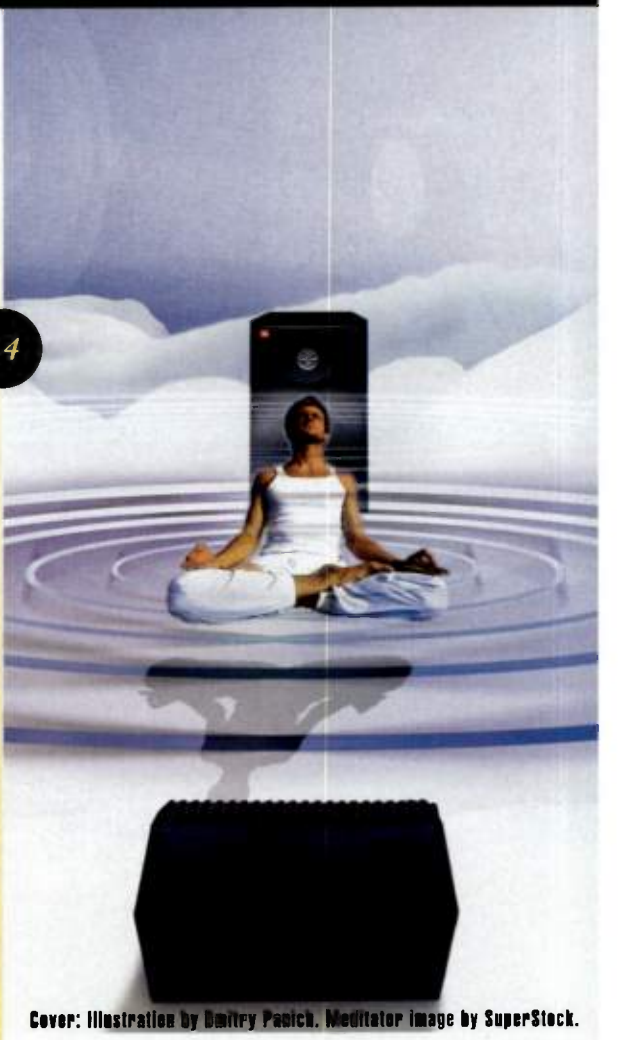

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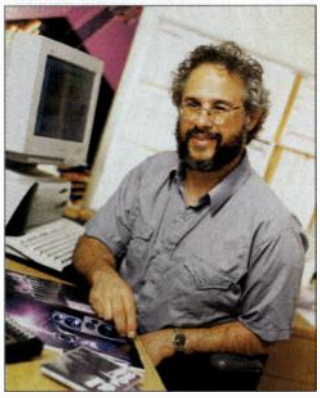
ANTHONY PIDGEON

Lately, I've watched several organizations experience growing pains. For starters, **EM**'s *Onstage* and *Remix* magazines are clearly gaining momentum, in response to which our editorial department is growing and adjusting to the workload. In an unrelated situation, a dog-rescue group I volunteer with (San Francisco Bay Area German Shepherd Rescue, www.sfgsrescue.org) recently jumped in size and is learning to deal with growth. Rapid growth is rarely stress-free, and many of us at **EM** and in the dog-rescue group have had to pause, catch our breaths, and do a reality check before moving forward. Observing the two situations, I reflected on the processes at work.

Dealing with creative people can be dicey, especially in a group situation. Business and personal relationships often get mixed up with questions of art. If you are in a leadership role, how do you manage to get sustainable, consistently good results? Admittedly, nobody bats 1.000 where people are involved, and sometimes you'll lose no matter what you do. But a combination of focus, flexibility, and patience can go a long way.

First, focus. You must keep your eyes on the prize, which is group success, however the group measures that. Unfortunately, a lot of bands fail to clearly establish goals. The objective might be commercial success, but maybe it's playing a few local clubs, jamming in the garage, or joining forces in the studio. Your goals make all the difference. Make sure that all agree—or be prepared for future shock. Sit down and talk it over. Then, be sure to stay focused on the group's goals. That means being patient and taking your emotional reactions into account as you make decisions. It's tough, but try it.

For instance, our dog-rescue group has more than doubled in size in a year, and we experienced our first open personal conflicts. That's not surprising; we suddenly have more people, and some members do not know each other well. We dealt with it by immediately reminding all concerned that our rescue mission is paramount. That resolved the problem, at least for now.

These next points should be obvious. One is to firmly establish the leadership structure. A benign dictatorship, open democracy, or limited partnership can each work well if done properly—but only if the lines of authority are clear.

The other point regards recruiting. We've all lost count of the bands we've seen ruined because people try to build serious groups with friends just because they're friends or because they'll drive their truck 500 miles for gigs. You want people who can play and who won't act out, of course, but also consider whether they'll work for group goals. If you're serious about a band, be sure your bandmates are, too.

I'm not suggesting being a hard case or doing absolutely anything to reach your goals; just be cool and keep your eyes on the prize. The key to staying focused is motivation; in general, people make music together because they want to, not because they have to. Setting goals, establishing reasonable rules, and being flexible and patient helps keep people positive and productive and will get you far. Staying focused is not easy, but you'll like the feeling you get when it works.

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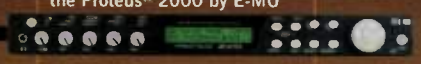

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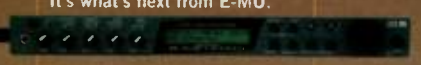

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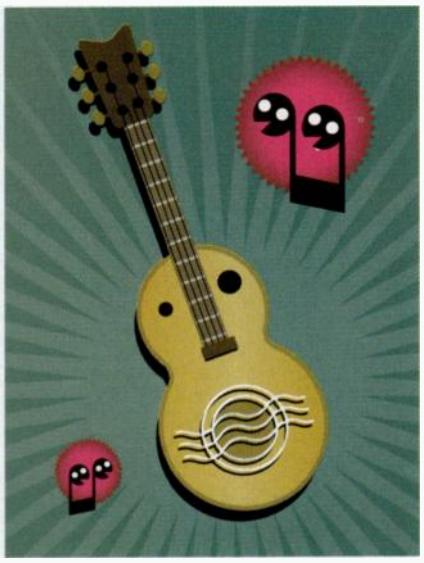


NUTRITIONAL SUPPLEMENT

After reading all the techno software and hardware talk in the July 2000 issue, Larry the O hit me square on the head with his column "Final Mix: Stale Beer and Bitter Fruit Make Poor Nutrition." Now here's a guy I could hang with. His comments come straight out of the real world and speak to composers and designers specifically.

As a composer and sound designer for a big facility, I've been experiencing the burnout phenomenon for the past couple of years. Everything that happens in these audio suites is based completely on subjective opinions, some delivered with the pure brutality of the advertising world and some based just on corporate obstinacy. The situation results in anger, resentment, and often a protective shield that inhibits productivity.

You work for three days on a score and sound design only to have a director, producer, client, or the like criticize them on the basis of priorities that have very little to do with the program—very frustrating. I've had to adopt Larry's technique of self-hypnosis just to get through the week on some occasions. You just put on that "work with it" hat, regardless of whether the criticism is right or wrong, and keep that positive attitude. As soon as you let the frustration take over you, you're gone for good.

Thank you, Larry. Your insights and viewpoints keep us going far beyond the trials and tribs.

Robert Pascarella
via e-mail

Just when I had been recently fired without an explanation from a major gig that I'd held for two years and was desperately looking for new work, and as I sat in a hospital room where both my parents are in what I'm afraid is the final stage of their lives, I was reading through your July issue. (Before I lost my gig, I purchased a DAW, and I'm trying to get educated on the subject.) I stumbled upon Larry the O's "Stale Beer and Bitter Fruit Make Poor Nutrition," and I was stunned at what he was saying. It was as though he wrote that for me. I got goose bumps as I read, and tears came to my eyes. Everything he said hit home with incredible accuracy.

I want to say thanks to Larry for his words of wisdom and insight. They have made a difference for me in dealing with the hurdles in my path. I have never written a letter to a mag before. That single issue provided me with not only a lot of information I'll use when recording with my new DAW, but also information I'll use when dealing with everyday life.

Thank you, Larry, for the good advice, and know that you did a great deal to help me understand why and how things happen. I know now that if I hadn't lost my gig, I would still be on the road instead of with the people who mean the most to me in their time of need.

God bless you, Larry. For some reason you wrote what you did, and it touched me. Best to you, and may you never feel that disappointment again.

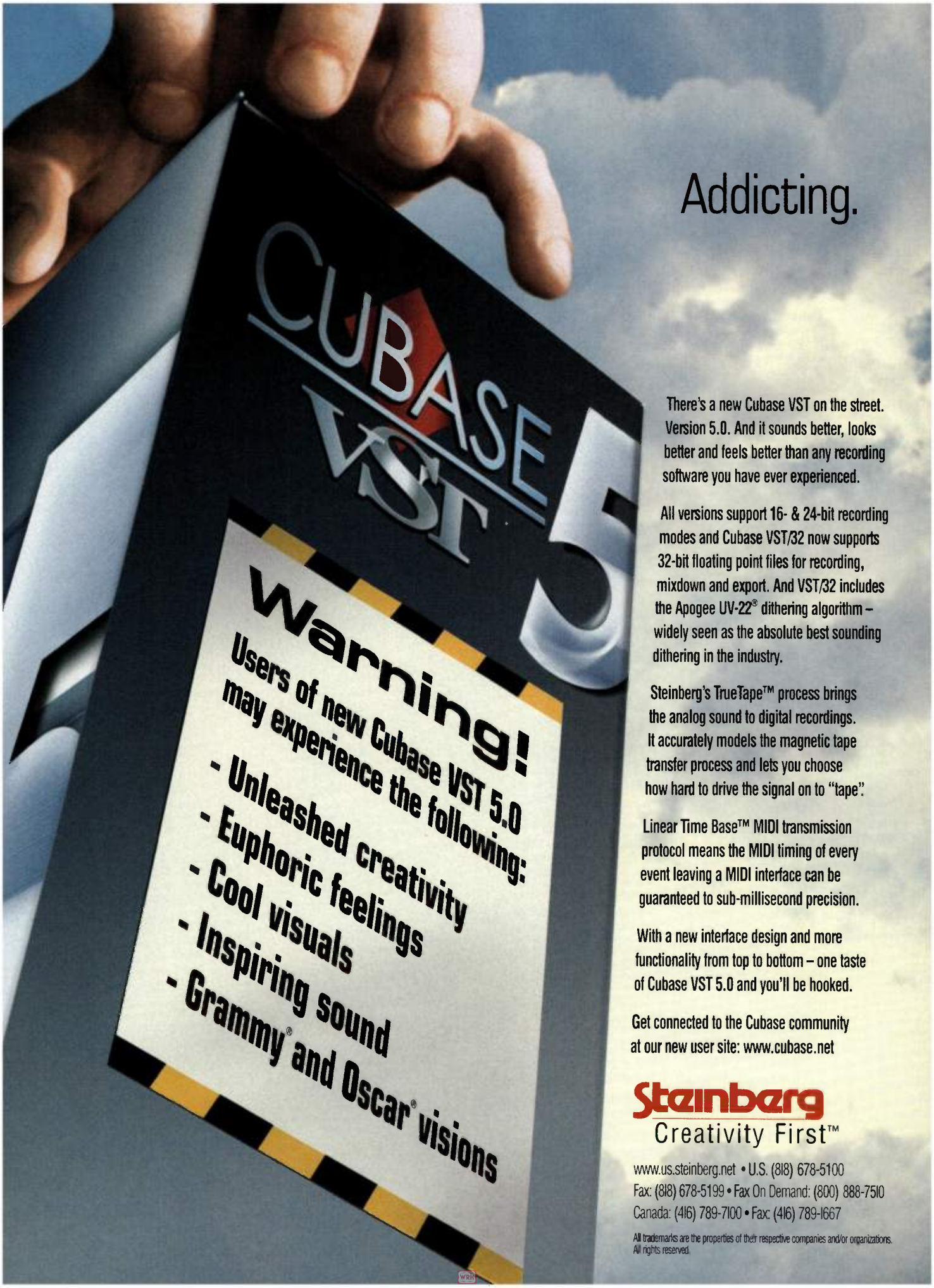
Adrian Garcia
via e-mail

WINDOWS 2000 SUPPORTER

In the July 2000 issue, a reader asked about dual-processor machines and the viability of Windows NT and 2000 as an OS for multimedia [see "Letters: Dual Personality"]. I have much experience in this area, and I thought I'd help clear up some misconceptions.

The reader thinks that Windows 2000 is not a reasonable choice because it does not provide adequate multimedia support, and that NT 4.0 is a better alternative. This is not true. Windows 2000 is NT 5.0, and it has far superior multimedia support. It comes out of the box with DirectX 7.0 and Plug and Play capabilities. Win2K is not a problem at all in itself. The problem is the slow rate at which hardware manufacturers are developing drivers for Win2K and NT. Many audio software packages, such as Cakewalk's *Pro Audio*, Sonic Foundry's *Vegas* and *Sound Forge*, and Steinberg's *ReBirth*, run wonderfully in Windows 2000. And every DirectX plug-in I know of works perfectly in Win2K.

Also, many people think that a second



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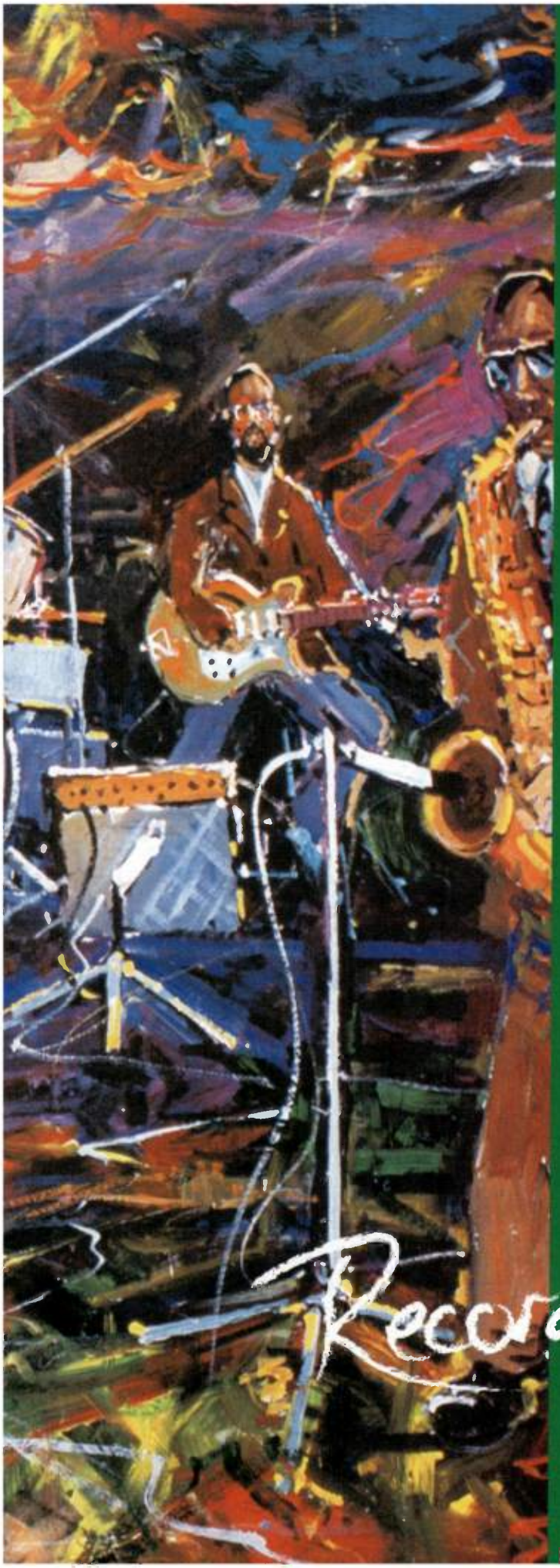

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processor will increase the performance of all applications. Actually, a second processor can share the load only if it has a separate "thread" to work with. So in most cases, an application has to be able to divide its various processes into separate threads. That brings me to the next misconception. *Pro Audio* isn't multithreaded, but it still benefits from a second processor. Why? Because DirectX plug-ins aren't part of the application. Each plug-in is a separate program, a separate thread. As a result, all applications using DirectX plug-ins typically get a 30 to 40 percent performance boost from a second processor.

I've found that audio applications are much more stable in Windows 2000 than in Windows 98. Not only do they crash less, but disk access is far more stable and fluid, allowing a song to play even when you're pushing your hard disk's data-throughput limit.

If the major hardware manufacturers realize this, we'll see real progress in stability and performance. It's sad that I can use my MOTU 2408 in the not-so-stable world of Windows 98, but I must resort to a SoundBlaster Live when using Windows 2000.

Scott Reams
Sacramento, CA

SECOND OPINION

I'm writing in response to the advice Gino Robair gave to Paulo Jorge da Silva Costa regarding a wobbly display on a computer monitor [see "Letters: I Need Some Space" in the July 2000 issue of **EM**]. Paulo's monitor setup consists of Alesis Monitor One speakers and an Alesis RA-100 amp located a few centimeters above his computer. When the amp is on, the display shakes.

I question the explanation that the problem is likely the result of EMI created by the speakers. While the trouble *is* apparently EMI, I suspect the amplifier, not the Monitor Ones, is the culprit.

The magnets on some speakers can cause color impurity and distortion when placed too near a CRT (this will not happen with LCD displays, as there is no beam which can be deflected improperly by the speaker magnets), but in such cases the problems are stationary. In other words, the display looks pretty crappy, but it doesn't jitter or shake.

Also, Paulo states that his display shakes when the amp is powered up, implying that it's fine when the amp is off. A magnet's effects will be static and present whether a connected amp is on or off.

Sounds to me as if the RA-100's power supply is emitting EMI that is interfering with either the scanning circuitry or the CRT beam itself, and moving the amp will eliminate the problem. Power supplies have more than enough stuff (transformers, electronic switching, and so on) that can interfere with a video display.

My point may seem pretty trivial, but with the near-field-monitor market being as competitive as it is, I believe that you are doing Alesis a disservice by leading your readers to believe that the Monitor One might cause display problems when in fact it may not. (Mr. Robair even went so far as to advise Paulo to consider investing in shielded monitors!) Hey Paulo, just move your amp.

Mark Klein, The Noise Floor
Cleveland, OH

REALITY CHECK

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A&R rep watches Keaton's band perform in a club. So blown away by the band is he that he phones his record company and tells them, "We gotta sign these guys!" I laughed and said, "Yeah right, only in the movies."

"Working Musician: A&R Primer" [see the June 2000 issue of EM] left me feeling the same way. Thank goodness indie musicians can now get their music to the masses without crossing record-company thresholds. I hope EM continues to explore and promote these other opportunities and gradually phases out articles about the lost art of butt-kissing A&R people and record companies.

An interesting article would be one about how A&R reps and record companies over the years have screwed countless bands who signed their lives away and ended up abandoning their musical careers forever.

Plenty of musicians like me have day jobs and families and are involved in music for the love of it. If revenues from my music will pay for my gear, I'll consider myself a success. I'll stick to selling my band's CDs on Amazon, MP3, and my Web site—and retain complete control, thank you very much!

Daniel Stecko
via e-mail

DISPLEASURES OF THE FLESH

Just a few words on the IUMA ad on page 21 of the August 2000 EM . . .

Ugh! I don't know whether to laugh, gag, or scream. Imagine my surprise when I was making the morning coffee, getting the mail, and sitting down to enjoy the latest issue of EM! (No, I don't live in Pleasantville, but this scenario does happen once in a while at my place.) I flipped along and yikes! That ad is a little too lifelike and gruesome.

I realize the issue I have is more with IUMA than you folks. And yes, I know the ad is supposed to get attention. But as someone who makes a living with my hands as well as my ears, I found the imagery very disturbing. The blood got my gorge (and coffee) to rise! The protruding bone is accurate enough to turn the strongest of stomachs. Are you guys that desperate for ad revenue?

We have enough death, violence, and dismemberment in our society as it is. This ad isn't appropriate at all. Sorry, I have to go hurl now.

Joe Hannigan
via e-mail

ERROR LOG

August 2000, "Master Class: The Bottom Line," p. 44: To get the DI signal and the miked signal in perfect phase alignment, a good rule of thumb is to delay the DI signal by 1 ms for each foot that the mic is placed away from the cabinet. Then listen to the combined mic and DI sounds while fine-tuning the DI signal's delay time in 0.1 ms—not 1 ms—increments.

August 2000, "What's New," p. 17: The correct contact information for PSP is tel. and fax 48-60-196-3173; e-mail contact@psp-audioware.com; Web www.psp-audioware.com.

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Rhythmic Journey

NEW

AUDIO/WAV-AKAI

Awarded Keyboard's KEY BUY AWARD, the review said - "As you'd expect, Steve's drumming is perfect - and the recording quality is among the best I've heard. In summary, this collection is first class. From start to finish, Steve's timing and feel are rock solid. 10/10" 2 CD collection from ex-Journey drummer STEVE SMITH. Demo at soundsonline.com (ID#1076)

NY Cutz Off Da Hook

NEW

2 AUDIO/1 WAV

Another exciting release from Zero-G. NY Cutz - Off Da Hook is the long awaited, eagerly anticipated follow up to the hugely successful NY Cutz (which was awarded Keyboard Magazine's KEY BUY AWARD). 8's huge slabs of fat cut street ready vibes and juicy grooves, are waiting for you to get your teeth into! So good, Zero-G named it twice! (ID#1093)

King Tone Grooves

NEW

AUDIO

King Tone Grooves provides the meanest and most varied collection of stylised breaks ever brought to you on one CD. From Killer 70's Funk loops to Slammin' Hardcore Dance Cutz, from huge Bonhamesque Beats to Deep Soul Grooves all played live on the finest quality kits with the biggest fattest sounds imaginable. Hot off the press, Check it out! AMG (ID#1091)

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ExtremeRockFunkRage

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Slam On the Breaks

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By Marty Cutler

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The Replex offers a choice between reverb, single-head delay, or multiple-head delay, all selectable with the tap of a footswitch. The number of delay repetitions can be varied with the Feedback knob. You can also change the delay time of the first simulated head with the Single Head Time knob. A separate knob sets the time of the second head when you're using the Multi Head Delay setting. The Vintage Factor control adjusts the amount of simulated tape-coloration. Hughes & Kettner; tel. (800) 452-6771 or (847) 439-6771; fax (847) 439-6781; e-mail info@hughes-and-kettner.com; Web www.hughes-and-kettner.com.

▶ FOSTEX VF-16

Fostex's VF-16 16-track digital multitracker ($1,399) offers a total of 24 tracks, 8 of which are virtual tracks for alternate takes. You can record 16 simultaneous, uncompressed tracks of 16-bit, 44.1 kHz audio, 8 with the analog inputs and 8 with the built-in ADAT interface. A/D conversion is 20-bit, D/A 24-bit.

Fostex's proprietary FDMS-3 technology promises more-efficient hard disk use, more recording time, and faster editing. The unit comes with an internal 5.1 GB EIDE hard drive. Files can be imported and exported in WAV format using external SCSI devices.

The VF-16 features two digital multieffects processors offering delay, reverb, chorus, and flanging. You can use 2-channel dynamics processing for input pairs 13 and 14 or 15 and 16, and dynamics processing is always available on the master stereo bus. All 16 channels offer 3-band EQ with parametric mids and highs, and a separate EQ is available for the master outputs. You can also raise or lower the pitch by 6 percent.

You get 17 60 mm faders: 8 for analog input, 8 for track input, and 1 stereo master fader. Each mic/line input has a trim pot, and inputs 7 and 8 offer XLR jacks with phantom power and insert points. All inputs have a peak LED indicator.

The unit has two effects sends and two aux sends. Both can be switched from prefader to postfader.

The VF-16 can send and synchronize to MIDI Time Code and can also respond to MIDI Machine Control. The unit's 99-scene mixer memory can be automated by MIDI Program Changes, and there are 90 Locate Points. You can also designate seven Mark Points on the fly. Fostex Corporation of America; tel. (562) 921-1112; fax (562) 802-1964; e-mail info@fostex.com; Web www.fostex.com.

▼ RAXXESS ISORAXX

Mechanical noise in the personal studio seems to be directly proportional to the number of disk-based devices and fans you use. Raxxess's IsoRaxx ($1,299) is an acoustic isolation rack that can reduce noise from MDMs, computer systems, and other disk-based devices by as much as 19 dB.

To contain drive noise, the IsoRaxx employs a gasketed, Lexan front door with a compression latch, ensuring a tight seal. The rear door has a keyed lock for security. Breakaway hinges allow easy equipment installation. The IsoRaxx's blower box, which has two variable-speed fans, keeps things cool. The electrostatic HEPA intake filter keeps dust away from your drive mechanisms.

Acoustical foam throughout the rack reduces noise, and two 14-space, shock-mounted rack rails reduce equipment vibration. You can run cables through the 4-inch knockouts.

The IsoRaxx acoustic isolation rack is available in ebony-fleck or maple finishes and measures 24.5"(W) × 32"(H) × 26" (D). Casters are included. Raxxess; tel. (800) 398-7299 or (973) 523-5105; fax (973) 523-5106; e-mail sales@raxxess.com; Web www.raxxess.com.

BITHEADZ PHRAZER

Mac users can now revel in drag-and-drop, loop-based music composition with BitHeadz's *Phrazer* for Mac OS ($399). The program allows you to combine audio files from a variety of different sources to create complex arrangements on the fly, without concern for tempo or key.

Phrazer can import digital audio files in AIFF, WAV, SD2, Audio CD, and ROM formats; files created with Sonic Foundry's *Acid;* and of course, native files created by *Phrazer.* You can trim and fine-tune audio files with the included sample editor.

Each track supports multiple loops. *Phrazer* can stream from the hard disk or audio can be loaded into RAM for optimum compatibility with iMac, G3, or G4 computers. The program can automatically time-stretch and pitch-shift audio material, and you can automate volume, panning, and DSP effects for each track. You get seven types of DSP: chorus, flange, parametric EQ, shelving EQ, filter, reverb, and distortion. Tracks can be triggered from your QWERTY keyboard or from a MIDI controller.

Once you have tweaked your mix to perfection, you can render *Phrazer*'s output to a stereo audio file for use with other applications. *Phrazer* can also lock to MIDI Clock for synching to your favorite sequencer. Additional third-party-software support includes MAS 2.0, ReWire, DirectConnect, ASIO, Sound Manager, OMS, and FreeMIDI. BitHeadz; tel. (831) 465-9898; fax (831) 465-9899; e-mail info@bitheadz.com; Web www.bitheadz.com.

ZOOM PS-02

If you have ever wanted to carry your digital multitrack recorder in your shirt pocket, here's your chance. The PS-02 Palmtop Studio ($624.99) from Zoom provides you with three audio tracks, a rhythm section, a multi-effects processor, and a microphone, all in a subpalmtop-size package.

You can record audio tracks directly to a SmartMedia card and bounce tracks with panning and effects. If you record in Hi-Fi mode, a 64 MB card will allow approximately 17 minutes of stereo recording time. In normal mode, a 128 MB card will yield more than two hours of recording time for a single track. The recorder offers virtual tracks, one of which can be selected for playback.

The PS-02 has a variety of built-in stereo drum kits, monophonic bass sounds, and preset patterns. The patterns can be sequenced and recorded into the audio tracks.

The multi-effects processor offers guitar- and bass-modeling effects for recording guitar tracks. It also has dedicated vocals effects for when you're using the PS-02's dynamic microphone. Effects include noise reduction, cabinet simulation, EQ, modulation effects, reverbs, and delays.

Among the PS-02's amenities are a backlit LCD, a chromatic tuner, an 8 MB SmartMedia card to get you started, and a belt carry strap. You can run the unit with four AA batteries or the furnished power supply. Zoom/Samson Technologies (distributor); tel. (800) 328-2882 or (516) 364-2244; fax (516) 932-3815; e-mail sales@samsontech.com; Web www.samsontech.com.

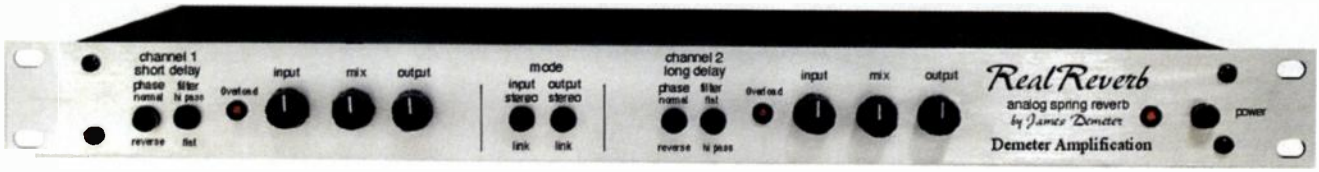
DEMETER RV-1 REALREVERB

Enthusiasts of vintage reverb sounds will be interested in Demeter Amplification's single-rackspace RV-1 RealReverb ($699). The genuine spring reverb offers a short (1.5-second) and a long (3.5-second) reverb tank. Each tank is full-size and has six springs.

The two tank channels can be used independently or linked. Switches controlling reverb phase and a low-cut filter give users further customizing options.

You can adjust the springs' input sensitivity, wet/dry mix, and output level for each channel. An overload indicator lets you know about any possible distortion.

The inputs and outputs are on balanced XLR and ¼-inch TRS connectors. The power supply employs a custom-built, shielded, toroidal transformer, and individually shielded reverb tanks help minimize noise. Demeter Amplification; tel. (818) 994-7658; fax (818) 994-0647; e-mail info@demeteramps; Web www.demeteramps.com.



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KORG

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SOUND ADVICE ▲▲▲▲

▲ SOUND BURST

Sound Burst's *Classic Synths* (Mac/Win, $149) gathers samples from 14 vintage synthesizers for BitHeadz's *Unity DS-1* and *Unity Player*. More than just a collection of raw waveforms, *Classic Synths* includes sampled pads, lead sounds, basses, electric piano, and brass.

Among the different synthesizers are Oberheim's Matrix and OB series synths, Roland's Juno and JX synths, and Yamaha's CS and DX synths. Some of the programs are voiced to resemble the original factory presets, but the majority use preset samples as springboards to totally new sounds.

The CD-ROM has more than 2,000 samples organized into more than 600 programs. The booklet lists all the banks, their sizes, and the number of programs they contain. Sound Burst by C.R.I.M.; tel. 39-11-650-1307; fax 39-11-650-7109; e-mail info@soundburst.com; Web www.soundburst.com.

KID NEPRO

More techno mayhem for the Korg Triton has arrived from Brooklyn in the form of Kid Nepro's *Techno/Dance Mix* patches ($25). This collection of programs and combinations leans heavily on rhythmic pulses and drum loops, with 64 programs that take advantage of the Triton's drum sounds and dual arpeggiators. Each key triggers a different component of the loop. By triggering different components in different orders, you can create a wealth of rhythmic and sonic variations. The second 64-program group consists of 16 analog-styled synth basses and 48 Hits and Bits. The combinations are where the real fun happens. Drum loops, rhythmic pulses, basses, and hits are accessed in a single combination, allowing you to compose entire songs from a single keyboard setup. Kid Nepro; tel. (246) 420-4504; fax (246) 418-0282; e-mail kidnepro@aol.com; Web www.kidnepro.com.

▼ SPECTRASONICS

Retro Funk (Akai, E-mu, Roland, and Kurzweil CD-ROM, $199; audio CD, $99) captures a variety of original grooves and fills with vintage drums and a live feel. To create the collection, Spectrasonics enlisted studio veterans Eric Boseman, Bob Wilson, John Ferraro,

and Greg Bissonette. All bring impressive résumés and chops to the table, but if their grooves don't quite fit the pocket you need, you can use the Groove Control feature to get the feel that your tracks demand. Spectrasonics recorded the sounds dry to tape, capturing the warmth of analog compression.

The package consists of two CD-ROMs of drum loops, in both unaltered and time-sliced versions. The included Groove Control MIDI sequences allow the time slices to be groove-quantized and pitch-shifted, and you can change the tempo without changing the groove's pitch. (The audio-CD version does not contain any Groove Control sequences.)

You also get an Add Ons section that includes sampled vinyl noise, cymbal samples, and claps. Spectrasonics/Ilio Entertainments (distributor); tel. (800) 747-4546 or (818) 707-7222; fax (818) 707-8552; e-mail ilioinfo@ilio.com; Web www.ilio.com.

▼ KEYFAX

Keyfax has engaged a group of notable studio musicians to create its *Funk Groove Library* for the Yamaha RM1x MIDI sequence remixer ($29.95). The collection of Standard MIDI Files was prepared from the live performances of Ron Beck, Dale Ockerman, and others. Keyfax recorded the loops in real time using MIDI guitars and MIDI drum kits.

Each segment consists of four to eight bars and contains complete grooves for drums, bass, and various other instruments. The grooves have multiple sections of variations, breakdowns, and fills. *Funk Groove Library* provides six additional single-instrument patterns for drums, guitar, bass, electric piano, clavinet, and synth.

The files are on a 3.5-inch floppy disk in the RM1x's RIP format. The disk and printed owner's manual come in a CD jewel case. Keyfax Software/Hardware; tel. (800) 752-2780 or (831) 460-0172; fax (831) 460-0173; e-mail us@keyfax.com; Web www.keyfax.com.

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▶ E-MU VIRTUOSO 2000

Sporting a set of sounds that are geared toward realistic orchestral arrangements, E-mu's Virtuoso 2000 sample-playback synthesizer ($1,395) offers 64 MB of ROM of orchestral instruments. The unit provides up to 128 notes of polyphony that can be distributed among 32 MIDI channels. E-mu promises accurate timing in dense sequences.

Each of the Virtuoso's samples was recorded with four microphone pairs, and additional samples were recorded from the perspective of the audience to capture the recording space's natural ambience. These separate samples can be blended to taste with the more conventionally recorded instruments. If 64 MB of sample ROM isn't enough, the Virtuoso 2000 has two ROM slots for up to 64 MB of additional samples. You can choose from E-mu's library of expansion ROMs ($249 to $395) or burn your own samples with the help of an E-mu E4 Ultra sampler workstation ($1,795).

The Virtuoso 2000's extensive synthesis features can take you well beyond orchestral emulation. You get 50 resonant-filter types switchable to 6th or 12th order, 2 16-waveform LFOs per voice, and 24 "patch cords" with 60 sources and 60 destinations. The unit has three 6-stage envelope generators: one hardwired to the amplifier, another controlling the filter, and an auxiliary envelope assignable to a wide variety of parameters. The two flexible effects processors have 30 available algorithms apiece. You can quickly tweak synthesis parameters, effects levels, and Velocity response from the front panel.

The Virtuoso 2000 has two discrete MIDI sections: MIDI A has In, Out, and Thru connections, and MIDI B has In and Thru. You can configure the six audio outputs as individual outputs or as three pairs of stereo outputs with programmable panning. The unit also sports coax S/PDIF I/O. You get a pair of stereo effects returns for use with external processors. E-mu/Ensoniq; tel. (831) 438-1921; fax (831) 438-8612; e-mail info@emu.com; Web www.emu.com.

▲ TRUTH AUDIO TA-1P

The quest for accurate-sounding near-field monitors is much like the quest for the Holy Grail, but Truth Audio believes it has the answer with its TA-1P near-field reference monitors ($999 per pair). The cabinet has two 5-inch mineral-filled polycone woofers with 14.6-ounce magnets and a 1-inch, tuned-chamber, cloth-dome dynamic tweeter.

According to the manufacturers, the TA-1P has a frequency response of 48 Hz to 20 kHz (±3 dB). The rear bass port helps produce a more powerful low end. Truth Audio's patented crossover design ensures clear midrange imaging. The monitor's impedance is 4Ω nominal and 3Ω minimum. Maximum power handling is 160W. Truth Audio; tel. and fax (850) 267-1255; e-mail truthaudiovp@netscape.net; Web www.truthaudio.com.

▼ KORG TRITON-RACK

Korg has packed the synthesis, effects, arpeggiation, and sampling features of the Triton keyboard sampler into the Triton-Rack ($2,500) and added major enhancements and expansion capabilities.

The Triton-Rack has room for up to 2,057 programs, 1,664 Combinations, and 200 Multis. The synth is GM Level 2 compatible and includes 256 GM programs and 9 GM drum sets. A library of 382 different riffs is available for auditioning sounds in their proper musical context.

When its eight PCM expansion slots are filled, the Triton-Rack can hold up to 160 MB of waveform memory. Each new 16 MB PCM board ($200) has a dedicated bank in the Triton-Rack for its programs and Combinations.

As a sampler, the Triton-Rack can hold up to 96 MB of samples with the help of three SIMM slots. Samples can be exported to your computer as WAV and AIFF files. The sampling architecture is the same as that of the Triton keyboard, with a fixed 16-bit-linear sampling rate of 48 kHz.

Digital I/O capabilities are versatile; optical S/PDIF connections come standard with the unit. Options include the EXB-DI ADAT Optical output ($200) and the EXB-SCSI SCSI interface ($200). Perhaps most interesting is the promise of future support for mLAN audio and MIDI via FireWire.

As with the updated Triton keyboard, you can use the Time Slice function to create patterns consisting of loops with rhythmic samples. The patterns can be played at any tempo without changes in pitch. You can also add Korg's EXB-MOSS board ($500) for six voices of physical modeling and other types of digital synthesis.

The Triton-Rack is 16-part multitimbral and includes a playback-only sequencer for songs created on the Triton keyboard. Songs can be stored in a Flash-Memory Demo area. Other features include six outputs and four assignable real-time knobs. Korg USA; tel. (516) 333-9100; fax (516) 333-9108; e-mail product_support@korgusa.com; Web www.korg.com.

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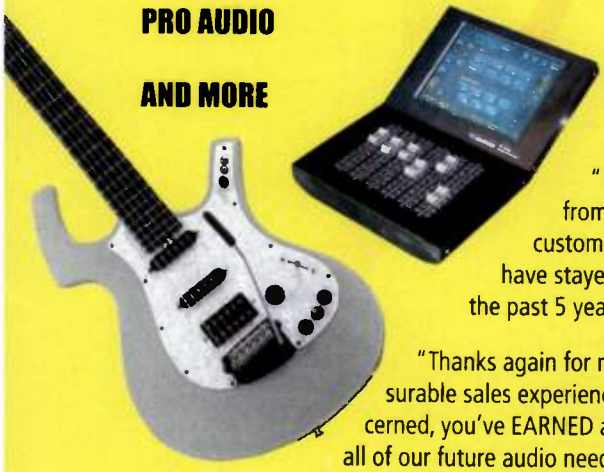

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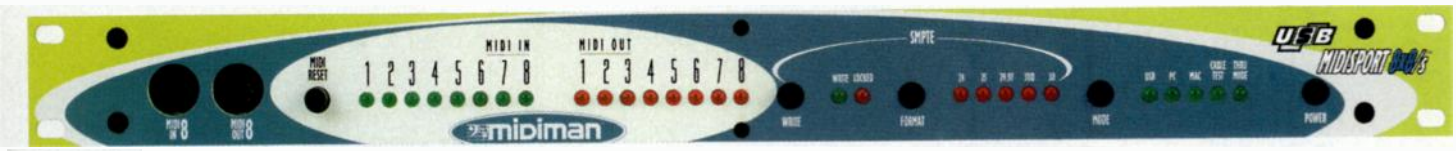

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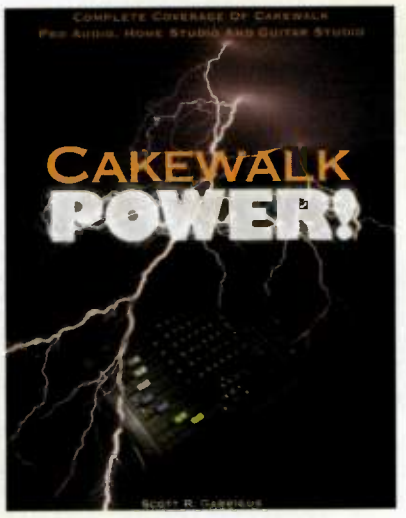


▲ MUSKA LIPMAN

Muska Lipman Books has just published Scott R. Garrigus's *Cakewalk Power* ($29.95), an extensive tour through Cakewalk's *Pro Audio*, *Guitar Studio*, and *Home Studio* MIDI and audio recording software. The book guides you through fundamental MIDI and audio operations, including recording, editing, mixing, and playing your tracks; adding effects; creating audio CDs; and producing music for video and Web distribution.

The examples and tutorial exercises are liberally illustrated with screen shots. Icons highlight tips, warnings about irreversible actions, and useful notes.

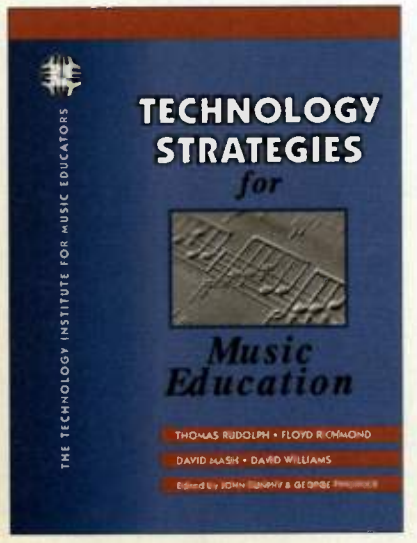
The author's Web site (www.garrigus.com) supports the book. There, you can subscribe to a free monthly newsletter containing tips and techniques not included in the book. Garrigus's site also features a discussion area where you can post questions and a live-chat area where you can talk with the author. Garrigus worked closely with Cakewalk to ensure that the information presented in *Cakewalk Power* was technically accurate. Muska Lipman Books; tel. (513) 924-9300; fax (513) 924-9333; e-mail publisher@muskalipman.com; Web www.muskalipman.com.

▼ TIME STRATEGIES

Music technology has enormous potential for use in education, but few know how to take full advantage of the technology. With this in mind, TIME (Technology Institute for Music Education) Strategies has published *Technology Strategies for Music Education* ($14.95, plus $3.50 shipping and handling), a guide to music-technology competency for music educators.

Authors Thomas Rudolph, Floyd Richmond, David Mash, and David Williams are well-known music-technology authorities. Their book discusses technology in music education, areas of competency, the national standards for arts education, student-teacher interaction strategies and activities, information processing and lab management, and music-technology benchmarks. The guide contains sections on electronic instruments, MIDI sequencing, notation software, and computer-assisted instruction.

The student-teacher activity section, targeted at students in grades K through 12, lists goals such as coming up with new ideas for using music technology. An appendix organizes activities into seven technology areas, including sequencing and Internet telecommunications. TIME Strategies; fax (610) 287-3971; e-mailtimemused@aol.com; Web www.ti-me.org.



▲ SYNTH-TEK

Some people buy synthesizers and, daunted by batteries of menus, buttons, or obtuse manuals, never explore the instruments' depths. The *K2500* CD-ROM tutorial series from Synth-Tek, for Macintosh and Windows ($99 each for beginner and intermediate levels, $149 each for advanced levels and *Super VAST*), provides assistance for stumped Kurzweil K2500 and K2600 users.

Each cross-platform CD-ROM contains more than 100 audiovisual tutorials covering topics ranging from how to connect the Kurzweil to how to transpose the instrument's MIDI output. The topics are logically developed in order of increasing complexity. The *Super VAST* CD-ROM contains tutorials on KDFX and controllers, sample DSP functions, FUNs, and other advanced features.

The discs also provide users with additional helpful information in the form of 104 frequently asked questions, 55 troubleshooting tips, and a 150-word glossary of terms. Synth-Tek; tel. (805) 955-0415; fax (805) 583-2654; e-mail info@synthtek.com; Web www.synth-tek.com. ●

By Roger Maycock

DOTDOTDOT.COM

Media Enforcer (http://mediaenforcer.tripod.com) is a line of defense for owners of digital media. The application helps you track your intellectual properties and prevent their theft. *Media Enforcer* is designed to put the responsibility back on the offending users. It checks all servers on all Internet services for illegal material and provides you with the identifying information you need to either ban the users from the service or, if necessary, hold them accountable in a court of law. The current version supports Napster and Gnutella, and the company is adding more services . . . Anticipating the release of DVD-Audio discs, **Dolby Laboratories** (www.dolby.com/tech) has posted a white paper titled "Guidelines for Producing Music in 5.1 Channel Surround" in the Technical Information section of its Web site. The paper, available as an Adobe *Acrobat* PDF file, describes the origin of multichannel audio for the cinema, the differences between mixing for film and mixing for music, and how to set up a 5.1-channel monitoring environment. It also provides suggestions on how music mixers might best employ the format's six channels . . . **HitQuarters.com** (www.hitquarters.com) is a new, free Internet contact service designed to help producers, songwriters, and musicians find the people they need in the constantly changing music industry. The site, which HitQuarters claims has the largest database of its kind, contains information about important music-industry figures, company descriptions and contact details, classifieds, and more. With HitTracker, the site's search feature, you can find out who is credited with breaking a particular artist. Still in its formative stages, HitQuarters plans to eventually provide newsletters, chat rooms, links, and articles.

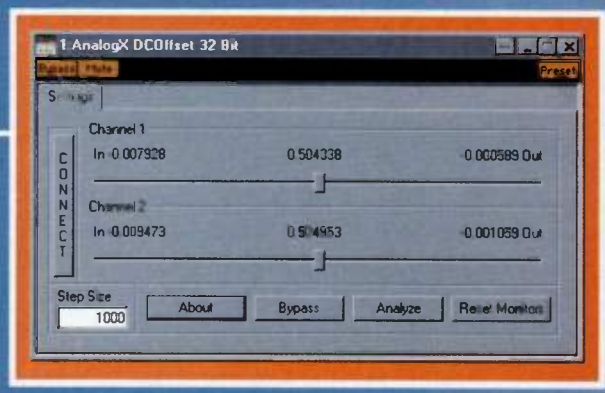
WEB SITE OF THE MONTH

If sheet music or MIDI files play an important role in your music making, Net4Music (www.net4music.com) is for you. Boasting a collection of more than 60,000 titles, this site is a wonderful resource for all musicians, regardless of musical preference.

Finding music is easy. The site's search function asks you for a title and delivery medium (sheet music, MIDI file, or both). Hot Picks in a wide range of musical genres are also listed. When you locate a composition of interest, you can audition it as either an audio or MIDI sample. Prices for individual compositions are on a par with those you would find in a traditional music store.

Net4Music's editorial content covers a broad range of topics. The site even has an innovative feature that helps you publish your own compositions.

DOWNLOAD OF THE MONTH



In working with computer-based digital audio systems, musicians commonly encounter DC offset, in which a file contains a static, nonperiodic waveform. This constant-voltage, DC waveform can cause audio artifacts—such as pops and clicks—and often appears during A/D conversion (among other operations). The shareware plug-in *DCOffset* from AnalogX (www.analogx.com) can correct DC-offset problems and serves as a good addition to a Windows-based audio engineer's toolbox.

AnalogX claims that *DCOffset* is the only DirectX plug-in capable of detecting and analyzing DC occurrences in real time. The plug-in can recognize DC offset after processing less than ten seconds of audio, so you don't need a separate program to scan the entire file.

DCOffset requires an application that supports DirectX audio plug-ins. It works with either real-time or non-real-time applications. In addition, you can employ this useful plug-in on mono or stereo files and with 16- and 32-bit data types.

WEBCAST

Virtuetv.com (www.virtuetv.com) is Europe's leading provider of Internet Webcasts. With a comprehensive palette ranging from rock concerts and sporting events to corporate product launches and press conferences, the site has a wealth of "channels" that address just about every possible interest a visitor might have.

Offerings include the Music, Lifestyle, Film, News, Business, Sports, and Radio channels, a weekly Webcast schedule, and a special-events section tailored specifically to broadband users. Virtuetv.com's interface is colorful, easy to navigate, and relatively free of advertising. With just a few exceptions, broadcasts are transmitted in both RealMedia and Microsoft Windows Media formats. Particularly notable is Virtuetv.com's events-notification system, which alerts you to upcoming broadcasts. Because the site caters to an international audience, it includes a Global Time Check feature that converts Greenwich Mean Time to your local time. The Time Check keeps you from missing any events.

After Virtuetv broadcasts an event, the company usually archives it and makes it available for on-demand access. Whether you're looking for concerts, foreign radio, or news, Virtuetv.com is a great place to begin your search.

WEB APP

As an increasing number of musicians find their niche in Web audio and music creation, new sound formats keep surfacing. U.K.-based SSEYO (www.sseyo.com), developer of the Koan interactive audio platform, recently introduced *Freedrum*, a combination 16-step sequencer, synthesizer, and beat organizer. *Freedrum* is a Koan-driven Web application with a Macromedia *Flash* front end. This combination allows you to create, mix, and process musical grooves for use on a Web page.

Freedrum uses no audio samples. Instead it leverages Koan's integrated software synthesizer with fully routable and Web page–programmable modules. The Koan system supports MP3, WAV, SoundFonts, Yamaha's XG, and MIDI. It provides full MIDI-file playback capability and lets authors apply synth sounds and MP3 samples, allowing generative augmentation of existing material. The system enables you to create small (as little as 1 KB), low-bandwidth files that load quickly into a visitor's computer system and sound surprisingly good. To experience your creations, a visitor simply downloads and installs the free *Koan* plug-in, which automatically adjusts its audio output to match the power of the computer.

Freedrum offers a variety of Kits and Patterns. You can customize your creations with effects such as distortion, delay, reverb, and ring modulation, and you can modify the grooves to your liking. Other features include Particle Systems (for granular synthesis and randomization) and the Virtual Wave Generators. Patterns can be automutated and mixed in real time.

On each track, you create your own beats and sounds. You then save them as audio files or Koan-Play files or simply use the system as an interactive, generative beat mixer. You can also export your settings to a text file that can be added to a Web page or shared with others.

The Koan Interactive Audio Platform includes the free Internet plug-in, both pro- and consumer-version authoring tools, and a licensable engine. It can drive Web animations (for example, Flash/DHTML) using JavaScript and features about 250 parameters and 100 functions. Additionally, the company's Web site includes numerous Koan templates, SoundFonts, MP3 samples, and support documentation to help you get up to speed quickly.

Freedrum is cross platform with modest system requirements. For Mac and Windows, your browser must be Netscape's *Navigator* or Microsoft's *Internet Explorer* 4.0. You must install the latest *Koan* plug-in and Macromedia's *Flash 4+* plug-in. PC users need Windows 95/98/2000/NT, a Pentium-compatible processor running at 133 MHz, DirectX 3, a codec supporting MP3, and a sound card. For pieces using SoundFonts, your sound card must be compatible with Creative Labs' AWE or SoundBlaster Live and have 512K of SoundFont memory. Mac requirements are a Power Mac or iMac, Mac OS 7, and QuickTime 4.0 (for MP3 support).

BAND ON THE WEB

Ultraphonic

Catching the attention of a record-company A&R scout has always been a daunting task. However, one band that has found a way to differentiate itself from the masses is Atlanta-based Ultraphonic (www.ultraphonic.com). The group did this by winning Garageband.com's $250,000 recording contract competition.

Garageband listeners choose a contest winner from the approximately 40,000 songs available on the site. Reviewers receive two randomly selected songs at a time, and they don't learn the artists' identity until after the review process is complete. Each artist gets the same number of reviews, and the artist with the largest positive count at the end of each quarter wins the recording contract.

Ultraphonic, whose members have been together for slightly more than two years, consists of Jason Fowler (guitar and lead vocals), Brent Wilson (bass), Brooks Wilson (guitar), and Tom Conroy (drums). They released their debut CD in early '99 to rave reviews. Their songs on Garageband.com are from their forthcoming CD, *Impression*.

Ultraphonic spokesman and producer Bryan Holmes comments, "We are very pleased to have won. Each of the five songs we uploaded made the top 30. In an industry where a few major-record-company executives pick and choose what the general public gets to hear, music fans now have an opportunity to voice their opinions. It's long overdue. It took the Internet and Garageband.com to provide a forum that assesses the talent and marketability of an artist with absolutely no political diversions affecting the outcome."

The song that put Ultraphonic over the top is "Our World." "Our fans' reaction to the song confirmed the positive gut feelings we had when we wrote it," notes Wilson. "Bryan Holmes always felt it had the greatest potential as a first single. We had already decided to put a remix of the song on our upcoming record, and the results on Garageband.com proved this was a good idea."

The Internet has been a tremendously helpful vehicle in promoting Ultraphonic, driving sales of the band's music and furthering their opportunities for live performance. With their excellent writing, performing, and production skills, their future looks very promising.

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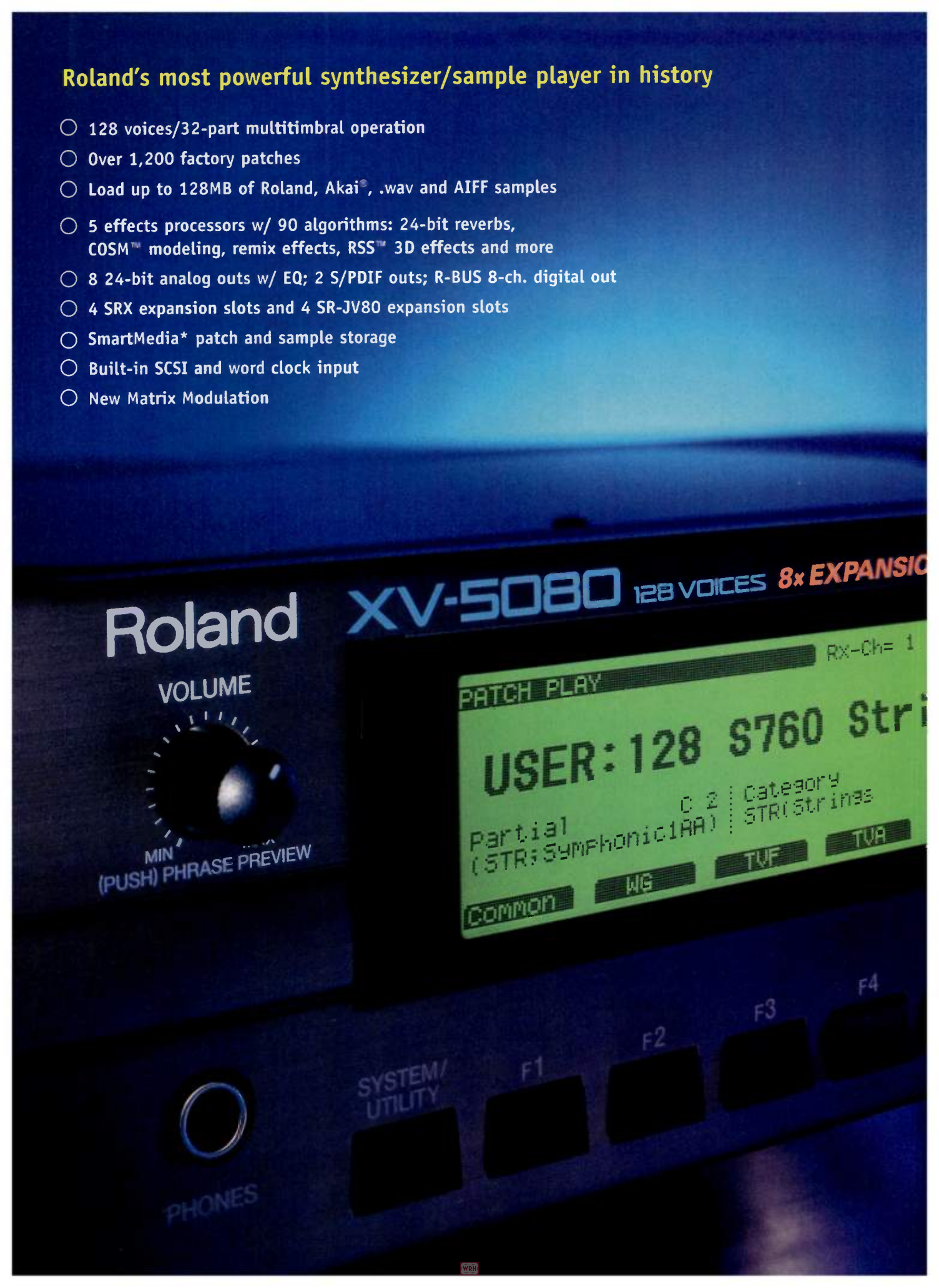

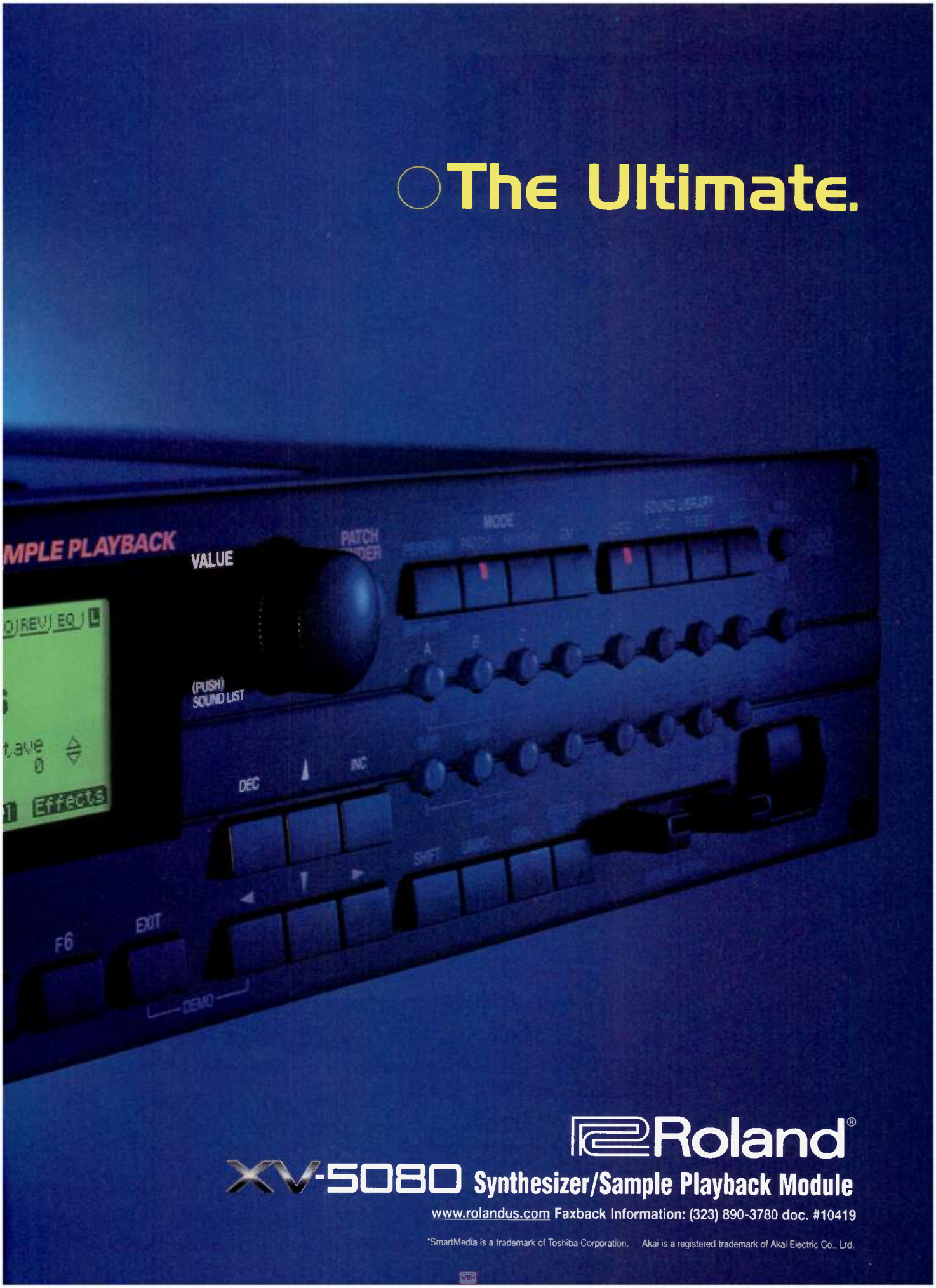
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VINTAGE PAGE

By Julian Colbeck

Each month "Vintage Page" ranks classic electric and electronic keyboard instruments according to their current popularity, based primarily on sales and rentals in the used-instrument market. We note in parentheses the previous month's rankings for keyboards that remain hot items, enabling you to follow the changes in the market. Our estimated street prices are categorized according to the instrument's condition.

