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AUGUST 1986

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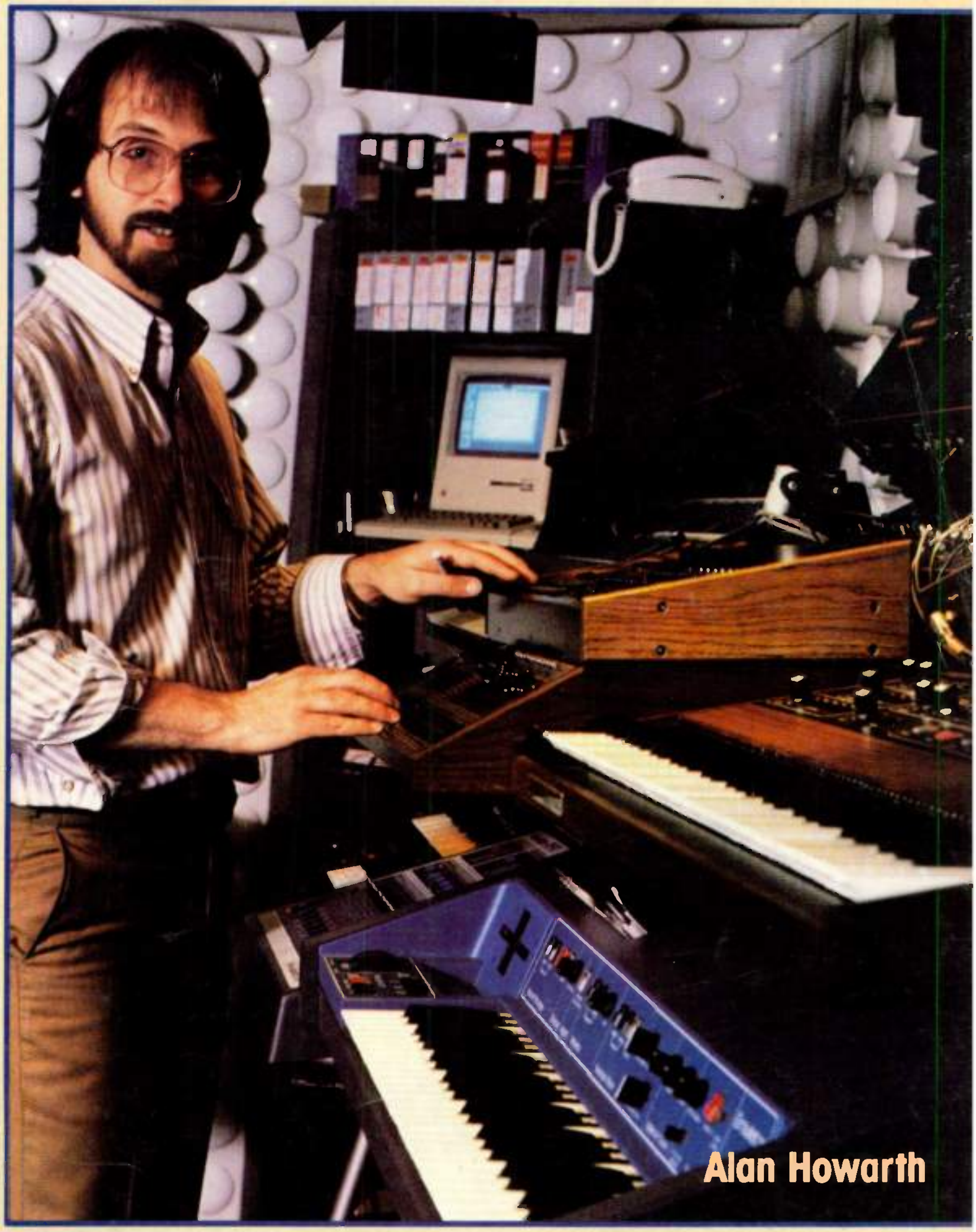
Interview:
Alan Howarth



Alan Howarth

Focus on Do-It-Yourself

A MIX PUBLICATION

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SOUND DESIGNER™ provides all the tools you need to design virtually any sound. Used with your digital sampler and a Macintosh,® SOUND DESIGNER offers many powerful *digital* sound editing and processing features previously available only in *very* expensive computer music systems. And unlike other systems, SOUND DESIGNER is easy to use. It works like this:

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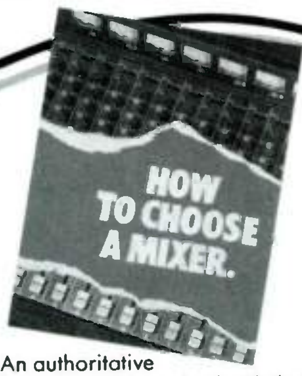

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VOL. 2, NO. 8 AUGUST 1986

Electronic MUSICIAN

A MIX PUBLICATION

DEPARTMENTS

COVER

It's a long road from working in a music store to working with the best Hollywood movie-makers, and not too many people can make the grade. But Alan Howarth's unique combination of technical and musical ability has landed him work on some of the biggest films of the decade. In this month's issue, Alan tells what it takes to reach the top of the Hollywood heap, and shares some of his secrets for successful cinematic synthesis. Photo by Ann Summa.

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ARTICLES

Editor's Note

It's summer...time to head to the lake or beach, catch up on some reading, watch a few well-hit balls work their way out to centerfield, or...tinker around with some of your musical tools.

Just about every musician I know has customized some aspect of his or her setup. Maybe it's replacement pickups for a guitar, new drum chips for an old drum machine, or something complex—like installing a memory expansion or MIDI retrofit in an older piece of gear. Whatever the case, musicians seem to have an urge to shape their tools to fit their specific needs, and all of us at *EM* encourage that kind of approach. Yeah! Give us some new sounds, some new ideas, and express your *own* personality through the way you make a musical instrument uniquely yours.

Many people, though, are still afraid to pop open their gear and poke around...and with good reason. It takes a certain amount of courage to void the warranty and risk the chance of blowing something up inadvertently. But one does not become a musician, or an equipment modifier or builder, overnight. Start your do-it-yourself career with something simple, like building your own cords or making a simple cable tester. If you make a mistake, you'll only be out a little bit of bucks and time. However, it has been my experience that if you start simple enough, and at least know the very basics of electronic construction, you can usually succeed with simple projects. If you're starting at ground zero in terms of knowledge, go to the library. There are several books on do-it-yourself electronics available.

Where most people run into problems is with their second or third project, when, flushed with success from previous efforts, they attempt something a little bit beyond their reach. The first reaction is anger ("how dare this thing not work!"), then reflection ("maybe there was a typo in the magazine...") to analysis. During the process of analysis, when you discover that you forgot a wire, or a connection was not good enough, or an IC was inserted backwards, or if indeed there *was* a typo, you learn how to think logically and solve the problem. This is actually the most educational aspect of doing-it-yourself, even though it's the most frustrating until you figure out what's wrong. I must confess I learned a lot more when a project *didn't* work the first time than when it did.

In addition to our regular editorial coverage, we have a bunch of articles to get you into the spirit of personalizing your equipment and customizing your setup. Yes, the extra performance *is* worth it, as is the hands-on experience in finding out what makes the gear really tick (which can literally save your gig if those skills ever help you do a quick repair someday). Most importantly, though, comes the sense of satisfaction and confidence that comes only from being able to say..."I did it myself."

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MIDI

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MIDI Overflow Mode for slaving units
8 simultaneous polyphonic channels
MIDI remote programming
MIDI guitar controller compatible

If you're interested in a high-performance synth, it's time to test drive an Ensoniq ESQ-1 Digital Wave Synthesizer. It puts 120 sounds at your fingertips as fast as you can switch it on and plug in a cartridge. But that's only the beginning.

In addition to standard synthesizer waveforms, the ESQ-1 features complex multi-sampled waves for a total of 32 waveforms on board. Each of the ESQ-1's 8 voices uses 3 digital oscillators with the ability to assign a different waveform to each oscillator. That's thousands of distinct sonic possibilities.

The ESQ-1 is simple to program because it lets you see what's really going on inside. Its 80-character lighted display shows ten programs or parameters simultaneously. So you'll spend less time writing down numbers and more time laying down music.

A built-in 8-track polyphonic sequencer makes the ESQ-1 an ideal MIDI studio. Each track can play internal voices, external MIDI instruments, or a combination of both. And each track can be assigned a separate program and MIDI channel. Like any good studio, the ESQ-1 can auto-correct timing, auto-locate passages and balance individual tracks during mixdown.

You can build songs made up of 30 different sequences and store them internally, externally on tape or on 3.5" diskettes using the Mirage Sampling Keyboard or Multi-Sampler.