In the Spotlight:

Roland Juno-106

Produced: 1984–1986
Made in: Japan
Number produced: 40,000
Synthesis system: analog, subtractive
Price new: $1,095
Today's prices: A=$600; B=$450; C=$300

Roland's polyphonic Juno-106 was one of the world's first MIDI synths. Its MIDI implementation is impressive: panel controls transmit SysEx; the unit receives and sends on all 16 channels, and you can selectively filter MIDI data.

Even so, the Juno-106's sounds and the speed of access it offers have kept the flame alive through all the years since it debuted in 1984. An edgy, almost digital-sounding analog synth, it sounds considerably less warm and fuzzy than its immediate and non-MIDI predecessor, the Juno-60, but the 106 garners just as much admiration as the 60 for simple and highly tweakable sound and processing parameters.

George Michael used the Juno-106 extensively for the warbling synth tone and the gated synth part of "I Want Your Sex," and William Orbit admits to merciless use of the Juno-106 throughout Madonna's *Ray of Light* album.

Popular throughout the 1990s as a cheap source of dance sounds, the Juno-106 achieved *de rigeur* status in the goa, trance, and industrial music scenes. Sadly, as a result, units occasionally command silly prices.

Features

The Juno-106's basic voice architecture is similar to that of the Juno-60 and JX-3P synthesizers. The Juno-106 begins with the digitally controlled oscillators (DCOs) that offer a choice of variable-pulse or sawtooth waveforms. Pitch is selectable with 16', 8', and 4' settings. You also get a sub-oscillator (which generates a square wave) and a noise generator. Individual sliders control the levels for each of these parameters.

The voltage-controlled resonant lowpass filter (VCF) can self-oscillate and can be shaped by a 4-stage envelope generator, albeit one shared with the VCA. You can also control the VCF cutoff point in real time using the pitch-bend wheel, and if you want to be really cool, you can transmit that information over MIDI. (Set the MIDI Control switch to 3.) The separate highpass filter is controlled by a single slider.

Roland made the most out of not much with the Juno-106. The two-position chorus fattens up the single-oscillator-per-voice architecture, and portamento adds interest and movement to the shared (filter/amplifier) envelope generator. A Unison mode—good when used in conjunction with portamento—harnesses all six oscillators and is excellent for rich, spiky bass lines. There are 128 patch memories.

Sounds

Definitely clearer and more defined than a Juno-60, the Juno-106's sound elicits strong opinions among vintage-synth users. What some people find clean and sharp, others hear as thin and flat. The Juno-106 lacks the timbral richness of, say, a Minimoog, but its deep sine-wavey bass sounds are classic in their own way.

COURTESY AUDIO PLAYGROUND SYNTHESIZER MUSEUM

Some people consider the Roland Juno-106's single-oscillator sound to be too thin, while others praise it as clean and sharp and employ its two-position chorus for fattening. It certainly is doing something right, though: popular in its mid-1980s heyday, the Juno-106 remains in demand in today's techno/electronica scene.

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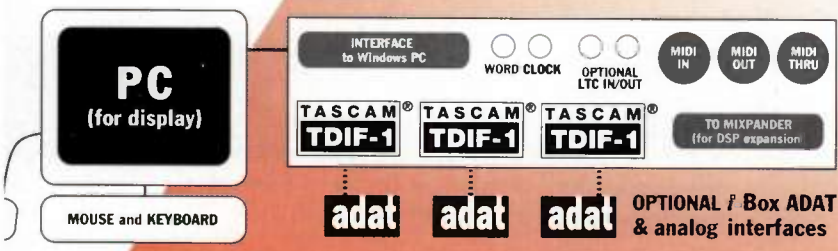

CONNECTIONS Right out of the box, R.Ed 24 can handle all three of your Tascam DA-88 digital recorders, as well as O2R and DA-7 consoles or other TDIF devices. No hassle with extra cards or extra cost. Optional *i*-Box interfaces allow ADAT and analog users to hookup with ease—and for a lot less than you'd expect.

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When it comes to pads, you won't get the drifting, shifting beds of a Prophet-5, but in shimmering atmospherics, the Juno-106 holds its own with the best. Although you can generate plenty of spectacular filter sweeps, self-oscillating whistles, and whines, the Juno-106's power is its fundamental simplicity and sonic precision. The Juno-106's sounds sit well in a mix; they tend not to clutter. Programming and hearing this instrument is not complex. "Amen to that," many say.

Problems

The Juno-106 generally enjoyed a trouble-free ride, but like any 15-year-old synthesizer, it's not immune to problems. Disappearing polyphony remains the most common glitch, thanks to a production run that used some bad ICs. The MIDI response can seem slow, and it will be slow if you're transmitting a lot of SysEx data. Keyboard problems (such as notes sticking or spiking) are also not unheard of, and caps often fall off the panel hardware. Solving these problems is still possible, though irritatingly expensive and time-consuming.

Roland made the most out of not much with the Juno-106.

Support

The Juno-106 is not a rare instrument. In fact, it was Roland's biggest seller until the D-50. Although it's unlikely you'll get much support from Roland itself, the Web is a rich resource for help, advice, and even a downloadable owner's manual. Check out the Juno-106 Connection (www.hinzen.de/midi/juno-106) for patches, editors, librarians, and more.

Orbit's much-publicized use of the Juno-106 poured fuel on an already hot market, and in spite of a very vocal Juno-106 backlash contingent—mainly Juno-60 users—this instrument remains deservedly, if not preeminently, desirable within the techno/electronica scene.

Julian Colbeck *has toured everywhere from Tokyo to São Paulo with artists as varied as ABWH/Yes, Steve Hackett, John Miles, and Charlie.*

THE LIST

1. **Korg MS-20.** Previous rank: 1. When it comes to power and flexibility, the MS-20 stays ahead of other single-oscillator mono synths, thanks to two oscillators and endless patch permutations, made possible by a real patch bay. It's a cool-looking customer, too.
   A=$1,200; B=$850; C=$500

2. **Roland Juno-106.** Previous rank: 2. See "In the Spotlight."

3. **Roland Juno-60.** Previous rank: 4. The warmer-sounding cousin of the Juno-106, the Juno-60 polyphonic synthesizer still has many loyal followers. Buy it for the arpeggiator and the evocative chorus.
   A=$475; B=$375; C=$275

4. **Roland SH-101.** Previously unranked. A good substitute for the now-overpriced TB-303, this monophonic synthesizer has excellent filtering and is perfect for the dance market, especially the cool colored models (blue or red) with performance-control grips.
   A=$500; B=$400; C=$325

5. **Roland TB-303.** Previous rank: 5. Prices are still crazy for the limited but sound-defining bass synth that launched the retro scene. It has been sampled and emulated to death, but owning a real TB-303 still carries considerable cachet. It can be a pig to figure out, and it has no memory locations.
   A=$1,100; B=$750; C=$550

6. **Sequential Circuits Prophet-5.** Previous rank: 3. The analog standard by which all vintage polyphonic synths are measured. There are good and bad models—version 3 is the model to buy—but for subtle, lush textures, the 5-voice Prophet-5 (now more than 20 years old) is still a hard act to beat.
   A=$1,750; B=$1,100; C=$400

7. **Hammond B-3.** Previous rank: 6. Expensive, big, and heavy, this tone-wheel organ is really only for people with cash, storage room, and roadies. The drawbars reveal a beautiful range of expressive tones, but you need to be a player—not a programmer—to get the most out of it. The more compact A-100 features essentially the same guts as the B-3 and may cost considerably less: $2,000 to $3,000.
   A=$5,000; B=$2,500; C=$1,500

8. **Fender Rhodes Stage 73.** Previous rank: 7. In the right hands, this player's instrument is capable of great subtlety. Extensively used by such jazzers as Herbie Hancock and Chick Corea in the '70s, the Rhodes electric piano remains cool and viable in the retro funk and dance scene.
   A=$600; B=$450; C=$250

9. **Korg M1.** Previously unranked. The top selling instrument in its day (it was released in 1988), Korg's polyphonic sample-playback subtractive synth remains popular due to its distinctive and exhaustive range of sounds and easy operation.
   A=$650; B=$550; C=$425

10. **Yamaha DX7.** Previous rank: 10. Prices are firming up for this sound-defining polyphonic FM synth of the mid-1980s. A pig to program, it sports no effects to speak of, but if you have a general interest in FM, this unit is a fascinating prospect for study.
    A=$425; B=$325; C=$200

Price Guide: The quoted prices reflect typical street prices you should expect to pay in U.S. dollars. The buy-in on vintage instruments, as with vintage cars, is just the beginning, though. Most of the original manufacturers are long gone, so maintenance and repairs are expensive.

A=like new; B=like, it's okay for its age; C=like hell

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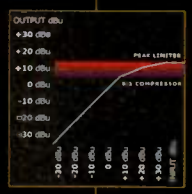

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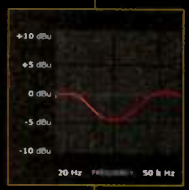

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By Scott Wilkinson

Musical Shoes

Since well before the earliest recorded history of humanity, music and dance have been inexorably intertwined. These two art forms probably developed together, each inspiring in the other the invention of new forms and modalities for expressing what words cannot. And yet, each discipline requires its own practitioners, who must learn their respective skills separately before they can collaborate to create a unified combination of sound and movement.

Dancers can now move to the sound of their own music.

Thanks to modern electronic technology, the nature of this collaboration might well change. Joe Paradiso, a principal research scientist and director of the Responsive Environments Group at the Massachusetts Institute of Technology (MIT) Media Lab (www.media.mit.edu), recently won the 2000 *Discover* Magazine Award for Technical Innovation in Entertainment for his Expressive Footwear system.

In the current version of this system, sensors are embedded within a pair of Nike sneakers with Dr. Scholl's foam inserts (see **Fig. 1**). These sensors include force-sensitive resistors (FSRs) in the toe that measure continuous pressure at two points under the toes and one point above them and a piezoelectric strip that measures changes in pressure under the heel. A pair of back-to-back resistive bend sensors measures the bidirectional bending of the sole.

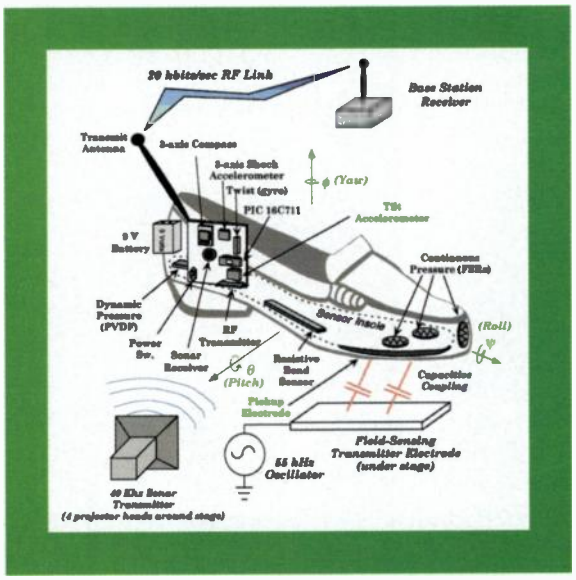
All other sensors are on a small circuit board mounted on the outside of the shoe. A dual-axis, micromechanical accelerometer measures slow rotations about the shoe's length and width axes. You can think of these as the *x* and *y* axes, both of which are parallel to the floor. A 3-axis piezoelectric accelerometer measures sharp impacts and fast kicks in three dimensions. The shoe's spatial orientation is detected by a magnetic sensor that acts like a 3-D compass within the Earth's magnetic field, and a vibrating-reed gyroscope measures the rate of rotation about the *z* axis, which is perpendicular to the floor.

The circuit board is also equipped with a detector that receives sonar "pings" from transmitters located around the dance floor and measures the shoe's relative position within the performance space. In addition, an electrode beneath the insole can measure how far the shoe is above a properly equipped floor.

All signals from the sensors are digitized by an onboard microprocessor and broadcast to an offstage base station from a small FM transmitter and antenna. Two shoes transmit on different frequencies to separate base stations. At a data rate of 20 Kbps from each shoe, all parameter values can be independently updated 50 times per second.

The base stations send the data over a serial connection to a Windows computer running *ROGUS*, a MIDI-processing program written at the Media Lab. This software maps the incoming data to any desired musical structure and sends the appropriate MIDI messages to various synths and other devices to generate music, which can be presequenced motifs and rhythmic loops as well as free-form notes and chords. Various controls modulate volume, pitch, panning, envelopes, tempo, effects, and any other musical parameters.

By measuring 16 different, continuous motion parameters in each shoe, this system offers great potential for generating highly expressive musical events. Of course, making good music still requires a musician to program the system, so the collaboration between musician and dancer will continue. But their roles are likely to change, and it should be very interesting to see how these changes affect the future of dance performance.



COURTESY MIT MEDIA LAB

FIG. 1: The Expressive Footwear system measures 16 different, continuous motion parameters, digitizes the signals, and transmits the data wirelessly to a base station.

A Shining Light
in a Sea of DAW's

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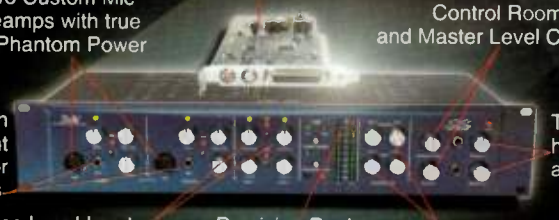

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Credits:
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Recorded with: Merl Saunders, Bob Welch,
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PRO FILE

By Matt Gallagher

Walls and Bridges

Emily Bezar's logistical nightmare becomes a big rock album.

Emily Bezar knows synthesis. Her music, which fuses classical and jazz elements with exotic electronic textures, owes much to her background in classical vocal technique and studies at Stanford University's famed Center for Computer Research in Music and Acoustics (CCRMA).

Faced with a limited recording budget, Bezar has to come up with ways to combine her personal studio with a commercial studio to get the best of both worlds. She perfected this tricky logistical approach for her third release, *Four Walls Bending* (DemiVox Records, 1999).

Four Walls Bending teems with intense, progressive pop. Its compositions and instrumental jams reflect a heightened collaboration with her band: guitarist Morris Acevedo, bassist Andrew Higgins, and drummer Steve Rossi. "I wanted it to be a band album," Bezar says. "It does have a lot of electronic sounds, but you hear the sounds only after you hear the song and the band. I create a world around that performance. The sound serves the song. 'Black Sand' has bleepy, analog, trippy things that remind me of an eerie kind of twilight on a beach."

However, Bezar believes her approach stands in sharp contrast with "a lot of music today, where the individual sounds seem to say more than the song itself." The production began at home with her acoustic piano, a Kurzweil PC88 and K2000, a Power Mac 8500 (upgraded to a G3) running Mark of the Unicorn's *Digital Performer* and BIAS *Peak*, and an Alesis ADAT-XT connected to the Mac through a Korg 1212 I/O PCI card. "It wasn't so much composing at the computer as *arranging*. I look at *Performer* as a big sketch pad." Bezar notes that the song "Sigh" was "a huge jigsaw puzzle. I used *Performer* to juggle song parts until it sounded like the tension and release was right." Next, she burned demo CDs for the band—"more or less blueprints of the final arrangement I wanted on the record"—and rehearsed the band.

***Four Walls Bending*/Emily Bezar**

Four Walls Bending was tracked onto 2-inch analog tape instead of ADAT tape to capture a warmer sound. The rhythm section and vocals were recorded at Coast Recorders in San Francisco, while the acoustic piano was tracked at Spark Studios in Emeryville. "Once we recorded the basics, I transferred them to ADAT tape along with a dub of the SMPTE time code and came home." Bezar returned to her personal studio for the final touches, which included editing, the addition of MIDI sequences, and guitar processing and overdubs. She also created a palette of electronic sounds by looping and processing her synthesizers—including her analog Sequential Circuits Six-Trak—in *Digital Performer*. All of these elements were transferred back to analog tape at Coast Recorders.

The biggest challenge for Bezar was working with the time code to ensure that her ADAT tracks were properly synched with the analog-tape tracks. When she brought her computer and ADAT-XT into Coast Recorders for final mixing, engineers Justin Phelps and Dan Phillips helped her overcome any problems with the system. "It was complicated to try and do this, " she says, "but it worked. When we mixed, we didn't have any sync problems.

"It's when you aspire to create a real electroacoustic sound that it becomes a complex project," Bezar says. "*Four Walls Bending* took a lot of planning, as all big multitracked albums do. I had to have a fair amount of foresight about the final sound of the song before I went in and recorded it. "

For more information, contact DemiVox Records; 2625 Alcatraz Ave., Suite 182, Berkeley, California, 94705; e-mail emily@emilybezar.com; Web www.emilybezar.com.

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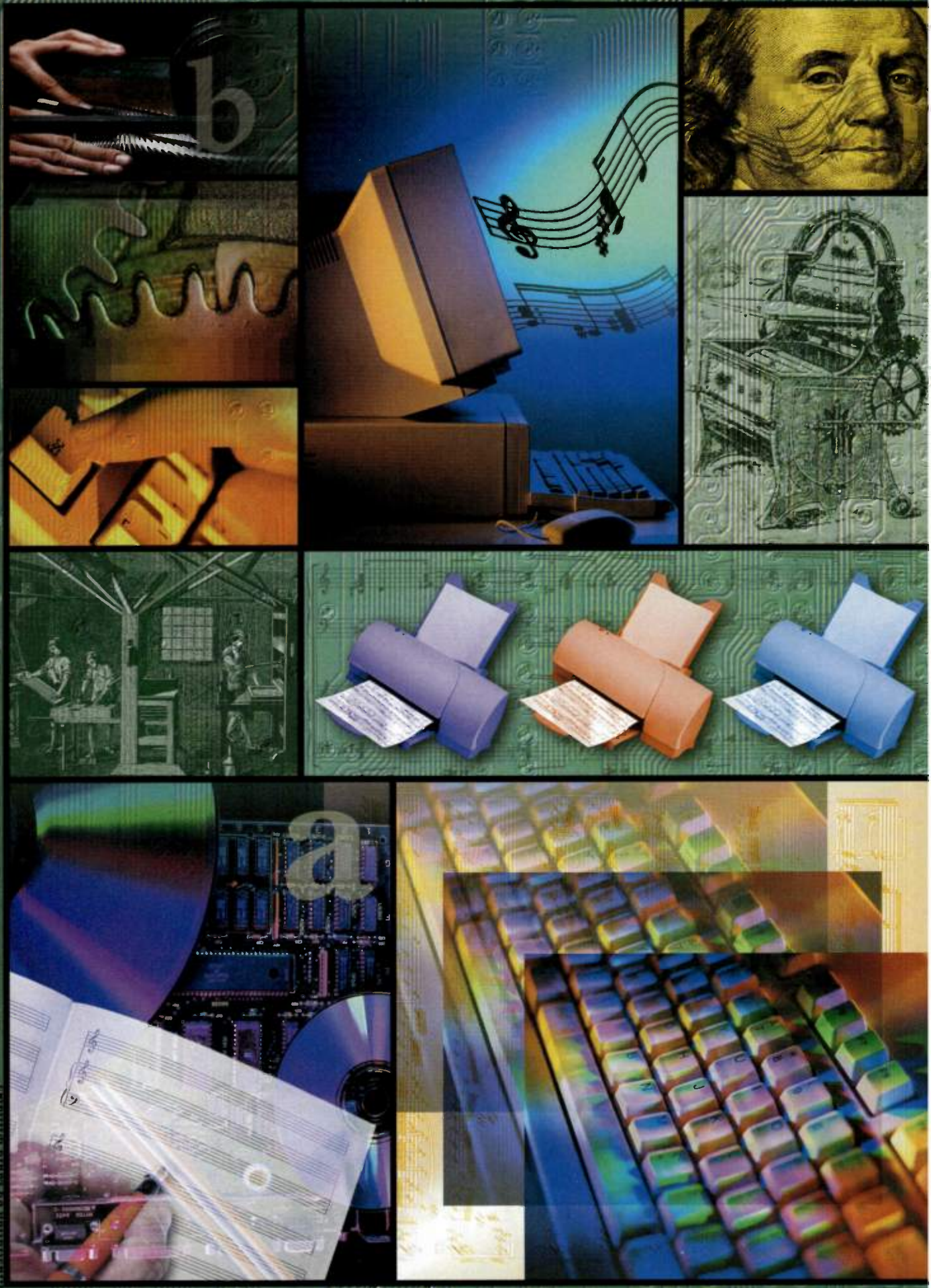


Illustration by Laura Williams

Get It in Print

Master CLASS

There's more to creating good scores and parts than just putting notes on paper.

By Rob Shirak

The quest to capture music in written form has been an exercise in frustration for centuries. As composers and players inevitably discover, quantifying most performance techniques is extremely difficult, and the rhythmic subtleties in many musical styles often defy accurate description. Nevertheless, music transcription remains an important method of transferring ideas, because it enables us to preserve our music and communicate it in a standardized form other performers can understand.

As with desktop-publishing programs, desktop-notation programs have grown more sophisticated while their user interfaces have become more friendly. The evolution has yielded several high-end programs that offer publisher-quality output and very high levels of flexibility. Yet expressing musical ideas through printed music takes more than simply putting notes and symbols on paper. It requires attention to detail and an understanding of how to effectively communicate with other musicians, especially under the stress and strain of recording sessions and stage performances.

With that in mind, I'll discuss some important tips and tricks for score and part preparation that over the years have helped maximize my end results. After covering some general considerations, I'll point out several of my favorite efficiency-boosting features in four popular programs: Cakewalk's *Overture*, Coda's *Finale 2000*, Mark of the Unicorn's *Mosaic*, and Sibelius Software's *Sibelius*.

NOTATION ETIQUETTE

The most important factor in creating music charts for other musicians is readability (see **Fig. 1**). Remember, the players often navigate their parts on the fly, so you'll get the best results from a thorough, well-planned score. In particular, if you're creating charts to be read only during a recording session or performance (rather than for professional publishing or archiving, for example, which may have more stringent guidelines), you can make it easier on the players and avoid slowing down a session or rehearsal by simply adhering to a few basic standards.

For starters, all charts need clear and consistently placed page numbers. Also, put the title (or an abbreviation of it) on each page of the score and parts in a smaller font size. The labels will prevent you from mixing up pages from different charts (see **Fig. 2**).

Be sure all of your scores and parts include rehearsal letters or marks at important sections or transitions in the piece. For example, you might use a separate rehearsal marking for each verse, chorus, and bridge of a pop song. The rehearsal marks don't have to be separate letters; some people prefer to use enlarged measure numbers instead. The idea is to provide a simple reference point for the musicians to use while they're rehearsing or recording. The rehearsal letter is typically large and surrounded by a box (see **Figs. 1** and **2**), although in some programs it's customizable in other ways.

When possible, adjust the layout of the individual parts so that rehearsal letters, key changes, and other important transitions line up with the beginning of a staff system. This lends the chart a more uniform appearance and makes it easier for the players to keep their place, as their eyes and ears perceive a common point of reference on the left side of the page. In addition, use double bar lines liberally; they also help point out significant sections and transitions within a piece.

Long runs of empty measures are typically combined into multimeasure rests, which help condense the parts into fewer pages and thereby reduce the number of page turns. You must still take care, however, to show where all rehearsal letters, double bars, tempo changes, meter changes, and other significant markings and transitions occur. In other words, multimeasure rests must be grouped in a way that makes musical sense, even if it means creating several multimeasure rests in succession to allow for double bars, rehearsal letters, and other musical landmarks. You can't simply throw a 97-bar rest at musicians and expect them to make their entrances reliably (see **Fig. 2**).

For some recording sessions, I don't combine rests into multimeasure rests unless I am absolutely certain that I don't want that particular instrument to play in that section. Instead, I leave the individual bars completely blank (without whole rests) in case I or the player decides to sketch in a new line or add a doubling at the last minute (see **Fig. 2**). It often pays to leave yourself some room for spontaneity or last-minute fixes.

To avoid confusion and wasted time during rehearsals and recording sessions, display measure numbers in every bar. Be sure to put in count-off

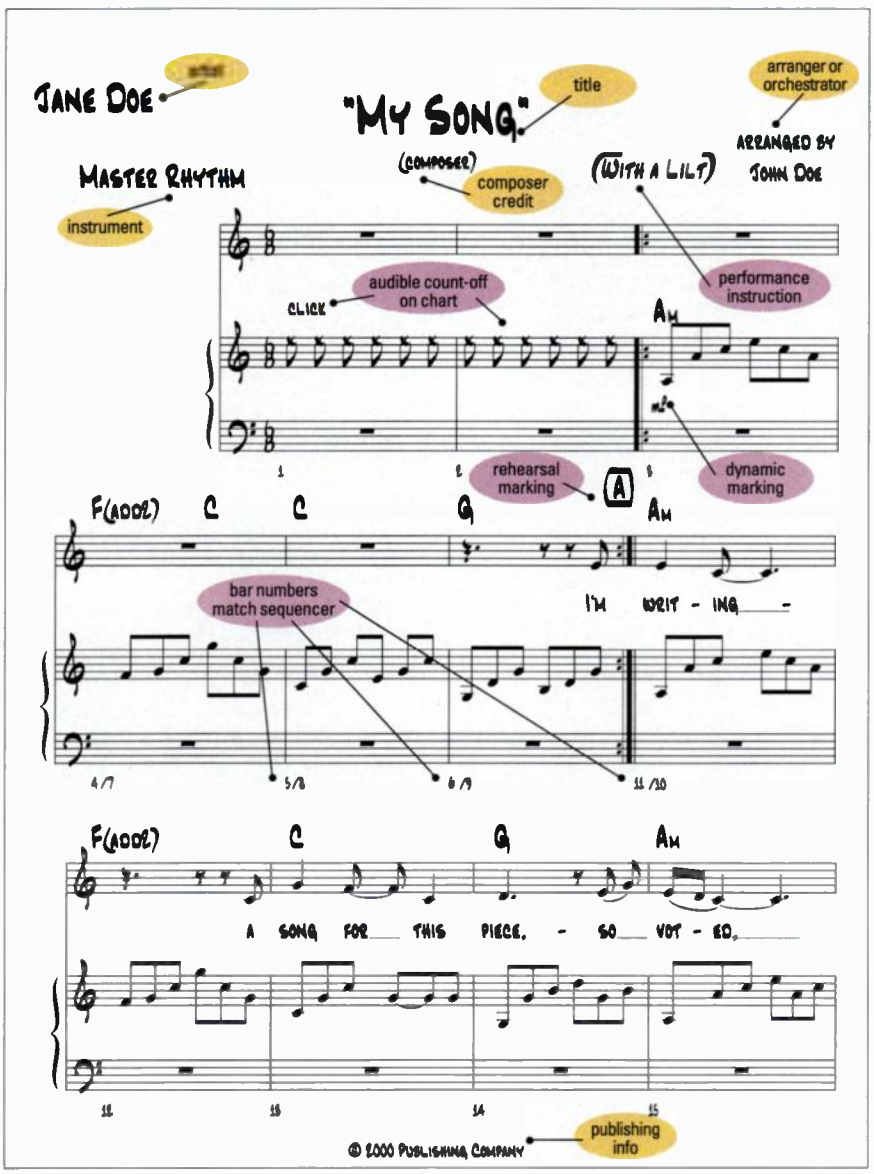
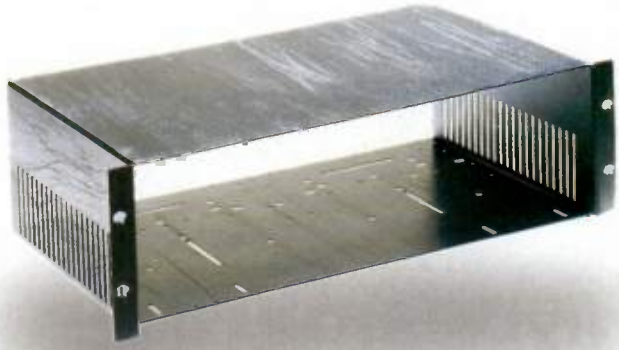


FIG. 1: Many important elements go into the design and layout of a professional score. This title page was created in Coda's *Finale 2000* with the handwritten-style JazzFont set.

ca. 70 million BC

Tyrannosaurus Rex
5.5 tons

ca. 1982 AD



Emmulatoris Akkaisus
13.4 lbs

ca. 2000 AD

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measures at the beginning and any duplicate measure numbers resulting from repeats, alternate endings, and codas. Don't forget to include the range of measure numbers within each multimeasure rest (see **Fig. 2**).

Although individually numbered measures clutter the chart a bit, nearly all professionally copied music parts include them. Especially when a chart has a MIDI-sequence counterpart, it's extremely convenient for the score and sequence to share the same measure structure and form. Measure numbers are most commonly located just under the staff to the immediate right of the bar line, although sometimes the numbers appear above the staff.

Include the initial tempo and any subsequent changes throughout the piece, especially if a click tempo has already been determined. It is also helpful, although not always necessary, to add a few cues of other instruments to help the player keep his or her place in the music and to provide a better overall view of the piece. Despite being more time-consuming, adding the cues ultimately helps ensure proper entrances from the players.

When preparing your score, don't forget the simple yet essential performance guides, such as dynamics markings, articulations, written directives, and other helpful information. Don't be afraid to use straightforward, descriptive text. Foreign language terms such as *adagio* and *molto rallentando* are now mostly relegated to traditional, high-art music literature. If you're writing a contemporary arrangement, film score, or jazz chart, and you want the music loud, put "LOUD!" in the score. If you want a guitar part to be creepy and spooky, or a synth part to sound obnoxious and buzzy, create a text box and place your descriptive phrase in big capital letters at the top of the chart. If you want a song to have the same rhythmic feel as a well-known tune, put that information in the chart too.

Don't underestimate the importance of these elements; professional charts always include them. Musicians who read competently "get" their parts much more quickly if you include stylistic direction.

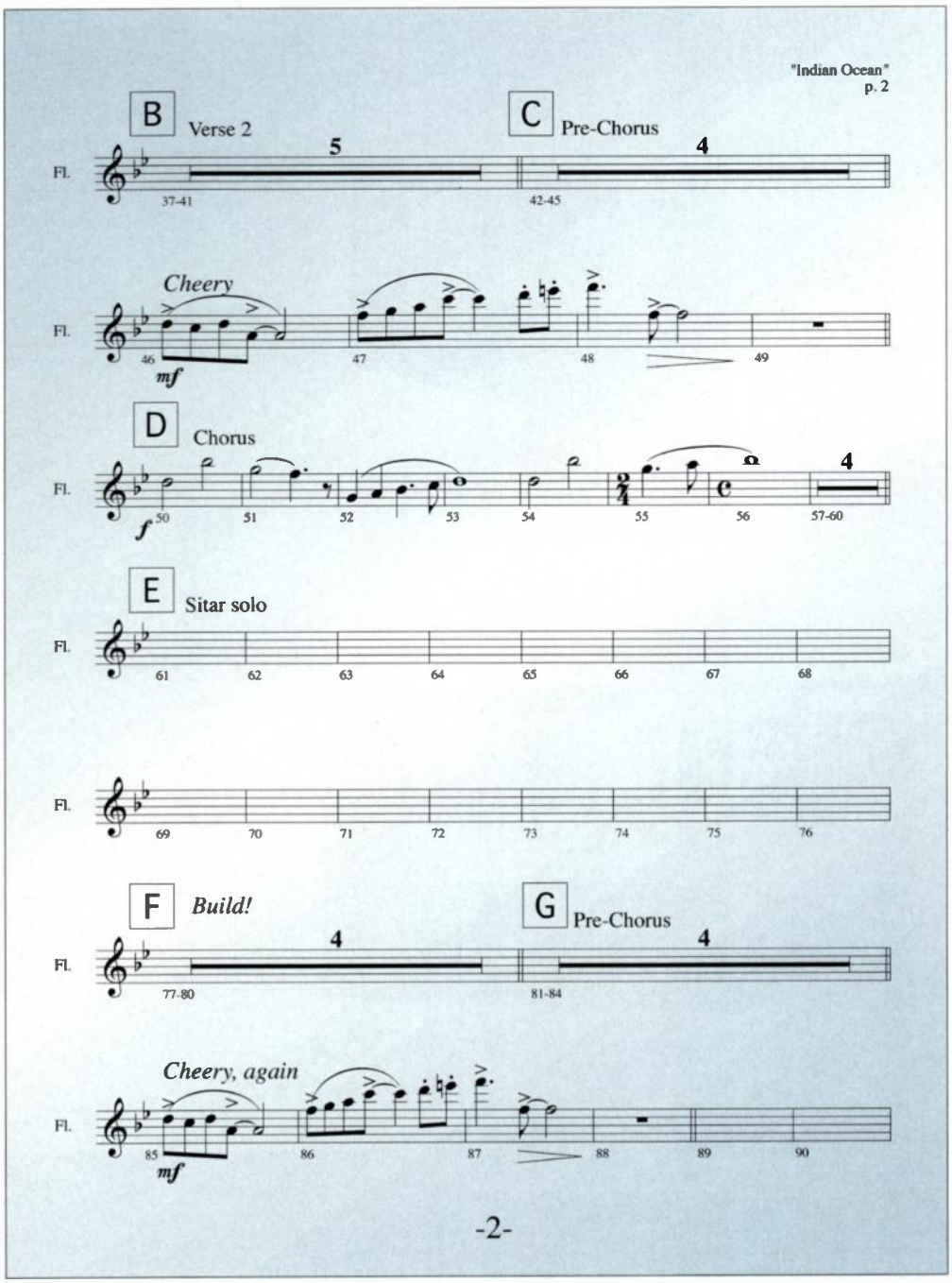

FIG. 2: You must clearly lay out individual parts in a way that makes visual and musical sense. In this example, I've left section E blank so the soloist can pencil in his or her part later. All measures are numbered, and multimeasure rests are divided to reflect important musical changes.

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Their initial level of confidence, especially on the first read-through, can set the tone and attitude for the whole session.

As a courtesy, include the names of the composer and arranger (and any copyright information) on the score as well as the parts. The composer and arranger will appreciate your acknowledgement of their work. A credit for the person responsible for the music preparation (in small print at the bottom of the title page) is also acceptable, especially in a work-for-hire project.

Use large, easily legible fonts for the title, rehearsal letters, and other text-based markings that affect the performance. Although cool fonts abound, the players shouldn't have to work unnecessarily hard to read the text. It's also a good idea to use your text styles consistently. For instance, use italics for performance guidelines, such as "softly" and "staccato," and nonitalics for more global text, such as "Verse" and "Chorus." Whatever your approach, make your style consistent within the piece and between pieces to prevent confusion.

Most professional music is copied onto special 9.5-by-12.5-inch paper, but this size is usually available only in cities with large music communities. Charts on 8.5-by-11-inch paper are gaining acceptance, but the roughly 20-percent size reduction is quite noticeable, especially under less-than-optimal performance conditions. Whichever size you choose, use a heavier stock to prevent visual bleed-through between pages. If you use letter-size paper, print on as much of the surface area as you can, and make sure your layout is clean and legible.

For parts, tape all of the pages together; this keeps the pages in order and saves the players from having to shuffle through loose pages. Professional copyists use a relatively inexpensive tool to tape up parts. Stapling parts is a big no-no. If your music charts serve an ongoing live situation, print and tape up a complete backup book of all your charts, and store them separately from your main charts in case of emergencies.

Most important, read through each part as a player would; check for accuracy, clear placement of vital elements, convenient page turns, and overall readability. This simple exercise will be your best feedback, and I guarantee the players will appreciate the effort.

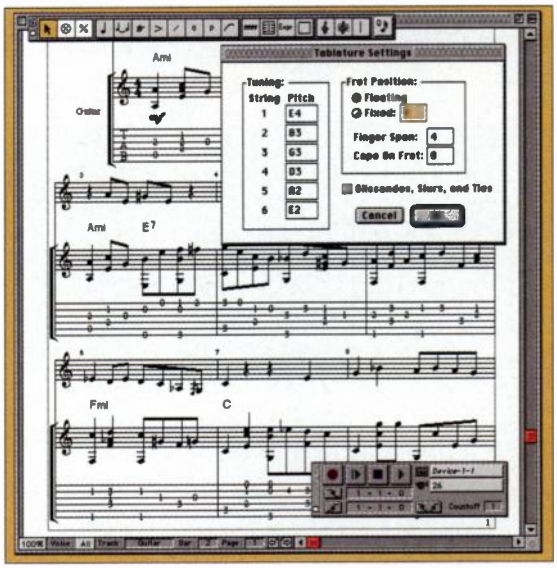

FIG. 4: *Overture*'s easy-to-use tablature capabilities have made the program popular with guitarists.

SET UP TEMPLATES

Nothing wastes more time than repeatedly building a new score from scratch. In all likelihood, you'll need several types of scores and parts, so it makes sense to set them up ahead of time with as much detail as possible. Even though some programs provide setup wizards and/or generic templates to help you set up scores more quickly, there are still plenty of personal touches you'll want to have in place. Creating detailed templates takes time in the beginning, but it saves time later on (when you probably need it more) and ultimately helps you create better charts.

First, create a blank score or open a generic template with the proper number of staves. Name the staves, choose the text font and size, and make other instrument-specific adjustments, such as staff order, staff spacing, and transpositions. Drop in a large dummy title and dummy text for the composer and arranger credits, part name, page number, and other text you'll use in most of your scores (see **Fig. 3**). If your scores contain lyrics, make sure they are properly dummied up as well.

If you'd like to use a specific font (and your notation software offers it), choose it while setting up your initial templates. Changing fonts in the middle of an actual score is extremely difficult; you have to spend substantial amounts of time reworking (or even replacing) large numbers of articulations, accidentals, and other markings. Over time, I have built separate templates for my common instrumentations with both engraver and handwritten font styles.

A good way to create your template is to delete all of the notation data from a previous score that you've fine-tuned to your satisfaction. Then save the score as your setup file. Your templates may not be absolutely perfect for all situations, but they will save you time by minimizing the number of repetitive tasks needed to generate a readable score and parts.

Print a few test scores or parts on the

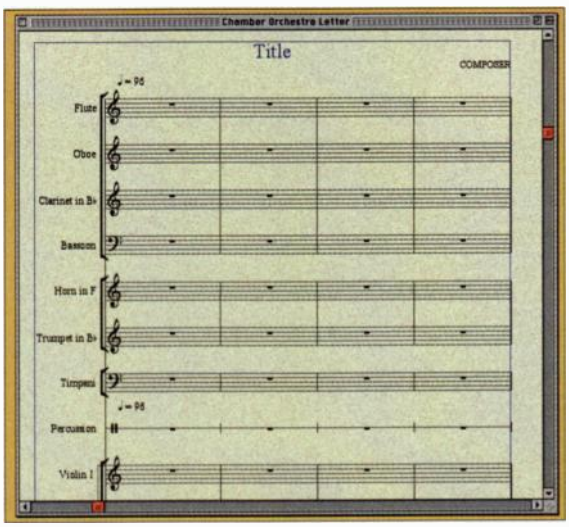

FIG. 3: Using templates can save you a great deal of setup time. You can customize this generic template from *Sibelius* to suit your specific needs.

same paper size you'll use for your final versions, and correct any new problems that unexpectedly show up on paper. If you anticipate using several paper sizes, make separate templates for each size; simply resizing a chart rarely works. Don't forget to read the parts as if you were someone seeing them for the first time. This test ensures that the fonts are sufficiently large and clear, that the text isn't overlapping staves or notes, and so forth. What looks great on one paper size may prove illegible on another. You can eliminate most readability problems with a solid collection of templates.

INPUT METHODS

All midlevel to high-end notation programs provide several methods for inputting notes into a score, including mouse clicking, computer- and MIDI-keyboard entry, and importing Standard MIDI Files. I use combinations of all four input methods, so the placement of my computer keyboard and mouse, monitor, and MIDI keyboard is extremely important. I must be able to keep my hands on both keyboards without looking away from the computer monitor. An ergonomic work environment is essential to the effective use of notation programs.

If you generate music charts primarily by extracting the notes from a MIDI sequence, keep in mind that what looks good on paper and what sounds good are seldom the same. Notation programs are getting better at translating MIDI files into usable notation, but trade-offs inevitably arise.

Using the built-in notation capabilities of most sequencers may enable you to generate adequate parts, but even with sequencers boasting professional-notation capabilities, you still must add many important details such as dynamics, articulations, repeats, text directives, and so forth. And you'll have to add these elements with a less-comprehensive or less-intuitive tool set than you'd find in most dedicated notation programs.

You may also be stuck with a single proprietary music font that doesn't look especially good or provide all of the necessary symbols. Most important, the crucial page-layout and fine-tuning features that ensure proper printing are typically missing. The situation is improving, but I'm usually disappointed with the results when I try to create great-looking charts with a sequencer.

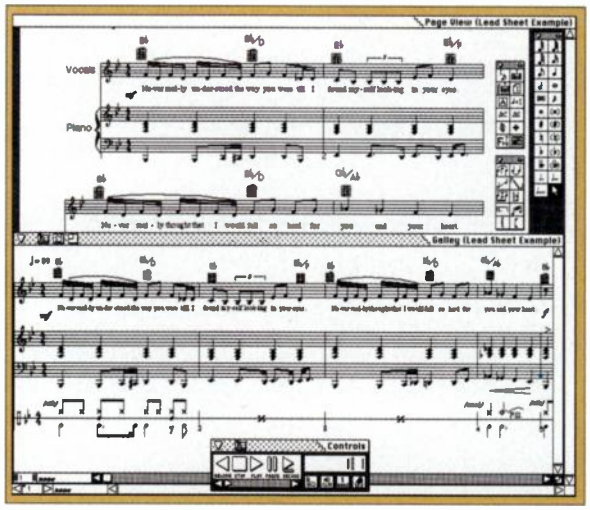

FIG. 6: *Mosaic* allows you to see different views of the same piece. In this example, Page view and Galley view appear together. Viewing scores and parts together can speed up many editing operations.

On the other hand, importing a MIDI file into a notation program offers the advantage of a more complete set of tools and an interface optimized for producing high-quality output. How you prepare the MIDI file before exporting it from your sequencer, however, can greatly impact the amount of time you spend tweaking the music once it's imported.

For the best results, create a copy of your sequence and rename it, so you'll know it's for notation purposes only. Working in the sequencer with the sequence copy, quantize any instrument parts that will be printed. In addition to having the attacks lined up on the beat, remember to quantize the note durations according to how you want the notation to look. Otherwise, you could end up with a bunch of unwanted 32nd- and 64th-note rests and tied notes scattered around the page. At this point, use your sequencer's built-in notation capabilities to check the quantization process.

Once you're satisfied with the quantization and basic transcription, export the sequence as a Standard MIDI File and open it in your notation software. The notation's overall appearance should be much better, although you'll still need to deal with the page layout and the other performance-related details.

SOME NOTES ON NOTATION

Speed versus power has always been a big issue with notation programs, but the truth is that the feature sets in high-end programs have all become quite extensive, and today's computers have largely erased most performance problems. From a practical standpoint, the real differences between programs lie in their user interfaces, ease of operation, and level of editing detail.

Current notation applications all have inherent strengths and weaknesses. For example, some don't link scores and individual parts in the same file, so if you make changes in an extracted part, you have to make the same change in the score as well. Some programs make part extraction a painful process by forcing you to readjust elements that were perfectly placed in the score. Other applications force you to use their own fonts, which may not have all the necessary symbols and may not look the way you want them to. Finally, some programs simply can't do all the things that you really need them to do.

Nevertheless, most programs today are surprisingly powerful and flexible. Here are some noteworthy features in several popular high-end programs that can help you work more effectively.

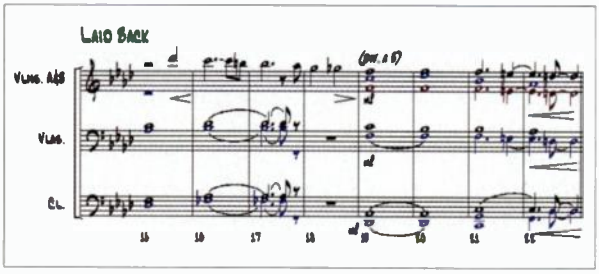

FIG. 5: *Finale 2000* makes excellent use of color. In this example, individual voices on a single staff are easily identified by their different colors. In addition, the performance instruction on the left stands out in green for easier editing.

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Cakewalk *Overture* (Mac/Win). *Overture* was originally Opcode's baby, but now Cakewalk is developing it. If you're familiar with *Studio Vision*, you'll feel right at home with *Overture*, which has a reputation for being fast and easy to use.

Overture lets you enter music solely with the computer's keyboard (you don't have to use the mouse). Once you get the hang of it, you can really fly, so some users prefer this method. It's often the fastest way to get notes into the computer.

Overture's shortcut keys allow you to add ties and slurs very quickly by simply dragging across or highlighting a group of notes and hitting the appropriate key combination. Other key combinations change enharmonic spellings, flip stems, and insert meter or key-signature changes. You can also cycle through the eight available voices per staff and toggle between the arrow cursor and whichever input cursor you last used—a handy feature.

Overture is particularly appealing to guitarists, because creating tablature is so easy. Simply enter an empty tablature staff below an existing staff (with standard notation) that you would like to convert, highlight the notes in the upper staff, and select Notes to Tablature. That's all there is to it. The music is instantly transcribed into tablature (see **Fig. 4**). You can even specify alternate tunings and other parameters such as allowable finger span. (*Sibelius* and *Finale* offer similar capabilities.)

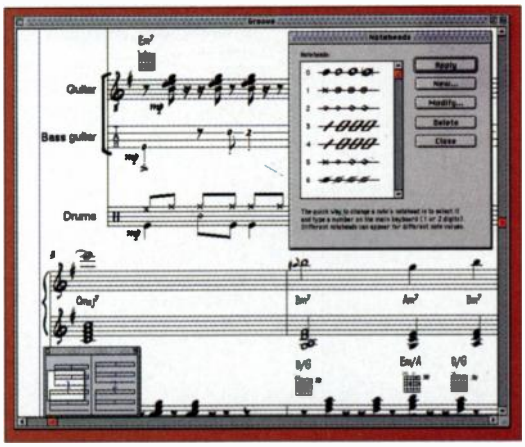

FIG. 7: *Sibelius* offers an extensive set of useful keyboard shortcuts. For example, you can replace note heads by simply selecting a new note-head type with the keyboard number keys.

Overture also lets you attach MIDI playback messages to its expression markings (as does *Finale*). Double-click on a marking and enter the data (tempo, patch, or controller value) in a dialog box. You can use this feature to create a Program Change between an open and muted trumpet sound, for instance, or to create a tempo change during playback.

Coda *Finale 2000* (Mac/Win). *Finale* is one of the most popular notation

programs among professional arrangers and copyists, music publishers, and educators. Learning to use *Finale*'s extensive set of features takes some time, but the recent addition of setup wizards and an improved interface have made the program more manageable than it used to be.

Finale allows you to assign colors to different elements, making them much easier to identify on your computer monitor. You can have up to four polyphonic voices (in either of two layers) on a staff at one time and assign different colors to the different instrumental lines sharing the staff. The colors make keeping track of the lines much easier, especially when the parts cross (see **Fig. 5**).

Finale automatically creates the correct transpositions for specific instruments, but it still allows you to work on the score in concert pitch.

▼

Multimeasure rests must be grouped in a way that makes musical sense.

You toggle the Display In Concert Pitch on and off in the Options menu. Input your notes at concert pitch, uncheck the command, and your instrumental lines are automatically transposed correctly. (*Sibelius* has a similar feature.)

The Program Options command opens a dialog box with more global parameters. One is a checkbox labeled "Load Application into RAM" (in the Mac version). If you have sufficient memory, checking this option significantly improves the overall program performance. If you ever have to trash your *Finale* Preferences file, be sure to recheck this box.

Finale's numerous plug-ins automate a lot of common tasks, but many users still shy away from these handy time-savers. The plug-ins extend *Finale*'s capabilities and greatly speed up operations that previously required wading through a variety of menus. The *Automatic Tablature* plug-in, for example, rivals *Overture*'s tablature capabilities, and the *Piano Reduction* plug-in reduces several staves into a piano grand staff. (*Sibelius* offers a similar plug-in.) You can nest your favorite plug-ins at the top of the list in the Plug-ins window, and you can create your own plug-ins by downloading a Plug-in Developer Kit from Coda's Web site.

For those using Mark of the Unicorn hardware or software, *Finale 2000* (as well as *Sibelius*) now supports MOTU's FreeMIDI. I use FreeMIDI, and it works great—although the original shipping version of *Finale 2000* had a bad habit of forgetting its MIDI settings (regardless of the chosen driver). Be sure to download the current update from the Coda Web site.

Mark of the Unicorn *Mosaic* (Mac). MOTU's notation program has a lot going for it, but development by the company has recently been lagging, and they've offered no indication of when another update might arrive. In spite of this, *Mosaic* has a host of loyal users around the world, and they have an active online user's list that posts work-arounds and other tips.

Some of *Mosaic*'s features are excellent, especially its extensive drag-and-drop capabilities. In addition, it has one of the best engines for converting MIDI files into usable notation. *Digital Performer* owners can synchronize the two applications in real time, allowing better-sounding MIDI playback in the background while they're working simultaneously on the notation file.

Mosaic draws a distinction between voices (instrumental lines) and staves. If you clone a voice onto different staves, editing one voice changes all of the others as well. This is a great feature for working with unison solo lines or doubled parts.

Because scores and parts are linked in the same file, changes to one are reflected in the other. However, when you extract a part, a lot of things tend to move around. If you're working on a score with only a few lines, you can minimize the amount of time you spend redoing your work by using the same staff and font size in both the score and in the individual parts.

First adjust your score to your satisfaction. Then create additional staves for each part that you want to extract. Assign the same voices to the new staves and use them for the Part views. This lets you preserve the score's layout, but still edit the parts as necessary for later printing. Changes made to notes in the voices are reflected in both parts, because the voices are shared.

Mosaic's Multiple Views feature allows you to simultaneously open windows for the score and parts, Galley view and Page view, and different zoom levels. If you have a large monitor or multiple monitors, try setting up the windows so that you can edit the score in Galley view and the part-layout in Page view at the same time (see **Fig. 6**).

Don't forget that *Mosaic*'s excellent custom-key-binding feature lets you assign any of the program's functions to almost any keystroke or key combination, which can significantly speed up many of your editing tasks.

Sibelius Software *Sibelius* (Mac/Win). A relative newcomer to the United States, *Sibelius* has been catching on like wildfire during the past couple of years. The program is fast and easy to use, its output looks great, and like *Finale*, it offers a handwritten-style font.

Sibelius has excellent customizable keyboard shortcuts. Many of them are quite easy to remember: K is for adding a key signature, T for a time signature, M for rehearsal marks, and so forth. For many other operations, *Sibelius* offers a similarly direct approach. For example, if you want to change several note heads—say, to percussion note heads—you highlight the notes you want to change and press one of the number keys to select one of several note-head types. A dialog box also displays all of the note-head options (see **Fig. 7**), but learning the number-key equivalents can greatly speed up rhythm-part editing.

Once you learn the program, you can perform almost every operation using the computer keyboard—often with just one hand. If you prefer, you can also work mostly with the mouse or combine the mouse and keyboard. (The other programs give you similar options.)

▼

Be sure to use large, easily legible fonts.

Sibelius also offers plug-ins that automate a lot of common operations. You can create your own plug-ins using the built-in ManuScript language. The *Keyboard Reduction* plug-in (similar to *Finale*'s *Piano Reduction* plug-in) is especially useful; it creates a piano reduction from a full score.

Sibelius handles tuplets better than just about any other program. Enter the first note, hit Control-3 (for triplets), and enter the rest of the notes. (*Finale* has a similar setup.) You can create any tuplet by typing the appropriate number; for example, Control-5 for quintuplets, Control-7 for septuplets, and so on. You can also easily nest tuplets within tuplets, which is very cool.

Composer/producer **Rob Shirak** *recently served as one of the music directors for the 72nd Academy Awards. He is the keyboardist and music director for Burt Bacharach and has performed with numerous recording artists.*

Introducing

WaveCenter/PCI

OK, you see what's happening: digital mixers are looking pretty cool. After all, they've got incredible sonics, built-in effects, and the automation capabilities you could only dream about before. But if you hook that puppy up to the NoiseRacket analog soundcard that came with your computer, you're right back in ****ville. (Rhymes with "Snapville.")

Here's a better idea: keep it digital with our new **WaveCenter/PCI** card. It has all the connections you need to integrate your digital mixer into your computer-based studio. Transfer up to 10 channels of digital audio simultaneously using ADAT lightpipe and S/PDIF, with 24-bit resolution on all channels. Use one of the built-in MIDI ports for mixer automation, and the other to connect your synthesizers.

Built from the same technological foundation that's made our Dakota card the leading digital I/O and MIDI solution for the PC, WaveCenter/PCI provides all the connections you need to integrate your PC or Mac with your digital mixer.

Your card

The perfect partner for your digital mixer

- 8 channels of ADAT optical I/O
- Stereo S/PDIF (coax or optical)
- High performance 2x2 MIDI interface
- Works with both Windows and Macintosh computers
- Fast PCI bus interface
- Direct interface for Nemesys GigaSampler software
- 24-bit resolution on all channels
- Includes Cool Edit Pro SE for out-of-the box multitrack recording
- ASIO drivers for PC and Mac

Amazing price: $389!

Check it out online:
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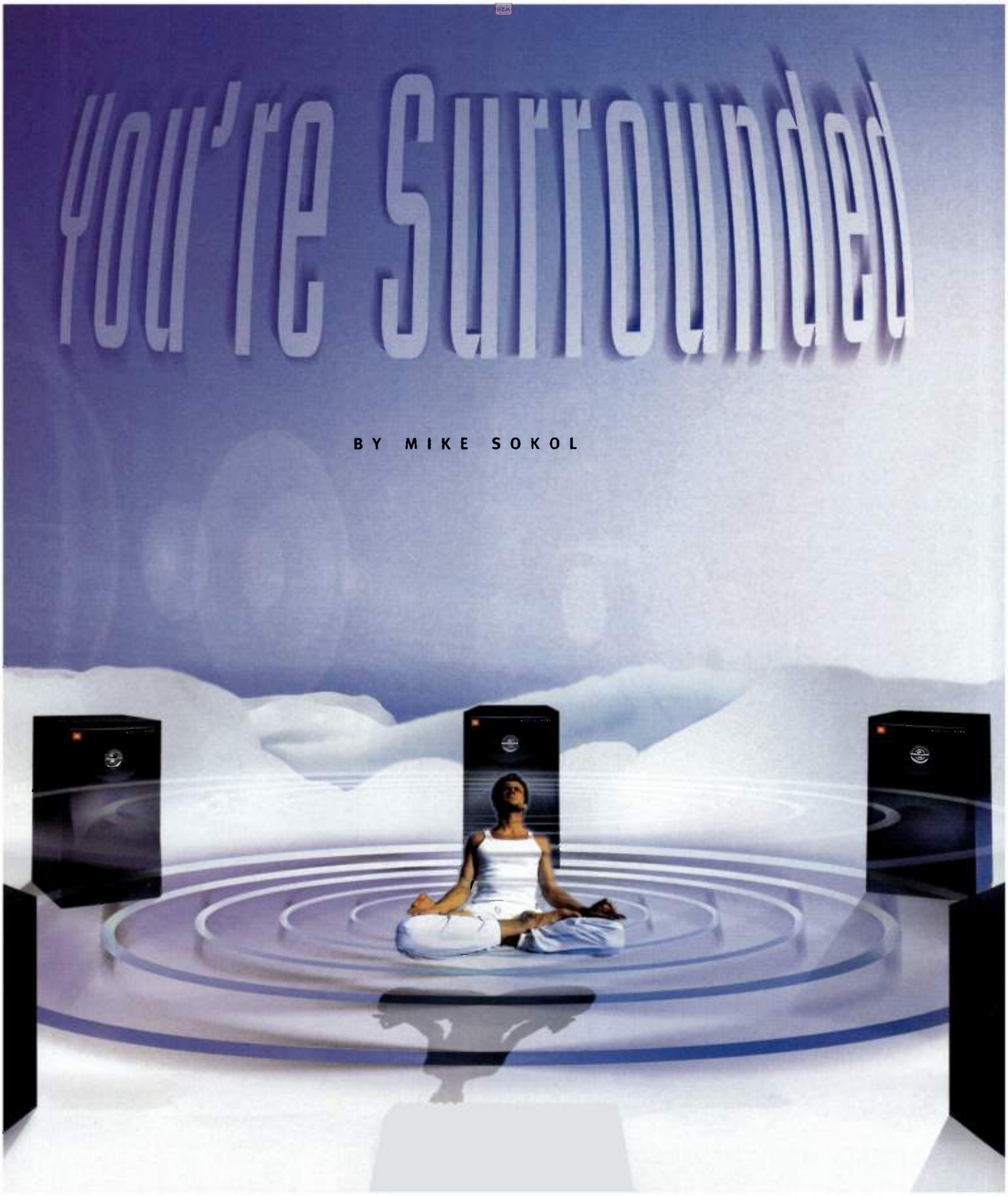
You're Surrounded

BY MIKE SOKOL



How to prepare for the coming storm of 5.1-surround projects.

Unless you've been asleep under a rock for the past ten years, you've at least heard about surround sound. During the past several years, it has had a serious impact on movie soundtracks, and it could replace stereo as the de facto music-listening standard. But surround is a new format, and lots of myths, misinformation, and misconceptions about it are floating around. So if you're not sure what terms like AC-3, DTS, DVD-A, and 5.1 mean, you're in good company.

You can provide customers with a multichannel surround-sound experience in two different ways. In one method, a number of audio processors synthesize a multichannel signal from 2-channel stereo sources, but no magic surround processor will transform an existing stereo mix into a proper surround mix. Most home-theater systems have some sort of "ambience" setting for this purpose, but it's the worst-sounding effect you can imagine. A few of the more advanced processors include "matrix" modes, some of which work pretty well, but these modes will never equal a true 5.1-surround mix.

A much better option is to create original multichannel mixes from scratch. Mixing in 5.1 surround requires some extra equipment, which means learning some new techniques. But it's worth the effort; mixing music in 5.1 surround is the most exciting thing I've been involved in. We are witnessing a milestone in audio history. Millions of home-theater systems in the United States can play back surround music mixes and movie soundtracks. It's time for the recording industry to put out surround product—and it's not as difficult as you might think.

ILLUSTRATION BY DMITRY PANICH/MEDITATOR IMAGE BY SUPERSTOCK

EARLY BIRDS AND WORMS

I'll start with a little historical perspective. You're probably familiar with monaural (aka mono) sound: you deal with only one channel of music information. The information can be as simple as a cheap clock radio with a tiny 1-inch plastic speaker or as complex as a concert sound system with dozens of tri-amped cabinets in large speaker stacks.

Of course, stereo is our standard listening format today. In stereo, you work with two distinct information channels, and you can position various instruments between two speakers typically arranged in front of the listener in an approximately 60-degree spread. This technique allows for some pretty cool psychoacoustic tricks which make the sound seem to emanate from a position between the speakers, even though no sound source exists in the center. This is called *virtual,* or *phantom, center.*

You can produce a stereo *sound field,* or *soundstage,* in a variety of ways. The easiest is to use a pair of microphones on a live group and direct these two channels to the speakers. But most music today is recorded in multitrack format. The channels are panned left, right, or in between at mixdown, creating an artificial soundstage.

Of course, stereo has been king for more than 30 years. The industry made a brief excursion into *quadraphonic* (4-channel) sound back in the 1970s, but squeezing four channels of music into a single record groove was beyond the technology's capabilities. A small group of home experimenters actually set up quad systems, and a few music recordings were released on LP and 4-track reel-to-reel tape. However, quad died an ignoble death, and many people ended up with expensive gear and nothing notable to play on it. The industry never really recovered from this brief affair with quad. To this day, you can make many record executives jump by mentioning the "Q word."

SURROUND REBORN

The movie industry revived the idea of surround sound. Aside from the incredibly ingenious multichannel soundtrack of Disney's *Fantasia,* the first real breakthrough was Dolby Surround, which offered left, center, and right front channels as well as a monaural, limited-bandwidth rear channel for special effects, such as the sound of Superman flying overhead. This mono rear channel was normally reproduced with two speakers to the sides of the listening area.

However, squeezing four channels of sound information onto the two audio channels of 35 mm film proved an imperfect solution. Playback with different Dolby Surround decoders could vary radically. More advanced decoders, such as Dolby Pro Logic, were designed, but they all suffered from the dreaded "phase-steering" problems, in which a level change in one channel could affect the mix in the other speakers.

Enter the digital age. The development of the compact disc in the early 1980s provided a way to deliver large amounts of digital data. Bits is bits, so the same bits could represent a graphic picture, your accounting information, or more audio. Tomlinson Holman (the "TH" in THX) was one of the leaders in surround sound in those days, and from his experiments with movie soundtracks, the term *5.1* (pronounced "five point one") was born.

The 5.1 format defines six discrete channels: five full-bandwidth channels (20 Hz to 20 kHz), and one low frequency effects (LFE) channel (the "point one" in 5.1) with a frequency response rated from 5 to 125 Hz. The LFE channel requires a specialized speaker, called a *subwoofer,* which reproduces only low frequencies. Few, if any, subwoofers can reproduce 5 Hz; most can reach down to 30 or 35 Hz before they roll off, and a few of the more expensive ones can go to 20 Hz. The channels are designated left, right, center, left surround, right surround, and LFE.

The bright people at Dolby Laboratories figured out how to digitally compress these six channels of information into a form that would take up less bandwidth than two stereo PCM tracks, and the Dolby Digital codec (coder-decoder) was born. Also known as Dolby AC-3, this codec is used on many current DVD movie soundtracks, and it is part of the High-Definition Television (HDTV) standard.

The situation remained static for a few years, but with the release of the movie *Jurassic Park,* a competing codec was introduced by Digital Theater Systems (DTS). The DTS codec (DTS is the name of both the format and the developer)

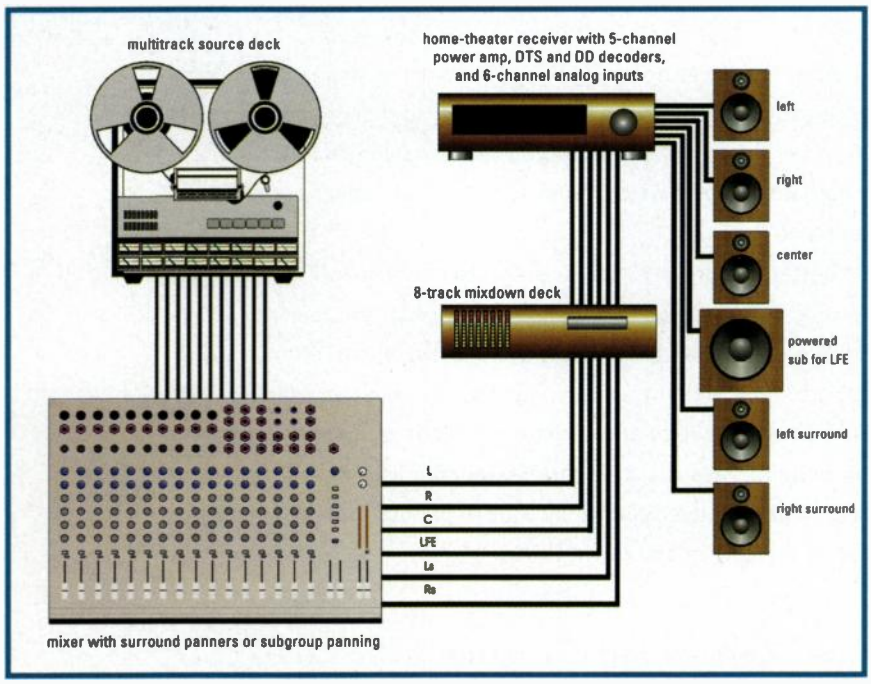


FIG. 1: You'll need two multitrack decks: one for the source tracks and an 8-track deck for five full-bandwidth tracks, an LFE track, and a separate stereo mix. The receiver includes bass management, which redirects low-frequency information in the main channels to the powered subwoofer for monitoring. This does not affect the main mix tracks, which must include audio down to 20 Hz.

For manufacturer information, please see Contact Sheet, p. 221.

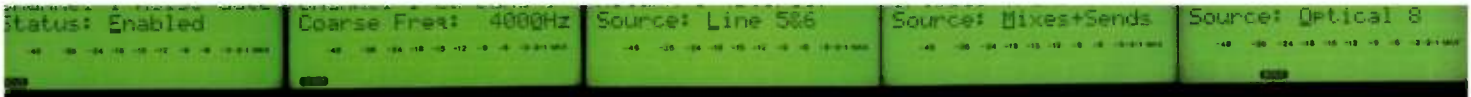
ezbus™

Computer Audio Recording Interface

Software Control Surface

Stand-Alone Digital Mixer

EVENT®
ELECTRONICS



Computer Audio Recording Interface

When a tune hits you need to respond *fast*—before the magic moment is lost forever. So when inspiration strikes, grab an EZbus, and hot-plug the latest and greatest recorder/processor/controller/mixer into your computer's (or laptop's) USB port and get busy making music.

(Or you could use a traditional interface and bust open your computer, wrestle with IRQ conflicts, and troubleshoot a couple of failed driver installations.)

May we suggest Option #1? Your creative juices will thank you....

EZ

Because it's really (*really*) EZ to use.

Because it makes getting precision 24-bit audio into (and out of) your computer a no-brainer.

Because it's got a ton of gozintas and gozoutas, and you've gotta ton of things that need gettininta and gettinouta. (Just look at the huge variety of connectors on the front and back panels—the EZbus can accommodate just about any source signal—analog or digital—you can imagine.)

Because you can create, store, and instantly recall custom mixes designed exclusively for your computer audio recording environment.

Because everything you need—from precision 24-bit/96kHz microphone, instrument, and line inputs to realtime, zero latency dynamics processing and EQ (use it while you're tracking!) to dual MIDI ports to remarkably flexible mixing, monitoring, and routing options—are all built-in, and available at the touch of a button.

(*Cue 50's teen movie dialog track*) "It's positively *dreamy*...."

"Vocal Record +EQ"

reate instantly recallable
tups for recording into
ur computer. Up to 32
lly-programmable mixes
n be stored and recalled
the fly from the front
nel, via footswitch, or
ing MIDI program
ange commands. Or
cord your EZbus mix
oves into your sequencer,
r sophisticated dynamic
tomated mixes.

Cut/boost: +12dB

Each EZbus channel and Main Mix bus offer super responsive, musical EQ—four bands worth, with high and low shelving plus two fully parametric bands. And just in case you go a little overboard, the EZbus's exclusive AudioAlert™ system will let you know if you exceed maximum digital levels and point you to the offending channels.

Attack: 002msec

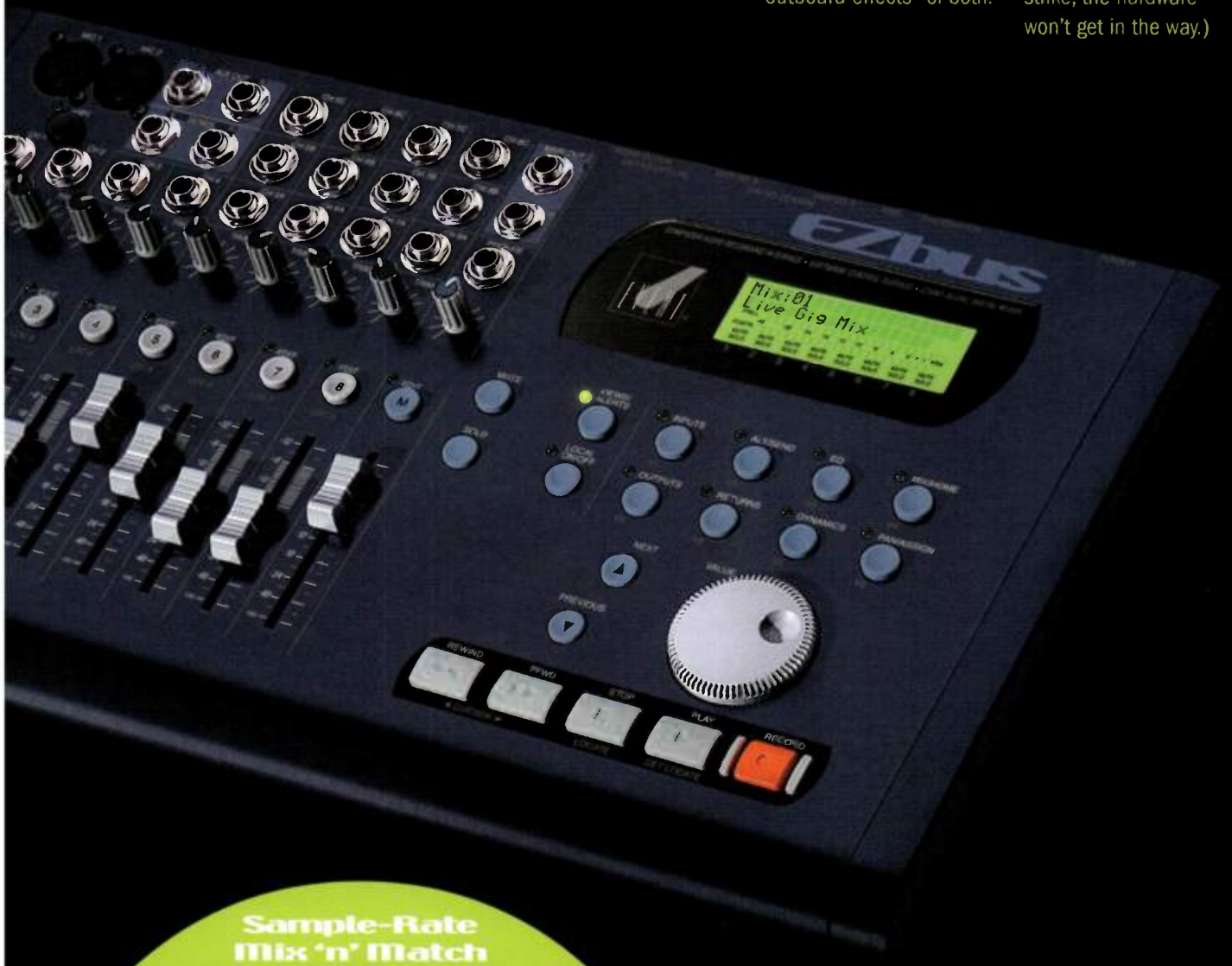
Realtime dynamics processing on every channel means never having to worry about clipping that perfect vocal take. Make your synths breathe with the Expander. Tame your vintage gear with the Noise Gate. And since the full complement of dynamics processing is also available on the Main Mix bus, you can give your final masters a smooth, polished sound.

Source: USB Right

Any analog or digital input can provide the source for an audio channel. Mix and match source signals with total freedom—you can even mix signals with different sample rates. Plus our unique TSI™ Triple Summed Input channels can each accept up to three analog input signals, so it's EZ to accommodate multiple synths, or a rack of outboard effects—or both.

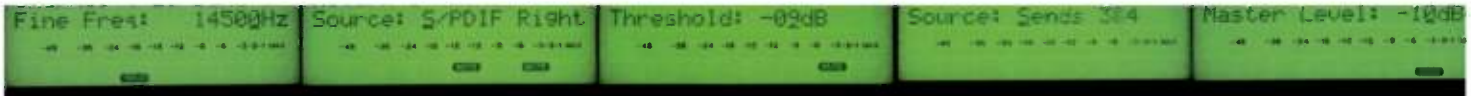
Source: Main Mix

What could be easier than hot-plugging the EZbus into your computer's USB port and getting your musical ideas recorded *fast*? No PCI cards to install. No IRQs to configure. No DMA channels to mess with. Just plug it in and go. (Sorry, you still have to provide the creativity. But we'll make sure that when inspiration does strike, the hardware won't get in the way.)



Sample-Rate Mix 'n' Match

The EZbus supports asynchronous sample rates (*i.e.*, two different sample rates at the same time), performing high quality sample-rate conversion behind the scenes. Why do you care? So you can record at 44.1kHz, for example, and simultaneously bring in DAT tracks via S/PDIF that were recorded at 48kHz. So *now* you care.



Software Control Surface

From Audio Mixer to Control Surface and Back Again

You're probably wondering, "How in the heck do I switch back and forth between audio control and MIDI control and still know where I am?" Fortunately, we have the answers: First, there's the front-panel Localizer™ button that provides instant access to either mode. Then, to let you reconcile between a fader's position and its actual value (assuming you've moved a "virtual" fader while doing some audio mixing), we let you select from four modes of fader operation: Actual, Relative, Null, and Smoothing. Each offers a different operational advantage—one of which is sure to match your creative approach.

EZ

Because you get to control your MIDI/audio sequencing software with its hardware controls.

Because it comes loaded with presets for controlling said software.

Because you can customize the MIDI command and control set to the nth degree.

Because you get full-featured transport controls for easily navigating through your projects.

Because those same transport controls can also generate MIDI Machine Control commands for operating your digital multitrack recorder.

Because you can set and recall multiple locate points.

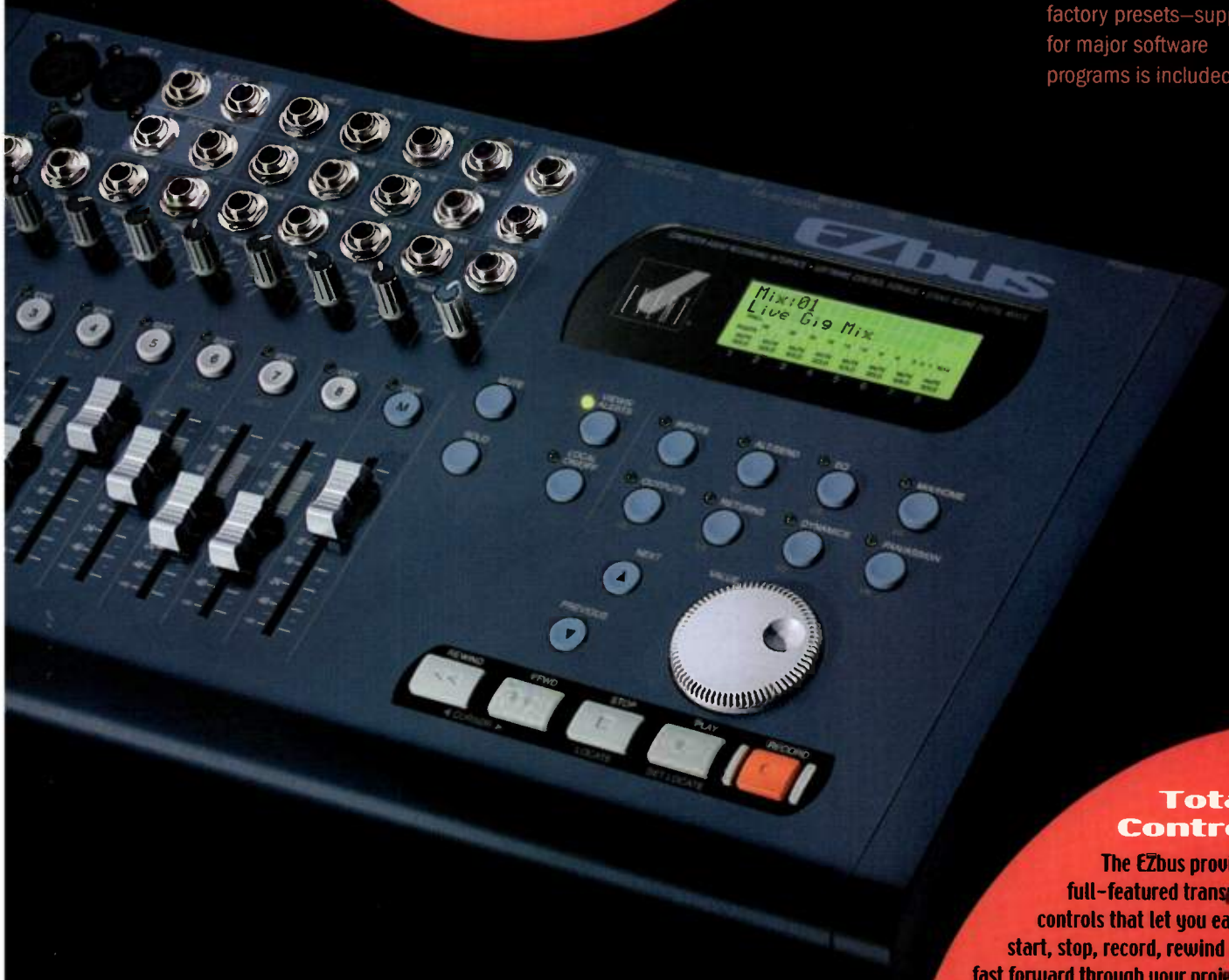
Because the data encoder wheel can control just about any inc/dec operation your sequencer supports—even audio scrubbing.

Because you get random access to fader banks.

Get Plugged in with TSI™

EZbus analog inputs five, six, seven, and eight feature our exclusive TSI™ (Triple Summed Input) architecture, so you can connect up to three audio sources to a single input channel. You could, for example, bring the signals from three stereo synths into inputs 5A and 6A (synth #1), 5B and 6B (synth#2), and 5C and 6C (synth #3). You could then bring the stereo returns from three external effects devices (or three more synths!) into inputs 7A and 8A, 7B and 8B, and 7C and 8C. All together, that's 12 audio signals brought into the EZbus using only four audio channels!

A single button-push is all it takes to instantly transform the EZbus from an Audio Recording Interface into a Control Surface. Then operate your favorite MIDI and audio software using the EZbus's hardware controls—even scrub audio tracks with the data wheel! (Software dependent.) Customize the controls with your own command set, or use the convenient factory presets—support for major software programs is included.



On screen: Steinberg Cubase VST

Total Control

The EZbus provides full-featured transport controls that let you easily start, stop, record, rewind and fast forward through your projects. The same transport controls can also generate MIDI Machine Control commands for operating your digital multitrack recorder(s). You can also set multiple locate points on the fly, and recall them at the push of a button.

Stand-Alone Digital Mixer

EZ

Because it's got enough ins and outs to hook up your entire rig.

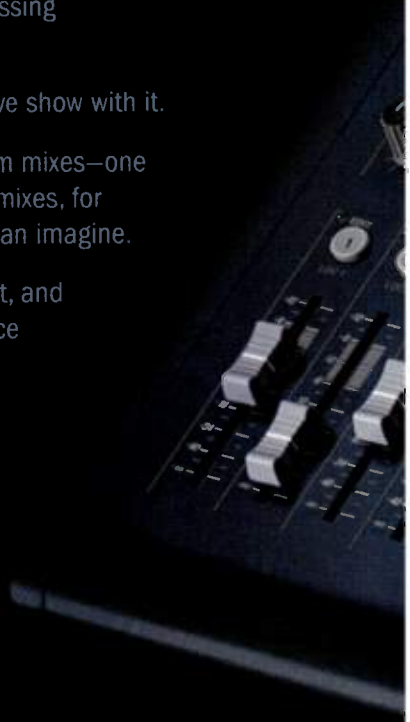
Because it lets you do amazingly sophisticated audio processing with zero hassle (and zero latency!).

Because you can slip it into a gig bag and run your whole live show with it.

Because you can create multiple, instantly recallable custom mixes—one for each tune, for independent live feed and stage monitor mixes, for headphone cues, for just about any audio application you can imagine.

Because its precision 24-bit/96kHz microphone, instrument, and line inputs and multiple format digital inputs let you interface with practically every piece of gear on the planet.

Because its word clock out provides sample-accurate clocking for your entire audio system.



Use Virtual Synths Live...

Simply connect the MIDI out of your controller keyboard to either of the MIDI inputs on the EZbus, then connect your computer to the EZbus via USB. Now, when you play a note on your controller, that MIDI signal goes out of EZbus via USB and triggers the virtual synth. Here's where it gets fun: The audio from the software synth comes back into a couple of EZbus channels via USB, and that signal is mixed with your regular hardware synths (which are plugged into other EZbus channels). Your entire keyboard rig—real and virtual—is now a single, integrated system!

...and Record Your Show!

At the gig, you send the EZbus Main Mix outputs to the nightclub's house console, while the EZbus Aux outputs feed your on-stage monitor system. Now you route the output of the house console back into the EZbus, and send that signal out USB to your laptop, on which you record your whole show. (Of course the recording becomes the CD you sell at the next gig!)

Source: Sends 1&2

Two independent S/PDIF outputs let you route signal to your digital effects processor while at the same time sending the full mix to your trusty DAT recorder. In the meantime, use the Aux and Send jacks to send 24-bit/96kHz signals to your esoteric mastering gear.

Level: -06dB

With four Sends, each independently configurable pre- or post-fader, you can easily create monitor and headphone mixes, patch into your outboard effects, and more. Sends can be routed to any analog or digital output—even to USB—so interfacing with all your outboard gear, old or new, is a snap.

Source: Optical 7&8

In keeping with the EZbus design philosophy of ultra-flexible audio routing, the four Returns can accept audio from any EZbus analog or digital source. Those signals are then automatically routed to the Main Mix bus. Need more inputs? The EZbus Returns provide you with four extra inputs to call on whenever you need them.

Alternate Mix: On

Generate independent control room, stage monitor, and front-of-house console mixes. Or create a four-bus setup for multitrack recording. The EZbus lets you route any signal from any source to the Main or Alternate (or both) Mix buses, with full control over level, bus assignment, and in the case of the Main Mix, EQ and dynamics processing.

Source: Direct Outs

Capture your live performance on your MDM by routing the individual EZbus channel outputs to it via a single optical cable. Now mix the tracks to stereo (using the EZbus, naturally) and burn a CD to sell at your next show! (Yes, the optical outputs can also be used for Send and Mix signals. What, you think we'd limit you now?)

Specifications

Computer Recording Interface

- 18 analog inputs (16 TRS balanced line inputs, two mic preamps with +48V phantom power, two instrument/line inputs)
- 24-bit/96kHz analog-to-digital and digital-to-analog conversion
- Optical I/O for ADAT Lightpipe and S/PDIF (automatically senses source type)
- Coaxial S/PDIF input
- Two independently assignable coaxial S/PDIF outputs
- Eight independent analog outputs (Main Mix Output L/R, Aux Output L/R, Sends 1 & 2, Headphone Output L/R)
- Audio recording via USB on Macintosh or PC computers; USB operation provides full-duplex two-channel 24-bit/48kHz recording/playback
- Two independent MIDI inputs and outputs, plus a third virtual I/O port for EZbus-specific communication
- Word clock output
- Programmable footswitch jack (punch-in, Mix advance, etc.)
- Hot-swappable; requires no IRQ (just plug it in and go!)

Control Surface

- Fully programmable front-panel faders, switches, encoder knob, and transport controls for sending MIDI Controller data via USB and the dual dedicated MIDI outputs
- Presets for Cubase VST (Mac/PC), Logic Audio (Mac/PC), Nuendo (PC), and Cakewalk (PC) included for controlling most mixing functions, including volume, pan, send levels, mute, and solo
- Front panel Localizer™ (Local On/Off) button for instant switching between on-board audio mixing and control surface mode
- Four selectable fader response operation modes (Actual, Null, Relative, and Smoothing)
- Transport controls with jog/shuttle wheel for controlling sequencers as well as any MMC device
- Set and recall up to eight Locate points

Stand-Alone Digital Mixer

- Eight primary audio channels; source signal from any analog or digital input
- Four virtual audio channels (source signal: EZbus Returns)
- Four multi-input analog channels; accept up to three independent source signals per channel
- EZbus Audio Routing Matrix™; provides super-flexible input/output routing
- 4-band EQ on each primary audio channel and on Main Mix; sweepable high and low shelving plus two fully parametric bands
- Programmable dynamics (compressor/expander/gate) on each primary audio channel and on Main Mix
- Mute and Solo on each primary audio channel
- Save and recall 32 internal snapshots of all mix and system parameters
- Two mono Returns and one stereo Return; Returns may be used as additional inputs
- Four Sends per channel, independently assignable pre- or post-fader
- Supports multiple bit-resolutions and sample rates up to 24-bit/96kHz
- Asynchronous sample-rate support via S/PDIF with high quality sample-rate conversion.
- AudioAlert™ function notifies user of errors, such as overloading an analog input, digital dropouts, or clipping due to excessive EQ.
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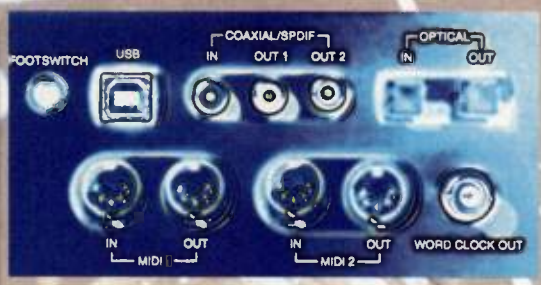

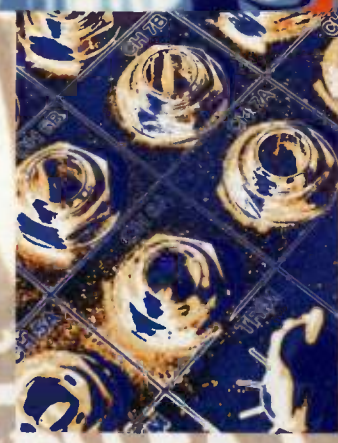

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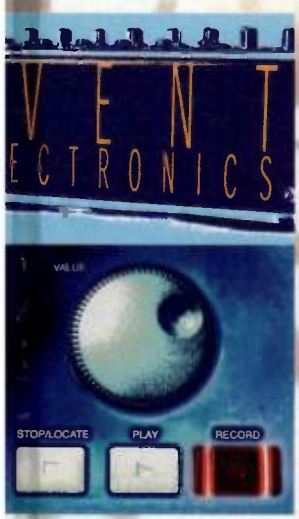

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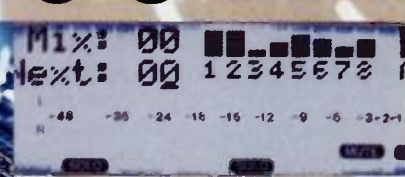

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uses less data compression and requires more bandwidth and data-storage space than Dolby Digital, so some DTS movies don't quite fit on a single DVD. However, the tracks have the potential to sound more like the discrete PCM tracks from which they were derived than is possible with Dolby Digital.

DTS pioneered a way to use the same format on a Red Book CD, but with compressed DTS data in place of PCM stereo audio. DTS also formed a record label to produce remixed 5.1-surround versions of stereo releases. Many of these remixes were done by the engineers who handled the original mixes. Currently, you can buy more than a hundred 5.1 DTS titles, including work from such artists as Steely Dan, Lyle Lovett, and the Eagles.

To play these CDs, you need a CD or DVD player with a digital audio output that can send the DTS bitstream to a DTS decoder, which extracts the six channels of information and converts them to analog. (Early DVD players have a digital output, but they don't recognize the DTS bitstream. Most consumer CD players don't have a digital output, and those that do might not recognize the DTS bitstream.) You also need six channels of amplification and speakers.

BACK TO BASICS

I'll start at the beginning of the mixing chain and go through it step-by-step. You need some special items to mix in 5.1 surround, but most studios already possess 90 percent of the needed equipment. Once you add a few select pieces, you could be mixing surround music in your own studio.

The first thing you need is a multitrack master of the tune you want to mix (see **Fig. 1**). The multitrack format is not an issue; it can be as simple as an 8-track analog tape deck or as complex as a pair of 48-track digital decks. I've done some really cool 5.1-surround mixes using 16- and 24-track ADAT systems. The source tracks can be in any digital or analog format, including a computer workstation. Of course, you want tracks with excellent production values.

You need to route the recorded tracks into a mixing console that lets you pan them between five output channels. (There should also be a sixth output for the LFE channel, but you don't pan anything from the main tracks to this output.) If you have a Yamaha O2R or O1V, Panasonic DA7, Tascam TDM4000, or Mackie Digital 8-Bus, you're already in business. Each of these digital consoles lets you patch the outputs from the *surround matrix* (outputs) to an 8-channel mixing deck (more on this shortly).

If you don't have a console with built-in surround panning, it's relatively simple to patch the equivalent using subgroups or aux sends (see the sidebar "Mixing Surround Without a Matrix"). But for ease of mixing, nothing beats a screen with a picture of the room and a cursor that shows where the sound ends up. Some consoles, such as the DA-7, provide a pair of controls on the work surface to pan left/right and front/rear, while others, such as the Mackie D8B, use a trackball or mouse.

Patch the console's outputs to an 8-track deck, where your final surround

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tracks will reside (see Fig. 1). The Tascam DA-88 has become the standard multitrack deck for surround due to its popularity in the film industry, but any common 8-track format will work, including a computer workstation. You don't need a Dolby Digital or DTS encoder to mix surround tracks; encoding is the last part of the process. Whatever you record your mix on, carefully note the track assignments. Unlike stereo, surround gives you many different track-assignment methods to choose from. The table "Surround Track Assignments" shows a list of the most common track-assignment systems, or modes.

Often you won't be able to choose which channels end up on what tracks. For instance, both the Panasonic DA7 and the Mackie D8B mixers are set up in mode 4, whereas many large-format consoles and some mixing programs (such as Minnetonka's *MX 51* for Windows) are designed to be used in Mode 1. Try to pick one output format and note it on the label of every 5.1 tape you make. Eventually, someone will have to figure out your track assignments to encode them on a disc, and you don't want your sloppy work habits to jeopardize a project.

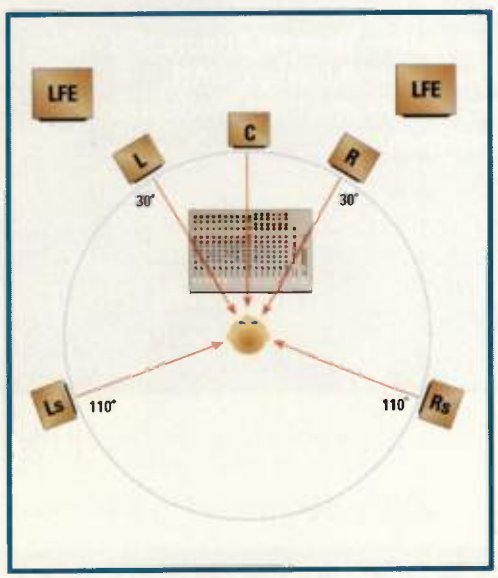

FIG. 2: According to the AES standard for 5.1 monitoring, the front left and right channels should be 30 degrees from center, and the two surround channels should be 110 degrees from the center.

SPEAKING OF SPEAKERS

You'll need to upgrade your monitoring system to include five speakers and a subwoofer so you can hear what you're doing in 5.1 surround. The speakers are arranged in a circle around the listener's head (see Fig. 2), and the channels are labeled L (left), C (center), R (right), Ls (left surround), Rs (right surround), and LFE. Feeding these speakers requires six channels of amplification. Keep the channel definitions in mind as I discuss patching options.

The simplest and perhaps best setup for music mixing is five matched near-field reference monitors used with a subwoofer. I like the JBL LSR-28P and M&K 150 speakers, but I've heard some great mixes on the little Alesis Monitor Ones and Yamaha NS-10s.

Try to match the main speakers as closely as possible, because the relationships between the center, left, and right levels are critical, and the relationships between the front and surround levels affect the final mix much more than you might imagine. You could mix and match speakers—many excellent surround mixes have been done with mismatched center and rear-surround speakers—but I recommend you use a matched set to avoid putting yourself at a disadvantage.

The physical speaker layout is pretty simple (see Fig. 2). Just put a mic stand at the mix position and measure the distance from that position to the location of the center-channel speaker. Then cut a piece of string to the same length and tie one end to the mic stand. Mark the center-speaker position and go 30 degrees to the left and right for the front L/R speaker positions.

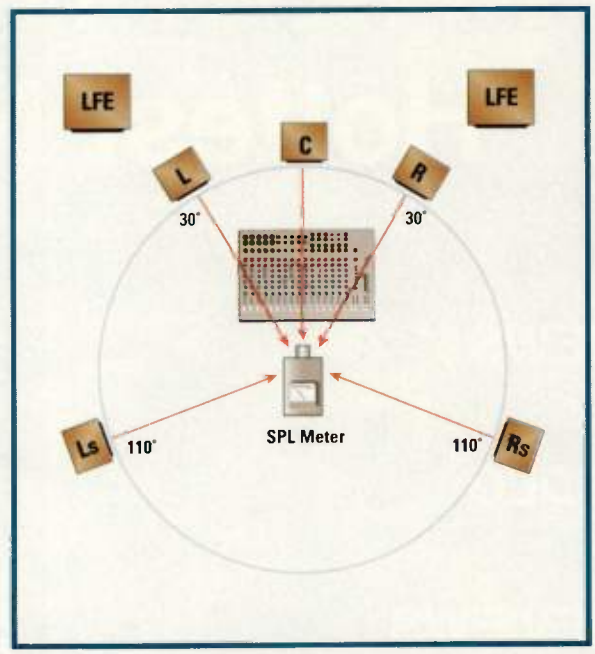

FIG. 3: Calibrating your speakers to ensure that their levels match is absolutely critical to 5.1 mixing. Feed pink noise through each speaker and check the level on an SPL meter. Full-bandwidth speakers should output 85 dB SPL, and the subwoofer should deliver 95 dB.

Next, go 110 degrees to the left and right of the center for the Ls and Rs (surround) channels. This setup is the AES standard for monitor placement.

There is one exception to the "five matched speakers" rule. When mixing movie soundtracks, you don't want direct-radiating, point-source surround speakers, because most movie surround effects need to be diffuse. As a result, the surround speakers should be *dipole* designs, which have drivers that fire forward and backward, creating a diffuse sound field. In this case, the surround speakers should be placed directly to the sides of the mixing position, 90 degrees from the center-speaker location. If you're mixing multichannel music, use a pair of direct-radiating speakers matched to the front three speakers and placed at the 110-degree positions.

YOU NEED MORE POWER

You'll probably need to upgrade your power amps, too. Many big studios can afford separate amplifiers and a bass-management controller to run them, but personal-studio owners should instead consider buying a large home-theater receiver. Priced between $500 and $1,000, such receivers provide more than 100 watts per channel and a

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All the main speakers should get the same amount of power, but the subwoofer probably needs as much juice as all the other speakers combined. Giving the subwoofer two to three times the power of one of the main speakers will probably suffice. It's important to understand that many subwoofers include their own internal power amps matched to their particular drivers. As a result, virtually all modern home-theater receivers include five channels of amplification and a line-level output for the subwoofer.

Make sure the receiver sports Dolby Digital and DTS decoding as well as discrete analog inputs that bypass the decoders. Meant to accommodate future decoders, these inputs are ideal for monitoring the six discrete channels of the mix. With such a receiver, you can compare your mix with commercial mixes from DVDs or DTS CDs. (To hear commercial discs, you also need a DVD or CD player with a digital audio output that passes both DTS and Dolby Digital bitstreams to the receiver.) If you're doing your own DTS or Dolby Digital encoding, the receiver provides the only way to listen to your final mix as the consumer will hear it. These receivers also offer bass management (more on that later) and speaker-calibration options.

I'LL LEVEL WITH YOU

Properly setting the relative volume level of each speaker is extremely important. It's very easy to tell when the left and right levels are wrong in a stereo mix; you can hear the sound leaning one way or the other. Hearing the balance with a surround system is not so simple. You'll need to purchase an SPL (Sound Pressure Level) meter to do this properly, and you'd be amazed at how many people use $50 Radio Shack meters for the job.

I could write a book just about speaker calibration, but for now, here's the quick and dirty. Grab some limited-band (100 Hz to 2 kHz) pink noise from the mixing-console noise generator, a home-theater receiver, or a test CD and patch it to an input channel on the console. Make sure to set the mixer's output gains to unity. Set the console's input strip so that the output level going to the mixing deck is at –20 dB (below 0) on the deck's meter. This is the standard reference level for surround mixes.

Feed the pink noise through one speaker at a time and point the SPL meter in the speaker's direction while holding the meter in the mixing location (see Fig. 3). Trim the gain of the corresponding amplifier channel so the meter reads 85 dB SPL. Repeat this process for each full-bandwidth speaker, one at a time, until you have the same output for each. (Most home-theater receivers include individual level controls for each channel, but they are often buried in a menu system that might require a video monitor for viewing and navigating.)

Next, run low-frequency pink noise (25 to 80 Hz) to the LFE channel. In theory, the gain of the LFE channel should be set 10 dB higher than that of

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SURROUND TRACK ASSIGNMENTS

No standard has been established for assigning channels to tape/disk tracks for surround mixing. However, these six modes are the most commonly used. Make sure to mark every mix tape or file so the track assignments are clear to all who need to work with them. Note that tracks 7 and 8 can be used for recording a separate stereo mix. That way, if the end users choose to listen in stereo, they will hear a real stereo mix rather than a downmixed version of the 5.1 mix.

Track	1	2	3	4	5	6	7	8
Mode 1	L	R	Ls	Rs	C	LFE	Lt	Rt
Mode 2	L	C	R	Ls	Rs	LFE	Lt	Rt
Mode 3	L	Ls	C	Rs	R	LFE	Lt	Rt
Mode 4	L	R	C	LFE	Ls	Rs	Lt	Rt
Mode 5	L	C	Rs	R	Ls	LFE	Lt	Rt
Mode 6	C	L	R	Ls	Rs	LFE	Lt	Rt

Key: L = left, C = center, R = right, Ls = left surround, Rs = right surround, LFE = low frequency effects. Lt and Rt = left and right stereo mix.

the main channels (95 dB SPL), as read by a real-time analyzer (RTA). But a Radio Shack meter has a lot of low-frequency rolloff, and you're feeding it less than two octaves of audio information in this case. These factors cause the meter to read lower than the true output level. As a result, when the LFE level is correct, the Radio Shack meter will show approximately 90 dB SPL—4 to 6 dB higher than the level for the five full-bandwidth speakers.

For diffuse surround speakers used in cinema mixes, the rear surround levels are set to 82 dB SPL (–3 dB relative to the other full-bandwidth speakers). For really small mixing rooms where you can literally reach out and touch the speakers, Dolby recommends setting the surround speakers 2 dB down; that is, at 83 dB SPL. It can be a bit confusing. For most music mixing, having all five speakers set at the same level is close enough.

If you don't get these levels correct, all the mixes you do will have incorrect surround- and center-channel levels, or the LFE level will be out of control. Those problems will force listeners to jump up and adjust the levels on their home systems.

BOTTOM FEEDING

Bass management is probably the least understood part of surround mixing. It's very important to comprehend how it works, lest you make mixes that sound great in your studio but prove unlistenable on a standard home-theater system. As noted earlier, in 5.1 surround, each of the main channels is rated at 20 Hz to 20 kHz, while the LFE channel is rated at 5 to 125 Hz.

Any of the main program channels can go as low as 20 Hz, but very few home-theater speakers can produce output that low. A clever circuit in the preamp/processor or receiver of a home-theater system removes any low-frequency information below a certain point (80 Hz is the THX standard) in the main channels and reroutes it to the subwoofer.

In 5.1 surround, the subwoofer does double duty: it handles all the bass below 80 Hz for all the main channels, as well as the "point one" LFE channel, which might be earthquakes or explosions.

Why do you need bass management (sometimes called *redirection*) in your studio? It's the law of inverse mixing. A speaker that's deficient in part of the audio spectrum forces you to overcompensate for the missing frequencies by adding them to the mix in disproportionate amounts. Try doing a mix with rolled-off tweeters and you'll see what I mean. Because you hear fewer highs than are really going to tape, you'll overcompensate with too much high-frequency level in the mix.

Let's apply this concept to bass management. Suppose you have five NS-10 speakers and a big subwoofer that reproduces *only* the LFE channel (explosions and such). Each speaker directly monitors a final output track. The NS-10 has a small woofer in a small cabinet, so it naturally rolls off anything below 60 Hz or so. If your source tracks have any sonic material with extra bass in the 20 to 40 Hz region, you'll never hear it on these monitors.

This is particularly troublesome if the tracks have some undesirable low-frequency information—maybe some vocal plosives or air-conditioner rumble you weren't aware of. The natural filtering action of the NS-10s might make you think all is well, but when the mix is played back in any home-theater system, the bass-management circuit will faithfully reroute this low-frequency garbage into the LFE subwoofer, where it will be available for all to hear. So if each of your main speakers can't produce down to 20 Hz and you don't have a bass-managed monitor system, you have a potential mixing disaster on your hands.

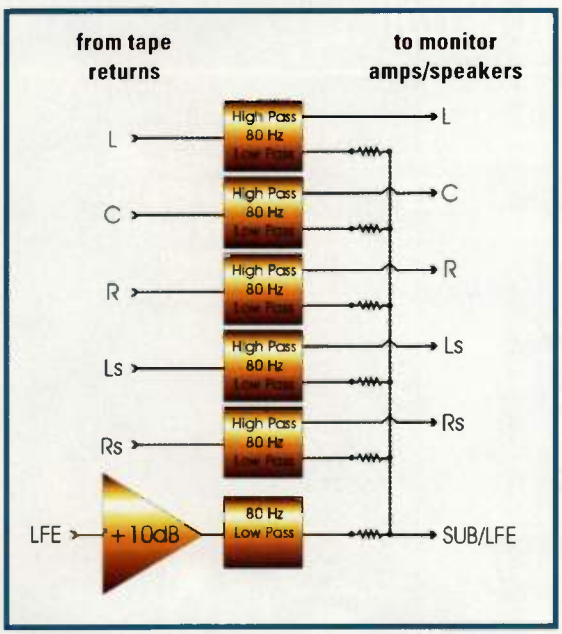

FIG. 4: Although the subwoofer handles the bulk of low-frequency (below 80 Hz) chores, you still need to feed full-bandwidth audio to the other five channels. The bass-management filter shown here affects only your monitor system; the recorded L, R, C, Ls, and Rs channels still contain the entire, unfiltered signal.

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Many mixing engineers think bass management has something to do with filtering the signal before it goes to the final mix-tape tracks. That's simply incorrect. In fact, each of the L, R, C, Ls, and Rs channels should get the full 20 Hz to 20 kHz program signal. You don't want to reroute the lows in these channels to the LFE channel; that's what the bass-management filter in the playback system is for. Instead, you want a bass-management filter in your *monitoring* system that emulates the home-theater playback system (see **Fig. 4**). This filter is placed *after* the mixing tape deck and directly feeds the monitor amplifiers, as shown in **Fig. 1**. If you're using a home-theater receiver for monitoring, it has an integral bass-management circuit, which does the job just fine.

Another misconception is that you must match the 80 or 120 Hz bass-management cutoff points in home systems to properly monitor in 5.1 surround. But because this filter is for playback and monitoring in your studio only, you have to do just what is needed to extend the low-frequency capability of your own monitoring system. Just as we don't care about the crossover frequency of the mid-range driver in a home speaker, we don't know and don't care about what the exact bass-management frequency is in a consumer system. We just know that somewhere around 100 Hz, all the bass energy will head to the big subwoofer cabinet.

My main monitors go down to 35 Hz, so I like to adjust my bass-management cutoff frequency down to 50 or even 40 Hz. This would limit the bass-localization effect (yes, you can localize 80 Hz bass, contrary to popular belief) and take some of the power load off the subwoofer, which attempts to reproduce the bass from the five main channels as well as the LFE information. That's why you should buy the largest subwoofer you can afford and fit in your studio. For excellent information on bass management, download Steve Harvey's paper *Secrets of Doing Surround Sound on Your Existing Console*, published by Martinsound (www. martinsound.com/lb_rp.htm).

Alas, there's no inexpensive way to put a separate speaker controller with bass management into a small studio. Martinsound makes something called the MultiMax controller, which does all the calibration, downmixing, and level control you could want, along with rudimentary bass management. (*Downmixing* refers to creating a stereo mix from a 5.1-surround mix.) It's certainly a great controller, and I take one along on my surround seminars because it's easy to use under stress. But its price of $3,000 is beyond many small-studio budgets.

Studio Technologies makes the Model 68/69 StudioComm ($1,599), which can be combined with the Model 65 Bass Management Controller ($899) to provide a $2,500 solution. These seem like expensive pieces of gear until you start shopping and find

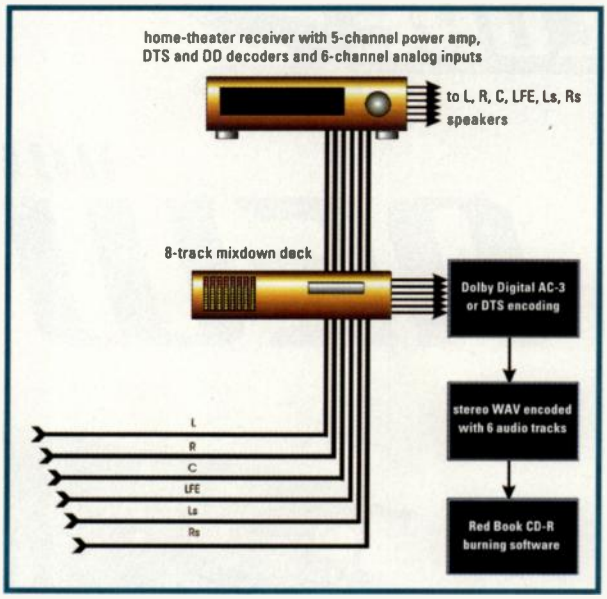

FIG. 5: If you are using a hardware-based system, as opposed to a computer-based system, you need to patch the output of your 8-track mixdown deck to a Dolby Digital or DTS encoder. With a computer-based DAW, of course, this is all done in software. The system shown here produces files that can be burned to CD-R.

MIXING SURROUND WITHOUT A MATRIX

Even if your console has only stereo outputs and some extra auxiliary sends, you can get into the surround-mixing game. Of course, doing fancy spins around the five main speakers is challenging, if not impossible, without a true surround panner that employs a joystick or mouse. But some of my first experiments with static surround mixes were done with limited tools.

For instance, on a console without subgroups, if you have four extra aux outputs, you can patch the stereo bus to the front left and right channels, aux 1 to the center channel, aux 2 to the LFE channel, aux 3 to the left surround, and aux 4 to the right surround. (Of course, if your console has only four aux sends, this leaves you no way to easily add effects.) The LFE output should be sent through a lowpass filter with the cutoff set somewhere around 80 Hz so that the LFE information is completely out at the 125 Hz "speed limit." This patching works fine for static mixes, such as symphonies, in which you're only setting the relative levels in each channel for the duration of the track.

Getting an audio source to pan across the sound field takes a few tricks with subgroups. On a console with at least four subgroups, you can assign buses 1 and 4 to the front left and right channels, bus 2 to the left rear, and bus 3 to the right rear. Patch separate aux buses for the center and LFE channels.

Now, by panning between odd and even buses, you can perform front-to-back moves in the surround sound field. Also, by selecting a combination of buses, such as 1, 2, and 4, you can even manage a diagonal pan. This system can work out quite well with consoles that have pan and subgroup automation.

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devices offering basic functions for up to $15,000. Gaaaaa! What's a personal-studio owner to do?

Using a home-theater receiver starts to look better all the time, because it handles bass management and speaker calibration, it decodes DTS and Dolby Digital discs, and it gives you a big knob that changes the levels of all channels at once. (Remember, you also want a receiver that includes six analog inputs that bypass the decoders so you can monitor the 6-channel mix directly.)

You could combine something like a Yamaha RV-1105 receiver ($700) with five small monitors and a self-powered subwoofer for an affordable speaker/amp/bass-management/calibration system.

DOWNMIXING BASICS

There's one other bugaboo to watch out for when doing surround. All 5.1 mixes might be *downmixed* to stereo at some point. For instance, if the consumer listens to a Dolby Digital DVD or DTS CD of one of your mixes and selects the stereo option on the receiver or surround processor, the six channels of information are mixed down to a pair of stereo channels and sent out the main left and right outputs.

In a downmix, the center-channel information is added into the left and right front channels equally, while the left and right surround channels are added into the left and right front channels. Some systems add the LFE channel into the stereo pair, and other systems throw away the LFE information.

This would be fine in a perfect world, but lots of potential phase conflicts crop up. For instance, if you put some sort of delay between the left-front and left-surround channels, you could end up with a huge, phasey sound when those channels combine into one. In such cases, a mix that sounds great in 5.1 surround can be unlistenable when downmixed to stereo.

If you think your carefully crafted 5.1 mix will never be heard in stereo, think again. Just as we need to check stereo mixes for mono compatibility, we also need to check 5.1 mixes for stereo compatibility. At the very least, a stereo version of your songs might be needed for radio play, and Lady Luck will probably choose the downmix of the one song that sounds as if it were mixed in your washing machine. Some very high-end surround mixes done by the most famous engineers on the planet sound horrible when auditioned in stereo. Guess what? Those engineers didn't understand the effects of downmixing.

What can you do about it? At the very least, you need a way to monitor the downmixing-cancellation effect. Many monitor controllers, such as the Multi-Max, have a button that engages Downmix mode. Then you can easily hear phase problems.

Furthermore, always do a separate stereo mix of any 5.1-surround mix. This is particularly important for the new DVD-Audio (DVD-A) format, which has enough data-storage space to include both a 5.1-surround and stereo version of your mixes. That way, when consumers select stereo mode on their receiver or processor, they hear

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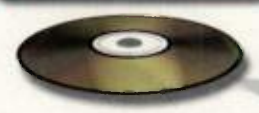

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your stereo PCM version rather than a downmixed version of the 5.1 mix.

The proper place for this true-stereo version is on the same 8-track master tape that holds the 5.1 mix, on tracks 7 and 8 (see the table "Surround Track Assignments"). Although you have to do the mix in two passes—one for surround and the other for stereo—you'll be way ahead of the game when someone requests a stereo version of the mix.

ENCODE THYSELF

The final part of making a 5.1 mix is the *encoding* process. **Fig. 5** shows where the encoder fits in the mixing chain in a hardware-based system.

Up until a few months ago, making a one-off DTS disc of a 5.1 mix required sending a six-track tape to DTS; the company would then encode the tracks and send you a DTS CD-ROM. Now you can get Minnetonka Audio's *SurCode DTS* ($499), a software DTS encoder for Windows that lets you encode your six discrete surround tracks as a DTS file with a WAV extension. This DTS file can then be burned onto a standard CD-R disc with any CD-burning application, such as Adaptec's *Easy CD Creator*. The file will play back through any home-theater system that includes a DTS decoder and a CD player with a digital output. (Don't try to play the CD from the analog outputs; all you'll hear is ugly noise.) Unfortunately, Minnetonka only develops software for Windows. I hope a Mac equivalent of *Surcode DTS* will be developed eventually.