If controlling other MIDI instruments is on your list of priorities, the ESQ-1 puts you in the driver's seat. It supports poly, omni and mono modes along with Ensoniq's multi and overflow modes that extend the MIDI capability of the ESQ-1 far beyond ordinary synths. You won't ever have to leave the comfort of its 61-note weighted, velocity sensitive keyboard to play any MIDI instrument in your setup.

Comparable high performance digital waveform synthesizers and MIDI sequencers can easily exceed the legal limits of your cash on hand. But the good news is that the ESQ-1 comes from Ensoniq—at a sane price of just $1395. For a glimpse of technology that's earned the name "advanced", put an ESQ-1 through its paces at your authorized Ensoniq dealer today.

Although you should always fasten your seat belt when playing the ESQ-1, you don't have to wear a helmet or obey the 55mph speed limit. ESQ-1 and Mirage are trademarks of ENSONIQ Corp.

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The Big Blue Blues

I just received my first copy of EM (May '86) and though overall it is a great and useful publication, J.B. Moore's article "The Big Blue Music Machine" was highly biased. Although I'm used to the 68000 microprocessor being put down in the usual corporate computer world, I don't want to see this happen in the music magazines too.

I'm mighty sick of the implication that the 68000 is only good for graphics and games. As for cost, I'd hate to think how much I'd have to pay for an IBM with 8 MHz clock speed, 1 Megabyte memory, two dual-sided 800K mini-floppies, windowing, color card, monitor, and MIDI interface. It certainly wouldn't be about $1,200, like my Atari 1040ST!

Worst was the DOS Diatribe. Freff may be able to teach beginners the basics of PC-DOS in 20 minutes, but I can teach someone those same functions on the ST or Mac in two minutes. I will grant Mr. Moore that there are more peripherals around for the IBM PC, and the documentation on my ST is seriously lacking. I'm not anti-IBM, in fact I just bought an IBM printer (although naturally the control codes are not Epson-compatible—Big Blue doesn't want to make it easy for other computer users).

Thanks for letting me rave awhile. I just didn't want some musicians to think that IBM is the only serious machine on the market.

Mick Seeley
Oceanport, NJ

Mick—The IBM story was one of a continuing series; we have also covered the Macintosh (February '86), Amiga (April '86), Atari ST (June '86), and this month, the CX5M. Each type of computer has its aficionados; the main point we're trying to make by running these articles is that there are many choices available in the world of computers, so choose the one that's best for you. Freff owns and uses both the Mac and PC, likes them both, and uses them for different purposes. I think his main motivation for writing the diatribe was because he didn't want some musicians to think that 6800-based devices are the only serious machines on the market.

MIDI Length Limits

I realize the MIDI spec calls for a maximum cable length of 50 feet, but is it possible to recondition or amplify the signal so that it can run between console and stage on a 120-foot snake? This would be excellent as our engineer could "perform" a number of programming tasks, thus freeing the musicians for playing music instead of pushing buttons.

Lloyd S. Mandola
Hollywood, FL

Lloyd—There are several ways to overcome MIDI cable length limits. One would be the Nady or similar wireless MIDI system, which is particularly applicable to fixed installations. Also, **EM** *author Jack Orman has used fiber-optic systems to transmit MIDI data over 4,000 feet, and has also experimented with a balanced line MIDI system that can send data over hundreds, if not thousands, of feet. He will be submitting an article on at least one of these approaches for publication in an upcoming issue.*

Hammond Condor

Having followed guitar synthesizers since their early days, I must take exception to the statement in "A Brief History of Guitar Synthesizers" (April '86 **EM**) that the Hammond Condor was only a fret-wired device. I own an entirely original Condor with the stock combination hex pick-up/tuneamatic bridge (mounts right on the bridge studs of a Gibson type guitar), circa 1969. Incidentally, I find **EM** very interesting and informative. Congratulations and continued success.

H. David Sarge
Abington, PA

(Those who are interested in Mr. Sarge's Hammond Condor can contact him at 2132 Rush Rd., Abington, PA 19001.)

Magnetic Questions

I was demagnetizing a tape deck and let the on-off switch slip while the tool was within about two inches of the head. I am now afraid that the heads may be re-magnetized or worse yet, permanently magnetized. In your *Home Recording for Musicians* book you mention the R.B. Annis company as a source for magnetometers, but I wrote them and my letter was returned. Can you help?

Mark Osburn
Fort Smith, AR

Mark—R.B. Annis products (the magnetometer and companion demagnetizer) are available from Polyline Corporation, 1233 Rand Road, Des Plaines, IL 60016. Tel. 312/297-0955.

Information Please

There I was just minding my own business when an issue of your magazine appeared in my mail. I liked it so much that I now subscribe. I currently own a Commodore 64 computer, but a lot of your articles appear at first glance to be beyond my beginner's knowledge. I will continue to grow and learn, but can you recommend any books that would be helpful to me on the MIDI system? Are there any books that pertain specifically to the C-64? Is there a User's Group that involves the C-64 and MIDI? If not, I would like to start one so various people could write to each other and hopefully help each other with their C-64 systems.

Please publish my name and address if this will help get C-64 users together. There are a lot of us out here who are just getting started and need all the help we can get. Thank you very much.

Lawrence D. Rogers
P.O. Box 4
Venice, FL 34284

Lawrence—Yes, working with computers does seem bewildering at first, but eventually the pieces will fall into place. For further MIDI information, see my book MIDI For Musicians *and check into the International MIDI Association (11857 Hartsook St., North Hollywood, CA 91607).* Run *and* Compute Gazette *magazines are devoted to the C-64;* Compute! *and several other magazines have articles of interest as well. These are available at major (and even some minor) newsstands. And of course. . .*Electronic Musician *will continue to cover musical applications of the C-64, and we expect to have more beginner-oriented articles as time goes on. Regarding books, there are several out for the C-64. A trip to a major metropolitan book store will unearth several titles, from books about hardware and interfacing to how to program in Commodore BASIC. Good luck!*

Complimentary Issue Compliments

When your complimentary March issue arrived some time ago, I must admit I was not at all pleased. I'd gotten so tired of the music/tech press, with its increasing bias towards sycophant hero-worship and equipment-envy.

It was, therefore, with heavy heart that I carried your issue home and grudgingly cracked the cover. Not until four hours later—having read every word, digesting even whole articles about equipment I'll never own and explanations of no personal use, did I realize I had just devoured the most enjoyable, informative, well-written and presented issue of *any* music publication I've ever seen. From Freff's elegant technical writing to the amazingly clear and complete circuit pieces, to the great layout and design, **EM** strikes a balance unachieved, to my knowledge, in any previous music-tech


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EARLY REFLECTIONS

Engineers talk about their new discovery: the Ibanez SDR1000 Stereo Digital Reverb

"The combination of performance and value is great! Rather than exhausting your entire reverb budget on one high-end unit, you can have a rack full of these. This puts many different environments at your disposal".

Jan Hammer: Jan literally has done it all. He came to public attention trading leads with John McLaughlin and the Mahavishnu Orchestra. Jan then released a series of LP's as a soloist and with his group "Hammer". Afterwards, Jan toured with the Jeff Beck Group, cut an LP with Neil Schon, and started composing for TV and movie soundtracks. Jan's most recent work for the "Miami Vice" TV series has won him even wider critical acclaim. Jan's soundtrack LP from the series won a Grammy in 1985.

"The SDR1000 is a great unit! To call it a digital reverb is too one-dimensional because of its across-the-board capabilities. It's the most cost-effective processor we have in the studio."

Lance Quinn/O.B. O'Brien: Lance and O.B. have been extremely busy since opening the Warehouse Studios. Some of their most recent credits include Bon Jovi's "7800 Degrees" gold LP, Lita Ford's "Gotta Let Go", Nils Lofgren's "Flip Ya Foot" and the first album by Cinderella".

"For today's mixes, where I use as many as 10 separate reverb units, the SDR1000 is an extremely effective, flexible tool for a wide array of effects. The reverb algorithms give smooth, fat sounds that work great, whether they're used for large chamber or small room sounds."

Ed Thacker: Ed is one of the most sought-after free-lance engineers on the west coast. Ed has worked on Glen Burtnick's "Talking In Code" LP and Glass Tiger's "Thin Red Line" album. Ed also engineered the soundtrack for the movie "Goonies", including Cindy Lauper's track "Good Enough".

The SDR1000 Stereo Digital Reverb is a true dual-processor that delivers strikingly warm, full reverb sounds. 30 factory presets, created by top-chart engineers, and 70 user presets put a virtual sonic universe at your fingertips. Fully programmable (there's even a programmable on-board EQ!), the SDR1000 has easy-edit sound creation software and MIDI-controlled patch recall to make it the "friendliest" reverb processor with this much flexibility. So whether you need a true stereo processor or two distinct reverb sounds simultaneously, the SDR1000 deserves your serious reflection—at your Ibanez dealer now!