Although a DTS disc looks like a Red Book CD-R (technically, it's an Orange Book disc), it probably won't play back in all DVD players. That's because the color of the dye and reflective layer in the CD-R medium itself might be incompatible with the wavelength of the laser in the DVD player's pickup. In addition, gold-colored CD-R media (such as the Kodak discs) seem to have a better chance of universal DVD playback than the dark-blue or green CD-R discs. Some of the newer Pioneer DVD players are advertised as having dual-laser pickups, and they seem to digest any color (and chemistry) CD-R I make. Interestingly, a CD-RW (rewritable) disc will play back on nearly every DVD player, even the old ones. So if you can't get a CD-R disc to play in your DVD player, burn a CD-RW disc and give it a spin. Of course, you can always play a DTS disc in any CD player with an S/PDIF output and a DTS-equipped receiver.

Those who want to use Dolby Digital encoding for surround mixes can burn a 44.1 kHz version of an AC-3 file and put it on a CD-R disc. You can get Dolby Digital encoders from Minnetonka Audio (*SurCode Dolby Digital*, $995) and Sonic Foundry (*SoftEncode 5.1 Channel*, $995). The encoding procedure is the same: load the six discrete audio tracks into the computer and toss them into the encoding software. Selecting the AC-3 WAV output creates an AC-3 file that's been padded out to fit in the exact same space as a stereo PCM file. Again, this file can be stored and burned on a CD-R or CD-RW for playback in many home-theater systems.

This method isn't recommended by Dolby Labs, and it's not 100 percent reliable, because some Dolby Digital decoders don't expect an AC-3 file to come from an S/PDIF bitstream with the Audio/Data flag bit set to Audio. Nevertheless, it works perfectly on many decoders and receivers. Until Minnetonka licensed the DTS encoder algorithms a few months back, this was the only way to hear a one-off version of your surround mixes on a home-theater system without dragging around a multitrack deck and mixer.

SURROUNDED BY INFO

There is much more to say about surround-mixing basics, but I've given you enough to get started. In future articles, I'll discuss subjects such as tracking for 5.1 mixdown and exotic surround microphones. But for now, it's time to start exploring the new world of multichannel music. It's a trip worth taking.

Mike Sokol *is a live-sound and recording engineer with some 30 years of experience on both sides of the console. He conducts free surround-mixing seminars at recording schools around the country; see www.modernrecording.com for tour dates.*

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The Real Thing

Thanks to the Internet, you can hire and work with real orchestras on a surprisingly modest budget.

By Gary Garritan

Many musicians dream of having their music performed by and recorded with a symphony orchestra. The vibrant cascading strings, majestic horns, and rising crescendos are enough to give any composer goose bumps. But these aural fantasies typically remain unfulfilled; unless, of course, you have a major record label behind you. Having a large orchestra at your disposal has always seemed beyond reach for most of us—until recently.

While working on my latest album, I needed a lush sound to complement the hundreds of layered harps in the recording. Hiring a symphony orchestra seemed like the perfect solution, so I contacted about a dozen orchestras in North America. As you might expect, I was hit with quite a sticker shock. The cost of hiring a first-rate orchestra for a day ranged from about $45,000 to more than $80,000. It was not exactly in my budget; in fact, it wasn't even close!

My next option for capturing a full symphonic sound was to purchase a sampler and an orchestral sampling library. Samplers from companies such as Akai, Kurzweil, and E-mu cost between $1,500 and $3,500. A software sampler such as *GigaSampler* with a separate dedicated computer system might also make a good choice.

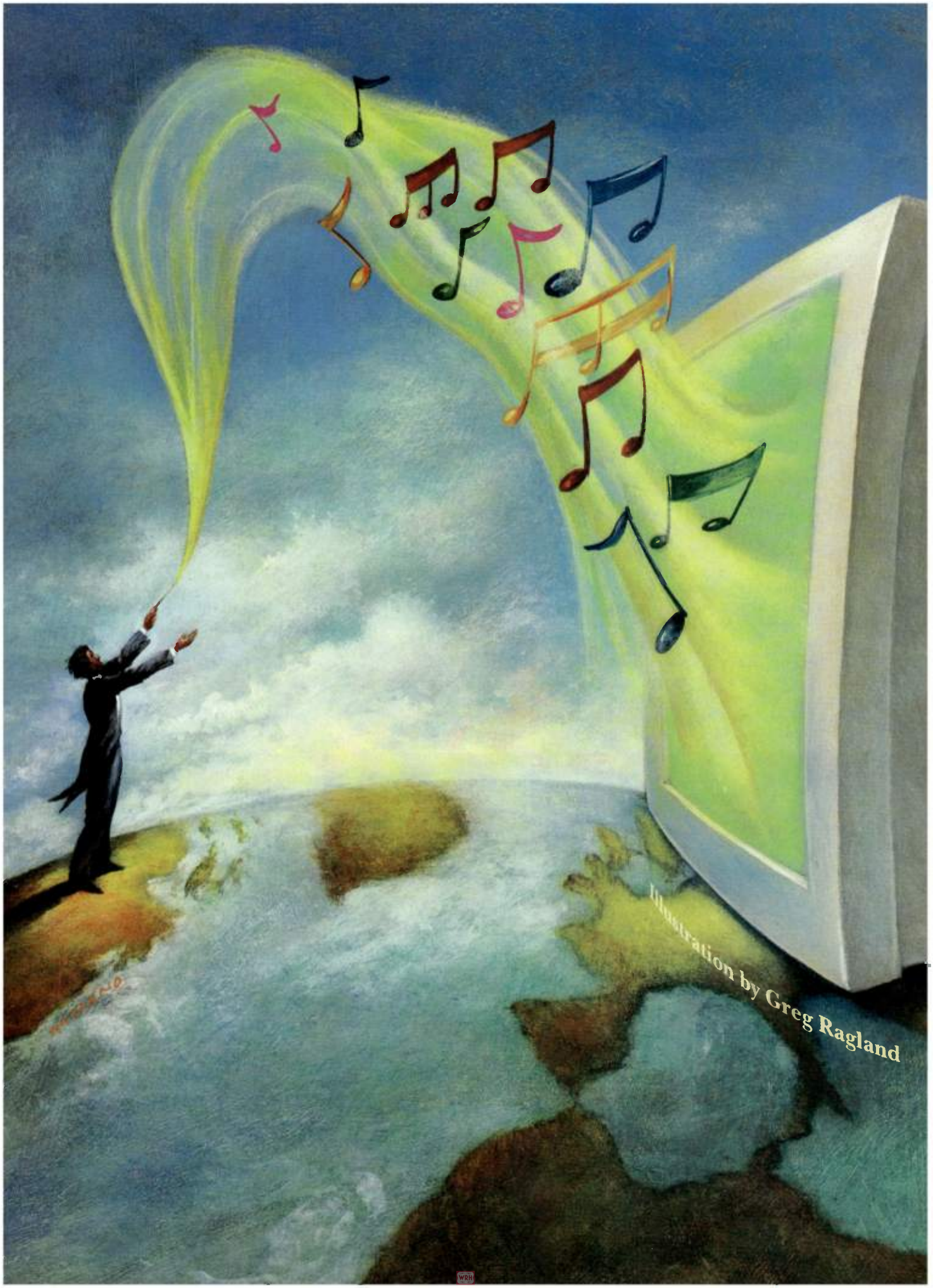
Illustration by Greg Ragland

There are currently several high-quality orchestral sample libraries to choose from, including the *Miroslav Vitous Orchestra* series, the *Siedlaczek Advanced Orchestra* set, the *Roland Orchestral Library*, and the Kirk Hunter *Virtuoso Series Strings*. These libraries typically excel at particular techniques (for example, legato playing) but may be weak at other techniques (such as fast bowing). After much investigation, I concluded that no single collection would be adequate. I needed a combination of sample libraries that would complement one another. For my project, the cost of the sampler and several libraries amounted to more than $6,000—much better than $60,000 for a real orchestra, but still a bit steep.

Furthermore, in spite of recent advances in sampler technology, sampled orchestral arrangements still don't match the grandeur of a real symphony. The powerful synergy of the orchestra members playing together simply can't be adequately captured through sampling. Uncomplicated legato passages may be acceptable for samplers; but when it comes to fast passages and certain types of articulations, sampling falls short. It seemed that none of my available options were satisfactory, and I was left on the horns of a dilemma until a chance encounter pointed me in a new direction.

While playing my harp at a Tulip Festival, I met a musician from Eastern Europe. She told me about how the orchestras of Central and Eastern Europe were becoming available to the West. After the collapse of the Iron Curtain, many symphony orchestras were no longer funded by their governments and were, therefore, offering their services on the open market. These orchestras needed hard currency and could be hired at a fraction of the cost of a comparable Western orchestra.

I embarked on an Internet search and soon discovered Symphonic Workshops (www.symphonicworkshops.com). Symphonic Workshops, the brainchild of Dr. Harry Hurwitz (see **Fig. 1**), began ten years ago as a conductor's workshop in Zlin, the Czech Republic. When Eastern Europe began opening up to the West, Hurwitz wanted to make some of Europe's finest orchestras available to conductors, so he developed an intensive conductor's training workshop. This workshop provided a rare opportunity for young conductors to gain experience with full-scale established orchestras while under the guidance of well-known professional conductors.

FIG. 1: Dr. Harry Hurwitz is the director and founder of Symphonic Workshops.

Apprentice conductors from more than 30 countries attended the workshop, which quickly grew from its origins as a conductor's boot camp into a variety of workshops throughout Eastern Europe. There are now opera workshops, a wind instrument workshop, piano workshops, a Strauss workshop, a repertoire workshop, symphonic workshops, and the Recording Fest. These workshops are held during the year in varying locations in the Czech Republic, Slovakia, Bulgaria, Romania, Poland, and the Ukraine.

I spoke with Hurwitz by phone at his office in Toronto about the feasibility of hiring an orchestra. My project consisted of ten songs totaling about 28 minutes of music with various degrees of orchestration. Hurwitz informed me that my recording project could be done at one of the biannual Recording Fests in the City of Olomouc, a small Moravian city in the Czech Republic, several hours north of Vienna.

The Recording Fest featured the Moravian Philharmonic Orchestra (40 to 75 players), and the recording could be done by an experienced, professional engineer in a concert hall with state-of-the-art recording equipment. This is going to cost a fortune, I thought. You can imagine my surprise when I received a quote of only $4,200 for a 50-piece orchestra!

For the Recording Fest, an orchestra is not hired on a per-player-per-hour

The Moravian Philharmonic Orchestra is a standard-size symphony orchestra with more than 70 players. The orchestra plays and records in its 600-seat hall and cooperates with Symphonic Workshops in the annual Recording Fest held in March and June.

For manufacturer information, please see Contact Sheet, p. 221.

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basis as in the United States. The charges are based on the finished recording: $3 per player per minute of final recording time. The above figure includes up to 18 hours of rehearsal for every hour of final recording. Bear in mind that not all of my songs required the orchestra throughout. Some songs needed only sections of orchestration. In other words, I could complete my project with almost 30 minutes of final music. The 50 players would cost less than $4,500, and that price might involve up to nine hours of rehearsal time leading up to the final recording. (In some cases, extra charges may be expected for engineering, hall rental, and the conductor.)

Packages are available that include all costs. For example, 40 minutes of final recording time for a 40-piece orchestra, including rehearsal time, hall rental, and engineer, would cost about $7,200. A 75-piece orchestra would cost $13,900. As mentioned earlier, my 28-minute project cost only $4,200. Prices for individual recordings often vary depending on the needs of the composer, but they are a small fraction of the costs for an orchestra in the United States.

The Liaison Link

It is very important to have an orchestral liaison who can represent you before the orchestra. He or she knows the laws of the country where the orchestra is located, knows the inner workings of the orchestra, and knows

INTERVIEW WITH A CONDUCTOR

Petr Pololanik belongs to a new generation of Czech conductors. After studying the violin and conducting at the Brno Conservatoire, he received a Master of Arts degree at the Janacek Academy of Music and Performing Arts in Brno, the Czech Republic. Pololanik's conducting career had an early start: when he was 18 he attracted considerable interest by conducting the Moravian Chamber Orchestra at the prestigious R. Wagner Festival in Germany. In 1992 he founded the chamber Capellen Orchestra, and in 1994 he became president of the Foundation for the Development of Musical Culture. That same year he was given a grant for further artistic studies at the University of North Texas. He has recorded CDs, composes and teaches music, and has worked in radio and television.

GG: It was a real pleasure for me to work with you. What was your experience like?

PP: I very much enjoyed working on your project. What was unique was how the Internet played such a strategic role. The Internet allowed us to discuss many technical and musical issues anytime we needed to.

GG: Does the Internet and technology help when working with people overseas?

PP: The Internet and new technologies are the best things that have developed in communication. Both help to pull down barriers, not only in distance, but also in the time it takes to spread knowledge and know-how. There are many positive things. The only shortcoming that I see is that a large gap is arising between people who are technologically enabled and those who are not.

GG: What has been your experience working with other Americans?

PP: I have conducted and have done recordings for Americans, British, Japanese, Austrians, and others. Generally, Americans are very easy to talk with; they make me feel comfortable and communicating with them is very friendly. Their music reflects this. American music is very free and "bohemian." Perhaps, Eastern Europeans would benefit by getting into some of the newer and freer music. Eastern European music is more traditional and structured, and Americans may benefit from this approach as well. However, it has always been enjoyable and mutually beneficial working with American composers and arrangers.

GG: How does hiring a conductor solve some of the problems relating to language and culture?

PP: In the Czech orchestras, you can find many players, and quite a few also know English. However, the main problems, I think, lie in the different mentality, different cultural traditions, and still different lifestyles. Therefore, a Czech conductor is the best solution if one wants to get the most from the orchestra and thereby save time and money.

GG: Are the musical traditions much different in Eastern Europe?

PP: No, I don't think so. The musical tradition of America springs from the European cultural traditions. So, there aren't any real fundamental differences.

GG: How many orchestras have you conducted?

PP: To date, I have worked with more than 20 orchestras around the world.

GG: What types of music have you worked with?

Petr Pololanik appears as guest conductor with many leading orchestras in the Czech Republic and abroad.

PP: Each conductor should be able to conduct Beethoven, Copland, Andrew Lloyd Webber, or any other style of music. In my case, I come from a family of musicians and composers who have written more then 300 works for theater, film, and television. But I think there is a more important matter: each conductor must have a taste for the subtle differences in the spirit of different types of music. In my conductor's role, I have worked with all the major musical styles: pop, folk, jazz, classical, contemporary, etc., and I always try to follow the spirit of the music.

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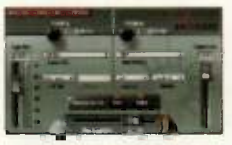

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H. D. Wells, *H. G. Wells's little-known older brother, shared his more famous siblings's visionary acumen but, due largely to his futile desire to be a rock star fully 50 years before the arrival of rock, lived most of his life in obscurity, playing in a succession of Gilbert & Sullivan cover bands in pubs in and around Bromley.***

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***OK, we made all that up. Think you can do better? Then send your own* H. D. Wells Biography *(in 100 words or less) to biography@antarestech.com (or to the address below) by October 1, 2000. Using arcane criteria known only to us, we'll pick our favorite and send the author a free Antares plug-in of his or her choice. Really.*

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how to negotiate with the parties involved. Stories abound about the hazards of negotiating within former communist countries. Prices rapidly fluctuate on a whim and new charges magically appear. You may also have to contend with the infamous black market, language and customs barriers, and a frustrating, Byzantine bureaucracy.

A liaison can also help with booking hotels, finding flights, arranging local transportation, exchanging currency, and hiring translators. It would be unwise for anyone to attempt to hire an orchestra in these countries without a liaison. Hurwitz and the staff at Symphonic Workshops have had ten years of experience dealing with the orchestras in these former communist countries, and they know how to make a project happen. Functioning like a personal diplomat, the orchestral liaison makes it possible to concentrate on the music and avoid an array of difficulties.

No Sputnik Gear

One of my concerns was the quality of the equipment that would be used to record the orchestra. You can have the greatest orchestra in the world, but how good will it sound if it's recorded with the wrong gear? I thought about the sound quality of old Russian cartoons and envisioned equipment that looked like it came from the Sputnik era.

The Brno Philharmonic is a well-known orchestra based in the Czech Republic city of Zlin. Some of the best players in the country belong to the orchestra.

When Hurwitz mailed a CD of the Moravian Philharmonic Orchestra, I was very impressed with the richness of the sound. I was also pleasantly surprised when I received a fax of the equipment list: a Neve 8128 or AMEK Angela mixing console; Schoeps, AKG, and UREI mics; Neumann tube mics; Dolby SR; Lexicon, Eventide, TL Audio, and Drawmer outboard units; Apogee converters; Pro Tools; and much more. If they didn't have the gear I wanted, they could probably find it. Some musicians have even chosen to bring some of their own equipment. Hurwitz provided references from several satisfied customers. I signed a simple contract with Symphonic Workshops and the adventure began.

Now that I had found an orchestra with the proper recording equipment, much work needed to be done; the Recording Fest was two months away. Due to my touring schedule, I could not travel to the Czech Republic, and there was still a lot of scoring work ahead. I contacted my lifelong friend Frank Spitznagel, an arranger in New York City, to help with the arranging and scoring. We soon realized we needed more help.

The individual scores require proper notation, which includes articulations, bowings, expression marks, and dynamics. I relayed my concerns to Hurwitz and he suggested that I hire an English-speaking Czech conductor to help with the scoring and arranging and also to represent me before the orchestra.

Czech Mate

Unless you've had experience conducting an orchestra (especially a foreign orchestra), it's best to pass the baton and hire a professional conductor. Choosing a conductor is like choosing a producer or a mixing engineer; the conductor must be in tune with your vision and understand what you are trying to get across to your audience. There were several conductors to choose from, but after corresponding by e-mail, I knew that Petr Pololanik was the man for the job (see the sidebar "Interview with a Conductor"). Pololanik has a strong command of the orchestra, knows the English language fairly well, and is technologically savvy. Besides being a prodigious young conductor, he is proficient with MIDI sequencing and digital audio.

I decided to exchange musical ideas and prepare the scores via the Internet. After all, Spitznagel and I had been doing that for years between New York and the Pacific Northwest island where I live. Pololanik and I both had PCs with Windows 95, a Creative Labs Sound Blaster AWE32 sound card, Coda *Finale* notation software, and Adobe *Acrobat Reader*. Pololanik uses

FIG. 2: This screenshot from Cakewalk's *Pro Audio* shows the different instrumental parts in track view and in standard notation.

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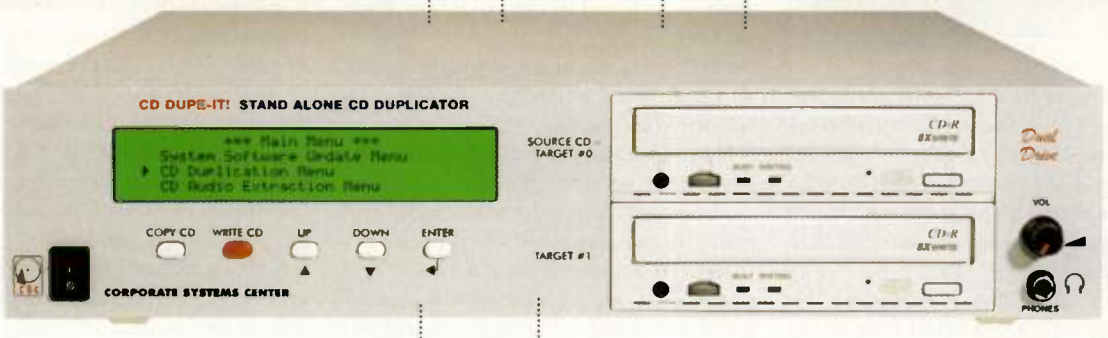

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Dual Drive Model is pictured above.

Voyetra Turtle Beach *Digital Orchestrator Pro* for his sequencing software; I use Cakewalk's *Pro Audio* for my sequencing. I also have a SEK'D audio card, *Samplitude 2496* software, and a Digidesign Pro Tools system on a Mac for final editing.

To keep our file sizes down, we decided to use General MIDI files to exchange musical ideas across the Internet. General MIDI files are small, and since we both had the same sound card, we knew that each of us would hear the same instrument sounds. What's more, our sequencers both had staff views so we could see the revisions in notation form while we listened to the MIDI files. Each part (violas, cellos, oboes, flutes, and so forth) was laid out on separate tracks, and each track was assigned a different channel (see **Fig. 2**).

My project had diverse instrumentation. Some songs required only a string orchestra to add a lush background; other songs required a full orchestra with strings, horns, woodwinds, and percussion. Certain parts of songs required individual solo parts; others called for ensembles.

Pro Audio allowed me to view the individual parts or the entire conductor's score in one view. MIDI timing resolution was not an issue, since we were using MIDI for scoring and not for recording a performance. MIDI made it easier to make modifications and communicate our ideas in an efficient manner. When the arrangements were agreed upon and it was time to score the project, the files were imported into *Finale* and sent in *Finale* format. That way, the individual parts could be printed in the Czech Republic without having to ship a mountain of scores for the 50 individual players. When special markings were needed, we printed out part of the score and hand notated it. Then we scanned the music and sent it as a JPEG file via e-mail, or we simply

IS THIS FOR REAL?

As he edited Gary Garritan's article, associate editor David Rubin became intrigued by the prospect of hiring a large orchestra on the other side of the world. Was it really feasible? Here's what he discovered:

As many composers have learned, it's difficult to find a full-sized orchestra to rehearse and record unpublished music. Local orchestras are busy playing Brahms and Beethoven and are understandably wary of unproven contemporary works. As Garritan points out, union orchestras in this country cost far more than most of us can afford, especially for a noncommercial venture such as a demo recording. Symphonic Workshops seemed to offer a viable alternative; I decided on the spur of the moment to give it a try.

The piece I had in mind was a 10-minute concert work that requires a 65-piece orchestra with a large percussion section (including timpani, xylophone, vibes, and piano). It's written in a freely chromatic idiom that employs mixed and changing meters as well as widely varying tempos.

I contacted Harry Hurwitz and as luck would have it, the next Recording Fest was scheduled to begin in just a couple of weeks. The timing was tight, but Hurwitz felt that it could be done if I sent the score and parts right away by FedEx to Olomouc, home of the Moravian Philharmonic Orchestra. Fortunately, the score and parts were already printed, bound, carefully notated, and ready to go, so I packed them up and trotted down to the local FedEx office. I was told the package would arrive in about three business days barring delays at customs; the shipping cost was just over $150.

Meanwhile, Hurwitz had made arrangements with Martin Braun, a skillful young Austrian conductor who was travelling north from Vienna to participate in the Recording Fest. He agreed to conduct the piece if the score arrived in time to prepare it. As it turned out, FedEx's estimate was a bit off; the score arrived nearly a week later. (The package traveled by air from Los Angeles to Paris and then on to Prague. It then went by truck to Olomouc. Each step introduced the possibility of a delay.) By the time the score arrived, Braun had only a few days to prepare, but he nevertheless agreed to tackle the job. In fact, he surprised me one afternoon with a phone call, so I could clarify a few points about the score.

By this time, Hurwitz and I had settled the details concerning the recording itself. I was offered the option of a high-quality multitrack (ADAT) master recorded with multiple mics, a mixing board, and a recording engineer, but that would add several hundred dollars to the project. To keep my experiment as inexpensive as possible, I chose the bare-bones route: a pair of microphones straight into a stereo DAT recorder—no mixing board; no extra personnel. We also agreed to limit the total session to about two hours (with a little latitude) to further curb the costs. The recording might not be of commercial quality, but it would provide a good sense of how the piece and the orchestra sounded.

The recording session went off without any apparent mishaps. Braun, who had clearly done his homework, rehearsed the orchestra for about an hour, focusing on problems such as articulations and rhythmic accuracy. (I know this because the DAT recorder was running throughout the session.) During the second hour, the orchestra played through the piece a total of three times; the final take was the best. It was certainly not a perfect performance, but it was remarkably good considering how little time had been spent on it. In spite of some tricky sections, Braun and the orchestra pulled the piece together and succeeded in properly capturing the spirit of the music.

The master tape arrived about two weeks after the recording session, and I quickly transferred the music onto my hard drive. After some editing and processing, I ought to have a pretty good demo of a 65-piece orchestra playing one of my compositions; it's hard to beat the real thing!

—David Rubin

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faxed it. For two months, Pololanik, Spitznagel, and I exchanged e-mails, MIDI files, *Finale* files, JPEG files, faxes, and audio files (on rare occasions) over the Internet.

Pololanik carefully scrutinized every note in the arrangements and made appropriate corrections as needed. In addition, he offered suggestions to make the scores more musical, offered better harmony changes, pointed out problems, and in some cases provided alternative arrangements. He listened to our arrangements and sent back revised MIDI files. We listened, offered comments, sent back the files, received revisions, and so on, until we "settled the score." Pololanik had a good feel for my music and understood the potential of an orchestra. Like the conductor's baton, I was in good hands.

As I mentioned earlier, for some of the songs the orchestra mainly provided background music for my harp leads. With other songs, however, the orchestra assumed more of a lead role. One of the issues that we discussed was whether to use a MIDI click track to sync to music that was already recorded. In some songs this was appropriate, but in other songs, Pololanik preferred to conduct in a free-form fashion rather than to the more restrictive MIDI beat. In the final mix, I could make any tempo adjustments in Pro Tools. Another concern was the pitch that the orchestra tuned to. Some orchestras don't use A=440, but Pololanik assured me that the orchestra would have the proper tuning.

We narrowly made the deadline in time for the recording dates. The two-month exchange with the conductor allowed us to refine the scores and to solve potential problems beforehand. More importantly, however, the conductor was intimately familiar with my music by the time the orchestra was ready to record.

The Grand Finale

I wish I could have been there. Nevertheless, the recording went without a hitch. All of the parts were properly notated, Pololanik knew the music like the back of his hand, the 50-piece orchestra had no problems, and the recording engineer was well prepared. Pololanik's father (Zdenek Pololanik, the well-known Czech composer) acted as musical director and sat in the recording room to monitor the musicality of the project. I spoke with the younger Pololanik by phone after both days of the recording session, and he assured me that all went well and that the final DAT tapes were on their way via FedEx. Backup DATs were made in case there were any problems with shipping.

I was astounded when I heard the tapes two days later. I had several takes to choose from, with different variations if the conductor was uncertain. Each take included detailed notes. The final product was amazing. I was particularly impressed with the richness of the string and horn sections, and the soloist parts—especially oboe and cello—were simply extraordinary. Placing the orchestral audio files in Pro Tools was easy, although some of the files required editing to match the timing. The end result was well worth it. You can hear samples of the orchestra at my Web site: www.harps.com.

Final Measure

If you ever wished you could have your music recorded by a real symphony orchestra, your dream can come true. You can hire an orchestra through the Internet at about the same cost as a sampling setup. Based on my own experience, however, I strongly recommend that you hire a liaison and get a good English-speaking conductor to represent you.

Hiring an orchestra in the Czech Republic or other Eastern European countries is not as hard as you might think, thanks to the Internet. In the final analysis, nothing sounds quite like a real orchestra, and as I discovered, you can truly warm up your music with orchestras from former Cold War countries.

Gary Garritan *is a pioneer in bringing the ancient into the modern world. He invented the MIDI harp (see "Pro/File: Harping on MIDI" in the July 1995 issue of* **EM**) *and developed GigaHarp. He is currently working on an orchestral sampling library in the GigaSampler format.*

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Sammy Peralta loves music. That pure and simple fact comes through strikingly clear as he sits at his keyboard tinkering with half-written tunes. Sammy's background includes work with talents including Tito Puente and Willie Bermudez. "I have to be careful because I can get so lost in the music, I sometimes forget I have a family that would like a little of my attention too".

One last point: Sammy Peralta's new CD ***On the One*** featuring Lenny White was mixed entirely with LSR monitors.

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The Csound programming language is virtually synonymous with "computer music." Musicians all over the world have used it for more than 20 years, and it has earned the reputation of being both incredibly powerful and incredibly difficult to learn and use. In recent years, an international group of dedicated hackers, composers, sound designers, and educators has brought new life to the language, transforming it into the most full-featured and high-powered synthesis and signal-processing software available. • Learning Csound has never been easier. The language is fully documented in a 500-page reference manual and supported by an active mailing list, dozens of online tutorials, a quarterly e-zine, several textbooks, and a growing number of graphical front ends and utilities. On top of that, Csound remains absolutely free of charge (see the sidebar "Online Resources"). All the innovation and online support are attracting a new generation of innovative desktop-based electronic musicians. • I'll review some of the basics of the Csound language, then move on to more-advanced elements. Granular synthesis, one of Csound's most robust synthesis methods, will get a special focus. I'll also look at scanned synthesis, one of the newest sound-creation techniques. When all is said and done, you'll have many new uses for this powerful language and a whole new outlook on this exciting form of music making.

BY RICHARD BOULANGER

WHAT EXACTLY IS CSOUND?

Csound is a programming language designed and optimized for sound rendering and signal processing. Written in the late '70s by Barry Vercoe of the MIT Media Lab, Csound is a direct descendant of the classic Music V language written in the late '60s by Max Mathews of Bell Labs. Today, it's available for nearly every computing platform. The language consists of more than 450 *opcodes* (operational codes), which are the software routines, or modules, used to build patches. In Csound, the patches are called *instruments*. The opcodes range in power and complexity from basic elements such as a table-lookup oscillator, linear envelope generator, and bandpass filter to the full-blown waveguide physical-modeling family. Opcodes exist for analog modeling, reading and processing samples, phase-vocoder resynthesis, FFT-based crossings, and more.

Typically, you design and play Csound instruments by creating two text files in a word processor or text editor: the *orchestra* file contains a description of one or more instruments, and the *score* file contains the notes, wavetables, samples, and any other information the instruments need. You tell Csound which score and orchestra files to use, and it *renders* (compiles) the resulting sound file to your hard drive. The instruments' complexity is limited only by your knowledge, interest, and need—never by Csound itself.

READY FOR LAUNCH

In its early years, Csound had a command-line interface, in which you typed the required parameters on a Unix console or personal-computer keyboard. Now, many *launcher* utilities produce or process a sound file and play a Csound instrument in real time. The launchers let you select the orchestra and score files using a simple graphic interface, and they generally provide menus, checkboxes, and text fields for setting and storing all the command-line options. Clicking the Render button starts the compiling process (see Fig. 1).

With some launchers, you can specify the output file's name, location, and type (AIFF or WAV at 16-, 24-, or 32-bit resolution). Many of the programs also display a graph of your waveforms and enable MIDI control or real-time audio input. You can hear the audio Csound creates directly through the sound card or create an audio file on disk.

For example, Matt Ingalls's *PPCsound* launcher (see Fig. 1) offers all these options and a console window for playing, pausing, or terminating the rendering. The termination function lets you evaluate your music before it has finished rendering and decide whether you want to continue or stop the rendering and edit your files. A graphic window displays oscillator waveshapes and sampled waveforms, and a list window displays the rendering's progress. The list window indicates when new instruments are allocated by showing their start times. It also displays the score's current amplitude, reports out-of-range samples, and shows other score and orchestra syntax errors. (See the sidebar "Graphical Front Ends and Utilities" for additional Csound helper applications.)

A SIMPLE OSCILLATOR/ENVELOPE INSTRUMENT

A block diagram helps users visualize Csound patches by charting signal flow, parameter settings, and interconnections between opcodes. Like Csound itself, these block diagrams read from top to bottom. In Fig. 2, you see an oscillator (oscil) whose amplitude is controlled by a linear envelope generator (linen). The oscillator's frequency and wavetable are supplied in the score and are indicated in the diagram as p5 and p6 (more on this in a moment). Here's the Csound code that creates this instrument (anything after a semicolon and before a carriage return is a comment, which Csound ignores):

```
; Example Instrument Definition

sr      =       44100           ; sample rate
kr      =       4410            ; control rate
ksmps   =       10              ; sr/kr
nchnls  =       1               ; output channels

        instr 1                 ; define instrument 1
k1      linen   p4, p7, p3, p8  ; amplitude envelope
a1      oscil   k1, p5, p6      ; table-lookup oscillator
        out     a1              ; output
        endin                   ; end instrument 1
```

Notice that the code comes in two main sections. The top four lines designate various global settings such as the sample and control rates (sr and kr) and the number of

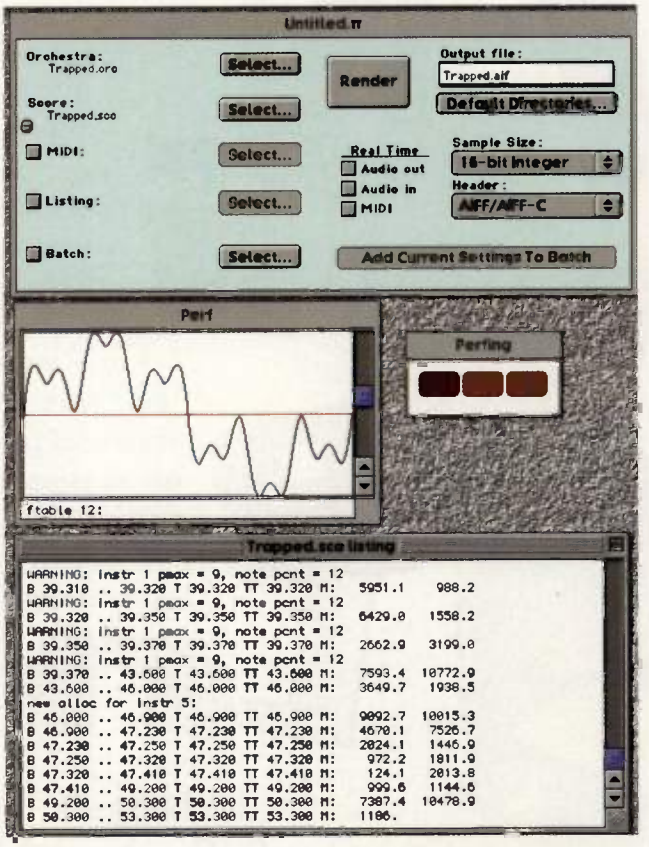

FIG. 1: Matt Ingalls's *PPCsound* is a graphical interface for specifying many Csound rendering settings. The utility also displays the rendering's progress and any waveforms used in the process.

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audio channels (nchnls). Generally slower than the sample rate, the control rate updates control parameters that don't need the high resolution of audio samples. The information in the top four lines is included only once in the orchestra file. The instrument itself is defined in the bottom section, starting with "instr 1." Because an orchestra can contain as many distinct instruments as you want, each must have its own number.

Each opcode has a unique set of *arguments,* parameters the opcode requires to perform its function. In general, the syntax of a statement defining an opcode is as follows:

```
output      opcode name          argument1, argument2, argument3 . . .
```

For example, Csound's linear ASR envelope generator, the linen opcode, has this syntax:

```
k/ar        linen        k/xamp, irise, idur, idec
```

K/ar indicates that you can have the opcode render its output at either the control (k) or audio (ar) rate. To the right of the opcode name are the four arguments that linen requires. The first, k/xamp, includes the k/x prefix, which tells you that the amplitude can be dynamically altered by another opcode at the control or audio rate or set to a constant value. The *x* represents any type of input: constant, control rate, or audio rate. Using a *k* as well as an *x* is redundant, but that's what the Csound reference manual does.

The *i* prefix on the next three arguments—rise time, duration, and decay time—means that they must be set to constant values in the instrument definition or from the score; you cannot alter them during the course of a note.

Now look at the arguments for Csound's table-lookup oscillator, the oscil opcode:

```
k/ar        oscil        k/xamp, k/xcps, ifn, [iphs]
```

Like linen, oscil can render its output at either the control or audio rate. Both the amplitude (k/xamp) and frequency (k/xcps) can be constant or modified by other opcodes at the control or audio rate. The wavetable that the oscillator reads (ifn) is set at initialization time and cannot change during the course of a note. The term *ifn* refers to the number of a function table, which Csound creates according to instructions in the score when it computes the sound. Arguments in brackets are optional. You can offset the oscillator's phase (iphs) if desired.

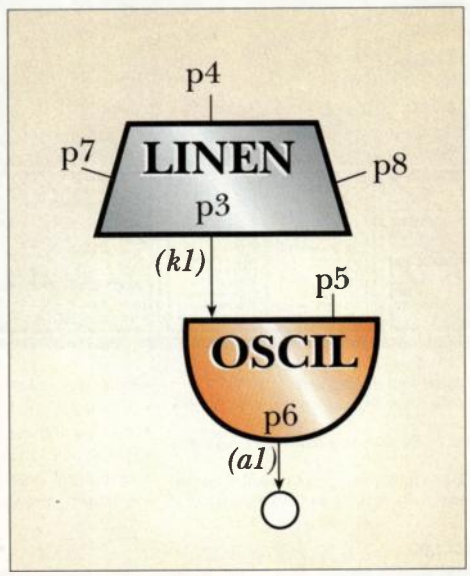

FIG. 2: Visualize your Csound instruments with block diagrams. This instrument uses a linear envelope to control an oscillator's amplitude. The oscillator's frequency and wavetable, represented by the p5 and p6 markings, are derived from the score.

An opcode's arguments, or input parameters, are to the right of the opcode's name, and the routine's output, or result, is on the left. This syntax makes it easy to "patch" the output of one opcode to the input of another. In addition, opcodes can render their outputs at either the lower-resolution control rate or the sampling (audio) rate.

Furthermore, you can set and reset an instrument's parameters directly in the score by using the *p* prefix in the instrument design (see the previous "Example Instrument Definition" code) to refer to a specific *p-field* in the score file. A p-field is a column in the score containing the different parameter values for each note in the music. For example, the first column (p1) is for the instrument number, p2 the note start time, p3 the duration, and so on. The first three p-fields are standardized, but the rest can be anything you want. Following is an example of a Csound score:

```
; Function 1 uses the GEN10 subroutine to compute a 4096-point sine wave
; Function 2 uses the GEN10 subroutine to compute first 16 partials of sawtooth
  f1    0 4096 10 1
  f2    0 4096 10 1 .5 .333 .25 .2 .166 .142 .125 .111 .1 .09 .083 .076 .071 .066 .062
;=============================================================
; p1  p2    p3   p4     p5     p6       p7      p8
; Ins Start Dur  Amp    Freq   F-table  Attack  Release
;=============================================================
  i1  0     2    20000  440    1        1       1
  i1  2.5   2    16000  220    2        0.1     1.99
  i1  5     4    11000  110    2        3.9     0.1
  i1  10    10   10000  138.6  2        9       1
  i1  10    10   9000   329.6  1        5       5
  i1  10    10   8000   440    1        1       9
```

In the oscil line of the "Example Instrument Definition" code, I assigned the oscillator's frequency argument to p5. This lets me set the instrument's frequency (in hertz) on a note-by-note basis in the score by simply changing the value in the fifth column of the score's note list. Similarly, because the ifn argument is assigned to p6, the sixth column in the note list controls the oscillator's waveshape for every note. Notice that columns p7 and p8 set the linen opcode's attack and release times, which can also be changed for every note.

In addition, one opcode's output can dynamically alter another opcode's input arguments during the course of a note, and this modulation can occur at the specified audio or control rate. The control rate is typically chosen because it is generally slower than the audio rate, but in some cases (such as with certain filtering or physical-modeling parameters), the audio rate is preferable. In the "Example Instrument Definition" code, the linen opcode's output (k1) is used as the oscil opcode's amplitude value.

MORPHING GRANULAR

Admittedly, applying an envelope to an oscillator is not that interesting, but using envelopes to dynamically alter the parameters of Csound's grain opcode during the course of a note can produce some unique textures. (For more on granular synthesis, see "Square One: A World in

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a Grain of Sound" in the November 1999 issue of **EM**.)

A granular-synthesis instrument is diagrammed in **Fig. 3**. Of course, one way to control the amplitude of the final output is to modulate the grain opcode's amplitude parameter (located at the far left of the grain block in **Fig. 3**) with an envelope generator. In **Fig. 3**, however, this parameter is set to a constant value as specified in the p4 column of the score, and the output of the grain opcode is sent through a linen opcode, where it is shaped by the attack and release settings in the score. (This block diagram doesn't show all of the opcodes' parameters, which is not uncommon. Have a reference manual on hand to review an opcode's arguments.) The linen opcode's audio-rate output is labeled "a2," and the grain opcode's audio output is labeled "a1."

In this granular patch, various grain-parameter values change as the note plays because they are under the control of the line and expon envelope opcodes (two types of envelope generators). Alternatively, you can assign p-fields in the score to various parameters; for example, by assigning two p-fields to the start and end grain-density arguments, the density could change from 12,000 grains per second to 4,000 grains per second during the first note. During the next note, density might increase from 5 to 2,000 grains per second. You can also modulate the grain duration by giving different start and end values to the grain-duration argument.

By overlapping these two notes in time, you can morph from one evolving grain "cloud" to another that is quite different in shape and composition. Even the basic waveform the grain opcode uses can change for each note—the possibilities are endless. (See the sidebar "Top Six Tips" for some general suggestions for composing with Csound.)

GRAPHICAL FRONT ENDS AND UTILITIES

For many musicians, harnessing the strength of Csound is difficult because of its cryptic and arcane terms and syntax. As a result, many enterprising Csound users have created graphical user interfaces (GUIs) that are more informative, intuitive, and quite frankly more appropriate than the plain-vanilla Csound commands. Here's a short overview of three such interfaces.

VISORC

By David Perry

http://cornelius.dhs.org/visorc/home.htm

This Windows GUI combines a Cycling '74*Max/MSP*-like graphical representation of each Csound opcode with a piano-roll score-file representation. It's easy to experiment with patches by simply "connecting" opcodes, and the powerful score editor has numerous features for rapidly generating extensive note lists. *VisOrc* lets you quickly toggle between text and icon views, so you can work with your files in the way most comfortable for you. It's a robust and powerful program for diving into Csound composing.

CECILIA

By Alexandré Burton and Jean Piché

www.musique.umontreal.ca/Org/CompoElectro/CEC

Cecilia is a complete Csound production environment for Linux and Power Macintosh. With the program's Grapher window, you can take full advantage of the synthesis and signal-processing power of Csound without even being aware of its existence (see **Fig. A**). In *Cecilia*, you'll find a large menu of modules, which are graphical interfaces to a huge library of incredible-sounding and extremely powerful Csound instruments.

VMCI

By Gabriel Maldonado

http://web.tiscalinet.it/G-Maldonado/vmciplus.html

This virtual MIDI-controller interface is oriented toward the real-time manipulation and performance of MIDI instruments. By moving onscreen sliders, you can adjust many Csound instrument parameters in real time, and it's easy to save parameter settings, which you can recall at any time. *VMCI* is best used as a front end to Maldonado's fast DirectX Csound port, *DirectCsound*. The combination makes a powerful performance duo.

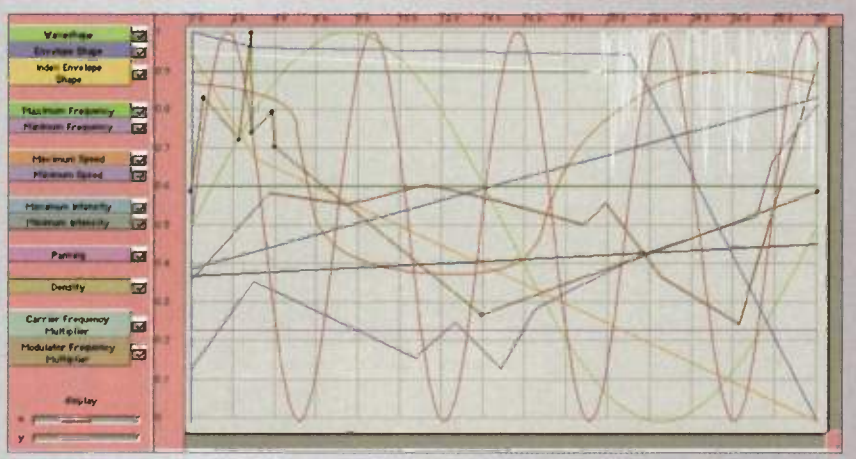

FIG. A: *Cecilia* is a graphical front end for Csound that makes the language itself invisible to the sound designer. The StochasticGrains module is one of the many available built-in tools.

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GRANULAR MIDI

The sonic landscape of granular synthesis is relatively new, so exploring it with a real-time Csound patch should be useful. To this end, I'll convert my granular synthesizer into a MIDI instrument and replace most of the envelope generators with scaled 7-bit controllers using the midic7 opcode. This opcode lets you map an arbitrary controller number to an arbitrary working range best suited for the instrument at hand. For example, consider the following line of code:

```
kfreq     midic7     60, 0, 12700
```

This code maps the value of Continuous Controller 60 to the variable called kfreq (audio frequency) and multiplies the CC value by 100. An incoming value of 0 would result in an output of 0, a value of 64 would result in an output of 6400, and a value of 127 would result in an output of 12700. (Don't insert a comma within any number you use as an argument.) The code maps the controller's value into a useful frequency range.

When you start playing the instrument, what will the initial values be? Using the ctrlinit opcode, you can set the starting values of all controllers on a certain MIDI channel. The following example sets the initial value of CC 60 to 64, CC 68 to 0, and CC 72 to 127, all on channel 1 (indicated by the 1 in the first argument). You decide what your instrument does with this CC data.

```
ctrlinit    1,        60, 64,        68, 0,     72, 127
```

Once you start playing the instrument, you can give these controllers any other value, but the instrument always starts from a preset, repeatable, and predictable value.

Now it's time to construct the real-time granular synthesizer. Here's the code:

```
ctrlinit    1,        80,0,        16,10,     17,64,     18,40,     19,40

            instr 1
icps        cpsmidi
iamp        ampmidi   10000
kcps        midic7    80, 0, 10000
kdens       midic7    16, .01, 1000
kampoff     midic7    17, .01, 1000
kptchoff    midic7    18, .01, 10000
kgdur       midic7    19, .0001, .1
a1          grain     iamp, icps+kcps, kdens, kampoff, kptchoff, kgdur, 1, 3, .1
a2          linenr    a1, .01, .4, .5
            out       a2
            endin
```

This orchestra file has two MIDI *value converters:* cpsmidi (labeled "icps") converts MIDI note numbers to their equivalent in hertz, and ampmidi (iamp) maps MIDI Velocity (0–127) to linear amplitude (0 to 10000), as the grain opcode requires. CC 80 (kcps) is mapped to a range of 0 to 10000 and added to the base frequency derived from the note played on the MIDI keyboard, which gives CC 80 a large pitch-bend range. This function shows up as "icps+kcps" in the grain opcode's arguments (in the line starting with "a1"). The other four controllers in this example (CC 16, 17, 18, and 19) are also mapped to specific ranges and assigned to grain density, grain amplitude, grain pitch offset, and grain duration. They appear in the lines labeled "kdens," "kampoff," "kptchoff," and "kgdur," respectively.

One of the important considerations in this and every MIDI instrument is the linenr opcode (a2). In the past, Csound instruments always "knew" the note's duration when the instrument was initialized because every note statement included a start time (p2) and duration (p3). All other parameters in the score were up to the sound designer. But in a MIDI instrument, the duration depends on how long the key is depressed. To address this issue, Csound creators added a new class of MIDI-based opcodes that go into a release phase when the program senses a Note Off. These opcodes end with the letter *r* (for example, linenr, linsegr, and exponr).

This MIDI instrument's score file is also unique in that it contains no notes. Instead, a "dummy" function table (f0) causes the instrument to actively sense real-time MIDI data for a user-specified amount of time; here, I've specified 30 seconds. The first two lines in the following score are for creating the grain's waveform and amplitude envelope, respectively.

```
f1        0         4096       10         1
f2        0         4097       20         2          1
f0        30
```

GRANULAR PROCESSING

Instead of granular synthesis, you can use the sndwarp opcode to independently time-scale and pitch-shift a sample

TOP SIX TIPS

As you have probably surmised by now, you can use the Csound language in many different ways to make music. Remember this short list of ideas for various applications.

1. Synthesize textures, pads, drones, and loops and use them as audio tracks in your digital audio sequencing or mixing software.

2. Create samples, then download them to your favorite MIDI sampler. You'll amass a huge store of new source material in no time.

3. Process existing samples or audio tracks at virtually every stage of the pre- and post-production process. Use Csound on your final mix for equalization, compression, reverberation, acoustic modeling, or auditory localization.

4. Design a family of efficient MIDI instruments in Csound and play them in real time in your sessions or concerts.

5. Design MIDI instruments whose parameters are assigned to MIDI controllers and render the instruments using a Standard MIDI File containing both notes and controller data.

6. Use a Csound score file to compose an entire piece using the traditional text-based note list.

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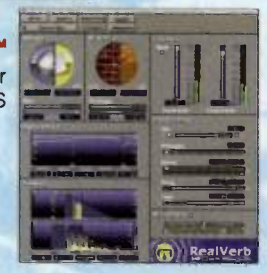

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in real time. A granular-processing instrument of this type uses the GEN01 function (Csound's table-generating functions are called GENs) to store the samples from a preexisting sound file indicated in the score. In the following instrument, CC 16 (ktwarp) time-scales the sound from 0.25 to 25 times its original length. CC 17 (kresamp) resamples the file and transposes it from 0.25 to 4 times the original pitch.

ONLINE RESOURCES

Here's a rundown of some sites that offer support for Csound activities. This list will continue to grow, so check back at the two first sites.

cSounds.com and Csound.org. The two main online Csound hubs are the commercial www.cSounds.com and the academic www.Csound.org. At cSounds.com, you'll find *The Csound eZine, The Csound Online Reference Manual, The Csound Catalog with Audio, The Csound Masters* (a series of Csound-based music CDs), *The Elements* (a series of Csound-based sampling CDs), and The Csound Sound Design & Composition contests.

At Csound.org, you'll be forwarded to The Csound FrontPage @ MIT Press, which has links to all Csound applications, sources, utilities, instruments, compositions, samples, tutorials, and people.

The mailing list and FAQ. One of the main reasons that Csound's popularity continues to grow is the wealth of information, suggestions, and solutions shared online via the Csound mailing list. I encourage you to join, introduce yourself, ask questions, make suggestions, and share your ideas. To subscribe, send an e-mail to csound-subscribe@lists.bath.ac.uk. However, before posting your first questions, check out The Csound FAQ at www.csounds.com/faq, where you might just find the answer you're looking for.

The reference manual. Csound is the ultimate educational environment for the computer musician. In the past, however, getting lost in Csound was usually par for the course. Today's *Public Csound Reference Manual* (www.lakewoodsound.com/csound) is a wonderfully readable and useful document, thanks to years of dedicated work by an international team led by David Boothe.

The magazine. Offering something for users at every level, the *Csound Magazine* (www.csounds.com/ezine) is by far the best online Csound material. It is edited and written primarily by Hans Mikelson, one of Csound's most prolific and brilliant sound designers and teachers. During the past few years, Mikelson published five issues, each containing the following tutorial columns: "Beginners," "Synthesis," "Processing," "Real-time," and "Internals." Each issue includes a feature; topics have ranged from score generation, composition, and psychoacoustics to cellular automata and ecological modeling.

Tutorials. Educators and sound designers have posted excellent tutorials online. A good place to start is my original 1989 TOOTorials (www.csounds.com/tutorials). Also, MIT Press has uploaded the first chapter of *The Csound Book,* (http://mitpress.mit.edu/e-books/csound/fpage/pub/csbook/Boulanger1/Boulang.html). The tutorial takes you from the basics to advanced concepts and designs. It includes more than 40 instruments and fully integrates and links with the HTML version of *The Public Csound Reference Manual.*

The book. MIT Press recently published *The Csound Book* (1996), the definitive Csound textbook, and posted several of the chapters online (www.csounds.com/book). The book is devoted exclusively to the study of computer music through Csound. More than 20 years in the making, this exhaustive guide contains 32 tutorial chapters in the printed text and 45 additional HTML chapters on the first of two CD-ROMs. All the chapters were written by the world's leading sound designers and teachers, and all contain complete working Csound instruments to illustrate the topics covered. Virtually everything is covered here: programming-language and software-synthesis basics, classic synthesis techniques, waveguides, chaos, studio effects, analog modeling, acoustical modeling, signal processing, reverberation, 3-D audio, C programming, and much more.

Instruments. Many instrument collections are posted for you to explore and modify. Among the finest online Csound libraries is a commercial set by Steven Cook (www.babcom.u-net.com/csound.html), a modeling set by Josep Comajuncosas (http://members.tripod.com/csound), and an incredible analog-emulation and studio-effects set by Hans Mikelson (www.werewolf.net/~hljmm/csound).

The Catalog with Audio. All the online collections, and hundreds more, are in *The Csound Instrument Catalog with Audio* (www.csounds.com/catalog). The advantage of this comprehensive collection is that all of the 2,000-plus instruments have been tested, edited, aligned, and prerendered. MP3 files of every orchestra are located alongside the text versions in this organized database. You can point and click to instantly hear a musical demo of any instrument. If something turns you on, simply click on the corresponding orchestra to launch Csound right from your browser and render an uncompressed version. *The Csound Catalog with Audio* is the best way to hear virtually everything that Csound is capable of doing. It also contains the most current version of the language, the first chapter of *The Csound Book, The Public Csound Reference Manual,* and step-by-step instructions on how to set up your computer and browser to run the program.

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```
ctrlinit    1, 16,40,17,10

            instr 1
icps        cpsmidi
iamp        ampmidi     10000
ktwarp      midic7      16, .25, 25
kresamp     midic7      17, .25, 4
a1          sndwarp     iamp, ktwarp, kresamp, 2, 0, 2000, 1000, 10, 3, 0
a2          linenr      a1, .01, .4, .5
            out         a2
            endin
```

The sndwarp opcode's time-based algorithm opens a moving window into your sample file and changes the sample rate of the contents according to the ktwarp and kresamp settings. This code transforms the source in a wide range of clean, grungy, and magical ways.

In the next score, I've placed a sample's file name in quotes ("hello.aif") to read it into the GEN01 routine. As before, I've also used the dummy function table and amplitude-envelope function.

```
f1      0       65536   1       "hello.aif" 0 4 0
f2      0       4097    20      2               1
f0      30
```

ACOUSTIC MODELING

Csound includes a large family of analysis/resynthesis opcodes and utilities. The convolve opcode provides a world of filtering, reverberation, room-simulation, and sound-morphing effects. Convolution is a generalized filtering algorithm that performs a *dynamic spectral intersection* between two sound files: a source file (which contains the material you wish to alter) and a filter file (which contains the impulse response of the filter).

Convolution is unique because it doesn't restrict how you define the filter's impulse response. Thus, the frequency characteristics of any sound file can filter any other sound file. You accomplish this by converting both files into the frequency domain and multiplying them together. Any frequency components they have in common will be emphasized, and any that are different will be attenuated or zeroed. If the frequency characteristics of either file change over time, as most do, the result will be a time-varying filter, or a dynamic spectral intersection between the two files. Csound's cvanal utility turns any sound file into a filter.

As an example, I'll convolve a voice with the cvanal-converted recording of a door slamming in a room. The result sounds like the voice speaking in that room. The following code shows a generic convolution orchestra with a variety of audio sources and filters, including the door slam ("door1.con") and my voice ("hellorcb.aif"):

```
            strset      10, "hellorcb.aif"
            strset      11, "allofme.aif"
            strset      12, "door1.con"
            strset      13, "Cymbal.con"
            strset      14, "WhiteNoise.con"
            strset      15, "drumloop.con"

            instr 1
iscale      =           p4
itrans      =           p5
isource     =           p6
ifilter     =           p7
a1          diskin      isource, itrans
a2          convolve    a1, ifilter
            out         a2*iscale
            endin
```

In this convolve instrument, the diskin opcode (on the line marked "a1") reads a sound file into the orchestra from disk. The diskin opcode is disk based rather than RAM based, and it lets you transpose and reverse the direction of the file or use another Csound instrument to process the file in various ways. The strset opcode at the top of the orchestra file contains the names of the source and filter files and their associated function-table numbers. Once a file has an associated number, you can refer to the file in p-fields in the score and change it on a note-by-note basis.

In the next example score, each of the first four notes uses a different filter file (indicated in p7); among the files is the door slam (12). The source file (p6) remains the same for the first four notes; the next four notes have a second source file. You can see in the convolve instrument that p6 is assigned to the isource variable.