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magazine. Sign me up! I look forward to your next arrival in my mailbox.

Paul Wagner
Santa Cruz, CA

CZ Multiple Outputs?

In the May '86 EM, the editor hints that Alan Gary Campbell has more "CZ Secrets" in store for us. Personally, I'm hoping that Alan or someone will come up with a "mod" that provides the Casio with four individual audio outputs when used in multi-timbral mode. Now *that* would be hot! Also, it would be nice to be able to choose which channels will be assigned to which voice, instead of always having them assigned in succession, but I'd be willing to bet that that function is buried in a chip somewhere and not easily changed. But check into that multiple output modification idea, will you? If not for me, then for the other 500,000 people who own a Casio!

Bob Tyler
San Diego, CA

Bob—We asked Alan about your question and he replied: "I frequently get requests for the individual output mod that Bob describes. Unfortunately, the polyphony of the CZ instruments is a software function; there is only a single DAC/expander/sample-hold circuit that outputs the composite audio signal for all voices/channels, combined. To obtain separate outputs, you would have to multiplex four DAC circuits, and rewrite the Casio's operating system to drive the additional circuitry—a major (read: expensive) undertaking, at best. With the continued price reductions on CZ synths—despite the yen/dollar fluctuations—it's probably more cost-effective to buy three additional CZs!

"The channel-assignment problem, however, is fixable if someone (not me, please!) creates a new CZ operating system in ROM (as described in the CZ Mods article). I've heard rumors of this sort of thing being in the works, but nothing definite.

"Bear in mind that Casio is not deaf to user feedback. It wouldn't surprise me if they came out with a multi-output, rack-mount, velocity-sensing CZ in the near future."

Error Log

Regarding Jim Chandler's Bassmod article (February '86), several people have asked for specs on diodes D1-D4. These can be any general-purpose silicon types, such as 1N4001, 1N4002, 1N914, 1N4148, etc.

Sorry to disappoint all you Steve Sagman fans, but Industry Update is taking a vacation this month so that he can finish up his feature article on telecommunications for our next issue. EM

Arnold Mathes, *Stranger from the Depths* (cassette). Arnold Mathes is possibly the world's most prolific experimental electronic composer. On this, his 16th tape, he gives a semblance of order to his organized chaos by introducing a Roland TR909 drumbox. He also makes use of an Electro-Harmonix Instant Replay for some bargain sampling. VCS-3, minimoog, ring-mod talking and various other sources fill up the tape with a lot of sonic variety. There's even some movement within the pieces this time. Mathes never fails to confound expectations. $5 postpaid from 2750 Homecrest Avenue, Brooklyn, NY 11235.

Dennis Andrew, *Concepts* (Day-01; cassette). For his debut tape, Andrew has chosen a synthesizer with a sequencer to burble out front while he does light extemps in the background. Generally using organ-like voices, the tape has almost a Terry Riley feel and the pieces fade in and fade out, indicating they could go on forever. Daylight Productions, P.O. Box 284, Metuchen, NJ 08840.

Jean Michel Jarre, *Rendez-vous* (Dreyfus 829125-2). There was a definite progression in Jarre's first five albums, moving from the simple and accessible toward elaborate sampling and some pretty "outside" composing. Perhaps feeling that he was leaving his listeners behind, for his sixth *Rendez-vous* the comet Jarre returns to such pre-Cambrian synthesizers as ARP 2600, Moog, Oberheim and Synthi AKS. The music is also a return to simpler days, although for those of us who revered his early works that isn't necessarily bad news. One piece here was going to feature Ron McNair in the first solo ever recorded in space, but the ill-fated Challenger dictated a change in saxophonist.

Cule Roebling, *Convenient Memory* (cassette). Roebling sings like a seasoned professional, and backing her are Des Modems, a studio band employing, altogether, 30 musicians. These eight tunes are consistently good despite, or maybe because of, twelve songwriters and seven engineers in six different studios. It plays like any of the better stuff on FM radio. I give it a 63. This tape is available for $6 from P.O. Box 7521, Stanford, CA 94305.

Lila, *Amethyst* (Lila 01). Peaceful multi-part music on DX7 and Prophet, perhaps best described as Tangerine Dream without the sequencers. A couple tunes do feature digital drums, so don't get the impression it's totally ethereal. Recording on a Tascam 38 gives noticeably more depth of field than their first release, 1984's "Illuminatus" single, recorded on a 3340. The album shares the single's rapidly-changing, intelligently-arranged character—not unlike Richard Souther (come to think of it) whom you may remember I praised shamelessly 4/86. U.S. distribution by EUROCK, P.O. Box 13718, Portland, OR 97213.

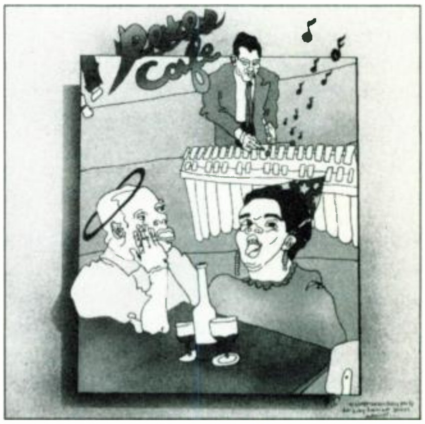
GTR, *GTR* (Arista 8-8400). The long-awaited collaboration between Steve Howe (ex-Yes) and Steve Hackett (ex-Genesis) seems an unlikely pairing on the surface—how many lead guitarists does a group need? But these days groups are often formed for business as much as artistic reasons, and the draw of this collision is unmistakable. Fortunately, they've hired a vocalist (as well as a bassist and drummer), though the vocal lines sound as though Hackett and Howe are writing to their own limitations. The music is fairly typical corporate rock—bland, unchallenging, lacking in any clear clues as to who's playing. Asia is probably the closest parallel, musically, financially and potentially.

Larry Carlton, *Alone/But Never Alone* (MCA 5689); **Al Di Meola, *Cielo e Terra*** (Manhattan 53002); **Marc Johnson, *Bass Desires*** (ECM 1299). Both Steve H.'s are credited with synthesizers in *GTR* above, but their guitars dominate the mix. By contrast, here are three albums by jazz guitarists dominated by synthesizers or guitar synthesizers. Carlton's is the most mainstream, playing Earl Klugh foppish jazz over delicately tasteful DX7 backgrounds by himself and Terry Trotter. Di Meola's Synclavier guitar blends in with his natural acoustics, creating precise environments with just himself and a percussionist. Johnson's a bass guitarist, but Bill Frisell's guitar synthesizer weaves non-stop textures behind the quartet. We're starting to see, as guitar synthesizers develop, bands with both a guitarist and a "guitar synthesist." 'Bout time.

Robert Greenidge/Michael Utley, *Mad Music* (MCA 5695). Synthesizers and steel drums may seem like strange bedfellows, but these two members of Jimmy Buffett's band make them seem like lifelong companions. Greenidge's style on the pans reflects his native Trinidad more than, say Andy Narell's jazz panwork, and digital drums lend a more rockin' rhythm than stretched-out jazz. Perhaps this is Buffett's influence in absentia.

John Jarvis, *So Fa So Good* (MCA 5690). Keyboardist Jarvis has contributed to sessions from blues, folk, soul, rock and country to his latest incarnation as a New Age pianist. These 11 tunes on DX, Rhodes and acoustic piano (drums by Linn) feature an elegant simplicity which most recalls Vangelis's theme from *Chariots of Fire*.

MCA Master Series, *Sampler '86* (MCA 5692). The broad appeal of "specialty labels" such as Windham Hill, Meadowlark, Fortuna, Elektra Musician and the granddaddy of them all, ECM, has encouraged corporate giant MCA to come out with their own high-quality coordinated sub-label, MCA Master Series. This one is dedicated to showcasing different sides of the instrumental talent around Nashville, including Carlton, Greenidge/Utley, and Jarvis above, as well as three other acts less suitable for *Electronic Musician*. It's nice to know big labels can still do it right when they've a mind to.



Peter McNutt, *Pete's Cafe* (T.M.O.Q. Spheen Records). Not all great productions come out of the majors, of course. McNutt's slick fusion-jazz LP was recorded at home on a Tascam 244 and features two percussionists and a reedman. McNutt, amazingly, plays everything else on a bevy of manually and computer-controlled keyboards. In sound quality and everywhere else—except maybe cover graphics—McNutt puts the shame to the majors. 5314 Avenue "H," Austin, TX 78751.