```
; ================================================
; p1    p2    p3    p4    p5    p6    p7
; Ins   Strt  Dur   Amp   Trans Source Filter
; ================================================
  i1    0     6     .01   1     10    12
s
  i1    0     7     .01   1.1   10    13
s
  i1    0     12    .01   0.6   10    14
s
  i1    0     5     .008  1.5   10    15
s
  i1    0     6     .01   1     11    12
s
  i1    0     7     .01   1.1   11    13
s
  i1    0     12    .01   0.6   11    14
s
  i1    0     5     .008  1.5   11    15
```

The convolve and cvanal opcodes can create exotic reverbs, complex resonators, and some incredible new effects. Try them with your favorite audio material.

SCANNED SYNTHESIS

Csound brings together the old and the new. Not only has it preserved a historical approach to computer music and retained a universal library of instruments, but it also encourages exploration of new areas. One of these new directions is *scanned synthesis*.

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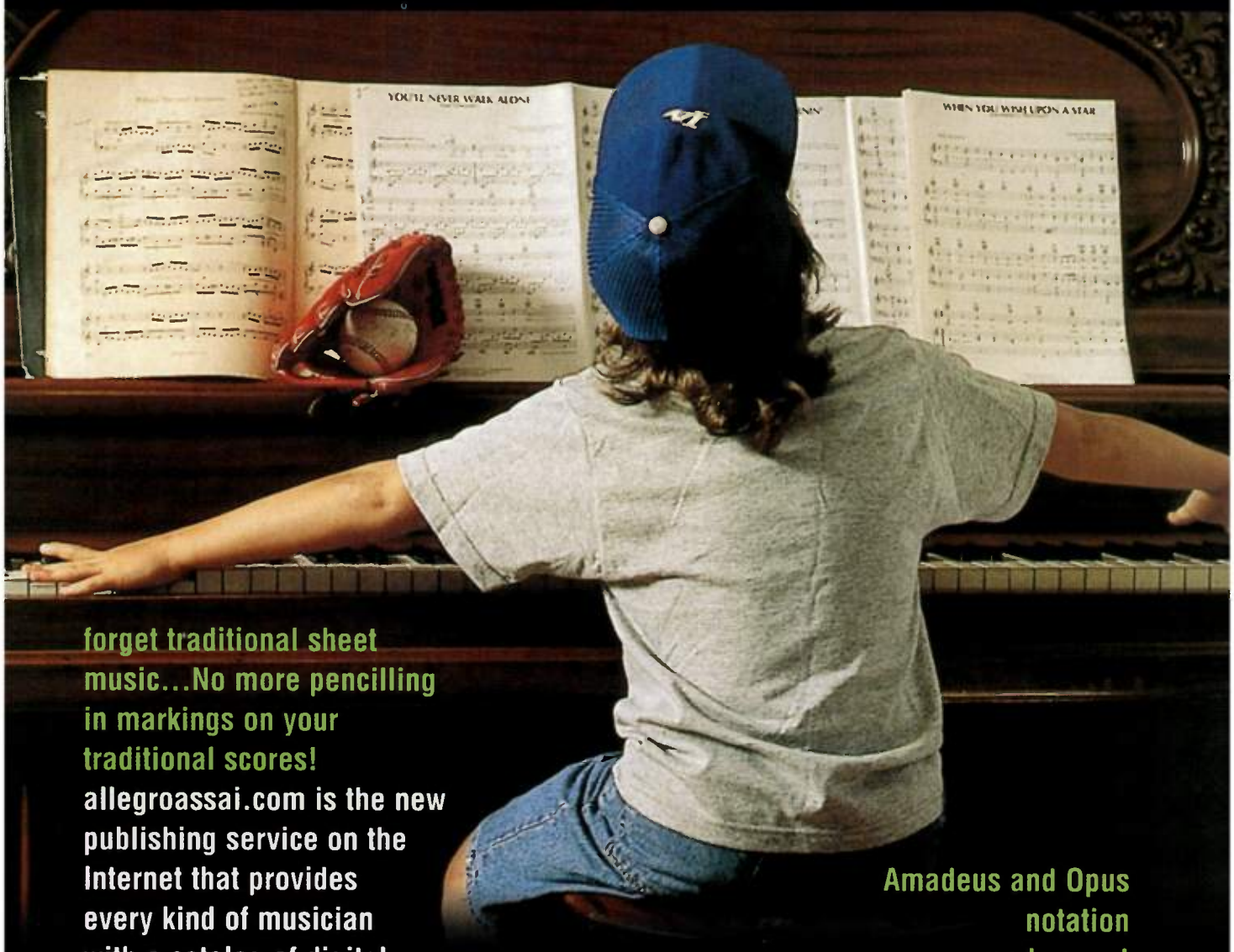

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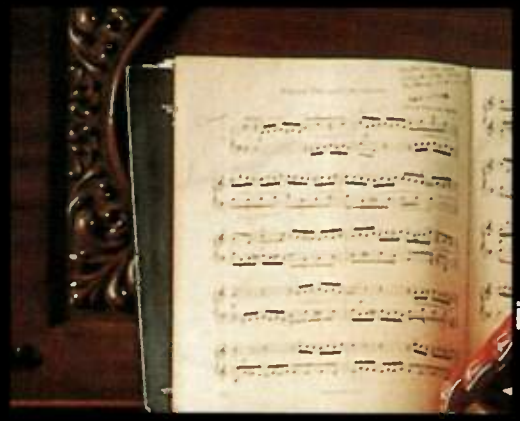

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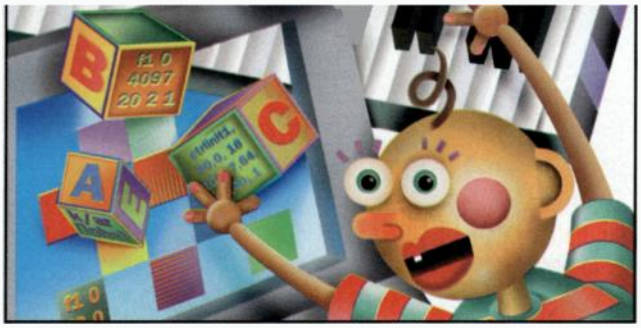

constantly evolving timbres using tiny wavetables. (Scanned synthesis is similar to wave sequencing in the Korg Wavestation, but it goes much further.) Unlike a traditional wavetable synthesizer, in which waveshapes are processed through filters, effects, and so on, scanned synthesis uses a physical model to generate a slowly varying waveshape that is scanned at a desired amplitude and rate. The shape of the scanned wave determines the timbre you hear, and the rate at which the wave is scanned determines the signal's pitch.

The waveshape used in scanned synthesis resides in a wavetable only 128 samples long. The wavetable can be any set of 128 individual samples, but the set based on a physical model of a vibrating string is particularly interesting. The string is modeled as a series of interconnected masses and springs. This "mass-spring" model is a very common way of looking at the physics of a string; researchers have used it for many years. (For more information on this model, see Curtis Roads's *Computer Music Tutorial* [MIT Press, 1996].)

Depending on how the model's components connect with each other, you can alter the wavetable so it represents the behavior of objects other than the string. For example, you can change the string into a cylinder, torus, or sphere, and you can also modify the scanning trajectory used to derive values from these models.

Csound implements scanned synthesis as a pair of opcodes: a dynamic wavetable generator called scanu and a wavetable scanner called scans. The dynamic wavetable that scanu generates evolves at a very slow rate, typically between 0 and 15 Hz. This rate lets you modify the waveshape as it evolves; in fact, you can control every component of the model individually—a very powerful capability indeed. By altering the effect of damping forces on the string, the manner in which it the string decays, the nature of the "pluck," and so on (all parameters of the scanu opcode), you can generate a vast range of waveshapes to scan. With a powerful computer, many of the opcode's parameters are controllable in real time, giving you a very powerful virtual instrument to perform.

The scans opcode can also be modified to take many different paths through the waveform. The basic trajectory is stored in a function table, which is simply a matrix that contains the locations of the points to be scanned and defines the order in which they are scanned.

Perhaps most important, because the opcode that scans the wavetable is separate from the one that computes and updates the wavetable, it's possible to apply the scanning system to other "classic" synthesis techniques and dynamically animate them. Many of Csound's signal generators use wavetables, so they can be scanned and animated in new, vivid, and controllable ways. Except for some highly specialized software, Csound is the only way to experiment with scanned synthesis. You can find a complete online tutorial on scanned synthesis with more than 20 model instruments at www.csounds.com/scanned.

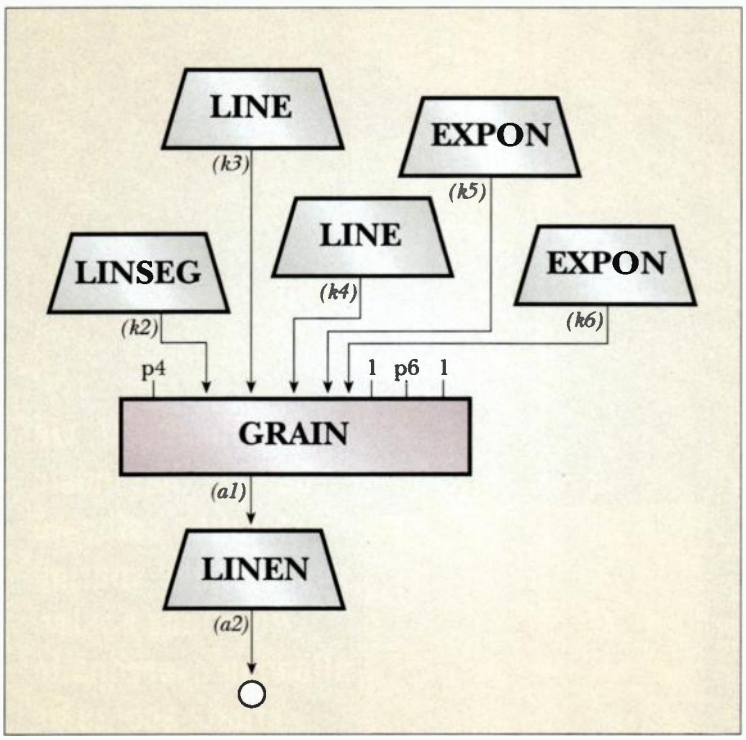

FIG. 3: This block diagram shows a granular-synthesis instrument in which linear and exponential envelopes dynamically modify the grain parameters. Each parameter is under the control of a different envelope. A linear envelope scales the grain opcode's audio output.

WHY CSOUND?

Csound provides every computer musician with new forms of musical expression and brings together the worlds of the composer and the sound designer. For the audio artist, Csound illuminates the underlying structure of the algorithm, the inner life of the sound. In Csound, the instrument is the synthesizer and the sound is the composition. As with any MIDI synth in your studio, you can play notes with the Csound synthesizer. In Csound, however, the note itself can be more than you've ever imagined.

Learning and using Csound has never been easier (see the sidebar "Online Resources" for a summary of online learning aids). Finally, your computer has the power to fully explore this limitless synthesis and signal-processing tool. So let your computer do some composing and start changing the way you make music. Join the international Csound community and take your music beyond MIDI, over the edge, into the ever-expanding Csound universe.

Richard Boulanger *is a professor of music synthesis at the Berklee College of Music and the editor of a new book on Csound called* The Csound Book *(MIT Press, 1996). He is also the editor of* The Csound Instrument Catalog with Audio, *available from cSounds.com.*

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'Buy Me A Rose' was recorded on an 8-track in the back bedroom of an old farm house. We only had one microphone. And every time a cow mooed or a plane flew over, we had to stop the tape. Not very high-tech, but it worked.

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Windows Media Audio

Microsoft ups the ante in streaming audio.

By Brian Smithers

When the computer industry's 500-pound gorilla turns its attention to streaming audio, you just have to stop and listen. Microsoft has introduced the latest version of its streaming media technologies, promising, among other things, significant reductions in file size with improved audio performance.

Windows Media Technologies (formerly known as NetShow) is a set of components that handles the preparation, delivery, and playback of audio and video. The audio component, Windows Media Audio (WMA), is Microsoft's newest codec for wringing more audio quality out of fewer bits. As usual, Microsoft has covered all the bases, offering *Windows Media Tools* for encoding audio and video, *Windows Media Services* for distribution, *Windows Media Digital Broadcast Manager* for e-commerce, and *Windows Media Player* for end-user playback (see **Fig. 1**). Before the sheer repetition of the name gives you the same tic it's given me, let's just say that for anyone trying to pump music through the still-inadequate cyber pipeline, WMA is definitely a step in the right direction.

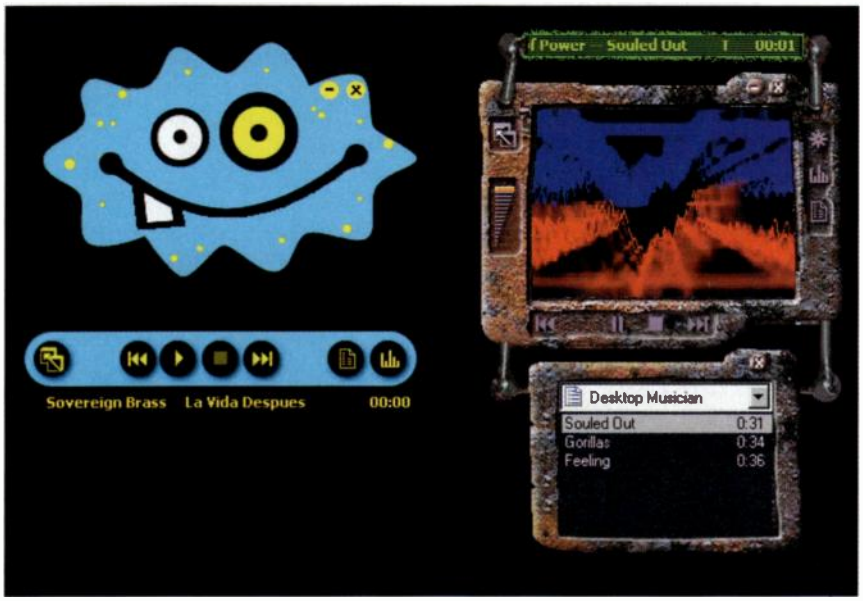

FIG. 1: Meet Toothy and Rusty. You could just chuckle at Microsoft for trying to keep up with Winamp and other players by adding skins and "visualizations" to *Media Player*, or you could design a cool skin with your band's logo and make it available to your fans on your Web site right beside your MP3s—I mean WMAs.

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type of data compression called *frequency coding* or *perceptual coding*. Sophisticated algorithms analyze the audio to determine what our ears will miss the least and discard it. Because some information is lost in the process, this is known as "lossy" compression. The most prevalent forms of lossy audio compression on the Web are currently RealNetworks G2, MP3, and WMA.

All three compression schemes are capable of delivering high fidelity at low compression ratios, but the more you squeeze the audio data, the more obvious the "lossiness" becomes. Telltale signs include diminished high- and low-frequency response, graininess, and a swirling sound not unlike the effect of a phase shifter or ring modulator. Each new generation of encoders further reduces the side effects of frequency coding, and the current version of WMA continues the trend.

HYPE, HYPE, HOORAY?

Microsoft's marketing machine boldly claims that WMA produces "sound quality at 22 Kbps that is roughly equivalent to G2's audio quality at 44 Kbps streamed, and it produces files that are half the size of equivalent-quality MP3 files." Of course, these are the same reality-challenged folks who keep telling us that the latest version of Windows is faster, easier, and more dependable than the last, so I was understandably skeptical about this claim. After careful listening tests, however, I was pleasantly surprised.

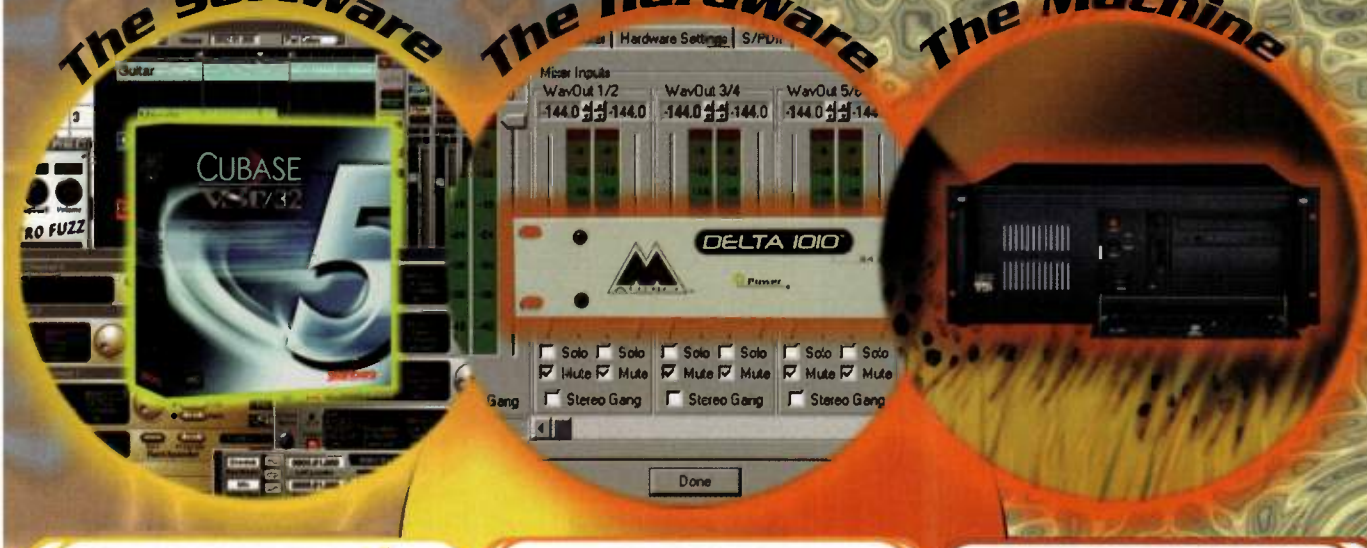
The comparison to MP3 is hard to argue with, especially when listening through typical computer speakers. At 64 Kbps, WMA exhibits little if any "swirl" and matches the reasonably smooth character of MP3 at 128 Kbps. At its highest bit rate of 96.7 Kbps, G2 achieves very similar audio quality. I would bet that most people listening through most common car stereos, boom boxes, or portable headphones would find it hard to distinguish audio encoded at these bit rates from the original source. At 128 Kbps (MP3's 10:1 data compression ratio), this is impressive enough, but at 64 Kbps, WMA manages better than a 20:1 ratio.

Microsoft's claims for WMA versus G2, however, are slightly overstated. At comparable bit rates, WMA did indeed sound better than G2, but it did not sound "roughly equivalent" at half the bit rate, in my opinion.

I encoded and auditioned clips in a variety of musical styles at 22 Kbps and 32 Kbps settings: typical for 28.8 and 56 Kbps modems, respectively. At these lower bit rates, the differences between the compression formats became readily apparent. MP3 sounds the worst by far at low bit rates, with the swirl and grit spoiling the music in a hurry. G2 at 32 Kbps, on the other hand, is pretty listenable. The frequency response seems comparable to AM radio, which makes this combination a perfectly viable medium for "taste-testing" new music. Given the proliferation of Web radio sites, plenty of people also find it adequate for background listening.

While WMA failed to match G2 at half the bit rate, it did sound clearly better at the same bit rate. There was less phasing and a much broader tonal spectrum. The music sounded more real and maintained some of the bottom end that is usually so decimated by frequency coding. For me, WMA at 32 Kbps turns the corner from tolerable to enjoyable. To put it in perspective, I have more than once put up

For manufacturer information, please see Contact Sheet, p. 221.



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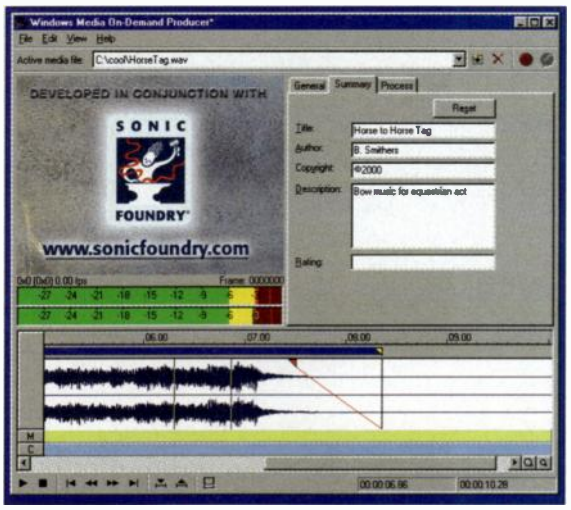


FIG. 2: *Windows Media On-Demand Producer* is a free utility from Microsoft and Sonic Foundry for encoding audio and video files in the Windows Media format. Audio files can be trimmed by dragging the yellow markers, and fades of varying lengths can be added by dragging the red markers.

with bad-sounding radio reception for an extended period if I was really interested in the music.

If you have to encode at 22 Kbps, Windows Media Audio does help, but it's still not a great choice for musical purposes other than auditioning music before buying it. While WMA's improved frequency response fills out the sound, it's still pretty rough. On the other hand, it's miraculous that you can even recognize music that is squished down to 1/47 of its original size. A 30-second clip that started out at 5 MB is only 80 KB at the 22 Kbps setting! If you have no choice but to squeeze your music through the eye of a needle, WMA is for you.

WE ARE BORG

Microsoft has thought of just about everything, assimilating the diverse functions of G2 and MP3 into a single creation-to-destination package. It has been common for online music lovers to audition new music via 32 Kbps RealAudio streams and then download 128 Kbps MP3 files for playback on their computers or portable MP3 players. Musicians hawking their wares online have had to encode, upload, and present their music in both formats.

To accomplish the same thing with WMA, bands may still want to use a lower bandwidth, such as 32 Kbps, for streaming samples, and a higher bandwidth, probably 64 or 128 Kbps, for download. They still have to manage two versions of the file, but will only have to deal with a single file format. Instead of using products from different companies to encode the files into different formats and then using RealNetworks' *RealPlayer* or Nullsoft's *Winamp* to play them, you can use *Windows Media On-Demand Producer* to encode both versions and *Media Player* to play both. What's more, dozens of companies representing many of the most popular audio players have already licensed the Windows Media Audio format, including Nullsoft, MusicMaker, Yahoo, and even RealNetworks. This widespread support should make it even easier for you to distribute your music online.

The impressive *Windows Media On-Demand Producer* utility (see **Fig. 2**) was developed by Microsoft and Sonic Foundry. It devours audio and video files and spits out Windows Media files. It can even generate HTML templates to simplify the publishing process. All you have to do is open a file and push the red button, and a wizard prompts you for bit rate, destination directory, and file name. I rarely find such wizards useful, because they often don't work, take too long, or don't cover the specific needs of my project. Happily, the *On-Demand Producer* team got it right, and the encoding wizard is simple, quick, and useful.

You can even encode multiple files in a single process, although you can't encode a group of files at mixed or multiple bit rates in a single pass. *On-Demand Producer* makes child's play of trimming a file and adding fade-ins and fade-outs. To change the beginning or end of a file, simply drag the yellow marker. To fade in or out, drag the red marker. You can zoom in to the sample level, add markers, and embed commands or text, including closed captions. Microsoft and Sonic Foundry couldn't have done much more to make encoding WMA files easier, and the best part is that *On-Demand Producer* is free.

Speaking of free stuff, I also tried out the new *Windows Media Player* 7 (see **Fig. 3**). It's big, it's slick-looking, and it wants to manage all of your audio and video files. It plays audio CDs, looks up artist and album information online, and even extracts CD audio tracks and converts them directly into WMA

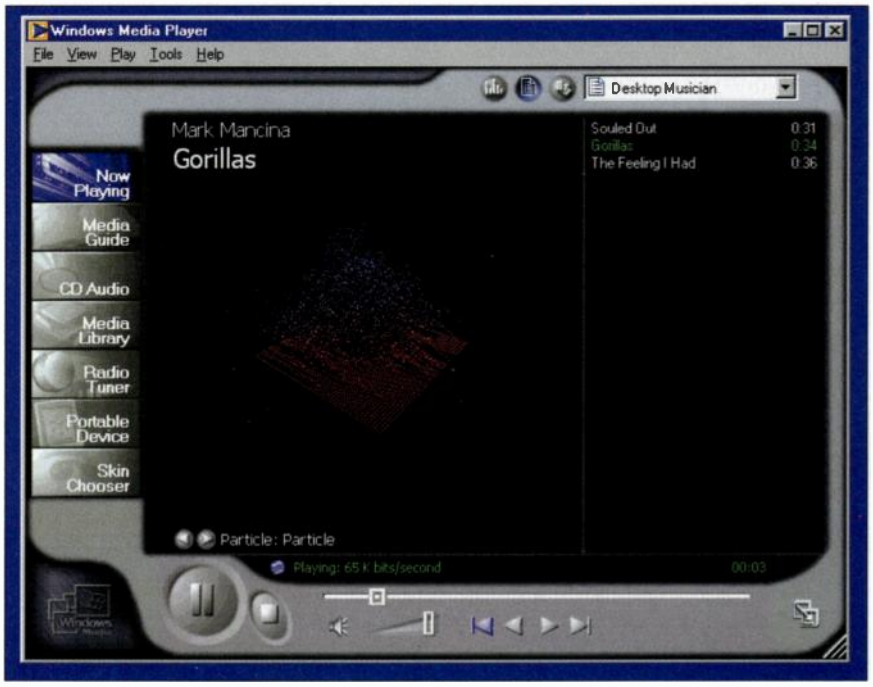
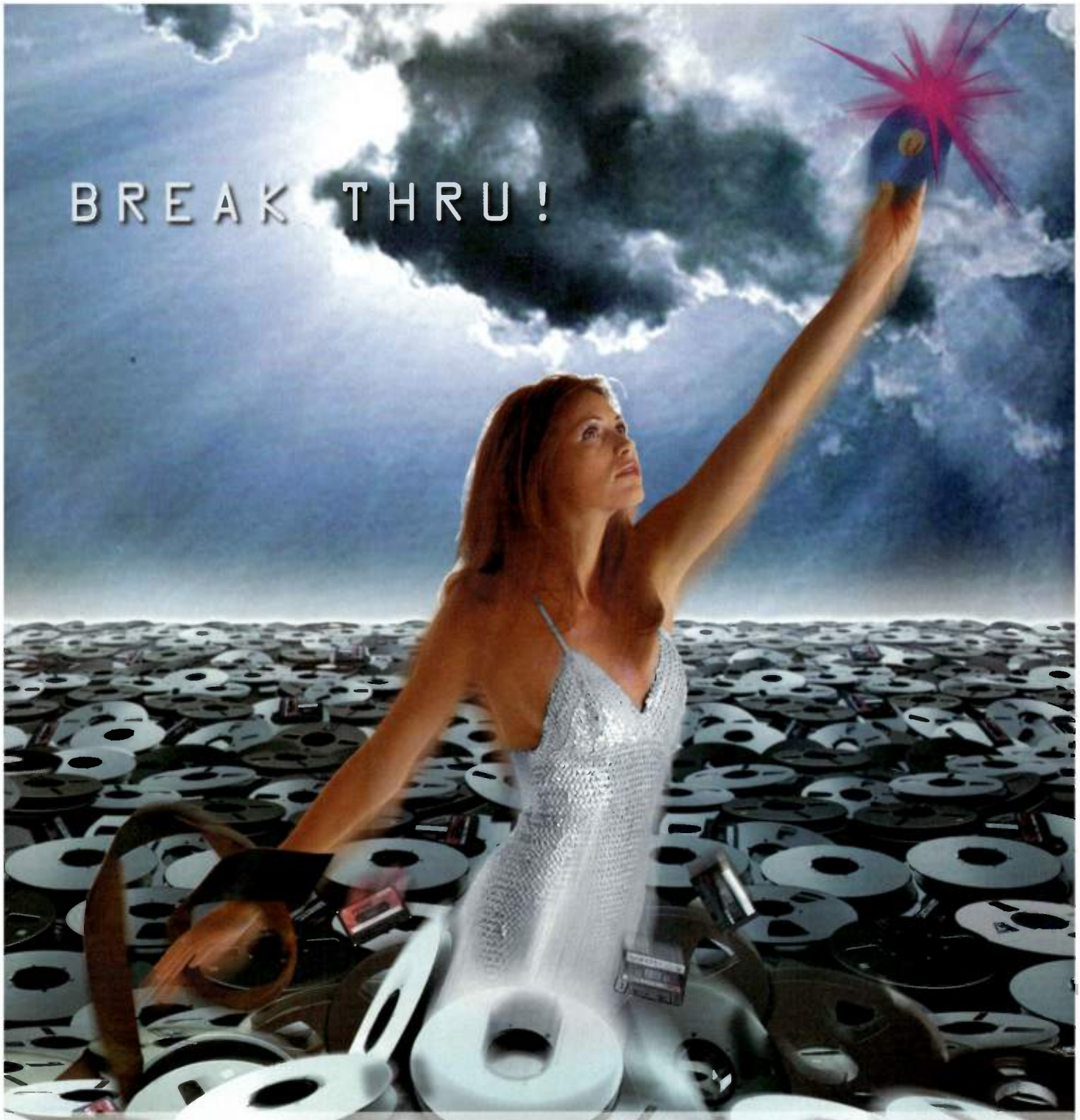


FIG. 3: Version 7 of the *Windows Media Player* features playlists and file management along with the ability to "rip" files directly from CD to WMA format. Along with direct links to Microsoft's online media guide, it offers facilities for loading WMA files into portable players.



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files. You can then use the program to load the resulting WMA files into a compatible portable player. New features include the ability to burn audio CDs from WMA and MP3-based content, and the new player also offers artists the option of allowing secured content to be burned to CD. What's more, the new version lets you create Content Packages: a kind of boxed set that includes graphics and audio in a single download. That means you can add such things as cover art, lyrics, Web site addresses, and even video clips to your works when you distribute them.

Media Player 7 also lets you create and manage playlists of audio and video files, and it keeps track of your Web radio stations. As if that weren't enough, it also looks cool. As with *RealJukebox* and *Winamp*, you can select from among a couple dozen "visualizations," which are more or less graphic permutations of a waveform display, and you can dress up *Media Player* in different "skins" (see **Fig. 1**).

Giving anyone the ability to transfer digital media so easily, however, is a double-edged sword, and Microsoft has included tools to give artists some chance of receiving financial compensation for their work. *Windows Media Rights Management* is an SDMI-capable mechanism that encrypts files to enable authors to track and license playback. (SDMI stands for Secure Digital Music Initiative.) Support for e-commerce and pay-per-view scenarios is also provided.

▼

Windows Media Audio stands a very real chance of upsetting the apple cart.

THE WORLD WILL BEAT A PATH

Third-party support for WMA is growing quickly, and Microsoft's list of "coming attractions" is impressive. Products from Sonic Foundry and Cakewalk already support the new format, and other companies are jumping onboard as the demand grows. Version 6.3 of *Media Player* for Macintosh is currently in beta release, so Mac-based end-users will be able to take advantage of WMA. Mac-based producers of audio content aren't quite so lucky, though, as *On-Demand Producer* is currently available only on the PC.

One of the driving forces behind the MP3 explosion is the proliferation of Rio-style portable playback devices. WMA has its sights set on such players; according to Microsoft, the next generation of many of these players will support WMA, including Diamond's Rio 600, RCA's Lyra, Creative's Nomad II, and Sony's VAIO Music Clip.

Microsoft has also built support for WMA into its Windows CE handheld-computer operating system. Now CE-based PDAs are poised to do double duty as portable music players. Palm-sized PCs from Casio, Compaq, and Hewlett-Packard already natively supports Windows Media Audio.

A number of major music Web sites have announced support for WMA as well, including giants Sony, BMG, and EMI. Microsoft has also launched its

own site full of multimedia content, called—surprisingly—WindowsMedia.com. Liquid Audio is building WMA support into its online offerings and the next major version of its multiformat player, and House of Blues is broadcasting concerts in WMA alongside RealAudio.

RealNetworks commands about 80 percent of the streaming audio market, and anyone on the planet who doesn't know how to play an MP3 file on their computer knows a 12-year-old who does. Nevertheless, Windows Media Audio stands a very real chance of upsetting the apple cart, especially when you consider WMA's improved audio quality, easy implementation, and Microsoft's knack for getting major players on its bandwagon. The fact that RealNetworks is releasing version 8 of its product line without any improvements in its audio characteristics gives WMA an additional boost.

CALL THE ATTORNEY GENERAL!

I must admit that it's impressive and startling to see Microsoft roll out something as comprehensive as Windows Media Technologies. The company had the foresight as well as the resources to cover all the bases and to do it in a single stroke. Those looking for a way to present and distribute music online hardly need to look beyond WMA. With a free high-quality encoder, a new codec that covers the ground previously split between RealNetworks G2 and MP3, support for digital rights management and e-commerce, a free playback browser plug-in, and growing support in consumer devices, it's as close to a complete solution as you can get.

As a musician preparing material for online distribution, it's great to have this sort of one-stop shopping. At the same time, you have to wonder what the Windows Media juggernaut will do to the lively competition that has brought streaming audio so far so fast. The one thing that's certain is that WMA has succeeded in upping the ante for compressed audio quality in both streaming and downloadable applications, and that's a good thing.

Brian Smithers *is an Orlando-based musician, conductor, and educator. His latest work,* La Vida Después, *was recently premiered by the Sovereign Brass.*

Spectral Vistas

Synthesizing realistic instruments for fun and profit.

By Stephen David Beck

You've just finished your latest symphony, but the New York Philharmonic isn't returning your calls, and you don't have enough money to hire an orchestra. So you decide to use your General MIDI synth to orchestrate the piece. But wait a minute; that violin doesn't sound very real, and that flute—who are you kidding! What's a modern composer to do?

The problem is, your ears and brain are just too good and too smart to be tricked into believing that a GM flute is really a flute. For us to believe that a synthesized sound is real, the sound must behave according to the fundamental laws of acoustics. This is a principle I like to call "acoustic viability." Let's talk about what it means to create an acoustically viable sound and what you can do to make your sounds more realistic.

When you start to examine the makeup of a particular sound, you discover that timbre is not static. In fact, an instrument's timbre varies tremendously with respect to time, amplitude, and pitch. These variations are extremely important in that they enable us to identify instrumental sounds.

Unless synthesized instruments account for these timbral fluctuations, they will not sound acoustically viable. Like natural instruments, synthesized instruments must reflect the changes in timbre that are consistent with the principles of acoustics. Many common synthesizers don't take these subtleties into consideration, but there are ways to incorporate timbre changes into your sounds.

JACK DESROCHER

GETTING TO KNOW YOU

So what are the properties of an acoustically viable synthesized instrument? Three important principles of acoustics come to mind: initial transients, resonance, and envelope variations caused by amplitude and pitch. These are not the only issues to worry about,

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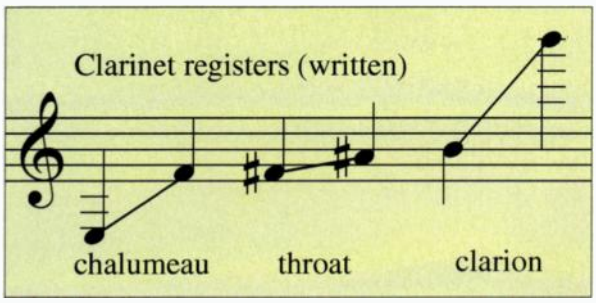

FIG. 1: The clarinet's range falls into three distinct registers: chalumeau, throat, and clarion.

but they're very important, and addressing them will dramatically improve your sounds.

Initial transients. In the first 50 to 100 milliseconds of an instrumental sound, its spectrum is very unstable. This is due to *inertia,* the law of motion that states that things at rest tend to stay at rest unless acted upon by an external force. For example, the air column inside a saxophone has a certain amount of inertial resistance that must be overcome before it will vibrate properly. During the first 50 milliseconds of a note, that inertial battle produces wild spectral fluctuations called the *initial transient.* Pitch goes haywire, chaos ensues, and then you hear "saxophone."

Initial transients can vary dramatically at different pitches and amplitudes. For stringed instruments, higher pitches are generally produced by shorter lengths of string and thus exhibit less inertial resistance and shorter transients. Low pitches are produced by longer lengths of string and thus exhibit more inertia and longer transients. These variations in initial-transient behavior provide important cues as to what instrument is making the sound.

The impact of initial transients on an instrument's sound is twofold. In wind instruments, there is some pitch fluctuation caused by the instability of the air column. Second, the timbre of the sound is incomplete, nonharmonic (that is, the partials are not multiples of the fundamental frequency), or both. As the air column or string overcomes the inertial resistance, both pitch and spectrum stabilize.

Because most wavetable synthesizers are based on acoustic-instrument samples, the initial transient is preserved in the recorded samples. But if the synth doesn't use multiple wavetables (different samples for different pitches), the transient you hear could be acoustically wrong.

Resonance. It's well known that the timbre of acoustic instruments varies greatly across the pitch range. That is, the timbre of a given instrument is different depending on what pitch it's playing. As illustrated in **Fig. 1,** for example, the clarinet has three distinct registers: *chalumeau* (low), *throat* (written F♯ to A♯ above middle C), and *clarion* (upper). Each register has a different color and feel, but they are all part of a single, dynamic timbre.

Registers sound different because the musical instrument itself shapes the timbre through resonance. Have you ever noticed that when you sing in the shower, some notes boom out much louder than others, even though you aren't singing those notes any louder than others? This is because the shower functions as a resonator. It naturally amplifies certain frequencies based on its size and shape.

All musical instruments exhibit the same effect: The shape and size of a musical instrument creates natural regions of resonance called *formants,* and any frequency that falls within a formant region is naturally amplified, altering the resulting spectrum (see **Fig. 2**).

Remember that instrumental sounds contain many *partials* or *harmonics.* Consider an instrument with a formant region centered at 600 Hz. If you play a note with a fundamental frequency of 200 Hz, the third harmonic (600 Hz) is amplified, contributing significantly to the sound's spectrum. A note with a fundamental frequency of 250 Hz will have a different timbre because the second harmonic (500 Hz) and third harmonic (750 Hz) fall outside the formant's range.

How can you simulate formant resonance? With samplers, you can use multiple samples of an instrument spread out over the keyboard. A sample every three to five notes should give good results. But what if you don't have a lot of RAM in your sampler, or your wavetable synth doesn't use multiple samples? Another solution is to put a resonant bandpass filter in the signal path.

Bandpass filters can reasonably simulate the effect of an instrument's formants. The filter boosts frequencies inside the band, simulating the formant effect. However, if the ratio of the filter's center frequency to the filter's bandwidth (often called Q) is greater than 20, the filter begins to ring like a bell. This can be a cool effect, but it takes the formant a step too far.

Envelopes, amplitude, and pitch. The speed of an instrument's pitch envelope (during initial transients), spectral envelope, and amplitude envelope varies with both the pitch of a note and its dynamic level. This seems rather intuitive, and it's also related to inertia. However, there are some interesting interactions worth considering.

For example, pitch affects the amplitude envelope in various ways. Higher pitches have to move less mass, so the attack and release times of the amplitude envelope are shorter. This is because energy dissipates more quickly from small masses. Lower pitches have to move lots of mass, so attack times are longer. They also dissipate energy more slowly, so release times are longer as well.

Amplitude also affects its own envelope. When a note is played loudly, there is a lot of immediate energy to overcome inertia, so even low notes have quicker attack times with loud dynamics. In addition, the inertial resistance is overcome more quickly, resulting in shorter initial transients.

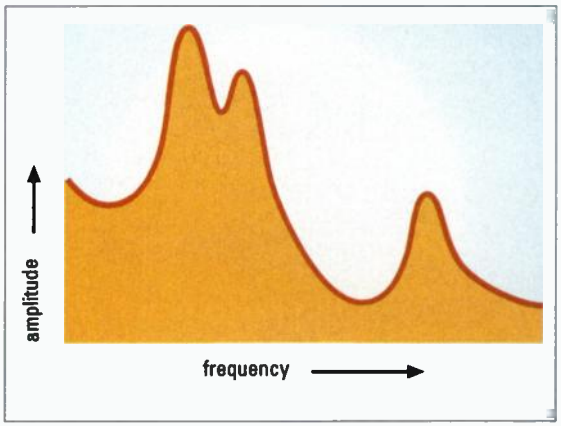

FIG. 2: Formant regions are traits that all musical instruments share, including the human voice. The vowel sound "ah" produces three distinct regions.

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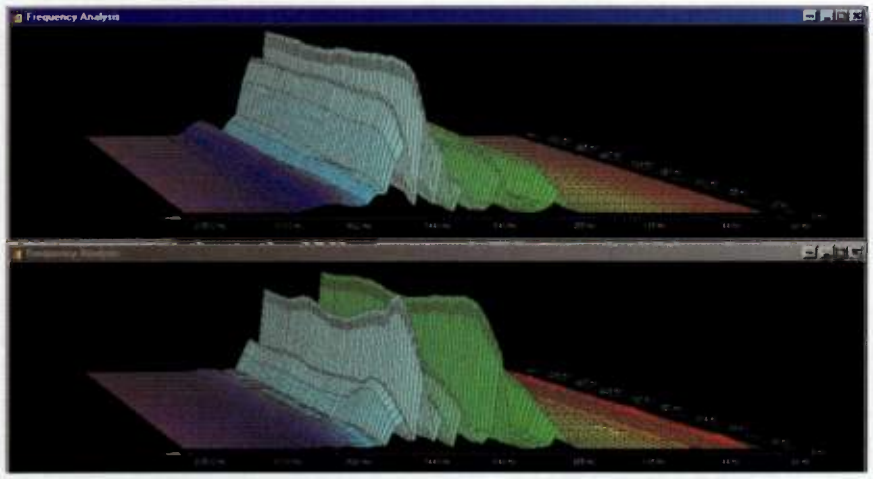


FIG. 3: An identical note played at different amplitude levels generates different amounts of upper partials in most instruments. When an oboe plays B♭3 loudly, there is significant energy above 3000 Hz (top). When the same note is played softly, there is very little energy above that point (bottom).

More important, loud notes have more partials than soft notes. In any wind instrument, the air column vibrates at several frequencies simultaneously, each a multiple of the fundamental frequency. The louder a note is played, the more energy is pumped into the instrument, which stimulates higher-frequency vibrations (see **Fig. 3**). Each of these frequency components has its own amplitude envelope (these are called *spectral envelopes* to distinguish them from the main amplitude envelope). As amplitude increases, so does the speed of the spectral envelopes.

TRY THIS AT HOME

To incorporate some of the acoustic properties described so far, you'll need to adjust various parameters on your synthesizer. For example, a note's pitch and Velocity can be assigned to modify the attack and release times of the pitch and amplitude envelopes; this is called *scaling*. Look for features called something like *envelope scaling, Velocity scaling,* or *amplitude scaling* in your synthesizer's manual to find out how to accomplish this. Basically, you want to set it up so that higher pitches and Velocities shorten the pitch- and amplitude-envelope attack and release times.

Many samplers let you switch between soft, medium, and loud samples based on Velocity. You can also assign different samples to different keyboard zones, which sounds more realistic as you move up and down the keyboard. Of course, RAM requirements increase with either of these techniques.

If you own an FM synthesizer, you can assign Velocity and/or pitch to scale the modulation index, which affects the brightness of the sound and is an important factor in producing realistic

sounds on these instruments. Finally, try assigning pitch and/or Velocity to control the cutoff frequency of a lowpass filter; the higher the pitch or Velocity, the higher the cutoff frequency of the lowpass filter and the brighter the sound.

The Roland JV series of synths provides a good model for experimentation. These instruments use multiple samples (Roland calls them Tones), each of which can be enveloped, filtered, chorused, and reverberated. Tones can be layered in multiple combinations, and you can activate different Tones based on note number or Velocity value. By scaling filter resonance and envelope attack and release times with Velocity and using different Tones for different pitch registers, you can put together compelling synthesized instruments.

Using pitch and Velocity to scale the timbre and envelopes is perhaps the most important technique you can use to create acoustically viable sounds. You'll be amazed at how powerful these techniques can be for suggesting realistic acoustic behaviors.

IS THAT ALL?

In addition to the general principles described here, all acoustic instruments have unique properties that you should get to know when attempting to synthesize them. For example, unlike other woodwinds, the clarinet's spectrum contains primarily odd-numbered multiples of the fundamental frequency. In addition, the timbre changes very little during the sustained portion of a note. Starting with nothing more than a simple square wave, a filter, and an envelope generator, you can quickly create some convincing clarinet sounds.

On the other hand, brass instruments share certain acoustic properties. For example, notes at different amplitudes produce radically different spectra. In addition, higher partials generally take longer to sound than the lower partials. Certain brass instruments, such as the trumpet and trombone, have a cylindrical shape, while the French horn and tuba have a conical shape. These are just a few of the many factors that affect the "acoustic signature" of a musical instrument.

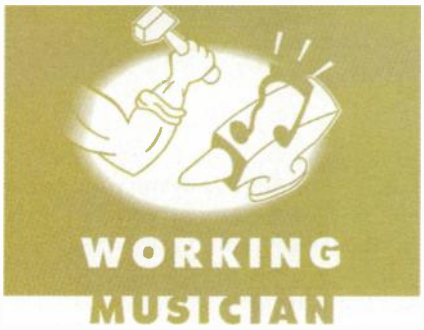
In addition, you need to understand how instruments are played, or more importantly, how they can't be played. A single piano note cannot crescendo, and you can't expect a woodwind or brass player to hold a note for more than 30 seconds. Knowing your acoustic instruments is as important as understanding your synthesizer.

GET VIABLE

The principle of acoustic viability is simple: If a sound behaves according to the laws of acoustics, people will believe the sound is real. Acoustic viability also helps convey those subtle changes in timbre that are necessary to create interesting and evocative music.

Incorporating the properties of acoustics in synthetic instruments is a powerful technique that will instantly improve your sounds. It does mean you'll be spending extra time programming your synth, but your music is worth the effort, right?

Stephen David Beck *composes music in both electronic and acoustic genres, and he codirects the Music & Art Digital Studio (the MADstudio) at Louisiana State University in Baton Rouge, Louisiana. He is also a past president of SEAMUS, the Society for Electro-Acoustic Music in the United States.*



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A primer on legally setting up your Internet-based business.

Michael A. Aczon

So, all those rags-to-riches stories you've heard have compelled you to start a company on the Internet. Before you leave the real world behind and launch your virtual company, take a step back and assess the various legal issues that await you. You need to devote some time and effort to making sure you cover all of your legal bases. As you probably know, the nature of online commerce changes almost daily, so you will also have to keep up with the new laws and legal rulings that will affect your business down the road.

JACK DESROCHER

To get you started, I have listed here several of the legal aspects of setting up an e-business. While this checklist is not exhaustive, it will make you aware of the major bodies of law that can affect an online entertainment-related business. These include intellectual property rights, personal property rights, and contract rights. Because every business is unique, you have to decide which legal issues require your attention.

THE RIGHT RIGHTS

Copyrights, trademarks, and domain names are among the most hotly contested issues in the entertainment industry. For entrepreneurs in online entertainment, establishing ownership of these rights is especially important—in fact, it's everything.

Trademark. In basic terms, a trademark tells the consumers the source of goods or services. Trademarks can take a number of forms, such as a name, a logo, or a phrase. For example, the Nike trademarks are "Nike," the Nike "Swoosh" and the phrase "Just Do It." Your first task is to decide exactly what you want to trademark.

Because an online business can cross state and international boundaries, you must seek federal trademark protection. The Web site of the U.S. Patent and Trademark Office (PTO) has very thorough and easy-to-follow instructions that walk you through the

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trademark-registration process online. (For the Web addresses of the PTO and other rights organizations, see the sidebar "The Rights Stuff.") After you've determined that your proposed trademark is available for use, the PTO requires you to (1) submit the words, phrases, or artwork you want to protect, (2) describe the class of business in which you are establishing your company, and (3) pay a filing fee.

The PTO then reviews the validity of your submission, taking in account challenges from other parties who feel your submission infringes upon their preexisting trademarks. Depending on the review's outcome, the PTO may register your trademark with the U.S. government. Once your trademark is registered, you may safely invest your time and money in making that name known to the public, armed with a good legal argument against anyone who claims to have owned it first.

Domain names. In e-business, your domain name is your calling card. It's how people remember your business, and it's what search engines pull up when people are searching for your type of product. Register your domain name at the same time you pursue your trademark protection. You should register it in as many configurations as possible and economically practical (for example, myname.com, my-name.com, myname.net, myname.org). Some online services can officially reserve a domain name for you. The Web site of the Internet Corporation for Assigned Names and Numbers lists all of the services that assign domain names.

Your desired domain name may already be registered by someone else, in which case you can either come up with a different name or purchase the domain name from the current owner. If you choose the latter option, be sure that the purchase contract spells out that you are the sole owner of the trademarks associated with the domain name. Also stipulate that the party selling you the name will not use or sell any domain name similar to or a variation of the name you just purchased.

Copyrights. The owner of any kind of copyrighted work has the exclusive rights to make copies of, display, perform, make revisions to, and publish that work. To use or sell third-party works on your Web site, you need the permission of the copyright holder. You may have to pay for those rights, either up front or based on the income you earn from those works. Payment arrangements will depend on the type of use. For example, if your site sells other artists' CDs and videos, your license agreement

The Rights Stuff

Check out the sites below for further information on copyrights, trademarks, domain name registration, and performance rights.

ASCAP	www.ascap.com
BMI	www.bmi.com
Internet Corporation for Assigned Names and Numbers (ICANN)	www.icann.org
SESAC	www.sesac.com
U.S. Copyright Office	www.lcweb.loc.gov/copyright
U.S. Patent and Trademark Office	www.uspto.gov

ONLINE RESEARCH AND ADVICE

Once you decide to take the plunge into the world of e-business, you must commit yourself to staying abreast of new technology and the laws involved in technology and business. Here are a few sites that might prove useful to you in establishing your business and keeping it legal.

There is no shortage of news outlets on the Web. In addition to the online arms of traditional news services such as the *New York Times* (www.nytimes.com) and CNN (www.cnn.com), you should frequently visit the news sections of such technology-specific sites as ZDNet (www.zdnet.com), Wired (www.wired.com), and Cnet (www.cnet.com).

You can also keep up with the major lawsuits that are brewing by checking in with the litigants themselves, the RIAA (www.riaa.com) and Napster (www.napster.com). The Napster site's "Press Room" area contains many articles about Napster and the RIAA's lawsuit against the company. These stories are from reputable news organizations from around the world. Many of Napster's court depositions are also available as downloadable PDF files.

The RIAA site has general information on copyright and piracy, as well as most of the organization's press releases regarding the suits against MP3.com and Napster. Bear in mind that these are press releases, not news accounts, so they only contain the RIAA side of the story. Still, reading them is a good way to follow the history of these cases.

Two legal sites worth checking out are FindLaw (www.findlaw.com) and FreeAdvice (www.freeadvice.com). The FindLaw site offers legal news, articles, discussion boards, and advice for small-business owners. The site's home page is text heavy and a little difficult to wade through, but clicking on the "Small Business" link takes you to a well-organized and helpful list of topics, such as step-by-step instructions on how to put together a business plan and how to deal with intellectual property and contracts.

FreeAdvice also provides help for small businesses. Topics covered include getting a tax identification number and partnership types. The site is heavily oriented to hooking you up with a lawyer (the enormous blue Find a Lawyer button after every paragraph is a tip-off), but it does give plenty of free helpful advice. FreeAdvice also sells legal forms at low prices (most cost about $10 or $11, and a few are $15).

—Mary Cosola

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with the artists, their labels, and their publishing companies may allow you to post their work for no fee up front, but require you to give them some of the money you get from unit sales. Some of the copyrighted works you want licenses for are photos, artwork, text, music, and videos.

The performance rights societies (ASCAP, BMI, and SESAC) try to keep up with the licensing and royalty collections for musical compositions delivered through Web sites. If you get a blanket license from each society, you can play any song from their catalogs on your site. Contact the performance rights societies to learn the procedures for obtaining performance licenses.

Of course, if your site will feature only your music and artwork, you won't need permission to use it. You should, however, copyright anything you have created for your site. Doing this through the U.S. Copyright Office is easy.

On each page of your Web site, include a tagline noting that all images, text, and music are the exclusive property of the copyright holders, and give the date of the copyrights. Some online businesses opt instead to devote a page to the various copyrights and trademarks connected to the works on the site. It's a simple way of dealing with the issue because updating one page is easy.

Trade secrets. In the fast-moving and highly competitive world of Internet commerce, protection of concepts, contacts, technology, and information—also known as trade secrets—is crucial. The nondisclosure agreement (NDA) is the legal ammunition used to safeguard trade secrets. Parties to NDAs disclose what information they obtained prior to their relationship and then promise that they will not disclose to others their respective secrets or use the information for their own ends. For example, if I disclose to you what I already know at the outset of our business relationship, you can't claim that I developed the idea *while* we were working together and then try to stop me from pursuing it on my own in the future. Our NDA would prevent you from claiming that it was *our* idea and not mine alone.

In e-business, your domain name is your calling card.

NDAs can also protect technology, information, and intellectual property developed during the course of a business relationship. If you've thought of a brand-new online business or a completely different approach to an existing business, it is worth your time and effort to prepare a well-drafted NDA and have it ready when you let others—such as potential partners and investors—know about your idea.

Personal rights. In setting up your online business, you'll have to consider the personal rights of the individuals appearing on your site. These rights stem from our constitutional rights to privacy. Furthermore, laws in many states protect an individual's right to the publicity and exploitation of his or her name, likeness, performances, and biographical material.

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So, before using any of those items online, obtain a release from the involved people that includes online display and distribution rights. Some audio/video production companies and record labels may have releases that were created before online rights were a consideration, so review the existing releases and negotiate to extend the rights to include online uses if necessary.

Keep in mind that so-called unofficial fan sites and flaming sites—which allow and even encourage people to share opinions and information about individuals—may open site owners to libel or disparagement lawsuits. Free speech online has been well respected and supported so far, but publishing false information that damages a person's character or reputation is a definite no-no. Check your site for any libelous material.

BUSINESS BASICS

Structuring your company as a sole proprietorship, general partnership, joint venture, limited liability company, or corporation can have enormous impact on the future of your business. Determining which of these forms of business is best for you is beyond the scope of this checklist, but here are some questions to ask yourself: Do you want to take on partners? Are you raising capital to get the business started? If you intend to sell the business later on, how much of your ownership and management share are you willing to give up?

▼

Review your deal with your Internet service provider.

The media has focused a great deal of attention on the "instant millionaires" who cashed in on their stock options after their Internet companies went public. If you are handing out pieces of your company in exchange for spec work, just be sure you retain enough of the ownership to control the management decisions and see your vision through.

Negotiate contracts with third parties with the Internet in mind. Review your deal with your Internet service provider. ISPs handle business accounts differently from personal accounts, and technical reliability, volume of use, and types of services vary widely. Don't get locked into a long-term contract with an ISP that might not be able to meet your needs as your business grows. Your contracts should spell out royalty and sales splits with content providers (including artists and record labels), online distributors, and other e-commerce businesses.

The government is currently encouraging sales on the Internet, but it's still determining federal and state taxes. Keep an eye out for any changes in tax policy. (See the sidebar "Online Research and Advice" to learn where to get the latest business and technology news and free legal and business advice online.)

FINE PRINT

Online entertainment is probably getting more publicity than any other type

of e-business. As I was preparing this column, the Recording Industry Association of America (RIAA) filed major lawsuits against MP3.com, MP3Boards.com, and Napster. The outcome of these lawsuits will directly affect how music is delivered to consumers. If you're not careful, any one of the court decisions could make your business legally liable to consumers, developers, or content providers.

One method of covering yourself is to include a carefully drafted disclaimer on your Web site. The disclaimer should state that you are

Your disclaimer and the business actions you take will put you on good legal footing.

aware of the various laws regarding trademarks, copyrights, privacy, and the like; that you are making good-faith efforts to comply with all of these rights; and that those using your site to conduct business can do so only if they comply with all of these rights. Taking this precaution won't guarantee that you'll be absolved of all wrongs, but your disclaimer and the business actions you take (for example, ceasing to do business with a person who knowingly violates another person's copyrights) will put you on good legal footing.

LOGGING OFF

Online business is one of the most exciting phenomena the entertainment industry has experienced in recent years, and it doesn't show any signs of slowing down. Many historical, business, and legal analysts compare the impact of online entertainment to that of the printing press, radio, and phonograph records. By keeping up with the legal issues that change with the technology, you and your cyberbusiness can ride this fast-moving wave more safely.

Artist manager and music-business educator **Michael A. Aczon** *recently logged more than 400 miles in an online cyberbicycle ride across the country.*

Expert Advice

Answers to readers' technical questions.

By Mike Sokol

XLR Q&A

Q: How are the three prongs on microphones such as the Shure SM58 related to the connections on a guitar cable? I'm trying to make an adapter that allows me to plug my mic into my guitar amp, but I'm not sure how the two are related. Any help would be appreciated.

Joe Reinhart
via e-mail

A: Your project is more complicated than just making a ¼-inch–to–XLR adapter cable. Because the output impedance of an SM58 microphone is less than 600Ω, its signal level is very low compared with that of a guitar's high-impedance pickups. So you need a microphone-input transformer (such as those made by Shure and Radio Shack) to step up the signal level enough to make the input channel happy. Because the transformer is a passive device, the output impedance is also increased.

The transformer looks like a little tube with an XLR female connector on one side and a male ¼-inch plug on the other. They cost about $15 and work quite well. Just plug the ¼-inch plug into your amp and the microphone cable into the female XLR jack. But here's a sound-engineer caveat: a guitar amp can be a poor substitute for a proper P.A. speaker. Guitar amps are intended for the midrange-heavy sound of a guitar and don't offer the full-frequency reproduction of a horn-equipped P.A. cabinet. As a result, you could be in for some serious feedback problems. However, try the adapter out and see how the guitar amp works in your application.

If your application doesn't require a transformer—for example, if you need to connect a mixer's speaker outs to a powered monitor's XLR ins, here's how you would wire the cable (see **Fig. 1**). With a TRS (Tip/Ring/ Sleeve) stereo plug, the tip goes to pin 2 on the XLR connector, the ring to pin 3, and the

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sleeve to pin 1. With unbalanced ¼-inch "mono" plugs, both pin 1 and 3 on the XLR are tied to the phone plug's sleeve, while pin 2 is tied to the phone plug's tip.

Although most audio equipment is wired "pin-2 hot," some foreign gear is wired "pin-3 hot." According to one of my British buddies, this is the U.S. manufacturers' fault: there have been several different official European standards over the past few decades, but U.S. builders would not agree on which pin should be hot. As a result, we sound folks carry around pin-2–to–pin-3 flipper cables for times when we need our gear to interface with gear from the other side of the pond. Not all gear is marked to indicate which pin is hot, so you may have to try your flipper cable both ways.

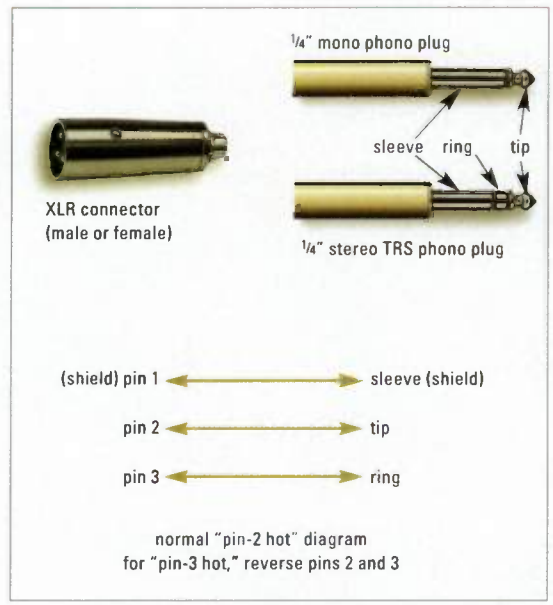

FIG. 1: This is the wiring scheme for making a basic XLR-to-TRS cable (with no transformer).

What Did You Say?

Q: I'm starting an internship as part of the music-production degree program at the Hartt school. I'll be working in sound reinforcement, and I'm considering getting custom-molded earplugs. Where should I search?

Mike Shear
via e-mail

A: Start your search at H.E.A.R. (Hearing Education and Awareness for Rockers) online (www.hearnet.com). Some of the best ear protection is for sale at the site. You can get anything from simple foam plugs to custom-molded earplugs with different attenuation filters. The price of custom in-ear hearing protection starts around $200 and really works great—no more ringing ears after a gig. If you take protective steps now, in a few decades you'll still be able to hear normal conversations. Nothing (and I mean nothing) is worse than a deaf audio engineer.

Getting FireWired

Q: Do any Macintosh audio interfaces use FireWire? At 400 Mbps, FireWire should be plenty fast enough to handle multichannel audio. It would be great if something could get eight or more channels of audio in and out of an iMac DV or G3 PowerBook. And what would really be awesome is a standard protocol for transmitting multichannel audio in real time over FireWire. Manufacturers of digital mixers could offer FireWire I/O as an option, and you could just hook up your console to your computer. This seems like such a natural way to go that I can't believe it's not happening—or is it?

David Baker
via e-mail

A: The short answer to your question is not yet, but hopefully soon. FireWire is a high-speed serial-transmission standard that was first developed by Apple Computer in the late 1980s. The company submitted it to the Institute of Electrical and Electronics Engineers (IEEE), which adopted it as international standard 1394. At the moment, IEEE 1394 transmission is limited to cable lengths of 4.5 meters between devices, but an addendum to the specification, called 1394B, will allow cable lengths of up to 100 meters using fiber-optic and CAT5 cable.

The protocols for sending media data over 1394 are defined by the International Electrotechnical Commission (IEC) in a standard called 61883. This standard has several parts: the first five parts govern the transmission of video, and the sixth part (IEC 61883-6) governs the transmission of multichannel

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Normally we don't name competitors in our ads. But in this case, Mix Magazine published the other nominees for the 1999 TEC Award for Outstanding Technical Achievement in Small Format Consoles: Allen & Heath's GS-3000, Digidesign's ProControl, Panasonic's WR-DA7, Spirit's Digital 328 and Yamaha's 01V. Thanks to all who helped us win this prestigious award.

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FIG. 2: KRK plans to develop two FireWire-equipped speaker systems: the E7Di and E8Di. A release date has not been announced.

digital audio and MIDI. This recently adopted 61883-6 standard was originally designed by Yamaha and called mLAN.

At several recent trade shows, notably the January 2000 Macworld Expo in San Francisco, Yamaha demonstrated three prototype mLAN products: an mLAN interface for the Yamaha 02R digital mixer, an interface for the CS6x and S80 keyboard synthesizers, and an mLAN-retrofit box. At the Macworld demo, these products interfaced with a FireWire-equipped Power Mac running a special version of Steinberg's *Cubase VST* sequencer.

Apple has not announced a schedule for adding native mLAN support to the Mac OS. However, several other music-technology companies, including Roland, Korg, and KRK (see Fig. 2), have announced that they will support mLAN. At the recent Summer NAMM show in Nashville, Korg showed a prototype Triton Rack module equipped with mLAN, but the mLAN features were not ready to be shown.

For more information, see "Tech Page: Fire in the Wire" in the July 1996 issue of **EM** and "Tech Page: Digital Harmony" in the July 2000 issue.

Impedance Inquiry

Q: I have a speaker cabinet with four 10-inch speakers rated for 400W @ 4Ω. My combo amp has an external

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For more on the MS2000, including links to reviews, go to www.korg.com/ms2000

KORG

speaker output that reads 300W @ 8Ω. Is it unsafe to plug this amp into the speakers?

Rick
via e-mail

A: I assume this is a bass or guitar rig of some sort. Judging from what you've told me, the setup should work fine. A transistor amplifier rated for 300W at 8Ω will probably output around 400W to 450W into 4Ω. Just make absolutely sure that the amp is capable of driving a 4Ω load. Check the documentation or call the manufacturer.

You may have to disconnect the amplifier's internal speakers to avoid overloading the amp, because an internal 8Ω load wired in parallel with an external 4Ω load will end up as less than 3Ω. You should ask the amplifier's manufacturer about the safety issues. You'll probably be okay with a transistor amp, but tube amps can cook the output tubes if the proper transformer tap isn't selected. From a wattage standpoint, a 450W amplifier should be fine with a 400W speaker cabinet.

Many professional engineers regularly "overpower" speakers with huge amplifiers to gain extra clarity and headroom. But this practice isn't recommended unless you really know what you're doing. Speaker ratings don't apply in all situations, so even amplifiers rated at half a speaker's acceptable wattage can fry the speaker, especially if you send frequencies outside of the driver's frequency cutoff. For this reason, it takes only a few watts of 40 Hz signal to take out a 100W compression driver in a horn. Also, an improperly tuned bass cabinet will let the speaker cone bottom out long before it reaches rated power. The rule is if it sounds bad, turn it down.

Of Mics and Guitars

Q: I need some microphone advice. I'm an accomplished musician, and I'm building a home recording studio centered around an Alesis ADAT LX-20. I am recording a multitrack CD, on which I'm playing numerous guitars, including a Fender Strat and 12- and 6-string acoustics. What microphone should I use to record the soft acoustic guitars, the electric guitars playing through a Peavey amp with a variety of distortion devices, and so on? I'm also looking for a retractable mic boom that can be built in to the ceiling and lowered and adjusted as necessary.

Don Benson
via e-mail

A: Although a lot of microphone choices are quite personal, some old standards have passed the test of time and are used by practically everyone. They're good guidelines to start with. First, for recording electric guitars played through a distorted amplifier, it's hard to beat the venerable Shure SM57. It's my automatic choice for standard guitar-amp combinations such as a Les Paul and a Marshall or a Strat and a Fender Twin. Some stage engineers put up Sennheiser 421s for this application. If you want a slightly brighter sound to cut through the rest of the mix, an Audio-Technica 4060 will work wonders. It's a side-address condenser microphone with a large diaphragm that has become quite popular for guitar recordings with some of the larger live-sound companies.

When miking any guitar amp, remember that each of the speakers will sound a little different, and moving the mic a few inches in any direction will make a world of difference. Typically, you should start with the mic pointed dead-on at the speaker and about one-third of the way in from the edge of the cone (see **Fig. 3**). Then get someone to move the mic around a little while you listen on your monitor speakers until you find the best spot. This method is better than equalization, and it won't muck up the phase linearity of the signal.

For acoustic guitars, the best choices are usually good condenser mics. Many of the new side-address condensers are quite affordable, but if you have a bit more money, it's tough to beat a pair of AKG C-414s. My favorite mics for acoustic guitar are a matched pair of Neumann U64s, tube-based, high-voltage, small-capsule condenser mics that I picked up about ten years ago. If you're looking for lower-priced alternatives, check out the Shure KSM32, the Audio-Technica 4047, and any of the excellent GT mics from Alesis.

Finally, remember that among outboard mic preamps the sound varies quite a bit, mostly due to various loading effects and interaction with the microphone. There is also a wide range of prices for mic preamps, with some units costing many thousands of dollars. On the lower end of the scale, a Tube MP from A.R.T. would be a good starter unit, and it costs less than $150. Personally, I like the preamps from SPL and D.W. Fern, and my absolute favorite is the Millennium Media HV-3. It's quite costly, but worth every penny.

As for a ceiling-mounted boom, I just screwed a standard mic-plate mount to the ceiling of my drum booth. When I need the mount for overhead, I screw on an Atlas Baby Boom. I don't know of any company that makes a boom specifically for your application, but with a little blacksmith work you should be able to bang together something out of Atlas parts. How about sending us a picture if you come up with anything interesting?

FIG. 3: The typical way to mic an electric-guitar amp is to position the mic about 1 to 2 inches in front of the speaker, offset from the center of the cone.

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Mike Sokol *is an audio engineer, computer geek, and rock musician with some 30 years of experience on both sides of the microphone and keyboard. E-mail your "Operation Help" questions to him at emeditorial@intertec.com.*

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YAMAHA

A5000

If at first you don't succeed, build a better mousetrap.

By Geary Yelton

Three years ago, Yamaha reentered the sampler wars, going up against the likes of Akai, E-mu, and Kurzweil with the A3000, the company's first new sampler in almost a decade. A formidable contender, the A3000 offered superior effects and easy user sampling, but it used an entirely new sampler format. In a world already crowded with competitors, the A-series format was poorly supported by third-party sample providers. As a result, A3000 sales suffered. Even though the sampler could import Akai, AIFF, and WAV samples right from the start, turning imported samples into usable programs took some effort.

Instead of throwing in the towel, Yamaha recently introduced two new models: the A4000 and A5000. By including nine CD-ROMs of Samples and Programs in the A-series format, Yamaha successfully took matters into its own hands. Yamaha has also instituted a third-party-sample program. Native CD-ROMs from East West, Sampleheads, Q Up Arts, and other companies are under way.

An extensive feature set and a 32-bit processor make the A5000 Yamaha's most impressive digital instrument ever. With 126 monophonic voices, 32 multitimbral parts, up to 128 MB of waveform memory, and six simultaneous effects blocks, the A5000 is one of the most powerful RAM-based samplers on the planet. (The A4000 is identical to the A5000, except that it has about half as many voices, multitimbral parts, and effects blocks. You can't upgrade an A4000 to an A5000.)

Providing 96 types of effects, 16 types of filters, Loop Divide, Loop Remix functions, and LFOs that can sync to MIDI,

FIG. 1: The Yamaha A5000 merges sampling with extensive synthesis features, a flexible multi-effects processor, and CD-burning capability. The five infinite-rotary encoders on the front panel double as buttons.

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the A5000 doesn't run short on new music-making tools. Its ability to alter a Sample's playback rate and pitch independently in real time is a creative time-saver. By connecting a CD burner to the unit, you can even create your own Sample discs and audio CDs directly from the A5000. In addition to the nine aforementioned Sample CD-ROMs, it also comes with a CD-ROM containing a full-fledged sampler editor for both Mac and Windows.

FIG. 2: The A5000 has two MIDI In ports for reception on up to 32 MIDI channels. The AIEB1 expansion board provides six additional analog outputs along with optical and coaxial S/PDIF I/O.

FIRST GLANCE

The front-panel display is large, bright, and easy to read. Sporting 320-by-80-pixel resolution, it displays plenty of text and graphics, and an LCD contrast control lets you adjust for changing lighting conditions. The knobs and buttons (see **Fig. 1**) mirror those of the A3000; five infinite-spin rotary encoders double as buttons, five mode buttons with green LEDs within them, and six function buttons with red LEDs above them. The names of the pages corresponding to each mode's six functions are silk-screened where the mode and function buttons intersect.

Three buttons lie below the mode and function buttons. The Command/Exit button opens and closes the Command menu, a screenful of instructions executed by turning and pressing the knobs. The Assignable button can perform one of six utility functions, such as turning off all notes. The Audition button plays the currently selected Sample.

A floppy drive is standard, and you can install a SCSI or IDE hard disk. Instructions for replacing the floppy drive with a Zip drive are spelled out in the owner's manual. The 2U case doesn't leave room for an internal CD-ROM or CD-R drive, but you can connect these externally via the standard SCSI port. Playback and track selection for a CD in an external drive are controllable from the A5000's front panel, a wonderful convenience when you're sampling from audio CDs. Unfortunately, the audio connection between the A5000 and a CD-ROM driver comes through the analog inputs.

On the back panel are two sets of MIDI In and Thru ports for 32-channel reception, and one MIDI Out port (see **Fig. 2**). SCSI devices can be connected to a half-pitch 50-pin connector. A pair of assignable, unbalanced ¼-inch audio-output jacks supplements the two main unbalanced ¼-inch outputs. The two ¼-inch balanced inputs on the front panel have gain knobs that run from mic to line level. Finally, a bay on the back of the unit makes room for the optional AIEB1 expansion board ($249.95), which provides six assignable analog outputs and S/PDIF digital I/O on both RCA coaxial and optical ports.

ARCHITECTURE

The A5000's Program hierarchy is identical to the A3000's—which is to say it's very straightforward. Many samplers' voice structures prove difficult to comprehend. If you're accustomed to instruments with Samples organized into keymaps organized into layers organized into Programs, you'll appreciate Yamaha's strategy. Samples are organized into Programs—it's as simple as that. Multisamples or drum kits can be combined into Sample banks and share sound-shaping parameters.

FIG. 3: The programmable StepWave LFO can feature up to 16 steps per cycle. Applied to oscillator pitch, it functions as a 16-step sequencer.

Of course, the Sample remains the basic building block. On most samplers, Samples are nothing more than waveform data. On the A5000, Samples include parameters usually left to Programs, such as looping, key assignment, Velocity range, and

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MIDI-channel assignment. You can also apply filtering, envelope shaping, and LFO modulation to individual Samples. Most samplers let you apply such sound-shaping parameters only to Programs, but the A5000 affords greater control over the details that make sounds more musical.

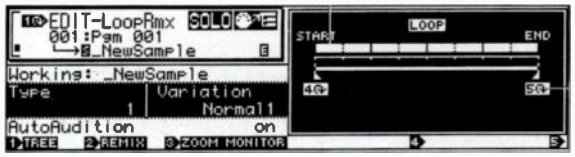

FIG. 4: The Loop Remix feature divides loops into segments and randomly reorders them. You can also save successful remix algorithms for later use.

The next level in the hierarchy is the Program. Samples must be contained in a Program before you play them under MIDI control. Within a Program, any Sample loaded into memory can be enabled or disabled. With up to 960 Samples in RAM, scrolling and selecting the ones you want might take a while. You enable Samples by assigning them to MIDI channels, so any Program can receive data on up to 16 simultaneous MIDI channels.

Sample parameters can be overridden in Programs using the Easy Edit functions. These functions include all the tweaks you'll likely need on the Program level, such as panning, filter cutoff and resonance, key range, Velocity range, portamento, and amplitude envelope. Easy Edits are made separately for each Program and only affect the sample data temporarily while a Program is called up.

You can play Programs one at a time by calling up each on a single MIDI channel. In Multi mode, the A5000 functions like a multitimbral synthesizer that receives Program changes on up to 32 independent MIDI channels.

IF MEMORY SERVES

Because the A5000's memory is volatile, all data must reside on internal or external disk drives, which hold up to 8 GB. Large disks can be divided into partitions. Data is stored on disk as Volumes, which contain Samples, Programs, and sequences. A Volume always contains 128 Programs, regardless of whether they contain any data. Everything in RAM can be saved in a Volume, and loading a Volume completely replaces all the data in RAM. Individual Samples and Programs can be loaded into RAM without affecting the rest of the data in memory.

The A5000's single-track, 16-channel MIDI sequencer can record all 16 channels in a single pass, but it can't record one channel at a time. You can also import Standard MIDI Files.

The unit's SCSI transfers aren't the fastest I've ever seen, but they're acceptable. Loading 60 MB from the internal hard disk takes almost ten minutes, and loading takes even longer from an external SCSI drive. That's still two or three times faster than SCSI transfers on a Yamaha EX5 synthesizer and 25 to 30 percent faster than transfers on a A3000. The A5000 copies data from CD-ROM to hard disk much faster than from CD-ROM to RAM, so it's better to load a Volume from CD-ROM to the hard disk before loading it into RAM.

The A5000 ships with 4 MB of RAM, upgradable to 128 MB via four 72-pin SIMM slots.

SAMPLES IN A HURRY

The A5000's best multisample feature is consecutive sampling. First seen on the A3000, this function automates recording and mapping new Samples as much as possible. It works perfectly with the way I like to sample synthesizers, and it performs equally well when I'm sampling from audio CDs.

The A5000 makes my synthesizer-sampling technique easier than ever. If you follow these instructions, you can sample any synth timbre over any pitch range at three different Velocities. First, create a sequence in an external sequencer that plays either every key or all the white keys at a low Velocity. Be sure to leave some silence between each note. Next, copy the note data, paste it into a new

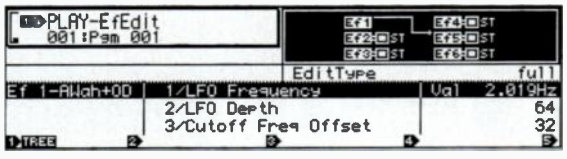

FIG. 5: The Effects Setup is extremely flexible. Here, the output of effect 1 feeds the input of effect 5, while the other four effects function independently.

A5000 Specifications

Maximum Polyphony	126 notes
Multitimbral Parts	32
A/D Conversion	20-bit
D/A Conversion	24-bit
Sampling Frequencies (analog input)	44.1, 22.05, 11.025, 5.5125 kHz; mono and stereo
Sampling Frequencies (digital input with AIEB1 expansion only)	48, 44.1, 32 kHz; stereo only
RAM	4 MB standard, expandable to 128 MB
Analog Inputs	(2) unbalanced ¼" TRS
Analog Outputs	(2) unbalanced ¼" TS (main); (2) unbalanced ¼" TS (assignable); stereo ¼" headphone jack
MIDI I/O	(2) In, (2) Thru, (1) Out
Effects Algorithms	96, including reverb, delay, chorus, flanger, compressor, amp simulator, rotating speaker, Beat Change, Low Resolution, Scratch, Voice Cancel, Auto Synth
Dimensions	2U × 3.56"(D)
Weight	17 lbs., 10 oz.

sequence, and change the Velocity to medium. Repeat the copying and pasting and set the Velocity to high.

Now set up the sampler's record parameters. In the setup display, define the Record Type as New+, which instructs the sampler to record a series of Samples consecutively. Specify the key range and whether you want the Samples mapped to all keys or only the white keys. Set the Map parameter to Auto and indicate the first pitch. Next go to the Trigger page and choose the Edge/Manual trigger type, set the threshold level, and specify the start and stop thresholds. Then go to the Record page, press Go, and start playing the sequence. As the A5000 records, it gives each Sample the same name but a different number, according to its place in the recording sequence.

When the sequence has finished playing, every note has been sampled at a fixed Velocity. The new Samples can be automatically assigned to the currently selected Program or a new Sample bank. Save your work, change the sequence to the next Velocity level, and return to the Setup display to specify the first pitch of the next series of Samples. Pretty cool, huh?

I really enjoy sampling with the A5000. You can set all the recording parameters in advance, so you don't have to concentrate too hard on button pushing when capturing new Samples. A pretrigger function ensures you'll never miss the beginning of a sound you're recording. A click track acts as a guide for recording or for matching the tempo of your source material, and the tempo is remembered as part of the Sample data. Waveforms are drawn on the display as you record.

The A5000 even lets you record Samples directly to disk, bypassing RAM, so you can record and burn CDs. A CD will hold only 650 MB, but the A5000's maximum Sample size is 1 GB (if your hard disk partition can handle it).

OF GREAT IMPORT

If you don't want to do your own sampling, the world is chock full of Sample discs, which the A5000 reads with varying degrees of success. Any Sample data created for the A3000 works with the A5000. It imports Samples in the Akai S1000 and S3000 formats, AIFF files, and WAV files, and it reads Sample data for the E-mu EIIIx, the Roland S-760, the Yamaha SU700, the antiquated Yamaha TX16W, and the Yamaha EX5, EX5R, and EX7 synthesizers.

Imported data includes keymapping and looping if they're part of the original Program. With the exception of E-mu and Roland Samples, the A5000 can read Samples from floppy and hard disks and most CD-ROMs. Oddly, the A5000 can't read CD-ROMs for the

Yamaha EX series unless you first copy the data to a hard disk.

AND THE ENVELOPES, PLEASE

The A5000's multimode filters sound extraordinary; they're so smooth, they mimic analog. The unit has three types of lowpass filters, two types of highpass filters, one bandpass filter, one band-eliminate filter, and two peak filters. These combine into seven dual-filter combinations. For dual filters, the Distance parameter determines the cutoff frequency of one filter relative to the other. The three lowpass and two highpass filters differ in depth of resonance instead of cutoff slope.

As you adjust filter parameters, a graphic representation of filter response is shown. The filter's resonance and cutoff frequency are Velocity-sensitive. When you hit the keyboard hard enough with the resonance cranked up, the filters really bark.

Each filter includes one band of parametric equalization, which can be peak/dip, low shelving, or high shelving. You specify the EQ frequency, gain or cut, and bandwidth. Again, the response is displayed graphically. Having individual EQ for each Sample is quite useful for balancing a series of Samples to get better-sounding multisamples.

Each Sample has three envelope generators and only one LFO. Amplitude, filter, and pitch envelopes are all basic ADSR types. The low-frequency oscillator offers sawtooth, triangle, square, and sample-and-hold waveforms and can control filter cutoff, amplitude modulation, or pitch-modulation depth. The LFO's phase can be inverted to reverse the direction of modulation.

Each Program also has an LFO, which can sync to MIDI Clock and has two waveforms: sine and stepped. The StepWave LFO is a 16-stage envelope generator for which you determine the value of each step (see **Fig. 3**). When applied to pitch, the StepWave LFO is essentially a step sequencer. Many years ago, I modified an analog synth so that I could route its step sequencer to its filter, which you can do with the A5000. This useful function can generate some very cool resonant-filter sounds.

THE DIGITAL DIVIDE

In techno and hip-hop, manipulating loops and breakbeats remains one of the most popular uses for sampling. The Yamaha A-series is probably better

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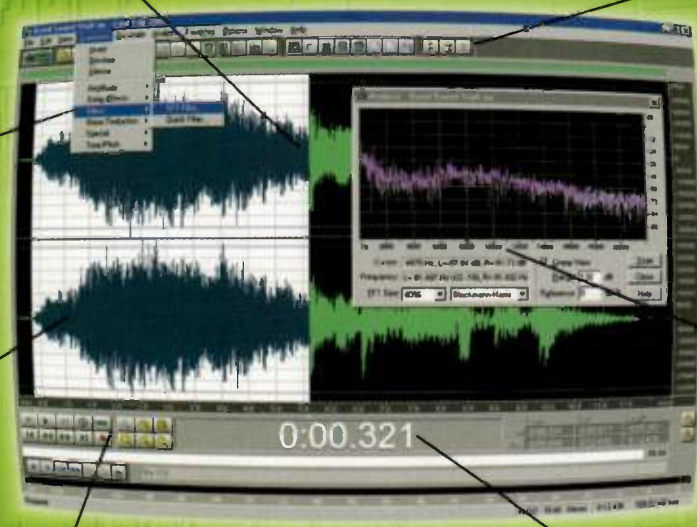

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than any other sampler for this kind of creative work. The Loop Remix and Loop Divide functions generate new material from existing loops. Both techniques work best with a steady beat, and might yield unusable results if any noticeable shuffle is in the source loop.

Loop Divide slices a loop into beat-based segments and automatically maps each segment to a MIDI note. The segments are assigned to a Sample bank, but because they're nothing more than playback pointers, they don't use any extra Sample memory. Like Steinberg's loop editor *ReCycle*, Loop Divide isolates the individual hits in a loop. Triggering the slices in a different order generates new loops.

Loop Remix slices a sampled loop into beat-based segments and randomly rearranges those segments (see **Fig. 4**). You determine whether the whole loop or part of the loop will be affected. Five preset and five user-defined remix types give you some degree of control over the results. You get eight variations per type, including Normal 1 and 2, Reverse 1 and 2, Break, LoFi, Pitch, and Gate. Some of the variations, for example, change the segments' playback speed and order.

Every time you press the Remix button, the segments shuffle into a new variation. Sometimes the result is useless, but when you hear something you like, you can add it to your Sample list. Once you complete the list, the A5000 automatically maps the Samples to different MIDI keys. You can save up to five algorithms and apply them to other sampled loops for similar results.

EFFECTIVE EFFECTS

The A5000's six independent, stereo effects processors offer 96 algorithms and lots of flexibility. Some of the algorithms are single effects; others are combinations. The processors are organized into two blocks of three. You can configure each block in series or in parallel, or two in series can be parallel to the third (see **Fig. 5**). In a parallel configuration, each block can apply totally different effects to different Samples. In a series setup, the output of the first effects block feeds the second, and the second feeds the third to produce really complex, original multi-effects. Thanks to the A5000's 32-bit processor, no bandwidth or processing-speed problems restrict the number of power-hungry effects you can apply simultaneously.

A Program includes up to six distinct effects. Each Sample and Sample bank in the Program can have any or all of the six effects. In Multi mode, the basic MIDI channel's effects settings override those of individual Programs.

The effects can process sounds as the sounds are being recorded; the A5000 can even be used as a stand-alone effects processor. To use the A5000 as a stand-alone effects processor, simply route the input to the effects and route the effects to the stereo output. (Make sure the Monitor function is enabled.) Because the output can be internally routed to the input, you can resample sounds while applying effects to the signal.

As many as 16 user-programmable parameters are available for each effect. When editing effects, you can view all of an effect's editable parameters or only your four favorite ones. This is very convenient because you don't have to wade through an extensive list of possibilities to adjust the most basic parameters. Up to four effects parameters per Program are

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- Analog and digital I/O expansion modules, including the new 24-bit analog and ADAT I/O modules for the RFX-32 Card
- The rock-solid Emulator Operating System that is regularly upgraded with powerful new features
- Legendary Z-Plane filters that make your sounds scream
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controllable in real time via external MIDI Control Change messages.

The A5000 definitely has a wider range of effects processing than any other sampler to date. All the standards are here, including reverb, delay, chorus, flanger, compressor, amp simulator, and rotating speaker. The A5000 also offers unconventional effects such as Beat Change, Low Resolution, Scratch, Voice Cancel, and Auto Synth. Many of these are downright bizarre and sure to be a hit with cutting-edge sound designers. All time-based effects can sync to MIDI Clock, so modulating LFOs and delays can play in time with the tempo of sequenced music.

An effect called Total EQ is applied to the stereo outputs and headphone jack. This 4-band equalizer lets you set frequency and gain for four bands and width for three of those bands. Total EQ has no effect on the assignable outputs or the optional digital outputs.

SOUND SUPPORT

The Samples and Programs on the CD-ROMs that ship with the A5000 are pretty impressively diverse, and predominantly of high quality. To use all the Volumes provided, you'll need at least 64 MB of RAM.

I especially like the two discs of orchestral instruments, *Strings/Choir* and *Brass/Wind Instruments,* which also include tuned percussion. Both discs contain various ensembles and solo instruments. Another of my favorites is *Real Drums,* which has five complete kits ranging in size from 8 MB to 32 MB, as well as loops and individual drums and cymbals. *World/Latin Instruments* offers a varied collection of loops, percussion, and acoustic instruments from Latin America, Arabia, Indonesia, China, and Japan.

Piano/Keyboards contains 4, 5, and 6 MB Yamaha grand piano selections. Among the keyboards represented are Rhodes and Wurlitzer pianos, a clavinet, FM synths, analog synths, a 12 MB harpsichord, various Hammond B-3s, and a 17 MB pipe organ. *Guitar/Bass* is the most disappointing disc; some of the basses sound decent, but the guitars suffer from short Samples and not enough variety.

Two discs are oriented toward modern dance music. *DJ/Producer Tool Kit* is full of synth basses and pads, edgy drum kits, and techno effects. Because the A5000 is so adept at dealing with loops, you'll find more than two dozen loops at three tempos on the *Syntraxx/Loops* disc, along with some great synth timbres. Most of the *Syntraxx/Loops* sounds make good use of the A5000's programming features.

EDITORIAL ASSISTANTS

Yamaha also includes a CD-ROM with Sample editors for both Mac and Windows. *TWE,* an audio editor that's been around for years, handles basic digital and audio-editing duties for Yamaha A-series samplers and EX-series synthesizers. The A5000 doesn't offer cut, copy, and paste operations, so turn to *TWE* if needed. TWE communicates with the sampler via SCSI. This can be problematic, because SCSI doesn't like two controllers on the same bus.

More specific to the A5000 is *A5000 Editor.* It displays all the A5000's parameters on your computer screen and controls the sampler remotely via MIDI. The Windows version is actually a plug-in for a Yamaha editing program called *XGWorks lite.* The Mac version requires OMS, a serious problem if you've upgraded to Mac OS 9.

SYNTH OR SAMPLER?

A sampler enables you to create a timbral palette that is yours alone. Unlike a ROM-based synthesizer, it lets you specify the raw Samples. Rather than selecting from a synthesizer's fixed set of waveforms, you choose the basic building blocks to load into the sampler's memory. With an enormous variety of unique effects and filter types, combined with the ability to sculpt sound down to the individual Sample level, the A5000 provides an extensive sound manipulation toolbox.

Three years ago, Yamaha introduced a wealth of great sampling ideas with its A3000. Despite the exciting new technology, it never caught on, mostly because it lacked a comprehensive Sample library. The A5000 improves on the A3000's technology and comes bundled with two computer-based editors and a library of CD-ROMs. Unless your life savings is tied up in a competitor, you have no excuse for not checking out the A5000.

Geary Yelton *lives in Charlotte, North Carolina, where he struggles to find time to spend in his personal project studio.*

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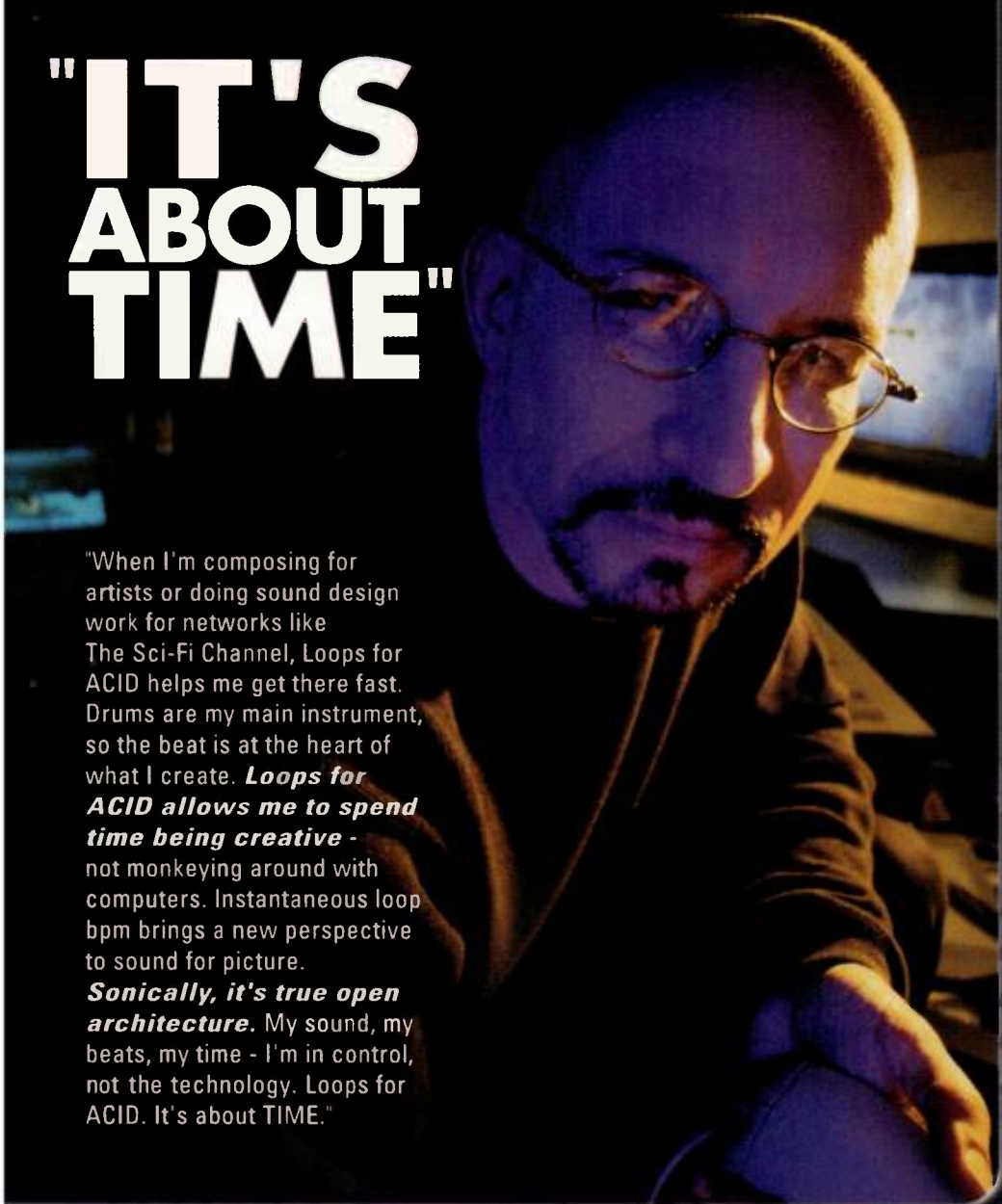

"When I'm composing for artists or doing sound design work for networks like The Sci-Fi Channel, Loops for ACID helps me get there fast. Drums are my main instrument, so the beat is at the heart of what I create. ***Loops for ACID allows me to spend time being creative -*** not monkeying around with computers. Instantaneous loop bpm brings a new perspective to sound for picture. ***Sonically, it's true open architecture.*** My sound, my beats, my time - I'm in control, not the technology. Loops for ACID. It's about TIME."

Willie Wilcox artist / producer / sound designer / composer
Utopia, Meat Loaf, Natalie Cole, Pointer Sisters, Sci-Fi Channel, USA Networks, MTV

- Ambient Atmospheres & Rhythms: *Robert Rich*
- Ambient Grooves: *Bill Laswell's Sample Material*
- Ambient Realms
- Blues Guitar: *Whiskey, Cigarettes & Gumbo*
- Classic Country
- Classic Drum Machines: *Syntonic Generator*
- Dark Ambient Soundscapes: *Pandora's Toolbox*
- Drum Components: *RADS*
- Drum Tools
- Electro Hip-Hop: *Mac Money*
- Electronica Grooves: *Cyclotronic Resonator*
- Essential Sounds II
- Ethnicity
- Futurist Drum 'n' Bass
- Hip-Hop/R&B Vocals: *Mac Money*
- Industrial Toolkit: *Methods of Mayhem*
- Jazz Solos and Sections
- New York Dance
- On the Jazz Tip
- Psychedelic Guitar: *Harvey Mandel*
- R&B Drums: *Groove Spectrum*
- Sounds of Asia: *Opium*
- Street Beats
- Techno Club Grooves I & II
- Techno Synth Loops I & II
- Textures and Soundscapes: *Rapoon*
- Twisted Reality: *Synthetic Sound Effects*
- Universal Groove Elements
- Voices of Native America
- World Percussion: *Marc Anderson*

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MARK OF THE UNICORN

DIGITAL PERFORMER 2.7 (MAC)

A major upgrade to a mature and elegant sequencer.

By Rob Shirak

Mark of the Unicorn is one of the oldest and most venerable music-software companies in the world. One of its first products, *Performer,* was among the first comprehensive MIDI sequencers for the Mac. Since *Performer*'s release, MOTU has continued to refine and enhance its flagship software, adding digital audio capabilities and many new features to create *Digital Performer.*

Version 2.7 of *Digital Performer* adds a multitude of features to an already powerful application. Updates include greatly enhanced automation, a new Drum Editor, multiple audio punches on the fly, real-time display of waveforms during recording, audio scrubbing during trimming, graphic time stretching, new and redesigned plug-ins, Mackie HUI and ReWire support, and an abundance of other features.

Digital Performer has always maintained a no-nonsense approach to sequencing and recording that appeals to many professionals. At the heart of *Digital Performer* lies the Tracks Overview window (see **Fig. 1**), in which you find the tracks, MIDI assignments, default-patch assignments, and related parameters.

One very cool feature of the Tracks Overview window is the Take box, which lets you record (and name) multiple passes within a MIDI or audio track. This works well for quickly recording repeated takes of a part, as it doesn't clutter the arrangement with redundant vertical tracks. Version 2.7 automatically creates new Takes for audio tracks while in Cycle Record, and the program displays audio waveforms in real time as tracks record.

Like most other digital audio sequencers, *Digital Performer* provides several editing windows. The Event Editor window displays MIDI data in an editable text list; the Graphic Editors provide visual editing of MIDI and audio data. The Audio Graphic Editor is as feature laden as the best stand-alone digital audio editor programs (see **Fig. 2**), and it can zoom all the way down to the sample level for incredibly detailed editing.

Digital Performer's dedicated Conductor track contains all tempo, key-signature, and time-signature data; you can easily view and edit these graphically or by event. The Conductor track can also be saved as a Standard MIDI File for importing into Digidesign's *Pro Tools.*

When the SMF is accompanied by an Open Media Format file and relevant audio files, *Pro Tools* can create a session from a *Digital Performer* project that contains sample-accurate alignment of audio events, a tempo map with bars and beats, text markers, and MIDI data. (MOTU plans to create a two-way handshake in a version of *Digital Performer* that can import OMF files.) I used this method with great success for several projects destined to be mixed in *Pro Tools.*

TURN THE BEAT AROUND

One feature missing in previous versions of *Digital Performer* was a dedicated editor for drum parts. Not anymore. The program's Drum Editor lets you view and edit notes within a rhythmic grid (see **Fig. 3**). Furthermore, you can create composite drum kits from multiple MIDI devices. The viewing and editing resolution ranges from whole notes to 64th notes. Triplets and dotted rhythms are not displayed in the grid, but, of course, they can still be present in the sequence.

Drum data is displayed and edited in four different panes: Note List, Note Grid, Note/Tool Settings, and Controller Grid. The Note List includes each drum sound and its name, note number, MIDI-channel assignment, and playback control. To the right of the Note List is the Note Grid, where you view and edit note placement and Velocity. A cool new feature is the Rhythm Brush, which lets you "paint" entire preset rhythmic patterns by dragging the Brush icon across the grid.

Under the Note List is the Note Tool Settings pane. Here, you can quantize a drum sound independently of the other sounds.

Below the Note Grid is the Controller Grid, similar to the Controller Grid in *Digital Performer*'s MIDI Graphic Editor window. Here Velocity, Volume, Modulation, and other controller data can be edited graphically.

I found the Drum Editor to be very useful, especially for styles of music incorporating grid-style, quantized percussion parts, such as common dance

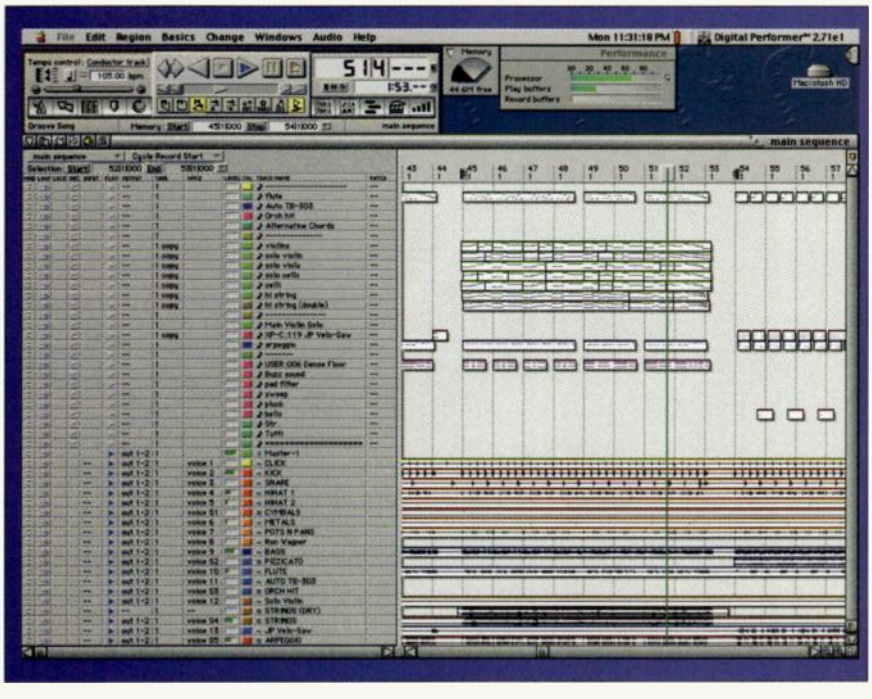

FIG. 1: The Tracks Overview window graphically displays MIDI content and audio-track waveforms. Tracks can contain multiple takes of alternative performances or duplicated data for editing.

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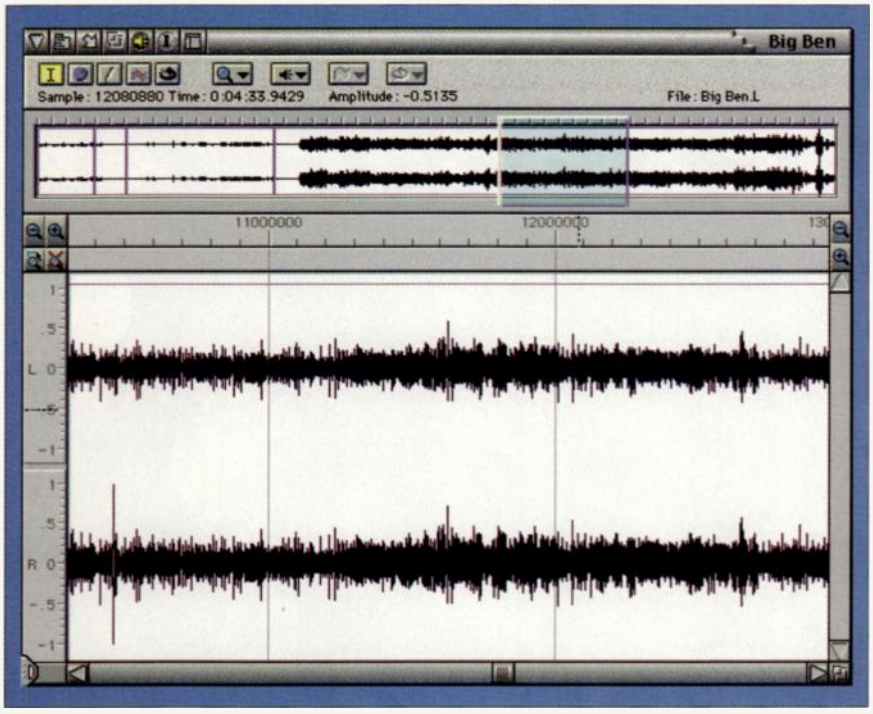

FIG. 2: The Audio Graphic Editor is as comprehensive as a stand-alone audio-editing program. You can view and edit waveforms all the way down to the individual sample.

music. Each pane allows extensive editing and displays a multitude of features that can be described only as comprehensive.

However, the overall look of the Drum Editor doesn't fit in with the clean, elegant graphics found elsewhere in *Digital Performer*. It's a little blurry and hard on the eyes, with too much dark color. I hope MOTU isn't moving *Digital Performer* in the sexy form-over-function graphic direction that has recently marred many music-related applications.

TIMING IS EVERYTHING

In version 2.61, MOTU upped *Digital Performer*'s internal MIDI resolution from 480 ppqn (parts per quarter note) to around 2 trillion ppqn, although you can specify any resolution up to 10,000 ppqn. Users comfortable with the standard 480 ppqn can simply add four decimal places for higher resolution.

Of course, Standard MIDI Files can't support such high resolutions, but *Digital Performer* creates minute changes in the tempo map as necessary when exporting Conductor tracks as SMFs to ensure accurate timing of audio events throughout an entire session.

As you might expect, *Digital Performer* supports MOTU's new hardware-based MIDI Time Stamping (MTS), which is accurate to within an amazing 0.33 ms when used with a USB-equipped MOTU MIDI interface. (It doesn't work with any interface connected to a serial port.)

Hearing is believing. I immediately noticed the timing improvement the first time I used a MIDI Time Piece AV/USB with my USB-equipped G3 PowerBook to play a dense MIDI sequence that I had originally created on my non-USB G3 desktop computer. When I use MTS, I spend less time realigning MIDI tracks to achieve the proper timing placements, which ultimately results in more productivity, a requirement of professional work.

Groove-quantized tracks actually hold their groove, even as tracks accumulate. Of course, MTS won't make any old, sloppy synth modules any less sluggish and inconsistent, but you can rest assured that any MIDI slop you hear comes directly from the synths and/or the serial nature of MIDI—not from the interface.

SOUNDBITES

Digital Performer has always had very good native DSP capabilities for audio data. The Spectral Effects processor provides high-quality pitch-shifting, formant-shifting, and time-compression/expansion functions that rival those of dedicated, stand-alone applications. All three effects can process simultaneously, and the results are immediately reversible if you don't like them.

Tracks can also be normalized, cross-faded (with user-specified times and shapes), stripped of silence, and faded. The dither feature affects all audio tracks. Soundbites (individual sections of audio) can now merge, and editing them during playback interrupts playback only

FIG. 3: The Drum Editor window allows extensive editing of percussion notes along a rhythmic grid.

For manufacturer information, please see Contact Sheet, p. 221.

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slightly. (In previous versions, Soundbites weren't editable during playback.) Tracks using a lot of real-time DSP can be bounced to disk as a new file to free up CPU resources. You can bounce entire mixes, complete with internal effects and automation, to a new file in less than real time on most computers.

Digital Performer cannot mix 16-bit and 24-bit files in the same song. If you need to import 24-bit audio into a 16-bit file, *Digital Performer* converts it down with a high-quality dithering algorithm. (When converting from 16 bits to 24, the software simply adds zeroes to the least-significant end of each digital word.) Again, *Digital Performer* handles these chores while you perform other editing and recording tasks.

Audio files are now time stamped, enabling the program to display their current SMPTE and real-time locations. All Soundbites can return to their original time-stamped locations, which is especially handy if you've accidentally moved a piece of audio without a grid enabled. Soundbites can also have sync points, places within the Soundbites that act as location reference points.

Version 2.7 boasts the very cool ability to graphically time-stretch Soundbites. The program automatically performs the time compression or expansion to fit the audio to the new start and end points. Simply grab the edge of a Soundbite with the hand tool and drag it to the desired location. *Digital Performer* does the rest.

▼ *Digital Performer* boasts the very cool ability to graphically time-stretch Soundbites.

When you're recording audio in Cycle mode (looping a section between Memory Cycle points) with the Overdub button engaged, *Digital Performer* automatically creates and numbers multiple takes with each pass until you stop recording. The *Polar* plug-in, which was introduced in version 2.6 (see **Fig. 4**), allows not only cycle recording of multiple takes, but also overdubbing into previous passes mixed in real time. *Polar* records its takes into RAM for faster, more glitch-free recording. You can place audio tracks created in *Polar* in *Digital Performer*'s Tracks Overview window or save them as a separate file for later use.

In the Sampler window, Soundbites can be exported via SCSI to external hardware samplers, such as the Roland S-760 or Akai S-series units. This is a very fast and convenient way to load WAV, AIFF, SDII, or audio samples from a CD into a hardware sampler.

MAS CONSTRUCTION

In addition to the destructive audio DSP functions I mentioned previously, *Digital Performer* ships with 27 nondestructive real-time audio plug-ins in the MOTU Audio System (MAS) format. That number actually includes 19 different types of plug-ins;

the other eight are variations. As yet, neither the plug-ins nor *Digital Performer* itself are optimized for Apple's G4 Velocity Engine, but according to MOTU, this enhancement lurks just around the corner.

For version 2.7, MOTU has added new plug-ins and redesigned several of the preexisting ones for better audio quality. All sport a beautiful new graphic face-lift. Some of the more elaborate plug-ins with lots of parameters have alternative layouts that dispense with the cool graphics and display only parameters with buttons, sliders, and numbers. However, what's most important is the sound quality—and these plug-ins deliver.

Of particular note are the *Sonic Modulator, Multimode Filter, Stereo Delay, Preamp-1, Phaser, Ring Modulator,* and *MasterWorks* mastering plug-ins. These processors are as good as—or better than—a lot of third-party offerings by major industry players, and I wouldn't hesitate to use any on professional projects. The reverbs, dynamics processors, and EQs are quite good, though they're useful rather than brilliant.

▼ **You can paint complete rhythmic patterns with the Rhythm Brush feature.**

Digital Performer now automates its plug-in parameters in real time. This can result in some incredibly wonderful tracks, which you can easily bounce to a new audio file if your CPU rebels against your natural desire to apply a Sonic Modulator to every track.

MOTU has really covered the nuts and bolts of effects. None of the plug-ins are too esoteric, but that's fine—all are extremely useful and high quality. If you need more (and who doesn't?), a wealth of third-party MAS plug-ins are compatible with *Digital Performer.*

MIX BLESSING

Digital Performer's mixer section is one of my favorite parts of the program. The logically organized and familiar-looking Mixer window graphically models a hardware console (see Fig. 5). It integrates MIDI and audio channels into the same mixer, further facilitating the recording process. (Any feature that blurs the line between MIDI and audio manipulation is good in my book.)

A drop-down menu and side window let you specify the number of visible channels and parameters, and you can resize the entire mixer to a smaller "narrow view," which still offers the functionality and parameter access of the normal-size mixer.

Each channel provides five Inserts for plug-in effects. Audio channels can accommodate DSP plug-ins, and MIDI channels can access special MIDI plug-ins, such as *Quantize, Arpeggiate, Transpose, Remove Duplicates, Time Shift,* and so on. These Mixer-window MIDI plug-ins are non destructive, and the original performance is restored when you disengage them.

In the Tracks Overview window, the same MIDI-editing functions are destructive if you don't undo them immediately. (The only ways to preserve

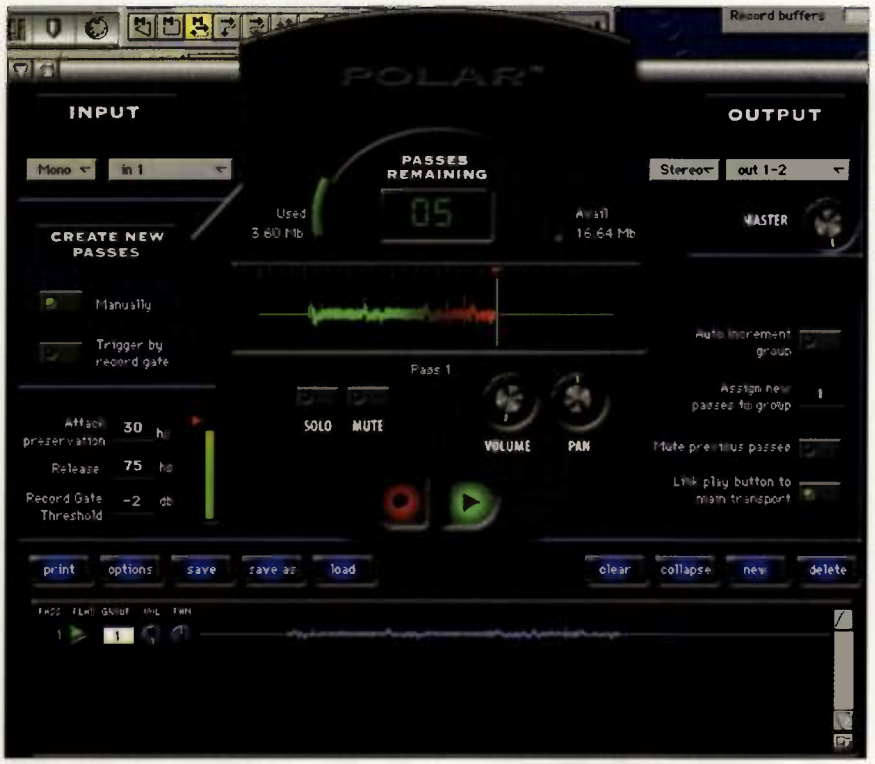

FIG. 4: The *Polar* plug-in's intuitive audio-recorder window is ideal for quickly recording multiple passes. You can create new takes on each pass or merge takes in real time, with the results easily imported into *Digital Performer*.

an original MIDI performance are to first duplicate the original take in the Tracks Overview window or use a MIDI plug-in within the Mixer.)

Option-clicking on an Insert (either audio or MIDI) lets you bypass the selected effect without having to open the plug-in's window. If you need more than five Inserts for a channel, just bus the signal to an Aux channel for further processing.

You can create up to 32 stereo buses, although only four sends are available per channel. Each send has its own Mute button. Setting a send to unity gain, a pan pot to center, or an audio fader to 0 dB is a simple matter of double-clicking on the desired item in the Mixer.

Each channel has dedicated Record, Solo, and Mute buttons, and the Master fader has a Mono summing button. A dedicated section of the channel strip is for controlling automation. Some sequencers embed the automation controls in the normal recording and playback functions but *Digital Performer*'s setting is much more appealing (more in a moment).

With both MIDI and audio channels, level meters appear alongside the faders, and appropriately spaced reference numbers indicate MIDI activity, Velocity levels, or decibels. In addition to a red Over indicator for audio tracks, *Digital Performer* now displays separate left and right meters for stereo tracks.

The elegant knobs, faders, and meters are just about as intuitive and easy to use as software controls can be. But if you aren't keen on tweaking a software mixer with your mouse, version 2.7 supports hardware interfaces and ships with a profile for the Mackie HUI control surface.

THE FADERS ARE FLYING

Digital Performer's automation section is awesome, reminiscent of the automation controls of high-end hardware consoles. Its greatly enhanced feature set is more extensive than version 2.6's, but it's still very easy to use.

The new Automation Setup window allows both global and track-specific automation setup and editing from one location. You can enable and disable audio and MIDI automation globally or by parameter type, and those settings can affect all tracks or individual ones. In addition, the same automation controls are accessible in the Mixer and Audio Graphic Editor windows, each of which has a button that changes color to indicate if automation is disabled, enabled, or recording.

Automation of individual plug-in parameters is available (but not for some third-party plug-ins). The Automation Setup window lets you choose the parameters you want to automate; it doesn't tax the computer's resources unnecessarily by activating all parameters.

Digital Performer has automation modes: Overwrite, Latch, Touch, Trim Touch, and Trim Latch. Overwrite starts writing automation data when playback begins. Latch starts as soon as you click a parameter control (fader, knob, and so forth). Both modes continue writing until playback stops or you choose another automation mode. Touch records data from the moment you first click on a

FIG. 5: The Mixer window is laid out like a hardware console. Simple to use, it displays both MIDI and audio tracks with faders, meters, sends, and Inserts for plug-ins.

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parameter control—until you let go of the fader or knob—at which point the parameter value reverts to the existing mix setting according to a user-definable ramp time.

The two Trim modes let you scale the automation moves you've already created; for example, Trim is ideal for adding that extra decibel of level you need to put the finishing touch on an otherwise perfectly automated track.

The audio processing when you're using automation is superb, completely devoid of artifacts such as zippering. Practically every control can be automated and easily edited. When coupled with a dedicated hardware control interface with real faders and knobs (such as the Mackie HUI, Radikal Technologies SAC-2K, or CM Automation MotorMix), *Digital Performer*'s automation capabilities rival the best automated hardware consoles. It certainly handles automation as well as or better than any software DAW on the market.

WISH LIST

In spite of the many things *Digital Performer* 2.7 does really well, it still lacks a few features. The majority of my complaints stem from slightly inconvenient editing methods that can frustrate power users. Most of the problems have work-arounds, however, and MOTU says it is working to address the shortcomings in upcoming versions.

Currently, the Tracks Overview window can only expand horizontally; I wish I could to expand it vertically and zoom in on specific tracks for a more detailed view of track contents. I'd also like the Tracks Overview window to provide a playback-quantize feature, or better yet, the ability to undo MIDI quantization back to the original performance. (You can use the *Quantize* plug-in within the Mixer for nondestructive quantization.)

The Show Marker Grid Lines feature should be available in the Tracks Overview, as it is in the Audio Editor. Then you could see specifically where a designated hit or musical transition happened in relation to the MIDI and/or audio. The enhancement would be especially helpful with film or commercial spots using a lot of Markers.

Many *Digital Performer* operations possess no keyboard equivalents, so some actions require several mouse clicks and/or wading through multiple menu layers. This approach is unproductive and time-consuming, and I eagerly await the promised addition of customizable keyboard commands.

When you're performing a punch-in, the user-selectable preroll measures provide only clicks; playback doesn't begin until you reach the punch-in point. The recorded material should play during the preroll measures, because punch-in without musical context is very difficult.

At the very least, you should be able to choose clicks or playback during the preroll. In all fairness, I must point out that you can work around this inconvenience by setting a punch-in point, locating to any point before it, and hitting the Record button. The material plays until you reach the punch-in point, at which time Record mode engages on the selected track.

And now for what is perhaps the biggest hole in *Digital Performer*'s feature set. For individual Soundbites, the program needs initial volume levels that are independent of MIDI Volume and automation. This feature would let you even out the levels between Soundbites before applying any mix-related volume moves to a track.

For instance, the program gives you only two ways to smooth out uneven

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Digital Performer 2.7 (Mac)
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1 2 3 4 5

PROS: Allows multiple punch-ins on the fly. Provides sample-accurate resolution of audio. Excellent native plug-ins. Extensive automation and mixer capabilities. Easily exports projects to *Pro Tools*.

CONS: Needs more keyboard shortcut commands. Some editing operations aren't intuitive. Drum Editor grid doesn't display triplets or dotted rhythms. Can't combine 16- and 24-bit audio files in the same song.

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notes in a phrase or create a level-matched composite of two performances recorded at different levels. You can destructively alter the levels of various Soundbites through a DSP process, or you can create a bunch of small MIDI Volume moves.

The latter technique makes the Mixer faders jump all over the place before you even begin mixing. Furthermore, if you copy and paste the same group of edited Soundbites into a different location, you must take extra care to copy the mix automation along with the audio; otherwise you'll have to establish the levels all over again.

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As a digital audio sequencer, *Digital Performer* 2.7 is hard to beat. It records and processes audio as well as or better than its competitors, and the mixer is top-notch. The automation is comprehensive and well designed, the audio plug-ins are professional quality, and the enhanced timing resolution of MTS is remarkable.

It must also be noted that *Digital Performer* boasts sample-accurate synchronization when used with its pro-level hardware and a USB-equipped computer. The MOTU system—with *Digital Performer* at its heart—is the best alternative to *Pro Tools*, especially if your recording technique demands more than a rudimentary level of MIDI.

The excellent features greatly outweigh the minor inconveniences. My wish list is inspired primarily by features I've found in other programs; naturally, I want them in *Digital Performer*, too. To MOTU's credit, the company has been very open to feedback and suggestions, and it is committed to incorporating the features that users want in future updates.

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JOEMEEK

PRO CHANNEL VC3Q

A versatile, dirt-simple voice processor for all occasions.

By John Ferenzik

The most critical aspect of the recording process is arguably the audio-signal path from the source to the final destination. The better the components composing that path, the better the final results.

One such pathway is Joemeek's Pro Channel VC3Q. This diminutive voice processor consists of a microphone preamp, an optical compressor, and three bands of fixed EQ, all tidily ensconced in a half-rack box. In size and appearance it is strikingly similar to the Joemeek C2 stereo compressor (reviewed in the September 1999 issue of **EM**).

The Pro Channel VC3Q arrived at my door for review just as I was finishing the MIDI sequences for three new songs. It was good timing, as the next item on my agenda was track-by-track overdubbing of vocals and a variety of instruments, and a voice processor was a logical candidate for the job. It would prove to be a project for which the Pro Channel was a natural.

UP FRONT

One glance at the Pro Channel VC3Q immediately tells you its lineage: the faceplate (see **Fig. 1**) is the telltale green color of all things Joemeek. A red LED on the left indicates phantom power on/off status. Next is the Input Gain control for the mic preamp or line input. Then comes the photo-optical compressor, with Compression, Attack, and Release knobs and a small red Compressor On button. Beside the compressor is the EQ section (or "Meequalizer," in Joemeek-speak), with Low, Mid, and High frequency knobs. As with the compressor section, a red button toggles the EQ on and off. On the far right of the faceplate, the Output Volume control regulates the VC3Q's output.

In the compressor section, Joemeek doesn't use the standard ratings; instead the Compression knob's settings are labeled 1 to 11. Attack ranges from 1 to 11 milliseconds and Release from 125 milliseconds to 1.5 seconds. Each EQ pot is variable from –16 to +16 dB, with the 0 dB setting at 12 o'clock. Both the High and Low controls are shelving EQ, while the Mid control has a Q value of one. A yellow LED lets you know when the EQ is enabled, and a red LED at the output of the EQ section flashes when the signal comes within 6 dB of the overload point.

A small 5-segment LED VU meter on the lower right indicates input level, with the lowest-level green LED doing double duty as a power-on indicator. (Note that the Output Volume control comes in after the input level has been established, so it does not affect the VC3Q's overload margin.)

'ROUND BACK

On the rear panel (see **Fig. 2**) you'll find an XLR input for the mic preamp, a ¼-inch instrument/line input, a ¼-inch insert point, a Mix In (a ¼-inch line level input routed directly to the output stage), and two balanced ¼-inch outputs. Plugging into the instrument/line input disables the XLR input. The unit has no on/off switch; it powers up via the input jack for the 12V wall-wart transformer provided with the unit.

A small red button turns the phantom power on and off, but unfortunately its placement next to the I/O connectors is potentially hazardous. For example, while setting up the VC3Q for the first time, I accidentally depressed this button while checking the mic-cable connection and was greeted with a thunderous *thunk* from the studio monitors.

The manual warns of this speaker-threatening situation and outlines a procedure for avoiding problems: first set the Input Gain to minimum, then make sure the mic is plugged in before switching on phantom power. But I wish the warning had been written in bold type. Also, although the VC3Q was clearly designed for use as a standalone component, this rear location of the phantom-power switch is discouraging to anyone needing to rack-mount the unit.

An outboard effect can be patched into the insert point using a ¼-inch TRS jack. The tip is send and the ring is return.

The Mix In provides a line-level "back door" to the VC3Q's signal chain, but it falls *after* the compressor and equalizer, and the signal is therefore not processed. Also, Mix In has no separate input-gain control; you must establish the optimum signal level prior to sending it through this input.

The two outputs on the back panel are wired in parallel, and you can make either output unbalanced by substituting a ¼-inch TS plug for a ¼-inch TRS plug. Having two outputs came in handy on more than one occasion. For example, while recording directly to an ADAT, I used the extra output to bypass my mixer and monitor directly from the VC3Q. Similarly, while recording track-by-track, I saved time repatching between takes by simply plugging both outputs into adjacent tracks on the ADAT.