Cosmic Debris, *While You're Asleep* (Non Compos Mentis Records). Cosmic Debris is a jazz quartet with Richard Bugg on E-mu modular (doubling on flute), Dennis Borycki on Oberheim 4-Voice (doubling on piano), John Powell on vibes and percussion, and Joel Young on drums. As you might expect from the lineup, the material is a bit more "electronic" than mainstream, but they do a good job of keeping it firmly in the jazz realm. 2703 NW 20th, Oklahoma City, OK 73107.

The New Hippies!, *It's Not What You Eat . . .* (cassette). You may remember David Snow from his *Passion and Transfiguration of a Post-Apocalyptic Eunuch*, reviewed in the very last issue of *Polyphony* (4/85). Well, he's still combining jazz piano and synthesizer with Zappaesque spoken interludes and assorted other lo-fi hardware. Snow advises that this tape was "mixed with low-level white noise and 60Hz hum to impart that warm 'analog' ambi-

More out of midi.

Axxess Unlimited's Mapper dramatically expands the performance capabilities of any *midi* instrument, no matter how sophisticated. Guaranteed.

The Mapper gets its name from the latest development in *midi* technology: *midi mapping. Maps* are electronic templates that redefine *midi* codes to perform tasks never before possible. And the Mapper is the only device that takes full advantage of *midi mapping*. Here are just a few of the basic functions of just one single *map:*

Note and Channel Reassignment

Any key or drum pad can be *mapped* to play any note or notes. Each note can be sent on its own combination of channels. Which means that you can have any number of layers and overlapping splits.

Example

MASTER NOTES

SIMULTANEOUS POSSIBILITIES

Channels 5 & 6

Channels 2 & 4

Channel 10

Performance Setup

The Mapper can send any *midi* messages on any channel at your command. That includes patch changes, control presets, and exclusive commands.

Example

MASTER PATCH SELECTION

SIMULTANEOUS POSSIBILITIES

Select patch 10 on channel 9

Select patch 38 on master keyboard

Set pitch range on channel 5 to a fifth

Set maximum volume on channel 8

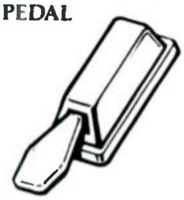
Set *midi* controlled lighting board

Switch Redefinition

Midi switches, such as the sustain pedal, can be assigned to any selected channels, or redefined to be any *midi* message.

Example

MASTER FOOT PEDAL



SIMULTANEOUS POSSIBILITIES

Decrease volume on channel 6

Turn on three *midi* controlled effects

Change program on channel 2, until pedal is lifted

Cross Modulation

A master control (mod wheel, pitch wheel, after touch, breath, foot modulator...) can affect each of its several slave controls in dozens of ways. Master controls can also be used to change exclusive parameters.

Example

MASTER PITCH WHEEL CONTROL

SIMULTANEOUS POSSIBILITIES

Reverse pitch channel 8

Increase volume channel 6 & 8

Increase exclusive parameter 83, channel 2

Using the Mapper

Your keyboard is the Mapper's main programming tool. The Mapper itself has two push buttons, a foot pedal, a 24 × 2 LCD display, two *midi* ins, two outs, and an RS232 port. The Mapper will store 128 *maps* in internal memory.

The Mapper is available in either a one-slot rack mount, or table version.

Why Wait?

This is the most exciting *midi* product available today. So if you want to expand your potential and have fun doing it, get your hands on a Mapper now!

THE MAPPER

P.O. Box 8435 • Fort Collins, CO 80525 • 303-482-5656 ence." $6 buys a bundle of fun from 9824 Maple Leaf Drive, Gaithersburg, MD 20879.

Hugh Le Caine, *Compositions and Demonstrations 1948-1972* (WRCI 4299). Hugh Le Caine (1914-1977) was a Canadian pioneer in electronic music instrument design, starting with 1948's Electronic Sackbut. In all, he designed some 15 one-of-a-kind instruments, everything from a touch-sensitive organ to a variable-speed Mellotron-like contraption. Although his lack of marketing acumen (or interest) kept him from becoming as well known as Robert Moog, Alan Pearlman or Donald Buchla, his "compositions and demonstrations" make a fascinating historical footnote, with exhaustive liner notes. $12.50 (Canadian) from The Hugh Le Caine Project, 146 Ridge Road West, Grimsby ONT, Canada L3M 4E7.