BEATING A PATH

With its wide dynamic range and variety of complex timbres, the drum kit is

FIG. 1: The Pro Channel VC3Q combines a mic preamp, a photo-optical compressor, and 3-band EQ in a half-rack unit.

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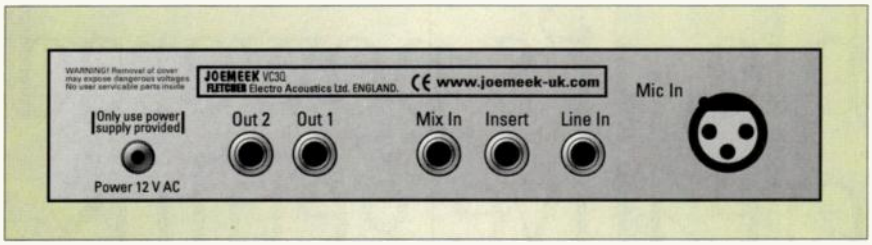

FIG. 2: Despite its diminutive size, the Pro Channel VC3Q provides dual outputs, a handy Insert, and more.

always a challenge to record, and it was in this arena that the VC3Q received its first workout. I had already dumped reference versions of my three new tunes to eight tracks of ADAT at 20-bit resolution, but I desperately wanted to replace the lifeless drum samples, track-by-track, with the real thing. Enter the VC3Q.

Unlike a brickwall VCA limiter, the VC3Q's photo-optical compressor can tame transients on all drums while still retaining lots of exciting dynamic detail. However, I had to experiment a bit before I found the best settings, as the compressor occasionally let transient spikes slip by, even when compression was set to 11.

After considerable time and effort, I ultimately contained some nasty snare-drum and hi-hat spiking. The hi-hat and crash cymbals also taxed my patience before I arrived at the right attack and release settings.

Yet patience has its rewards. I eventually culled some spectacular results. The VC3Q excelled at bringing out the cymbal wash that follows the initial stick attack of the ride. It fattened up an anemic hi-hat track, and it had the final word with troublesome cymbal crashes, producing a fuller sound with a longer, more natural decay.

VC3Q's EQ offers more immediate gratification. For a fixed EQ, it's particularly well voiced for drums. The low-end shelving EQ lent a hearty 80 Hz to the bass drum, rides and hi-hat overdubs benefited from a little 8.5 kHz high-EQ shimmer, and I spiced up a floor-tom overdub with some 1.5 kHz midrange *thwack*. I was also able to quickly point up one particularly dull-sounding snare with a modest addition of mids and highs.

BASSING OFF

For me, getting a bass track to sit in the mix from the get-go is right up there in importance with mother, flag, and apple pie. After I finished tracking the drums, I auditioned a variety of electric basses (passive and active, fretted and fretless) using the VC3Q as the "front end."

With basses, the VC3Q performed splendidly. Whether I plugged the instruments into the line input or (using a direct box) the mic preamp, I had lots of undistorted level while overdubbing to ADAT. The compressor gave me access to a wide palette of colors, ranging from mildly compressed (transparent) to heavily compressed (obvious), and I didn't spend an inordinate amount of time wandering the streets of Tweaksville. If you're in the market for a bass-recording solution, look no further.

Throughout all the bass overdub sessions, I found little need to add EQ because in most cases the compressor had already delivered the goods. On one song, however, I did have a little trouble adding midrange to hide a problem with one of the basses. The VC3Q's 1.8 kHz midrange point was a tad north of where the bass needed an 800 Hz boost, but I remedied the situation quickly by patching an outboard parametric EQ into the insert—a nice routing option.

STRINGING ALONG

Next I auditioned the acoustic guitars, and the VC3Q treated all comers with kindly respect. All three of my songs demanded both bright steel-stringed and darker nylon-stringed instruments, so I used a variety of microphones (dynamic mics and both small- and large-diaphragm condensers) to capture their respective sounds. When I ran an Audio-Technica AT4033 condenser through the VC3Q's compressor for one exceptionally bright-sounding steel-stringed Ovation, the guitar simply bloomed. For all the acoustic guitars, the compressor proved adept at wedding quiet finger picking to loud strumming.

When I auditioned the electric guitars, I first went for the squeaky-clean DI tone by plugging a series of low- and high-output axes into the VC3Q's instrument/line input. With one exceptionally low-output 12-string Stratocaster I was able to crank the input gain to get a respectable level. By applying massive compression with a quick attack and moderate release, I easily sauntered into Roger McGuinn

VC3Q Specifications

Audio Inputs	(1) balanced ¼" TRS (line); (1) balanced XLR (mic); (1) ¼" TRS insert; (1) ¼" Mix In
Audio Outputs	(2) balanced ¼" balanced TRS (line level)
Compressor type	photo-optical
Attack Time	1–11 ms
Release Time	125 ms–1.5 sec
Equalizer type	fixed
Low	80 Hz, shelving
Mid	1.8 kHz (Q=1)
High	8 kHz, shelving
Meter	LED (combined input level and gain reduction)
Maximum Input Level	+30 dB
Maximum Output Level	+26 dB
Frequency Response	10 Hz–50 kHz (+0, –0.5 dB)
Distortion	0.01%
Noise	–100 dB (line); –125.5 dB (mic)
Power Supply	12 VAC adapter; phantom power
Dimensions	9.5"(W) × 1.75"(H) × 5"(D)
Weight	2.5 lbs.

For manufacturer information, please see Contact Sheet, p. 221.


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territory with a nicely flattened (but not flat) sound—*exactly* what the tune needed.

Finally, I whipped out a trusty Shure SM57 and aimed it at the grill of a cranked amp for some true electric guitar *sturm und drang*. Any initial anxiety I may have harbored dissolved as the VC3Q compressed rhythm and lead guitars with aplomb. I found the unit equally adept at going over the top to coax some gnarly pumping and breathing from chunky staccato rhythm parts, and dropping back to gently smooth out the rough edges on a solo.

On the other hand, using the EQ was a hit-or-miss proposition. Whereas some of the guitars I recorded benefited from a little low- or high-end goosing, the fixed-shelf low and high EQ controls were usually too broadband to be used subtly. And when it came to dealing with what is arguably the guitar's Achilles heel—abundant midrange—I longed for the flexibility and precision offered by parametric EQ. However, in the VC3Q's defense, I wasn't stranded without an alternative. As with the bass guitar example, I was able to patch in an outboard parametric EQ when I needed to perform sound surgery.

VOCAL DRILL

Cutting vocals is probably the application that tests a voice processor the most. With this in mind, I moved on to

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overdubbing both lead and background vocals for the demo, using the VC3Q in conjunction with an MBNM 608 C-L large-diaphragm condenser and a Shure SM58. I was anxious to hear what the VC3Q could do to tame and/or enhance my voice.

Because I was tracking to ADAT, I wanted to rein in any spikes that would cause digital clipping. Only after many repeated passes did I get the compressor to quit squirming and behave. Even after I thought I had sussed out the winning combination of compression, attack, and release, the VC3Q would annoyingly let a transient spike through. I found myself singing with one eye on the meter—an exasperating process. However, once the smoke cleared, the end result was impressive. Thanks to the VC3Q's compressor, the vocal tracks boasted a vintage photo-optical sheen that didn't squelch a dynamic performance. The VC3Q compressed, not repressed.

On the EQ side, the High control could be nudged to add upper-end sparkle at 8.5 kHz and above, but it was easy to go overboard. When I tried for too much "sparkle," the VC3Q imparted the vocal equivalent of blinding glare. The situation was the same with the midrange. A little bit lent a nice 1.5 kHz presence boost (especially with the SM58), but too much produced harshness. I was pleased when I rolled off the low-end shelving EQ for some background tracks on one of the songs. The result was a slimmed down (but not anorexic) quality that helped the background vocals cut through a particularly dense mix. It was an exaggerated special effect, to be sure, but a very cool one.

TINKERING WITH TINKLING

The last free track on the last tune presented me with a dilemma. I could either wimp out and punt with a sampled grand part from the original sequenced MIDI track, or crack my knuckles and pound out the part on the real deal. Going with the latter impulse, I propped wide the lid of my baby grand, mounted a Shure SM81 small-diaphragm condenser on a boom stand just outside and pointing toward the middle of the harp, then ran the signal through the VC3Q's gauntlet. After playing through the song a few times to get an input level (and to relearn the part), I was set to do a little knob twiddling.

Finding the right dose of compression took some time, but the compressor ultimately obliged by providing me with a transparent performance. The VC3Q tamed the busy rock-and-roll section without squashing the sustained chords in the bridge.

Although the piano sounded, well, *grand* without any equalization, I did resort to doing a little "thickening" with the low EQ. (The manual states that the Low control's frequency is fixed at

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80 Hz, but I liked the unmuddy detail the VC3Q imparted at around 150 Hz.)

WRAPPING UP

Joemeek's VC3Q is an exceptionally versatile and compact voice processor. The controls are intuitively laid out and dirt-simple to use. The unit was deceptively quiet throughout my demo project, such that I had to check more than once to see if the VC3Q was powered up while recording. The mic preamp is first rate, offering lots of headroom and little tonal coloration. Additionally, the VC3Q's portability makes it a prime candidate for remote location recording, whether at a road gig or just down the hall.

The VC3Q also provides some unique routing options unavailable on other voice processors, including some that sport higher price tags. The insert proved a godsend on more than one occasion, allowing me to quickly patch in another compressor, parametric EQ, or oddball effect as need (or whim) dictated. The VC3Q's dual outputs made for some welcome patching shortcuts in the studio.

Although I used the VC3Q mainly to cut tracks, I had no qualms about patching in the unit during mixdown, especially when I thought a particular track would benefit from a little VC3Q flavoring.

In addition to the awkward location of the phantom-power switch, I have a few more minor complaints. Getting comfortable with the VC3Q's photo-optical compressor does take patience, and it can be more fidgety than an eight-year-old in front of a Pokémon-card display—especially when tracking vocals. Accordingly, you might think twice before using the VC3Q to record that talented-yet-impatient vocalist.

I found using the VC3Q's EQ to be a hit-or-miss proposition.

In some applications (such as processing guitar and vocals), I found the fixed EQ to be somewhat limiting. Also, although the dark-green front panel is attractive, reading the settings in low-light situations can be difficult.

But minor reservations aside, I was totally impressed by the aural results. The VC3Q's compressor, for all its quirks, really does a good job of conjuring up that vintage photo-optical vibe. The EQ was especially well voiced for pointing up the drums. Best of all I didn't have to rerecord any of the demo tracks I cut with VC3Q with some "better" voice processor.

Lastly, consider the bottom line. At its suggested retail price of $399, the Pro Channel VC3Q is among the least expensive dedicated voice processors—an important factor if your home-studio budget is a little compressed itself.

John Ferenzik *has toured and/or recorded with Todd Rundgren and Jefferson Starship, among others. He is currently back to drinking too much coffee while promoting his latest CD release* Zero Points for Zeus. *You can check out his progress at www.ferenzik.com.*

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ANTARES

MICROPHONE MODELER 1.1 (MAC/WIN)

A virtual microphone closet for the masses.

By Erik Hawkins

Do you dream of your own personal cabinet full of rare and expensive vintage microphones? I do. Unfortunately, economic circumstances prevent many of us from realizing this dream. Indeed, most personal studio owners don't have even $1,000 budgeted for their main mic, much less $5,000 (or more) for one esoteric model. You'll excuse me for being skeptical when a manufacturer claims its plug-in can turn an inexpensive dynamic mic into a high-end condenser mic for far less than a grand.

On the other hand, digital audio workstations have fulfilled the music-making fantasies of millions of musicians for a fraction of what comparable analog recording systems used to cost. And the science of plug-ins is rapidly progressing and producing amazing results. Perhaps a virtual microphone cabinet isn't such a wild idea after all.

The *Microphone Modeler* plug-in from Antares offers an impressive selection of microphone models, including classic tube condenser mics, industry-standard dynamics, and popular mainstream FETs. According to Antares, the company meticulously crafted each model from an original microphone—right down to the pickup patterns and stock windscreens—using its patented Spectral Shaping Tool technology.

Microphone Modeler supports the main plug-in platforms: TDM, RTAS, VST, MAS, and DirectX. Antares deserves a round of applause for getting its software to work with all these systems. On the other hand, only the DirectX version of *Microphone Modeler* works on the PC; I'd like to see a VST version for Windows. I used the TDM version on a Mac running Digidesign's Pro Tools/24 Mixplus system.

INSTALLATION JIG

The *Microphone Modeler* CD-ROM includes an installer for the plug-in format you purchased, as well as demo versions of other Antares plug-ins. Although beginning the installation process is as easy as double-clicking on the installer, the next step—authorization—isn't easy at all.

Microphone Modeler
Minimum System Requirements
Mac: Power Mac; Mac OS 7; host program that supports TDM, RTAS, VST or MAS
PC: Pentium; Windows 95/98/2000/NT; DirectX-compatible host program

The program employs a challenge-and-response copy-protection scheme. *Microphone Modeler* issues the challenge when you boot the unauthorized plug-in and click on the Authorize button. *Microphone Modeler* will run on your system for ten days, during which time you need to send Antares an image of the serial number (which is affixed to the product-registration card) by snail mail, e-mail, or fax. Antares, in turn, sends you a response to your software's challenge. Once you have entered the response, you can breathe a sigh of relief and move on. This is more complex than most challenge-and-response schemes, and it's frankly a drag.

FULLY STOCKED CABINET

Microphone Modeler comes with more than 90 different models of vintage and contemporary mics. In the vintage category, you'll find models of mics by AKG (C 12A), Neumann (U47, U87), RCA (BK 5A), and Sennheiser (MD 421).

The newer mics include the Alesis AT61; Audio-Technica AT4033 and AT4055; beyerdynamic M 500 LE Classic; B&K 4007; Earthworks Z30X; Lawson L47MP; Manley Reference Gold; Neumann KM 184 and TLM 103; Oktava MC 219; Røde NT1, NT2, and NTV; Royer R-121; and Shure Beta series. The full list of models is on the Antares Web site.

Antares regularly adds new models and makes them available as downloads. If the program is missing a mic that you feel should be included, the folks at Antares say they're open to suggestions.

Downloading the microphone models is straightforward, but locating the right folder to put them in threw me for a loop. I expected to dump the files into the Digidesign DAE folder's Plug-Ins folder. But to my surprise, they go into the Antares Models folder in the System Preferences folder.

FIG. 1: The Antares *Microphone Modeler* gives you the opportunity to create a hybrid of two mics with the Preserve Source buttons.

For manufacturer information, please see Contact Sheet, p. 221.

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Two folders are inside the Antares Models folder: one for the source mics (the mics with which your tracks were recorded) and one for the modeled mics (the mics you want the source mics to sound like). Source and modeled mics always come together in a new download.

ON THE SURFACE

Microphone Modeler doesn't have a lot of parameters, so the interface is clean and easy to comprehend. Only a few controls defy immediate understanding and force you to consult the manual, which is informative and well written.

The interface is divided into halves (see Fig. 1). The controls on the left-hand side adjust the source-mic input; the right-hand controls tweak the modeled-mic output. An input-gain fader is on the left, and an output-level fader is on the right. Menus for selecting mic models, pickup patterns, and low-cut filter values are on both sides of the interface.

Each side also has a knob for controlling proximity. On the source-mic side, this parameter value is the exact distance between the original mic and the sound source. On the modeled-mic side, proximity controls the sonic picture painted by the virtual mic model (for example, frequency response, proximity effect, and so on.).

At the bottom of the interface is Tube Saturation, which simulates the high-quality preamp of a triode tube circuit. Its effect ranges from a barely perceptible warming to a wonderful, thick tube coloration. Overall, Tube Saturation convincingly adds realism to the modeled mics. A small replica of a tube element is to the immediate right of the fader and becomes increasingly orange as you increase the Drive setting.

Above the Drive fader lie a pair of buttons labeled Preserve Source. These buttons let you retain the bass or treble frequencies of the original source microphone. For example, if you like the high end of the modeled microphone but the low end of the source mic, you enable the Bass button. According to Antares, the crossover between high and low frequencies varies from mic to mic.

The Preserve Source feature is simple but handy, especially for fine-tuning the modeled mic's effect.

KNOW THY SOURCE

You must know the details about your source mic to get the best results from *Microphone Modeler*. Otherwise, you can't expect *Microphone Modeler* to give you an ideal representation of a different mic. Nor can you tell the plug-in that a $49 mic is actually a Shure SM57 and hope to hear a vintage AKG C 12A from the output.

▼

The Tube Saturation effect adds realism to the modeled mics.

Of course, there's no rule against specifying a source mic that's different from the mic you actually used during recording. Perhaps the mic you recorded with isn't available in the Source Mic menu, in which case you might select a substitute with characteristics similar to those of the original mic.

But remember, a poorly recorded track made with a cheap microphone will always be a poorly recorded track. In addition, *Microphone Modeler* assumes that your recordings were made on-axis. An off-axis recording paints a very different sonic picture and undermines the software's basic operating premise.

Microphone Modeler uses source-mic information to eliminate the coloration produced by the original mic. It's up to you to provide the proper information. Selecting the original recording mic and its parameters (such as pattern and proximity) in the Source Mic section gives *Microphone Modeler* a starting point from which to synthesize a new, flat base signal in preparation for the Modeled Mic stage of the process.

For example, if you recorded with a low-cut filter of 75 Hz, choosing 75 Hz from the Low Cut menu reverses the original low-cut filter's effect. Think of the Source Mic settings as a way of leveling the playing field so that the modeling algorithms can work their magic.

The Source Mic Bypass feature is for recordings made via a direct line input. It's perfect for making instruments recorded directly sound as if they were miked. With Bypass on, just select the modeled mic you want on your track. You could, for instance, apply a modeled Royer R-121 to your direct-recorded fuzz-guitar track for punchier mids.

MIC CHECK

Most of the tracks I've been mixing and remixing lately have been flown into Pro Tools from ADAT. Many of the ADAT session tapes were recorded at different high-end studios in order to take advantage of their extensive mic collections and tracking environments. Many of the rough vocal and instrumental tracks, however, were done in personal studios to save money. These, too, were flown into Pro Tools. As a result, I've amassed quite a collection of tracks recorded on a wide range of microphones: the Sennheiser MD 421 and Shure SM58 on the low end, the CAD VSM and Neumann TLM 103 in the midrange, and the AKG C 12A and Neumann U 47 on the high end. In short, I was in an ideal position to give *Microphone Modeler* a thorough workout.

I attempted as many combinations as I could think of. For example, I changed the Sennheiser MD 421 to a Neumann U 47, and the AKG C 12A to a Shure SM57. The results ranged from subtle to dramatic. Turning a

ANTARES

Microphone Modeler mic-modeling plug-in

$599 (TDM)

$399 (RTAS, VST, MAS)

$299 (DirectX)

FEATURES ■■■■

AUDIO QUALITY ■■■■

EASE OF USE ■■■■■

VALUE ■■■■■

1 2 3 4 5

PROS: Innovative. Numerous mic models available. Excellent sound. Tube saturation feature. Available for all platforms.

CONS: You must know what the original source mic was. Copy-protection scheme is convoluted.

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Neumann TLM 103 into a Neumann U 47 created a track with more air, adding tube coloration to the low mids and making the sibilances sound much sweeter. Changing the MD 421 into a Coles 4038 produced a larger, beefier sound with extra boom in the low mids and much more smearing of the high end.

When I compared the modeled-mic tracks with tracks I had recorded with their real-world counterparts, the results were quite impressive. Many of the microphone models were not totally convincing right out of the gate, but I was able to tweak the proximity parameters and adjust the Tube Saturation to come up with some incredible likenesses.

QUASI DI TO DIE FOR

One of my favorite uses of *Microphone Modeler* is processing instrumental tracks that were recorded direct, as I did with my Tacoma CC-10E4 nylon-string electric/acoustic guitar. The instrument's active Fishman Prefix Plus pickup system sounds wonderful but doesn't capture the warmth and body you'd get by miking the guitar's sound hole.

When I'm writing a song, I'm usually too lazy to set up a mic for recording. I prefer to just "plug and play." Often these rough performances are excellent, and I always try to doctor them up in hopes of keeping them for the final mix. By processing the tracks with the Neumann KM 184 model, I created a fairly credible emulation of a miked sound hole, without some of the overtones.

Playing with the Proximity and Tube Saturation controls and adding room ambience with Waves' *TrueVerb* plug-in transformed a cheesy, flat, direct recording into a stunning live sound.

Remember, a poorly recorded track will always be a poorly recorded track.

PLUG THIS IN

Microphone Modeler does what Antares claims: it provides a cabinet of extraordinarily solid-sounding virtual mics. You're bound to be pleased as long as you properly match your real-world source microphones to your modeled mic. If you stray from *Microphone Modeler's* matchup guidelines, however, you may not like the results, depending on how much of a stickler you are for realism.

The affordable *Microphone Modeler* is a great tool for DAW-based personal studios with limited mic resources. It sounds great and is perfect for everything from mixing to tracking. *Microphone Modeler* definitely gets my vote as one of the most innovative plug-ins of the year.

Erik Hawkins *is a musician/producer working in Los Angeles County and the San Francisco Bay Area. Visit his Web site at www.erikhawkins.com for more equipment chitchat and tips on what's hot for the personal studio.*

K O R G

OASYS 1.01 (MAC/WIN)

Lost in a desert of me-too music technology? OASYS is no mirage.

John Duesenberry

Rumors of a Korg product called OASYS have been circulating in the audio industry for years. Depending on whom you talked to, OASYS was a keyboard instrument, a sound card, a software synthesizer, a dessert topping, or a floor wax. In the mid-'90s, an OASYS keyboard instrument was actually announced; then it quietly disappeared. At last, the "legendary OASYS project" (to quote Korg's own publicity) has delivered a product: the OASYS PCI system. As it turns out, reality has proven to be more intriguing than the rumors.

The OASYS system combines a high-quality, 12-channel, 24-bit DSP PCI card with software hosted on the Mac and, by the time you read this, the PC. (I tested the Mac version.) The system benefits from several lines of research and development undertaken by Korg. The OASYS synthesis algorithms are largely based on the physical-modeling technologies used to create such Korg products as the Prophecy, Trinity, Z1, WaveDrum, and MS2000. The OASYS effects catalog expands on the Trinity effects set. The system hardware resembles the 1212 I/O card, but OASYS has superior audio specifications and much more DSP power.

VIEW FROM THE TOP

OASYS stands for Open Architecture Synthesis System. "Open architecture" means the system is software configured and therefore versatile and easily upgraded. Depending on how you set it up, OASYS can be a MIDI synthesizer, a MIDI-controlled mixer, an effects processor, a 24-bit sound card, or any combination of the four. Whenever Korg devises a new synthesis algorithm, it will be available to OASYS users as an Internet download.

OASYS divides its processing tasks between hardware and software. The hardware component is a PCI board with five Motorola DSPs, which handle all the audio processing by running code downloaded from the host computer. The board supports 12 audio channels, configured as stereo analog, stereo S/PDIF, and 8-channel ADAT Optical. All channels are available concurrently.

OASYS PCI Editor is the centerpiece of the host software. *Editor* presents a polished user interface for the OASYS mixer functions and also acts as a synthesizer and effects editor/librarian. In *Editor,* you construct system configurations called Multis. A Multi encapsulates synthesis and effects algorithms, signal routing and mixing, MIDI channel and modulation assignments, and all other settings. *Editor* and the OASYS PCI Engine driver translate a Multi into DSP code and download it to the OASYS board. As the code runs, you control it in real time via MIDI or with onscreen faders, knobs, and other gadgets.

The host software also includes audio and MIDI drivers. Korg provides drivers for ASIO, Sound Manager, and Windows MME. DirectX is not supported. These drivers ensure compatibility with all major audio programs, including digital audio sequencers.

FreeMIDI, OMS, and serial-port drivers let you control OASYS from Mac MIDI sequencers and external hardware, and the system offers full MIDI support in Windows as well. Under FreeMIDI or OMS, OASYS appears as two 16-channel devices, labeled "OASYS PCI A" and "OASYS PCI B." The Mac serial-port driver supports only 16 channels, but digital audio sequencers such as Mark of the Unicorn's *Digital Performer* and Steinberg's *Cubase VST* can use OASYS as a MIDI device and simultaneously process and mix audio tracks via ASIO.

OASYS integrates smoothly with host-based effects, such as MAS, DirectX, and VST plug-ins running in your host audio software. Because the OASYS board handles the system's audio computations, the OASYS software makes few demands on your computer. Therefore, you can run host-based plug-ins with OASYS effects and synthesizers without swamping your CPU.

THE HARDWARE

The OASYS hardware consists of a full-length PCI card with four connectors (see **Fig. 1**). The analog I/O connector accommodates a breakout cable that provides four ¼-inch jacks (two in, two out). The analog inputs and outputs are unbalanced, and they are referenced to a +4 dBu signal level. OASYS employs 24-bit, 128× oversampling sigma-delta converters for analog I/O.

All digital audio connections are also 24-bit. The ADAT Optical connectors each carry eight channels of audio to or from ADAT-compatible devices. The digital-breakout connector fans out to a cable with three pairs of jacks: a pair of RCA S/PDIF connectors, a pair of BNC word-clock connectors, and a pair of 9-pin ADAT-sync (time-code) connectors. OASYS has no MIDI ports;

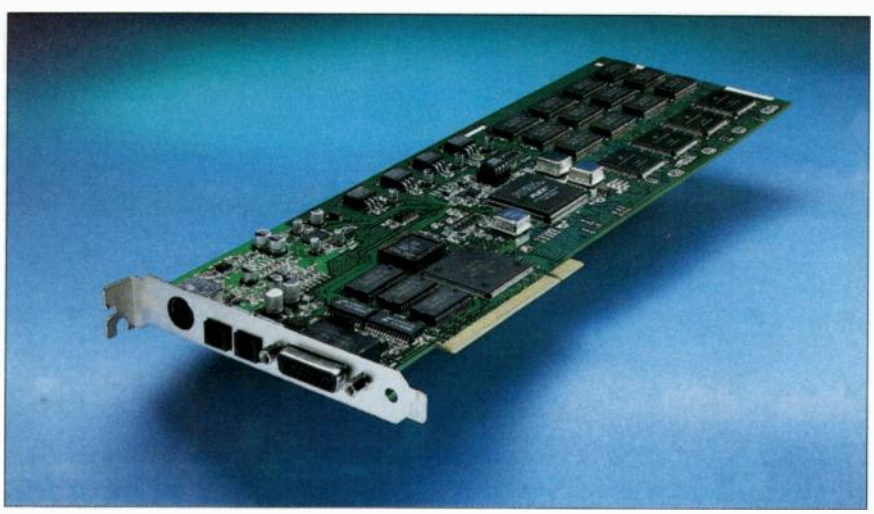

FIG. 1: Korg's OASYS is a long-awaited multipurpose synthesis, effects, and audio system. Five Motorola DSP processors power the single-slot PCI card.

For manufacturer information, please see Contact Sheet, p. 221.

CLASSIC SOUND

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FOLIO FX8

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- Built-in programmable Lexicon effects unit
- 3-band mid sweep EQ (with high pass filter)
- 4-bus structure
- 19" Rackmount option

FOLIO F1

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- 4 stereo inputs
- Professional 100mm 'long throw' faders
- 3-band mid sweep EQ (with high pass filter)
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- 19" Rackmount option

FOLIO FX16

- 16 mono inputs with high quality UltraMic™ mic preamps
- Professional 100mm 'long throw' faders
- Built-in programmable Lexicon effects unit
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- 4-bus structure
- 26 inputs to mix
- 19" Rackmount option

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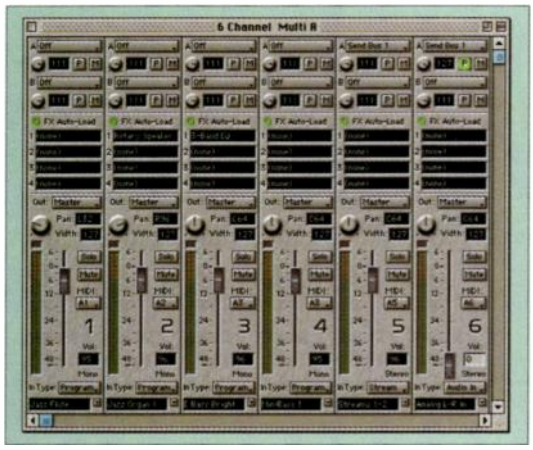

FIG. 2: The OASYS Mixer offers very flexible routing options and provides four insert effects per channel. In this example, inputs to channels 1 through 4 are synthesis Programs, channel 5 processes a stereo ASIO stream, and channel 6 processes the stereo analog inputs.

the system can use any outboard interface to send and receive MIDI via OMS or FreeMIDI.

The board's audio engine includes four 100 MHz Motorola 56303 DSP chips. An additional 80 MHz Motorola 56301 chip communicates with the host and performs other nonaudio chores. The processors execute DSP code at 480 million instructions per second. The system uses 24-bit buses throughout; it computes internal arithmetic with 56-bit precision. In some cases, OASYS uses 96-bit precision in its calculations!

The board's internal clock supports 44.1 and 48 kHz sampling rates. You can clock the board from an external device via the word-clock input, or you can provide a master clock to other devices from the word-clock output.

OASYS has no breakout box, but you can obtain up to ten additional analog I/O channels by connecting external converters to the ADAT Optical and S/PDIF ins and outs. The board is very quiet, but consider external converters if you're ultrafussy about noise picked up from the host computer.

PCI cards share the ground and power with the rest of the computer and are susceptible to a certain amount of system noise. Fortunately, I could hear system noise at the OASYS analog outputs only when I cranked my mixer and amplifier way beyond their normal operating ranges and listened on headphones.

GETTING STARTED

I installed OASYS on a 333 MHz Power Mac G3 running Mac OS 8.6. I followed the installation guide and experienced no problems installing, connecting, or running the hardware and software.

To install the hardware, just open your computer, snap in the card, and connect the required cables. Installing the software is also easy. The OASYS installer program loads *OASYS PCI Editor*, ASIO 2.0 and Sound Manager drivers, MIDI drivers, and other low-level components into your system. Note that some audio programs, such as BIAS *Peak*, support only ASIO 1.0. The OASYS Web site has an ASIO 1.0 driver for use with such programs.

Next, set the sample rate, audio-buffer size, and other audio parameters. The final step is to configure *OASYS PCI Editor* to communicate with either FreeMIDI, OMS, or the serial-port driver. If you choose FreeMIDI or OMS, you use their respective setup programs to install OASYS as a pair of 16-channel devices.

I tried a variety of software with OASYS. My main applications were MOTU *Digital Performer* 2.61 and BIAS *Peak* 2.10. I also took OASYS for brief spins with TC Works' *Spark* 1.5 and Tom Erbe's *SoundHack* (using Sound Manager). In addition, I used OASYS as a stand-alone, keyboard-driven, FreeMIDI and OMS device and as an OMS device driven by Opcode's *Studio Vision Pro* 3.5.4. In all cases, OASYS performed its audio and MIDI functions flawlessly.

MEET A MULTI

I set up a simple Multi to introduce you to the main elements of the OASYS user interface and to demonstrate some of the aspects of the system's versatility. This Multi is designed to work with FreeMIDI and *Digital Performer*, which aren't shown.

Fig. 2 displays six channels of the 12-channel Mixer window. At the bottom of each channel strip is a slot indicating the input source. Signal paths throughout the Mixer are stereo, but input sources can be either mono or stereo. In this example, inputs 1 through 4 are synthesis programs that model various instruments: Jazz Flute, Jazz Organ, E Bass Bright, and MiniBass 1.

The input to channel 5 is a *stream* of audio data. Streams are audio "pipelines" through which ASIO-compatible programs can send signals in real time. A stream can be mono or stereo. The input stream to channel 5 in this example, which is stereo, comes from two *Digital Performer* tracks. The input to channel 6 comes from the OASYS analog inputs.

In terms of MIDI, Mixer channels are normally assigned to a FreeMIDI or OMS device (A or B) and a MIDI channel number. The MIDI assignment routes MIDI messages to any synthesizer or effects on the channel, as well as to all the channel's faders, panpots, switches, and other controls.

In the example Multi in **Fig. 2**, all the Mixer channels are assigned to device A. The flute and organ are assigned to MIDI channels A1 and A2, respectively,

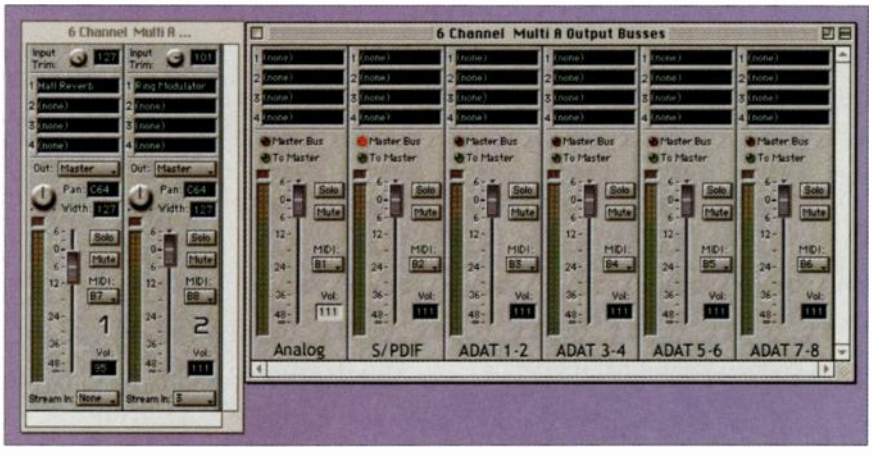

FIG. 3: OASYS offers four send buses, two of which are shown here. In addition, any of the OASYS physical outputs (stereo analog, S/PDIF, or ADAT) can serve as the Master Bus.


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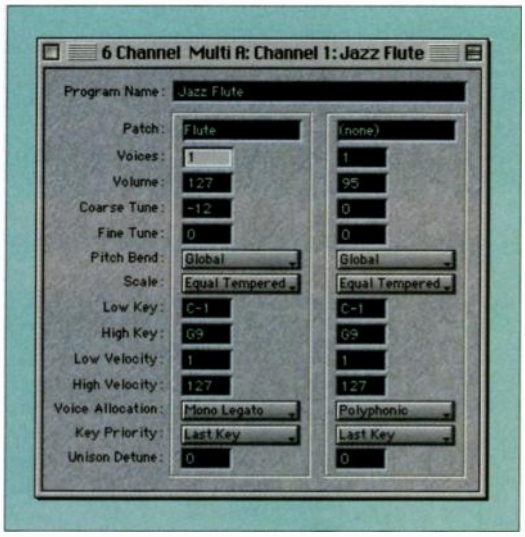

FIG. 4: The Program Editor specifies the Patches that make up a synthesis Program. This window sets the polyphony, tuning, key, and Velocity boundaries, and other parameters common to all Patches.

and the two bass programs are layered on channel A3. (Unlike a hardware synth, OASYS can have more than one patch on a single channel.) In effect, Mixer channels 1 through 4 constitute a 3-channel, multitimbral synthesizer. Channels 5 and 6 are assigned to MIDI channels A5 and A6, respectively.

Each channel strip has a panel with four slots for insert effects, as shown in the FX Auto-load section in **Fig. 2.** These slots let you add a chain of effects to each channel. Effects are inserted in series, prefader. The example Multi uses only a few insert effects: a rotary-speaker emulation for the organ on channel 2 and an equalizer on the bass on channel 3.

The Out selector in the center of each channel strip routes each Mixer channel to an output bus. In the example, all signals go to the Master output (more about the Master outputs shortly).

Each Mixer channel can also be routed to any two of the system's four send buses, usually for effects processing. Two send-bus selectors (marked "A" and "B") are located at the top of each channel strip. Each has a level control, a mute, and a pre/postfader send selector. In this example, channels 5 and 6 go to send bus 1. The green "P" indicator at the top of channel 6 shows that its signal is sent prefader.

RIDING THE BUS

OASYS has four send buses; to save space, I've only shown send buses 1 and 2 (see **Fig. 3**). Like Mixer channels, send buses support up to four effects. In **Fig. 3,** a hall reverb is plugged into bus 1 and a ring modulator is on bus 2. The sends are routed to the Master output. All send controls, such as the return fader and panpot, can be MIDI-automated.

In **Fig. 3,** send bus 1 receives signals from Mixer channels 5 and 6, while send bus 2's signal comes from a mono ASIO stream from *Digital Performer.* The Stream In menu at the bottom of each send bus selects a mono stream, which is summed with any signals bussed in from the Mixer.

Finally, notice that each output bus connects to a stereo pair of physical outputs (see **Fig. 3,** right). Each output bus strip has four slots for a final chain of effects, and controls can again be automated via MIDI.

You can route any Mixer channel or send bus output directly to one of the six output buses. Alternatively, you can route signals to a logical output called the Master Bus, as in **Fig. 2.** One of the six physical buses is always designated as the Master Bus, and all signals directed to the Master Bus go there. By simply designating a different output as the Master Bus, you can instantly reroute all signals to a different physical output. The example Multi uses the S/PDIF output, as indicated by the red Master Bus LED on the S/PDIF strip.

In this example, OASYS serves as MIDI synthesizer, mixer, and effects device, as well as a "plug-in" for processing *Digital Performer* tracks. This might seem complicated, but it took only a few minutes to set up.

PATCHES AND PROGRAMS

Before considering OASYS as a synthesizer, two basic OASYS terms must be explained: Patch and Program. An OASYS Patch implements a basic synthesis algorithm, such as a physical model of a trumpet. A Patch is a fundamental construct—you can tweak a Patch's parameters, but you can't create a new Patch in the *OASYS PCI Editor.* To do that, you must use *SynthKit,* an unsupported algorithm-development tool available free from the OASYS Web site (see sidebar, "Roll Your Own").

Programs are constructed from Patches and/or effects. A *synthesis* Program combines one or two Patches with insert effects. An *effects-only* Program contains effects but no Patches. This section discusses only synthesis Programs; I'll cover OASYS as an effects processor later.

Korg provides 28 Patches and hundreds of Programs based on these Patches. You select Patches and Programs from the Catalog window, where you'll find them organized into folders by type.

Patches function within Programs, and Programs function within Multis. To edit Patch parameters or just hear what a Patch sounds like, you must first create a Program containing the Patch. Then, you load the Program into one of the channels of a Multi.

In order to create or edit a Program, you open a Program Editor window (see **Fig. 4**). Here, you can specify the Patches that make up the Program. For each Patch, you set the number of voices, voice-allocation method, overall tuning, and other parameters. You can

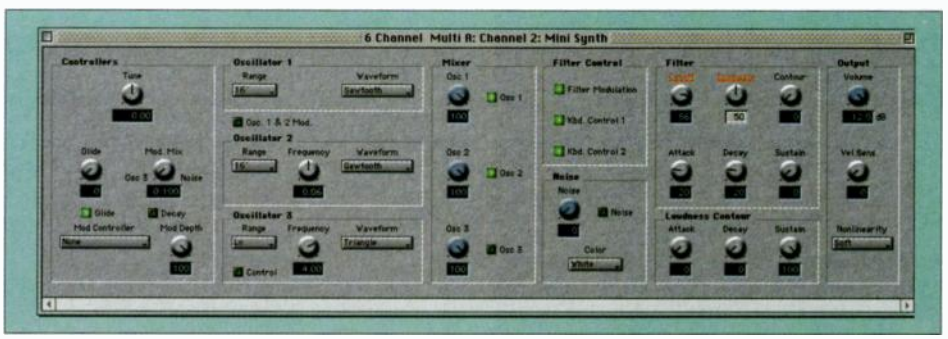

FIG. 5: The Control Panel for the Mini Synth patch will bring a nostalgic tear to many an eye. Parameters displayed in red (filter Cutoff and Emphasis) have MIDI modulation controllers assigned to them.

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FIG. 6: Control Panels, such as this one for the O-Verb LE reverb effect, make effects programming a pleasure.

also set up microtonal scaling and key or Velocity splits.

To edit Patch parameters, you open a Control Panel window (see **Fig. 5**). Each type of Patch has different parameters, so each has a different Control Panel layout; **Fig. 5** displays the Control Panel for the Mini Synth Patch. The Control Panel layouts are clear, the parameters and controls are grouped logically, and the controls (knobs, switches, for example.) are appropriate for each parameter. These serve as terrific patch-editor windows, and they make editing even the most arcane physical models intuitive and enjoyable.

The Control Panels provide extensive MIDI-modulation and automation possibilities. Double-click any parameter name, and the Modulation Palette pops up, which lets you assign two controllers to any parameter. OASYS supports many interesting combined or modified controllers, such as Mod Wheel + Aftertouch and Exponential Velocity. You can also record and play back the movement of knobs, switches, and other controls in the Control Panel via System Exclusive messages.

OASYS provides a great editing environment, but it's not perfect. There is only one level of Undo and no edit/compare function. In addition, you can't select multiple parameters for copying and pasting.

I liked the spacious layouts, but some might feel that the OASYS windows are too large. It would be nice to have an option to display windows at a reduced size.

OASYS AS A SYNTHESIZER

OASYS stands out from the common run of synthesizers because of its synthesis algorithms. The system employs several traditional synthesis methods and also offers a number of physical models. Many OASYS models are highly realistic and responsive to expression controls, and they show unusual attention to the details of sound production. I can only touch on a few favorites here. For detailed information on the OASYS models, download the *Patches and Effects Guide* from the OASYS Web site.

Among the electroacoustic instrument models, the Tonewheel Organ Patch takes top honors. This Patch models the tonewheels (61 of them!), drawbars, and key-click and percussion mechanisms of a B3-style organ. The Tonewheel Organ Patch achieves phase coherency between the tonewheels by modeling their individual behavior. (Sampling individual B3 notes can't do this, because the phase of any sample is unrelated to the phase of any other sample.)

The key-click and percussion emulations constitute a mini synthesizer in themselves. OASYS also includes an equally detailed rotary-speaker effect based on the Korg Toneworks G4. Korg has programmed many variants of the Tonewheel Organ, ranging from a funky Jimmy Smith sound to a distorted, churchy timbre recalling The Band's *Chest Fever*.

Another outstanding electro-acoustic keyboard Patch is the Reed Piano. This model sounds exactly like my first electric piano, a late-1960s Wurlitzer.

ROLL YOUR OWN

If you want to develop your own OASYS Patches from scratch, you need to get *SynthKit* from the OASYS Web site. *SynthKit* is a Mac-only environment for developing synthesis and effects algorithms that Korg engineers have used for several years as an in-house development tool. Now it's available as a free, unsupported adjunct to the OASYS PCI system.

SynthKit presents a graphical environment that superficially resembles many modular software synthesizers (see **Fig. A**). You grab icons representing algorithm elements, or "blocks," and connect them using virtual "wires." Many blocks are familiar elements of synthesis: oscillators, filters, ADSRs, mixers, noise generators, delays, and more. But *SynthKit* also gives you access to low-level physical-modeling components, such as Hammer, Bow, Cylinder, Glottal, and Reed.

Once you have connected your blocks, the algorithm is compiled into DSP code and downloaded to the OASYS hardware. You can then listen to it and control it via MIDI. When you're satisfied with the design, you can save it as a Patch and use it in the OASYS *PCI Editor* environment.

Remember that *SynthKit* is an engineer's tool, not a sugar-coated collection of audio toys. Despite an extensive manual, you're on your own; forget about calling a tech-support line to help develop your physical model of a sarangi. Regardless, *SynthKit* promises sound hackers endless hours of fun.

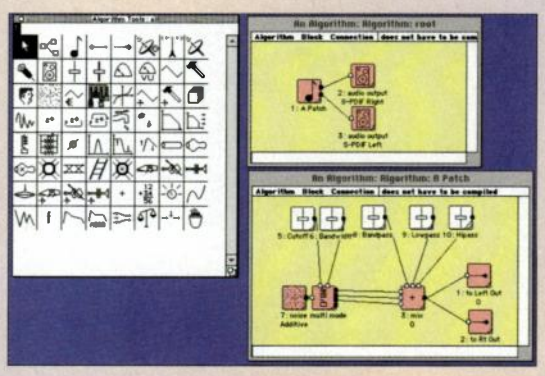

FIG. A: A free, unsupported algorithm-development tool for the Mac, Korg's *SynthKit* can be used to build new OASYS Patches.

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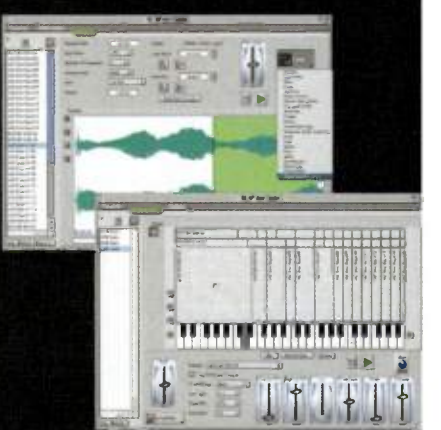

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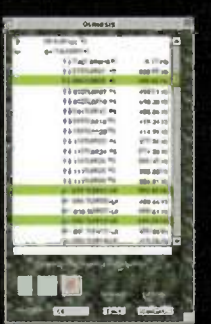

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Happily, the reeds don't break, corrode, or detune.

I've spent countless hours programming analog synthesizers, so I can attest that Korg did a lot of homework in designing its analog-synth models. Because they run at full audio rate, the OASYS envelope generators really sound analog. As a result, they are extremely smooth and free of "zippering." OASYS uses audio-rate envelopes in all Patches, not just in analog emulations.

OASYS gets hosts of other analog details right, too. For example, you can modulate the cutoff frequency of the Mini Synth (Minimoog) model's filter at audio rates. The Mini Synth also emulates "soft" VCA distortion, generates correct pink noise, and even has oscilator-range switches labeled like organ stops (2', 4', 8', for example)! The Pro Synth (Prophet 5) model is equally accurate, including great-sounding hard oscillator sync and full 0 to 100 percent pulse-width modulation. How about an ARP 2600 model, Korg?

The Percussion Synth and Beat Box Patches emulate analog electronic percussion; the Beat Box has a 16-step sequencer. These models sound as cheesy as the machines they emulate; I don't care for them, but no doubt they'd be useful in your next dance production. I wish Korg had provided physical models of drums or orchestral percussion instead. Surely the company that gave us the WaveDrum can do better?

While I enjoyed the simulations of great keyboards from the past, I found the acoustic-instrument models more interesting. These include winds (flute, sax), brass (trumpet, trombone), plucked strings, slap bass, and vocals. My particular favorite is the Tenor Sax. Like the other wind/brass models, this Patch emulates several subsystems of the instrument (reed, bell, bore) and player (embouchure, breath pressure), as well as the interaction and feedback between them. The sax model becomes extremely expressive and flexible once you've learned to manipulate its parameters. It can give you everything from a dry, breathy Gerry Mulligan tone to a raspy, honkin' R&B sound.

At extreme settings, Tenor Sax can also produce strange sounds bearing little resemblance to a saxophone. Working with three simple parameters—Breath Pressure, Breath Noise, and Growl—I was able to morph from an almost unpitched "blowing into a bottle" sound to a snarling, nearly inharmonic timbre. This startling effect was enhanced by using a Yamaha BC3 coupled with a MIDI Solutions Breath Controller (which converts breath-pressure signals from the BC3 into MIDI Breath Controller messages) to control these parameters. In fact, all the wind/brass models are more expressive under breath control, even with my crude technique.

I like the plucked-string models almost as much as the winds and brass. My only complaint about the acoustic models is that Korg didn't provide more of them. In addition to percussion, OASYS models of bowed strings, clarinets, double-reed woodwinds, and world-music exotica will hopefully surface in the future.

OASYS
Minimum System Requirements
(For running OASYS with concurrent MIDI and digital audio applications)
Mac: G3/200; 32 MB RAM; Mac OS 8.5.1; full-length PCI slot (must support bus mastering)
PC: Pentium II/300; 32 MB RAM; Windows 95/98; full-length PCI slot

Some OASYS Patches don't use modeling technology. The Z1 Organ Patch is based on additive synthesis, which is an effective way to "model" a pipe organ. The Virtual Phase Modulation (VPM) Patches are equivalent to 2- and 4-operator FM algorithms. Phase and frequency modulation are similar techniques, generating spectra that sound almost identical. (For more information on these techniques, see Curtis Roads's book *The Computer Music Tutorial* [MIT Press, 1996].) The VPM Patches and Programs sound quite a bit like early Yamaha 4-operator FM synthesizers, but with better envelopes and 24-bit dynamics.

AS AN EFFECTS PROCESSOR

I've long been a fan of Korg effects (I still use my Wavestation A/D), so I wasn't surprised by the high quality of the OASYS effects library. The catalog contains over 130 effects, including the complete Korg Trinity effects set. Effects types include reverb, delay, phasing and flanging, pitch shift and chorus, EQ and filtering, distortion and lo-fi, dynamics processing, amp/speaker and Leslie emulations, panning, ring modulation, tremolo, vibrato and other modulation effects.

Like Patches, effects are fully MIDI-controllable, and they are edited from Control Panels. In my book, this is a big bonus. If you were programming a reverb, would you rather use the tiny LED on an effects box or the OASYS O-Verb Control Panel (see **Fig. 6**)?

Effects-only Programs store effects configurations that are independent of any particular input source. Korg provides a large number of effects-only Programs. Some are ready-made effects

OASYS Specifications

Polyphonic Voices	variable
Effects Processors	variable
Number of Effects Algorithms	135
Types of Effects	myriad time-based, modulation, distortion, spectral, and pitch-shifting effects
Input Resolution	24-bit
Output Resolution	24-bit
Internal Processing	56-bit; 96-bit when needed
Analog Inputs	(2) ¼" unbalanced
Analog Outputs	(2) ¼" unbalanced
Digital I/O (stock)	ADAT optical; S/PDIF coaxial
Sampling Rates	44.1, 48 kHz
Other Ports	word clock (BNC)
Signal-to-Noise Ratio	92 dB
Frequency Response	20 Hz–20 kHz (±0.02 dB)
THD+Noise	0.005%

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chains, designed for processing particular types of signals. For example, Programs in the Electric Guitar group exploit various combinations of overdrive, EQ, amp simulation, and other effects suitable for making your Strat cry the blues.

Other effects-only Programs capture particular settings of one effect. For instance, the O-Verb group contains 36 assorted hall, plate, and other reverb and ambience programs. This group of Programs derives from Korg's new O-Verb algorithm, which makes its debut in OASYS. The O-Verb reverbs are, to use a technical term, gorgeous. They are spacious and transparent, with smooth, natural-sounding tails. You want them.

If you use many host-based plug-in effects, your CPU undoubtedly runs out of processing power sometimes. An overloaded processor might refuse to run the effects you want, or it might simply crash. OASYS effects can really help with this problem. You can run the less-demanding effects (such as small *Pluggo* effects) in your digital audio program while streaming other tracks to OASYS for more computation-intensive processing (such as reverb). The OASYS effect may very well sound better than the plug-in it replaces, so you can win on two fronts with this strategy.

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EASE OF USE ■■■■■
VALUE ■■■■
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PROS: High-quality audio hardware. Excellent acoustic and electronic instruments emulations. High-quality effects. Intuitive user interface. Full MIDI automation functions. Extensively programmable. ASIO, FreeMIDI, and OMS compatible.
CONS: Nonexpandable DSP architecture. Documentation needs enhancement. No DirectX support.

PERFORMANCE ISSUES

Speaking of processor performance, OASYS faces limits that apply to any real-time system with a fixed amount of computing power. For example, consider polyphony. The number of voices you can squeeze out of a given Patch type depends on the complexity of the algorithm, the sample rate, and most of all, the number of other patches and effects running on the DSPs. Korg documents the maximum polyphony available from each Patch type; at 44.1 kHz, this ranges from a low of eight voices (Pro Synth) to a high of 61 (Tonewheel Organ). Korg gives similar statistics for the number of copies of various effects you can theoretically use.

However, these are best-case figures, based on dedicating the entire system to one Patch or effect. In practice, you typically have multiple Patches and effects trying to share the OASYS resources. The available number of voices or effects is often far less than these figures indicate.

For example, the Multi in **Fig. 2** employs three monophonic synthesizers, a 61-voice organ, and four effects. This Multi gobbles up almost all the resources of the DSPs, leaving little room for anything else.

When a Multi exceeds the available resources, OASYS gives an error message. In this event, you must reduce the polyphony of a Patch, remove an effect, or otherwise reduce the DSP load. Until you do, OASYS won't play any audio. This is desirable behavior; the system is smart enough to know it's about to be swamped *before* it blows up. All the same, it's often difficult to predict whether a given combination of Patches and effects will work. Balancing the load of voices and effects is the trickiest and most frustrating aspect of working with OASYS.

THE WRITTEN WORD

OASYS includes three manuals totaling about 700 pages: the *Installation Guide*, the *Users Guide*, and the *Patches and Effects Guide*. In addition, there is an OASYS FAQ that addresses all sorts of performance and compatibility issues; it is essential reading.

The OASYS documentation is pretty thorough and clearly written, but it needs more tutorials and graphics. The tutorials in the MOTU *Performer Getting Started* manual would serve as a good model (pun intended) for Korg's

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writers. Also, the factory synthesis and effects Programs should be documented, at least in summary form. As of now, you have to actually edit a Program to find out about its Patches, effects, controller assignments, and so on.

You can download all the OASYS documents from the OASYS Web site, free of charge. The site has lots of good info, and many sound demos in MP3 format. You can also download the full *OASYS PCI Editor* as a soundless demo.

CLOSING ARGUMENTS

A fully professional product, OASYS boasts excellent audio quality and elegant and reliable software that interfaces neatly with other MIDI and audio programs. Most of the factory Patches and effects sound great, and there's a lot of potential for programming and sonic experimentation. On musical and sonic grounds, OASYS more than stands up to the competition.

The Electric Guitar's effects can make your Stratocaster cry the blues.

Given the steep list price of $2,200, however, you need to ask whether OASYS is cost-effective compared to alternative setups. But don't make the mistake of simply comparing OASYS to other sound cards; OASYS is much more than a sound card. Could you buy an equivalent sound card, a few dozen VST Instruments, and an equivalent collection of VST or MAS effects plug-ins for the same price? Not likely; a high-quality reverb plug-in alone can cost more than $400. Another contender might be CreamWare's Pulsar II, which, like OASYS, is an evolving system. Pulsar II uses a very different user interface, however, and doesn't feature the physical models that OASYS offers.

Only you can decide if OASYS fits your needs. As for me, I see an OASYS in my future, and I'm heading for it right now.

John Duesenberry's *electronic music is available through the Electronic Music Foundation. Check the EMF catalog at www.emf.org.*

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SOFTWARE TECHNOLOGY

VAZ MODULAR 2.1 (WIN)

Flexible, modular analog-synthesis software.

By Eric Bell

Analog synth-emulation software has recently come of age, with prices ranging from dirt-cheap to a few hundred dollars. Some products are locked into a single, fixed-synth design, but the more sophisticated programs allow you to build your own virtual instruments using a modular approach. (For a roundup of software synthesizers, see "Going Soft" in the July 2000 **EM.**)

VAZ Modular 2.1, a stand-alone software synthesizer from Software Technology, is a worthy candidate to satisfy your analog-synth cravings. It offers a well-organized environment and more than 60 highly configurable modules with which to build a wide variety of analog and other sounds. Unlike most of its competitors, *VAZ Modular* shuns screen-cluttering virtual patch cords, opting instead for a convenient and precise menu system to set up connections among its modules. A large number of example patches come with the program and more can be found on the manufacturer's Web site.

ENVIRONMENTALLY FRIENDLY

A breeze to install, *VAZ Modular* takes up a paltry 3 MB of disk space. Configuration is straightforward and includes support for MME, DirectX, and ASIO drivers. If your hardware supports it, opt for DirectX or ASIO, and you'll get much lower latency. This translates into a better feel when playing *VAZ Modular* as a "live" instrument from an external MIDI controller, and it also pays off when recording to an audio track in sync with existing tracks. (Some sound cards, such as the Echo series, provide lower latency with MME than with DirectSound.)

You can adjust latency time using the buffer size setting in the Set Preferences window. To find the minimum latency time, load a patch and start it playing. Then lower the buffer size until the sound starts to break up, and back it off a bit. You can also choose either stereo or mono output as well as select a variety of sample rates. As with all soft synths, the exact sample-rate options depend on the sound hardware in your system.

The settings for MIDI input are basic but workable. Besides choosing an input device, you can select a MIDI note number to trigger *VAZ Modular*'s internal step sequencer remotely, a handy feature for live applications. You can also configure *VAZ Modular* to associate a specific sequence with a synth patch so the sequence automatically loads along with the patch.

Documentation for *VAZ Modular* consists primarily of a Windows-style help file. Pressing the F1 key reveals context-sensitive help, which covers all areas of the program. However, more examples and details would make it easier to use and understand the program. A terse, two-part tutorial included with the software can get you started designing your own synth patches.

IT'S A MOD, MOD WORLD

VAZ Modular splits its functions into several dedicated work areas. The Mixer is the key to loading and running multiple synth patches. Open the Mixer window, and by default a new Synth channel appears (see **Fig. 1**). (In *VAZ Modular*, a Synth consists of a collection of modules, and a Patch means a specific group of parameter settings for those modules.) To add a second Synth, simply click on the number at the top of a channel strip and a new window pops up. Here you can build a new Synth or load one from disk. The Mixer window has 16 channel strips, one for each Synth in a Bank. Individual mixer settings can be saved and recalled separately from Bank and Patch files. The Mixer also offers two master effects and a single insert effect for each channel.

Synths are built in the Synth window, where you obtain access to the program's extensive sound-generating and processing modules. To open a module, right-click anywhere in the window and a pop-up list appears (see **Fig. 2**). (You can also reach the modules through the menu system.) A unique aspect of *VAZ Modular* is its ability to "roll up" individual modules into title bars. This saves screen space and lets you concentrate on just the modules you want to tweak. I would like it even better if vertically adjacent modules would move automatically to fill up vacated space. Similarly, when you "roll down" a closed module, others beneath it become obscured. It seems logical for existing modules to move down to make room. To further aid organization, you can color-code the different modules in a Synth design.

To route a signal to a module, click on its input control and a pop-up list containing appropriate sources appears. In **Fig. 3** you can see the list of sources available to the Oscillator module. This quick and precise routing

FIG. 1: The Mixer window in *VAZ Modular* is the first stop for creating new Synth designs. It also provides access to the Aux, Insert, and Master effects.

Small, but effective

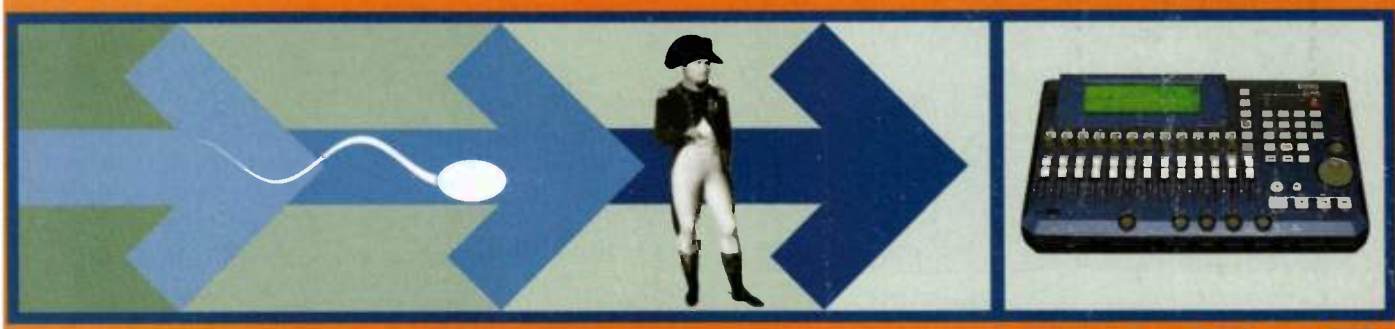

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FIG. 2: The VAZ Modular interface avoids clutter by allowing you to "roll-up" each module so that only its name appears. It also uses an efficient menu system for picking the various modules you'll use in your synth designs.

system lends itself to serendipitous, "last-second" choices. Unlike some synth programs in which you drag virtual patch cords between a source and its destination to make a connection, you don't have to search the screen to find a particular source device.

However, the menu has a few drawbacks. For example, you can't see a graphic representation of the signal path in a patch. In addition, there is no easy way to find out what other modules a particular module is routed to. It would be nice to have the option of a display that showed connections graphically.

By default, a Synth responds to MIDI commands on the channel matching the number in its Synth window. This can be overridden by setting the MIDI channel explicitly on the Synth's Master Controller module (more on the modules later). As with many soft synths, you can have more than one patch on the same MIDI channel.

Once you've created a Synth, you can save its Patch settings to a file. You can also save groups of Synths as Bank files, which hold up to 16 Synths. A snapshot feature lets you quickly save the current Patch. Snapshots are immediately available for recall from the main menu, which saves you from having to go through the File/Load dialog. Snapshots are also assigned automatically to the Patch list and can be selected via Program Change commands. One caveat: loading a Patch snapshot also loads the Synth modules that it uses. In other words, the synth modules needed are created, even if the exact set of modules was used in the previous Patch. When possible, use the Speed Load option, which checks if the modules of the new Patch match the current one and if so, just loads the parameters of the existing modules.

Finally, you can trigger Synths in a variety of ways: live from the computer keyboard or a MIDI input, from an external MIDI sequencer, or from the built-in step sequencer. If you want to use a sequencer running on the same computer, you'll need a "virtual MIDI router," such as *Hubi's MIDI LoopBack* or *MIDI-OX*. These and many other routers are available wherever fine shareware is sold.

MAKE A NOTE OF THIS

VAZ Modular's final main work area is the Step Sequencer window. The sequencer is based primarily on the Roland TB-303. Each step has a slider to set the note's pitch, and there are two additional sliders

FIG. 3: Assigning an input signal to a module proves as simple as clicking on a parameter, then choosing from a list of sources. Available inputs for the Oscillator module are shown here.

for adjusting the two multifunction controls (Control A and B). These two controls can be used to transmit various types of MIDI data, or for different types of "internal" purposes, such as modifying the tempo of a pattern. There's also a group of buttons that toggle the Double, Rest, Slide, and Accent options for each step. The sequencer supports up to 16 patterns of 16 notes each, and a rudimentary song editor allows you to create a piece of up to 256 bars.

Though you can't type in exact values for a step's pitch, you can program notes by moving the note sliders one at a time (a very tedious task) or enter them in step-time via MIDI. The sequencer offers a completely configurable randomizer, which provides some very powerful options for altering the playback order and content of patterns. Because you can toggle the randomizer while a sequence is playing, it's easy to experiment until you find something you like. You can also loop individual patterns or play them forward or backward. These functions can be toggled in real time.

I'm not a fan of emulating hardware in software programs because the tactile things that make hardware great—such as smooth faders, weighty knobs, or ribbon controllers—often can't be well replicated on a screen. It can be downright clumsy to try to do some things with a mouse that are intuitive and easy in hardware, like turn a knob or move more than one fader at once. That said, *VAZ Modular*'s sequencer is well implemented and easy to use.

Once you have a funky bass line tripping along, you may want to dump audio to a file and use it in a program like Sonic Foundry *Acid* or Cakewalk *Pro Audio*. *VAZ Modular* has a Capture command for this purpose. You'll have to crop the audio file after using the Capture command to remove any unwanted leading and trailing sound data, however.

More user-friendly is the Capture Sequence command. It writes a WAV file from the moment the sequencer starts until it stops. This is a great way to capture loops. If you want to grab individual notes to use in a sampler, set the sequencer tempo to "1" and capture a "sequence" consisting of a single note.

AROUND THE BLOCKS

VAZ modules for synth design are sorted into the following categories: Audio Sources, which includes oscillators, sample playback, a noise generator, and line input; Audio Processing, such as filters, EQ, amplifiers, and ring modulator and Modulation Sources, like LFOs and envelopes. You'll also find Mod Processing, which includes the Logic Gate and Sample and Hold modules; Routing, such as the Mixers, Panners, and Routing Matrices; and of course, Effects, which I'll discuss later. Additional modules appear in the Visualization category, such as the Oscilloscope and Meter, and various types of MIDI to "CV" converters are also on hand. (Used internally by the program, these converters are not intended to control external, voltage-based hardware.) There are no arbitrary limits on the number or type of modules in a Synth, a freedom that adds to your creative palette.

At the core of any analog synth lie its oscillators, and *VAZ Modular* boasts a wide selection. The basic Oscillator module produces either sawtooth or square

VAZ PLUS 1.7

If you want to get your feet wet with the *VAZ* style of emulated analog synthesis but don't want to jump completely into the pond, take a look at the budget-priced *VAZ Plus 1.7* (Win; $35 download; $40 on CD). *VAZ Plus* is more akin to a dedicated hardware synth than *VAZ Modular*; in effect, the signal path of its oscillators and filters can't be altered. You can, of course, modulate any of the audio modules, and *VAZ Plus* also sports the same internal algorithms as its big sibling, as well as a similar look and feel. What you give up is the modular, building-block approach to synth design that distinguishes *VAZ Modular*.

What kind of a synth does *VAZ Plus* offer? A very versatile one, as it turns out. Among the modules are two oscillators, two LFOs, a resonant filter, an amplifier, a portamento control, and a sequencer. The second of the two oscillators offers more parameters than the first; for example, there are more tuning options and the option of wavetable playback.

The complex filter yields loads of options to spice up your sounds. As with *VAZ Modular*, many controls can be modulated with LFOs or other inputs to give you deep, kaleidoscopic electronic colors. The sequencer window is just like the one in *VAZ Modular*, minus one or two controls. There is no mixer window, which is no big loss, as *VAZ Plus* supports only one synth at a time. However, like *VAZ Modular*, *VAZ Plus* is polyphonic. The polyphony setting hides in the Preferences dialog—once set, it applies to whatever Patch settings you load.

Want more? There's an arpeggiator similar to that in the full-blown product, with options to select direction, number of octaves, randomization types, and more. Another addition, which *VAZ Modular* doesn't include, is a list of built-in presets that are conveniently available from the main menu. *VAZ Plus* was meant to be played and enjoyed right out of the box (or download file, as it were). And of course, you can save and load your own customized patches.

Besides being an inexpensive way to test the *VAZ* approach, *VAZ Plus* serves as a viable, inexpensive analog synth that gives you good bang for the buck.

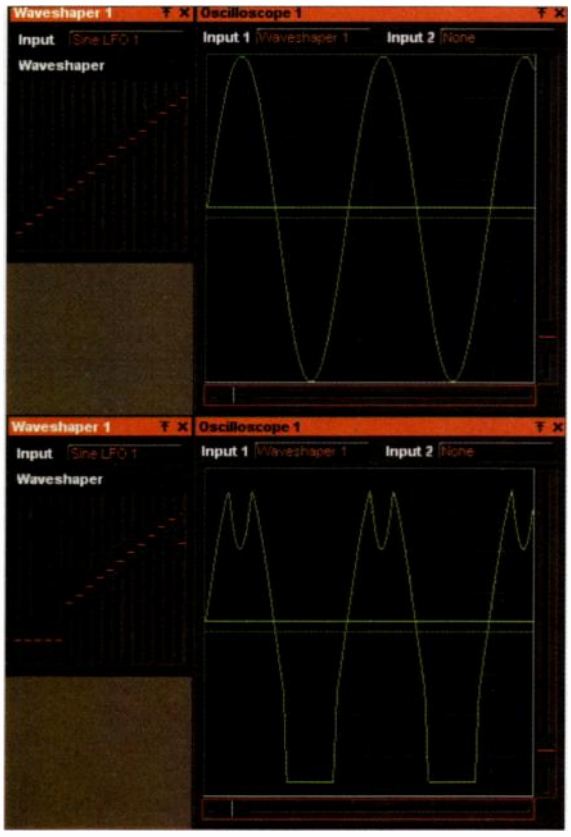

FIG. 4: The Waveshaper module is used to modify the harmonic content of an audio source. Here, a linear transfer function (top left) produces no change to the output sine waveform (top right), while a nonlinear curve (bottom left) alters the waveform to produce clipping (bottom right).

waveforms. A signal (such as an LFO) fed to the Pulsewidth Modulation control changes the oscillator's waveform over time, adding sonic interest. The Multi Oscillator module generates four waveforms that can be detuned relative to each other. The detune amount itself can be modulated by a control signal.

The Granular Oscillator offers an unusual way of making sound. It takes a 16-bit WAV file as its sound source and breaks the sample into tiny pieces called "grains." It then plays them consecutively, automatically crossfading between them. This allows you to time stretch a sample without altering the apparent pitch of the sound. You can control the length of the grains, the crossfade length, and the starting point of the sample; for example, you can stretch only the last portion of a sample, leaving the attack unchanged.

A CHANGE IS GONNA COME

VAZ Modular offers a huge toolbox full of things you can throw at—and into—your Audio Source modules.

There are very flexible modulation-routing options, and you can modulate nearly any input on the different modules. For starters, there are several variations on the old standby LFO. The basic module has variable Rate and Waveshape controls, both of which can be modulated. Two additional LFOs offer only sine and triangle shapes, along with a single control for rate.

The Multi Phase LFO outputs four saw and four triangle waveforms. You can set the delay between the waveforms (in fractions of a cycle) for some very cool control possibilities. Each output is typically routed to a different destination, but if you want to apply a combined signal to one input, simply route the four LFOs into a four-input mixer and route the mixer's output to the destination. I tried this and quickly came up with an interesting "heartbeat" throb effect with a single-oscillator synthesizer.

In addition to internal modulation sources, you can alter a module's values via MIDI. Setting this up couldn't be easier. Just right-click on a control, choose Learn MIDI, then move the slider or knob on your MIDI device that you want to assign to the control. If you have an external MIDI control surface or if you like to record your automation data in a sequencer, you'll find this feature very useful.

THERE'S MORE IN THE BOX

VAZ Modular offers many flexible filters, such as Resonant, Comb, One-Pole, State Variable, and the very cool Vowel filter. The Vowel filter is a 5-stage module, each stage of which offers ten different vowel sounds to shape the input signal. The filter interpolates from one stage/vowel sound to the next for smooth timbre changes. You can also choose from five "voice-type" presets, including male and female speech and singing.

The Waveshaper module is quite interesting but is poorly explained in the documentation. It provides a set of 17 sliders that control how the input-amplitude values of your source will be remapped to new output values. The position of the sliders themselves forms the transfer function, so it's easy to visualize what type of modification you're going to apply to the sound. You can see at the top of **Fig. 4** that a diagonal line extending from the lower left to the upper right of the screen produces no change to the sine-waveform source file because every value simply passes unaltered. But the transfer curve shown at the bottom of **Fig. 4** will produce some nice distortion. Once you get the hang of it, you'll find the Waveshaper lets you create very complex and rich-sounding waveforms from any input source.

Other Audio Processing modules include the Amplifier, Decimator, Parametric EQ, Frequency Shifter, and Sub Oscillator (which produces waveforms an octave lower than the input).

VAZ Modular also boasts a small group of built-in effects: Chorus, Delay, Flanger, Phaser, and Reverb. These are very basic, and each offers only a few parameters, but the program supports both DirectX and VST plug-ins. In fact, plans are afoot to release *VAZ Modular* as a VST Instrument. (Software Technologies also expects to release a VST version of *VAZ Plus*; see the sidebar "VAZ Plus 1.7".)

PATCHMAKER, PATCHMAKER

I found it quite easy to create Synth designs and Patches with *VAZ Modular*. If you're a little rusty at working with LFOs and oscillators or a just a bit green, hone your chops with the tutorial and save a few simple Synth configurations to disk. You can then go back and use these as the building blocks for more complex designs. (It would be helpful if examples of this type were provided with the program.)

The minimum configuration for a playable *VAZ Modular* synth consists of a sound source, such as an Oscillator module; a CV Converter to translate

VAZ Modular
Minimum System Requirements
Pentium II/100; 32 MB RAM; Windows 95/98/NT

For manufacturer information, please see Contact Sheet, p. 221.

computer-keyboard or MIDI notes into pitches and control signals, provide arpeggiation, and so forth; an Amplifier module to control output levels; and a Master Controller module where MIDI channels, polyphony, and other settings are made.

Starting from this simple design, I built my first patch in about an hour. I exchanged the Oscillator for a Multi Oscillator and added depth by detuning its four waveforms. Taking a cue from the tutorial, I used a simple ADSR Envelope to shape the Multi Oscillator's output. This gave the sound a more organic feel. Then I dropped in a Mixer module, so I could add a Flanger in parallel with the Oscillator's "clean" output and control the levels of both sounds. I also used an LFO to modulate the frequency of the Oscillator. With just a bit more tweaking, I had a searing lead sound with character and tonal depth and a nasty bite to its attack.

When designing patches in *VAZ Modular*, you can change any parameter or connection while the synth you are working on continues to play. You can hear your changes as you go, and the Snapshot feature is great for quickly saving promising settings. These interactive capabilities have some serious performance implications though, which I'll discuss next.

COMMAND CPU PERFORMANCE

VAZ Modular (like all software-only synthesizer programs) uses your computer's CPU and RAM as resources for sound generation. The demands placed on your system increase with the complexity and number of voices and effects you employ.

Running a single polyphonic voice in *VAZ Modular* on a Pentium II/366 MHz Celeron-based PC raised no performance issues and left me plenty of untapped CPU power, according to the Windows 98 system monitor. When I created a bank with four polyphonic patches of four voices each, I noticed two things that initially troubled me: *VAZ Modular* was using a huge chunk of CPU power, even when it wasn't in the foreground or playing any notes. In addition, it grabbed a large chunk of memory and hung on to it until I terminated the program.

SOFTWARE TECHNOLOGY

VAZ Modular 2.1 (Win) software synthesizer

download $299; CD $339

FEATURES ■■■■■

EASE OF USE ■■■■

QUALITY OF SOUNDS ■■■■

VALUE ■■■■■

1 2 3 4 5

PROS: Excellent analog sound. Great value. Large variety of modules. DirectX and VST plug-in support.

CONS: Heavy CPU usage. Live input using ASIO drivers only. Slow screen updates. Limited documentation.

Martin Fay, the creator of *VAZ Modular*, explained that the *VAZ* synth engine is always "running," similar to the circuits of a powered up analog synth. For example, LFOs are still oscillating, whether or not any notes are being

played. This is a conscious design decision on the developer's part and, as it turns out, one with distinct advantages. Among other things, it makes *VAZ Modular* incredibly responsive in terms of latency and user interactivity. When designing synths, you get instant and constant feedback to the changes and tweaks you make.

On the down side, this "background processing" can have an adverse effect. Screen redraw times, for example, suffer out of proportion to the system load. (I tested with a state-of-the-art AGP graphics card, so the load on the CPU should have been minimal.)

Fortunately, *VAZ Modular* gives you several options to deal with the processing issue. These include a Mute hot key, which quickly turns off all of *VAZ Modular*'s processing, thereby freeing the CPU for other tasks. (This also closes any open audio drivers.) Also, a Background Mute setting stops all processing whenever you switch from *VAZ Modular* to another program. Individual synths can be muted through the On button found on each channel strip in the Mixer window, which also lowers the demand on the CPU. This can also be accomplished via MIDI, so with careful planning, you can have many synths loaded but turn them on and off as needed.

THE SHAPE OF SOUND TO COME

VAZ Modular offers an infinite set of design possibilities. Its sonic quality is very high, and the synths sound authentic. (Try the comparison test, which pits *VAZ Modular* against recordings of actual hardware synths at the company's Web site.)

Another strong point of *VAZ* is the reuse of components in its user interface. For example, most of the Audio

▼

With just a bit more tweaking, I had a searing lead sound with a nasty bite to its attack.

Source modules share common parameters. These include the pitch section, in which you can select a modulation input to control pitch; and the tuning section, which is configurable over six octaves and adjustable in 1-cent increments. This makes it easier to learn the program, a task that can appear daunting at first, given the number of module types available and their apparent complexity.

If you love to get elbow-deep into synth design or require a huge amount of creative flexibility in your vintage synth sounds, *VAZ Modular* may be the answer for you. However, make sure you have lots of CPU power on hand if you plan to use a number of synth patches or voices at the same time or if you plan to drive *VAZ* from a digital audio program while playing audio off the hard disk.

If designing synth patches simply isn't your bag, are the modular capabilities of *VAZ Modular* overkill? Not at all. Even if you never design a synth of your own, the wealth of presets make *VAZ Modular* a worthwhile and cost-effective tool. It's a great way to expand the sonic palette of your desktop studio.

Eric Bell *is a writer, programmer, and musician living in Canada.*

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B O S S

VF-1

Amp modeling and effects processing in a little red box.

By Rick DiFonzo

An attractive bright-red unit, the half-rackspace Boss VF-1 multi-effects processor will draw attention in even the most crowded rack. Besides its good looks, the VF-1 offers reverb, delay, chorus, intelligent pitch-shifting, ring modulation, guitar-amp modeling, keyboard effects, EQ, compression, a tuner, and more, all at a surprisingly affordable price.

The VF-1's architecture proved so intuitive that I easily navigated and edited patches without consulting the 142-page manual first—a good thing, because the manual is unclear (even comical) in some sections, with spelling and grammatical errors aplenty. Documentation problems aside, it's always a pleasure to plug in a new toy and begin making music right away.

ONCE AROUND THE BOX

The input-level knob on the front panel adjusts the right and left channels separately and doubles as a push-button power switch. Directly below it lies the Hi-Z high-impedance guitar input, which enables guitarists (or bassists) to easily plug in to the VF-1 even when the unit is mounted in a rack. This input overrides the rear-panel left and right line inputs.

Next to the input section is a bright-green LCD that displays program and editing parameters. It's easy to read if you're close enough to turn the knobs, but the level meters are too small to be seen from a distance. On the panel's right side are six buttons—Utility, Write, Exit, Parameter Left, Parameter Right, and Category—for navigating, tweaking, and saving patches. A medium-size data-value knob doubles as an effects-bypass switch.

On the rear panel (see **Fig. 1**) are an AC receptacle for the "lump in the line" transformer and a useful hook for securing the power cord. Beside them are MIDI In and Out connectors, a coaxial S/PDIF digital output, ¼-inch left and right analog inputs and outputs, and a level switch (–20 and +4). A ¼-inch TRS controller input lets you use an expression pedal or footswitch for real-time parameter control, and two footswitches can be connected simultaneously by using a Y-connector such as the Roland PCS-31.

INSIDE STORY

The VF-1 employs a very flexible and easy-to-understand architecture. It has four banks, each containing 100 patches. Two banks contain factory presets, and two store user-edited patches. The basis of each patch is a multi-effect algorithm. You have 36 algorithms to choose from, each with its own set of effects. For example, the algorithm Guitar Multi 1 includes compressor/limiter, wah, amp simulator, 4-band EQ, noise suppressor, modulator, delay, and reverb. Each effect's parameters can be individually disabled. The order of effects in the chain is also modifiable.

With the Control Assign feature, you select the parameters to be altered by a footswitch, pedal, or other external MIDI controller. Different value ranges on the same controller can affect different parameters. For example, values 0 through 126 on an expression pedal can alter wah, and 127 can switch wah off. The VF-1 offers additional MIDI functions, including Program Change, Bulk Dump patch storage, and synchronization of time-based parameters (delay time, flanger modulation speed, and so on) to MIDI Clock.

The Utility menu displays global, system, MIDI, and tuner settings (including output mute for the guitar tuner). Here you adjust global output levels, the display contrast, global bpm (for time-based effects), and so on, and select the key signature for the intelligent pitch-shifter. You can also reinitialize the unit to its factory settings in the Utility menu.

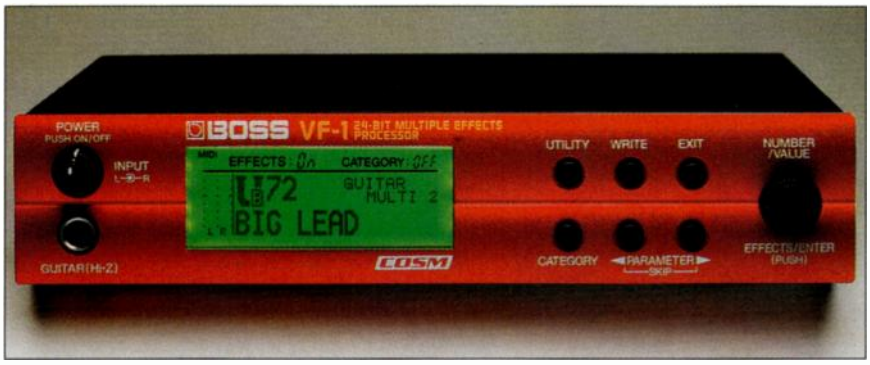
The VF-1 lets you assign each patch to a category: Guitar, Keyboard, Vocal, Groove, and so forth. When you're searching through the 400 patches, you can use the Category Search function to display only the patches from the currently selected category.

SOUNDS ABOUND

Space does not permit me to cover each of the VF-1's 400 effects patches, so instead I'll touch on the more interesting and useful ones. I'll start with the guitar and bass models, which were created with Roland's Composite Object Sound Modeling (COSM) technology. You get quite a few vintage and modern amps to choose from. Boss didn't include any manufacturer names, but some of the display icons bear a distinct resemblance to amps by Vox, Fender, Marshall, and others. The manual also hints broadly at the type of amp each model emulates.

I am fortunate enough to have some great vintage amps. I compared the VF-1 with them, and in every case the real amp moving air across a microphone sounded better. But not everyone has a collection of great amps at their disposal (or a studio where they can record them at suitable volumes), and the ability to dial up sounds instantly and record direct is a great convenience.

Using the VF-1, I created a nice-sounding, clean Vox patch, but the model's range was limited. When I tried to push it hard, as I do my AC-30,



The Boss VF-1 multi-effects processor offers a wide variety of effects in a small, moderately priced unit.

For manufacturer information, please see Contact Sheet, p. 221.

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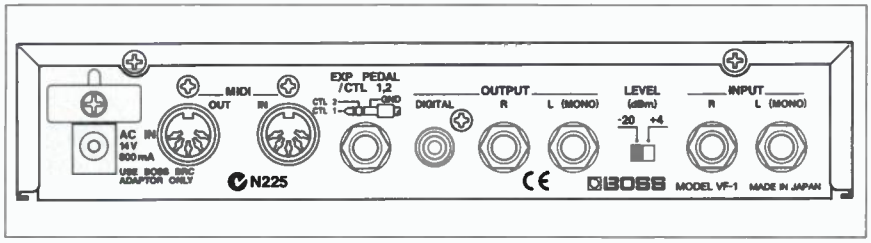

FIG. 1: Connections on the VF-1 include analog inputs and outputs, a coaxial S/PDIF digital output, MIDI In and Out, and an expression pedal or footswitch input.

it turned ugly, and not in a good way. Studio Lead and many other patches aimed at big crunch sounds exhibited what I call *the mosquito effect:* a lot of buzz with little sting. After a little tweaking I came up with some usable power-guitar patches, but overall I found that the models sounded most realistic on clean settings.

Most of the VF-1's guitar patches swim in delay, chorus, and reverb, which make their sound more appealing and provide immediate gratification to the first-time listener (especially the potential buyer checking out the device in a music store). There is no quick way to lose these effects while auditioning amp patches, because the amp models *are* effects. If you activate the front-panel bypass switch, you end up with only a direct guitar signal. The only way to turn off specific effects within a patch is to defeat each one individually, and that requires more button-pushing and LCD-reading than I care to do.

The effects, amp simulators, and pickup models were most effective when I plugged the VF-1 into a guitar amp (instead of going direct). I could set a neutral sound on my Vox, and then easily dial up a wide variety of sounds ranging from clean to mean. I had to adjust the factory patches to accommodate the guitar and amp being used, but there are plenty of parameters to fiddle with.

Although the acoustic-guitar models fell short of making any of my electric guitars sound like real flat-tops, I was able to make a single-coil pickup sound enough like an acoustic/electric to make it useful onstage. I also successfully used these models to add shine to actual acoustic-guitar tracks in the studio.

I then put on my bass-playing hat (and my four-string) and tried the bass patches. Some of the effects-laden patches were interesting (for example, Funkenstein and Dinosaur), but again, I had to tweak quite a bit to get usable bass sounds. Even after I'd fiddled and found sounds I liked, I preferred my regular bass-recording technique: through an amplifier with a Sennheiser D-112 mic on a 15-inch speaker cabinet. But bass guitar is often recorded direct, and if you think of the VF-1 as a direct box with built-in effects, then it will be useful to you. It won't replace or even compare favorably with a decent amplifier, but it is eminently practical in situations in which a loud bass amp is unwelcome.

Keyboard players will also like this box, especially the Ring Modulation, Step Phaser, and Rotary Speaker simulator patches. The Rotary Speaker effect makes a plain Hammond-organ sample really sing. The rotation speed and acceleration of both the rotor and horn are independently adjustable. I used a Boss EV-5 expression pedal to change the rotary speeds during a performance, and it worked great, but I wish there were a way to transmit the pedal values by MIDI to automate the moves with my sequencer. I also used Rotary Speaker on electric guitar and background vocals, and loved the results.

IN THE STUDIO

As an outboard multi-effects unit, the VF-1 packs a lot of wallop. Varied and very customizable, the reverbs generally sound smooth and natural. A number of reverbs are designed for specific uses (kick, acoustic guitar, vocal, piano, and so on) and acoustic environments, such as hall, club, arena, and Roland Surround. In a well-equipped studio the VF-1 would be useful as an auxiliary unit, and in a more modest setup it would be a perfectly acceptable main reverb.

A wide assortment of phaser, flanger, and chorus patches are available, and overall they are quite good. I got plenty of use out of the Dimension 1 Space Chorus patch; it's nice and rich and has that Roland sound. Available delay types include tap, tape (à la Roland Space Echo), and 3-D.

TRANSFORMED MAN?

The Male > Female patch alters a signal's formants instead of its pitch. It can achieve some interesting effects, but it certainly didn't convincingly transform a male vocal into a female vocal. Just to see what would happen, I "transformed" a male vocal track to female, recorded it to tape, and transformed it back to male. The resulting track brimmed with weird artifacts and possessed a bizarre, otherworldly character.

Radio Voice filters out lows and highs and gives you a very cool megaphone effect. Vocoder operates pretty much as you'd expect: a voice is input on the right channel and the signal to be altered is input on the left. Robot Voice tunes all notes in the incoming signal

VF-1 Specifications

Inputs	(1) ¼" Hi-Z guitar; (2) ¼" line; (1) ¼" TRS expression pedal/footswitches
Outputs	(2) ¼" line; (1) RCA coaxial S/PDIF
MIDI Connectors	In, Out
A/D Conversion	24-bit; 64× oversampling
D/A Conversion	24-bit; 128× oversampling
Sampling Rate	44.1 kHz
Dynamic Range	>97 dB
Program Memories	200 preset; 200 user
Display	backlit LCD
Dimensions	8.63" (W) × 1.75" (H) × 9.81" (D)
Weight	3.1 lbs.

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(which is preferably from a monophonic source) to a user-selected pitch. The formants can be tuned as well, making for some unusual vocal and rhythmic effects.

The Mic Simulator algorithm is a great idea, but it needs a little more work to live up to its billing. The idea is that smaller studios with limited mic closets can record an instrument with an inexpensive mic, and then process the track to make it sound as if it were recorded with an expensive condenser. I doubt that this will tempt musicians to forsake their condenser mics for a Shure SM57, but the Mic Simulator remains a flexible sound-modifying tool.

▼

The VF-1 employs a very flexible and easy-to-understand architecture.

I used the feature on an acoustic-guitar track (which was recorded with a Neumann TLM 103 mic) and easily shaped the sound by choosing different source and destination mics. But what really makes the Mic Simulator stand out is that it's stereo, with all left and right parameters independently variable. You can select different microphone models for each side and change the apparent distances between the mics and the source, thereby generating stereo from mono without having to chorus, delay, or double. You can also throw one side out of phase for a wider effect. This is very cool.

Other noteworthy effects include an intelligent pitch-shifter, hum eliminators (both 50 and 60 Hz), and vocal eliminators. Isolator splits a signal into three bands and then applies effects to each band. Lo-fi processors include bit filters that really grunge up a track and patches that simulate the sounds of an AM radio and a scratched vinyl record.

BOSS
VF-1 multi-effects processor
$495

FEATURES ■■■■
AUDIO QUALITY ■■■■
EASE OF USE ■■■■■
VALUE ■■■■■
1 2 3 4 5

PROS: Highly programmable. Clean, quiet 24-bit sounds. Plenty of user-patch storage. Lots of features and an intuitive interface.
CONS: Some of the amp models are disappointing. Manual is unclear on some subjects. Provides no way to quickly bypass effects on amp models.

THE BOTTOM LINE

I tested the VF-1 extensively in a wide variety of real-world applications, and in most cases I got good results. I did not expect it to replace a rack full of gear and a truckload of amps, and it didn't. However, it has some great sounds and is a very powerful tool, especially for musicians on a budget. I was a bit disappointed with some of the amp models, but I am very particular about guitar sounds, and I was comparing them with some of the best amps ever made. In most applications, particularly live performance, the models will do quite well, although you may have to tweak them.

The VF-1 packs a lot of power into a small box, and it will definitely find a niche in many personal studios. If all it provided were good-quality reverb, delay, and chorus effects, it would easily be worth the price. When you consider that it also offers amp modeling; pickup, microphone, and cabinet simulators; bass DI capabilities; and all of the other effects I've mentioned, you realize that it's quite a value indeed. Will large commercial recording studios buy lots of these units for their racks? I doubt it. For the rest of us, however, this is an extremely useful piece of gear and a valuable addition to any sound palette. I don't want to give it back, and that says it all!

Rick DiFonzo *has worked with many amazing artists, including the session singers who sang the themes of the* Gilligan's Island *and* Batman *television shows. He hopes one day to record with William Shatner.*


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A K G

C 3000B

One of the best affordable condenser mics just got better.

By Karen Stackpole

To B or not to B—that is the question. I, for one, was surprised when AKG replaced its enduringly popular C 3000 microphone with the new C 3000B. I have used two C 3000s in my standard location-recording rig for years and I have invariably captured good-sounding recordings with them.

Evidently, AKG felt that a little nip and tuck would spur the C 3000's sonic abilities to the next level. The company has done more than just add an additional letter to the mic's model number—it has also improved upon the C 3000's sound.

The AKG C 3000B has smoother mids and highs, fuller lows, and a hotter output than its predecessor, the C 3000. However, it offers only one polar pattern—cardioid—to the C 3000's two.

WHAT UP?

Like its predecessor, the C 3000B is a side-address, large-diaphragm condenser mic. The original C 3000 also contained a second, smaller-diaphragm capsule that sat on top of the larger diaphragm and enabled hypercardioid pattern selection. The design of the C 3000B dispenses with the secondary capsule, which means the new mic offers only one polar pattern—cardioid. But this is hardly a limitation. It seems that most C 3000 users rarely utilized the hypercardioid pattern, and considering the improvements in the mic's sound, I don't miss the hypercardioid option, either.

The C 3000B sports a brighter look than its matte-black forerunner, stepping out in AKG's snazzy silver-gray finish. The general shape of the new mic remains similar to that of the original, but the sturdy double-mesh grille is a bit larger, and it encompasses an improved, integrated, foam pop filter designed to better handle plosives. Internal shock-mounting shields the capsule from handling noise and stand-borne rumble.

The two-piece, die-cast metal body features a retaining flange on the connector shaft to prevent slippage from a stand adapter. Two recessed switches—one on each flank—allow selection of a 10 dB attenuation pad and a low-cut filter. What's novel about this particular filter is that it starts rolling off the lows at a rather high frequency—500 Hz. Typically, rolloffs start at 150 Hz or below (more on this later).

PACKAGE DEAL

The C 3000B comes with the H100 shock-mount, a spider suspension system that is now standard fare with many of AKG's studio mics. The fact that the mount is made primarily of plastic (including the threading on the stand adapter), and the fact that a simple fractional turn of the base is all it takes to lock the mic into the mount, left me with some doubts initially. When used correctly, though, the device holds the mic firmly and works well for its intended purpose. However, the strain-relief cable slots are too small to be of much use with cables thicker than average (Monster cables, for example), and the shock-mount's bulk makes it difficult to position the shock-mounted mic in tight places or to position the mics in a close XY pattern for stereo miking. Therefore, in stereo-miking configurations, I ended up using AKG's SA 41/1 stand adapters (not included).

Another "accessory" that comes with the C 3000B is a foam-lined cardboard box—now standard packaging for AKG's less-expensive mics. I find this new cost-cutting maneuver disappointing. Who wants to carry a mic around in a cardboard box? Bulky, easily torn or crushed, not water proof, and prone to deterioration from frequent use, this type of "case" is less than adequate for my purposes. I used the handy plastic carrying cases from my C 3000s to cart the C 3000Bs from venue to venue—a move that left me feeling better about my charges than if I'd carried them in a crappy cardboard box.

LOCATION IS EVERYTHING

I used a pair of C 3000Bs for a variety of applications—both on location-recording jobs and in studio sessions. I also conducted some controlled tests, comparing the C 3000B with an C 3000 and a C 414 B-ULS, and comparing one C 3000B with the other (to test for consistency). In most cases, I employed Focusrite Green-series preamps and Monster cables, and I recorded directly to DAT on a Panasonic SV3800. For the comparison tests, I routed the preamplified signals flat through a Mackie 1202-VLZ mixer.

My first gig with the C 3000Bs was at Yoshi's Jazz House at Jack London Square in Oakland, California, where I did a live recording of Full Throttle Orchestra, a large jazz ensemble containing saxes, flute, electric guitars, amplified double bass, and a drum set. During sound check, I put both the C 3000 pair (my standard workhorses) and the C 3000B pair side by side on stereo bars in front of the stage, about six feet from the forward row of musicians (horns) and at face level with the players. I routed the signals through my Mackie mixer to DAT for an A/B comparison.

The difference in sound was astonishing. Not only does the C 3000B have a smoother-sounding high end, it also has fuller lows, a better-represented

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midrange, and a hotter output than its predecessor. I quickly opted to use the C 3000Bs for the rest of the show. Indeed, I used them for all subsequent stereo-miking jobs throughout the test period.

Because I had occasion to work in jazz, rock, chamber, and film-music settings, I heard a multitude of instruments recorded through the C 3000Bs: saxophone, clarinet, trumpet, trombone, flute, electric guitar, acoustic upright bass, electric bass, drum set, marimba, electric violin, and cello. These instruments were primarily gathered in ensembles, and the mics were usually positioned three to eight feet away. In each case, the blend of sounds proved exemplary, and the client was pleased with the sonic quality of the finished recording.

When mastering live stereo recordings captured by my C 3000s, I usually temper the frequencies around 3 kHz, which is where the C 3000's presence boost is centered. The C 3000B's frequency response, however, is noticeably different, both on paper and to the ear. Rather than a boost, this mic shows a slight dip between 1 and 3 kHz on the frequency-response chart, as well as a rise that peaks between 6 and 7 kHz. These response characteristics jibed with what I heard and are in keeping with the smoother—yet still vividly defined—highs provided by the C 3000B.

Of course, the fact that I was working with talented musicians who used good instruments and had great tone also helped. A harsh-sounding saxophone or violin probably wouldn't get much help from the C 3000B, thanks to that 7 kHz peak.

LOWDOWN ON THE LOWS

In general, the C 3000B's lows are full and beefy without being muddy. In some cases, however, they can sound a bit boomy. Again, a comparison of the mic's frequency-response chart with that of the C 3000's is illuminating. On the C 3000B, the bass response is slightly—and very evenly—boosted from about 50 to 600 Hz; on the C 3000, there is a dip in bass response below 100 Hz. You can really hear this difference between the two mics.

I tested the C 3000B's 500 Hz rolloff on a location job in which the room was particularly resonant and the amplified upright bass somewhat unruly. Although the filter did an admirable job of taming the boominess, the rolloff frequency proved a bit high. It significantly reduced the presence of the upright, making it sound more like an electric bass, and it pulled all of the guts out of the bass drum. I opted to use the 75 Hz rolloff of the Focusrite Green preamp instead. The C 3000B's bass-cut filter might be put to better use in reducing proximity effect when you're recording, say, vocalists at distances of less than four inches.

SESSION WIZ

In studio sessions, I used the C 3000B to record mbira (a Zimbabwean thumb piano), electric sarod (an Indian stringed instrument), male and female vocals, and upright bass. The C 3000B captured the unique timbres of mbira without coloration. The attack on the metal keys and the warm resonance from the thumb piano were both well represented, and the buzz of the vibrating bottle caps attached to the instrument's surface didn't sound harsh or brittle. On electric sarod, I miked the amplifier, and the sound was accurately reproduced. However, the amp hiss was very prominent, accentuated as it was by the mic's 7 kHz peak. But it was a simple matter to notch out that frequency a bit, which helped cleaned up the sound.

▼

On vocals, the C 3000B sounded more like a C 414 than a C 3000.

For further comparison, I again put up the C 3000B next to the C 3000 and C 414 and then recorded drum set (with the mics positioned as overheads), an assortment of percussion instruments (including hand drums), acoustic and electric guitars, and male and female vocals. The C 3000B performed impressively in each application. As drum overheads, the mics captured a more open sound than the C 3000s, with fuller lows. In contrast, the C 3000s produced more definition in the high mids—which is not a bad thing when you need the drums to cut through a mix. Overall, though, I preferred the sound of the C 3000Bs: there was still plenty of definition, but the sound was richer and smoother.

The C 3000B came through with flying colors on a variety of percussion instruments, too, including shakers, wood blocks, gongs, dumbek, and tambourine. The mic captured lots of tone, good definition and attack, and a smooth, agreeable high end. The high mids were less "crispy" sounding than the C 3000's, which is to say the C 3000B has a warmer sound—more akin to the C 414's, albeit with a bit less control in the frequencies below 150 Hz.

The C 3000B's fuller lows can work for or against you, depending on the

C 3000B Specifications

Type	side-address, external-DC polarized ("true" condenser), pressure-gradient transducer
Diaphragm	1", gold-sputtered Mylar
Polar Pattern	cardioid
Frequency Range	20 Hz–20 kHz
Dynamic Range	126 dBA
Signal-to-Noise Ratio	80 dBA
Sensitivity	25 mV/Pa (132 dBV)
Self-Noise	14 dBA
Maximum SPL (for 0.5% THD)	140 dB (150 dB with pad)
Highpass Filter	6 dB/octave below 500 Hz
Attenuation Pad	–10 dB, switchable
Power Requirement	48V phantom
Dimensions	6.4" (L) × 2.1" (D)
Weight	11.3 oz.

For manufacturer information, please see Contact Sheet, p. 221.

situation. For example, they helped impart a beefier tone to a biting electric bass guitar that I stereo-miked for a film-music ensemble. On the other hand, the C 3000B sounded a bit boomy (though not unpleasant) on a steel-string acoustic guitar. In comparison, the C 3000 better accentuated the pick sound (which would serve well in a mix) and captured more high-end sparkle. The C 414 was again more controlled in the lows, and it possessed smoother highs than either of its less-expensive counterparts. Still, although each microphone offered a different perspective of the same instrument, they all sounded good.

On nylon-string classical guitar, the C 3000B sounded rich and warm, and here the low-end resonance was not so overpowering. However, on an electric guitar I recorded, the C 3000B accentuated a midrange-type quality that was a bit grating to my ears. (The C 414 had a smoother, more controlled sound in this application.)

SMOOTH VOCALS

I was very impressed by the C 3000B's performance on male vocals. My subject had a smooth, deep voice, and the C 3000B complemented his vocals beautifully. The smoothness in the upper mids (between 2 and 4 kHz) resulted in the mic's sounding more like the C 414 than the C 3000. In fact, I actually preferred the C 3000B over the C 414 on this particular singer. The mic provided a nice "breathy" quality without sounding overly bright, and it also offered good low-end support.

Female vocals also fared well in the comparison tests, with the C 3000B again sounding more like a C 414 than a C 3000. Also, compared with the C 3000, the C 3000B handles plosives better, thanks, evidently, to the built-in windscreen.

PETE AND REPEAT

My two C 3000Bs sounded virtually identical, but I did discover a 3 to 5 dB discrepancy between the two mics' output levels. This was somewhat disconcerting, but I compensated for it with preamp gain. Just the same, if you intend to buy two C 3000Bs for stereo-miking purposes, I suggest that you make the effort to find two with closely matching output levels.

WELL, I'LL B

In addition to being an affordable and great-sounding microphone in its own right, the AKG C 3000B is a real step up from its popular predecessor, the C 3000. Thanks to its full low end, warm mids, and smooth yet well-defined highs, this mic is well suited for many applications. Although the mildly boosted lows can sound boomy in certain cases, they are an asset in other situations. Likewise, the C 3000B's 500 Hz low-cut filter, though radical for some applications (stereo-miking ensembles, for example), could serve well to reduce proximity effect during close-miking. Most preamps and consoles, after all, provide low-cut filters positioned somewhere between 75 and 100 Hz, so the C 3000B's 500 Hz rolloff can be seen as a feature that offers another sound-shaping option.

AKG has done a great service to the recording community by upgrading the C 3000. A very universal microphone, the C 3000B would be a great addition to any mic collection and should have special appeal to personal-studio owners and other budget-conscious recordists. I'll B, I've decided.

AKG
C 3000B large-diaphragm condenser microphone
$520

AUDIO QUALITY ■■■■■
VALUE ■■■■■
1 2 3 4 5

PROS: Full low end. Warm-sounding mids. Smooth, detailed highs. Durable. Integrated windscreen. Includes H100 shockmount.

CONS: Cardboard-box carrying case. Low end can be boomy in certain situations. 500 Hz bass rolloff is quite high. Output difference of 3 to 5 dB between the two test microphones.

Karen Stackpole *is a recording/mastering engineer and an active drummer/percussionist. She operates Stray Dog Recording Services and is an instructor at the Ex'pression Center for New Media in Emeryville, California.*

ARBORETUM

HARMONY 1.0 (MAC)

Backup vocals at the push of a button.

By Thomas Wells

Arboretum's *Harmony* is a formant-based pitch shifter, corrector, and harmony processor that the company calls a "word processor for vocals." The software offers 9-voice mixing and the ability to make both macro and micro adjustments to audio files through a simple and intuitive graphical user interface. These adjustments are never destructive; you can always return to the original file.

Using the company's proprietary formant-shifting processing, *Harmony* retains realistic vocal and instrumental sound quality even after transposition (avoiding the "Chipmunks" effect). It also opens up a wonderful world of possibilities for creating fantastic vocal sounds for all kinds of sound-design applications. Not intended for real-time processing, *Harmony* is an offline arranging tool.

LET'S HARMONIZE

Harmony is not an independent application, but a plug-in for Arboretum's *HyperEngine,* a simple freeware program that allows rudimentary editing of sound files and acts as a manager for several Arboretum plug-ins. *HyperEngine* can also process sounds from various sources in real time, depending on the capability of the plug-in. A simple panner, lowpass filter, and ring modulator come free with *HyperEngine,* and demo versions of other plug-ins, including *Harmony,* are available at the Arboretum Web site.

Harmony 1.0 is designed to process files with single-voice, monophonic melodic material. The mono restriction is absolute: *Harmony* will not process stereo files. On the other hand, it will happily accept single-channel, polyphonic sound files, with some unpredictable and intriguing results. Although it is billed as a vocal processor, it does a great job on single-line instrumental parts as well. In addition, the program's speech processing—which lets you change inflection, gender, age, and so on—is terrific. *HyperEngine*'s I/O capabilities include support for the Korg 1212 I/O, Digidesign Audiomedia II and III sound cards, and Apple's Sound Manager.

Typically, you begin a *Harmony* session by loading a sound file into *HyperEngine* and selecting the region you want to work on. (The program supports both AIFF and SDII formats.) Next, you choose *Harmony* from *HyperEngine*'s Plug-Ins menu, and the main *Harmony* window appears with the Notes subwindow (the default subwindow) selected (see **Fig. 1**).

Harmony
Minimum System Requirements
Mac PPC/200; 22 MB RAM; OS 8.0; Web browser for navigating documentation and help files

Along the window's left side are buttons for selecting among the four subwindows (Notes, Mixer, Voices, and Process), as well as the Effect In/Out button. To the right of the buttons, consuming most of the display area, is a graph plotting pitch over time. You create a new plot by pressing the Calc button. Across the top of the graph are the tool and Voice selection buttons.

PICK A TOOL

To change the pitch graph, simply select a segment of the graph, which turns yellow. When you move the mouse over the selected segment, the Selection tool's crosshairs turn into a hand, which lets you transpose the segment up or down. The transposed segment then appears in its new location, while the original turns purple—a thoughtful touch. Equally useful is the "pilot light," a small green dot that appears at the bottom left of each tool button you select.

Use the Pencil tool to draw pitch curves for selected segments. The pitch curve is highlighted in yellow, and the original pitch segment turns purple. If you're not satisfied with the effect, redraw it or hit the Redo button to cancel the last edit. The Reset button lets you return to the original pitch graph and start over.

Use the Resize tool to reduce a segment's frequency deviations around a center, or average, pitch. This tool is useful for editing vibrato on individual notes and compressing or expanding a segment's range.

The Zoom tool, represented by a magnifying-glass icon, is simple to use. Click-dragging a rectangle around an area enlarges the area to fill the display; holding down the Option key and clicking in the display area zooms it back out. The small thumbwheel at the bottom of the display lets you scroll

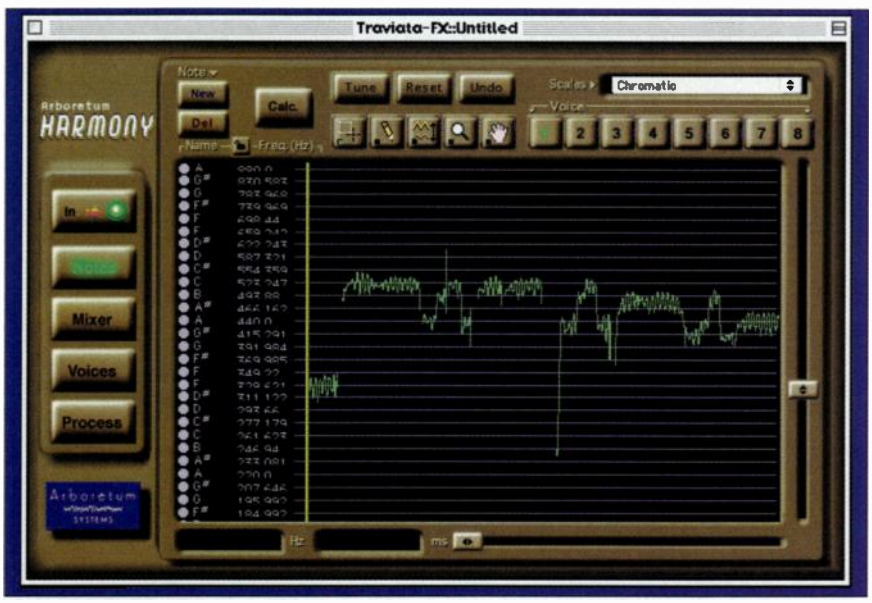

FIG. 1: *Harmony*'s Notes window shows the pitch curve of the currently loaded sound file. Files can conform to numerous scales, which you choose from the Scales pop-up menu in the upper right of the screen.