Ricky Starbuster, *Live The Dream* (cassette). Back in 1983-84, Starbuster released three cassettes and an album of his heavily-orchestrated electronic music. After some dissolution and disillusion, he's back with 90 new minutes on Prophet 5, Pro-1 and TR-606. He still plans on being "the best synthesizer artist in the world," though the competition has gotten pretty stiff in the last two years—*Live The Dream* is promising but not as elaborate as Lila. Quick, send him $10 for the cassette so he can get the PCM and sampling keyboard he deserves. P.O. Box 5582, Madison, WI 53705.

Steven Cooper, *Crystal Garden* (AM 132; cassette). Mastered to PCM and using several digital keyboards, Cooper shows what high-technology can do for New Age. Natural and synthesized environmental recordings are combined with slow-moving keyboard meditations which, due to their digital complexity, manage to maintain interest. Like his *Transcendence* (reviewed 2/86), *Crystal Garden* shows the intelligent side of the new minimalism. $8.98 from Valley of the Sun, P.O. Box 38, Malibu, CA 90265.

Steve Roach, *Empetus* (Fortuna 036). Roach has already had a tape *(Now)* and album *(Traveler)* of sequencer-driven synthesizer music, and an album *(Structures from Silence)* of meditative drones. His new release has been eagerly awaited to see what direction he'd take it. *Empetus* is rhythmic, using digital drums and fast sequencer patterns, but also shows something of the extended pursuit of absolute sound in *Structures*. The EUROCK catalog described him as "one of America's most adventurous and interesting synthesists," and it'd be hard to argue with that viewpoint. EUROCK Distribution, P.O. Box 13718, Portland, OR 97213. EM

Operation Help

Operation Help *is dedicated to helping musicians help each other. If you need technical assistance, a schematic for some old piece of gear, or just want to connect with people having similar interests, send your name, address, phone number (optional), and nature of your request to* Operation Help, Electronic Musician, *2608 Ninth St., Berkeley, CA 94710. There is no charge for this service, but we cannot guarantee that all requests will be published.*

Legend Guitar Amp: I'm looking for a schematic for my guitar amplifier, made by Legend Musical Instruments of Syracuse, NY. The model number is G12505C (50 Watt amp), serial number 00619, purchased new in 1982. Can anyone help, and does anyone know what happened to that company? Ben Sherman, 79 W. Main St., Apt. U, Westminster, MD 21157.

Minimoog mods: I am interested in any modifications (MIDI or otherwise) that apply to early model minimoogs. I am also interested in how well the J.L. Cooper MIDI-to-CV box works with minimoogs. Jim Logan, 339 Burns Ave., Syracuse, NY 13206.

Foxx Tone: I am in search of schematics for the legendary Foxx Tone (used by Adrian Belew, among others). Foxx Tones are hard to find in Quebec so with the schematic and parts list, I hope to build one myself. Any Foxx Tone users out there? Luc Lachance, 310 Vassal, Drummondville (Quebec), J2B 5H3 Canada.

Attention User Groups: EM wants to run a listing of all current User Groups related to musical electronics. These can be individual operations or groups sponsored by manufacturers. Here's all you need to do:

1. Describe who and what you are: name, address, phone, areas of interest, membership dues and qualifications (if applicable), and services offered to members. **Even if you have already contacted EM, this information must (MUST!) be re-submitted in writing. We will not accept listings over the telephone.**

2. Notice of specific, dated events must arrive at EM's offices three months prior to cover date. For example, if you plan to have a meeting in October, we would need written notice before July 1.

3. We hope to make this a monthly feature. To be listed every month, you must re-submit the info in Step 1 **every month** (hopefully updated to reflect current activities). Sorry to put you to the extra effort, but otherwise we have no idea of which groups are still active.

4. Send your letter or postcard to USER GROUP LISTINGS, *Electronic Musician*, 2608 Ninth St., Berkeley, CA 94710. There is no charge for this service.

EM

ACCESSORIES

The **SPX-2** ($295) retrofit expands DX7 memory to 512 slots and transmits over any MIDI channel. The **SPX-2F** ($295) is identical but sacrifices two banks of sounds to store function controls for each sound. The **MEX-1** ($695) rack-mount mass storage device holds either 3,072 sounds, or 1,536 sounds with individual function control parameters for each sound. The **DX Programmer** ($1,195) plugs into a DX7's MIDI connectors and enables all parameters to be changed by turning knobs or flipping switches—just like older analog synths. Monster Memory, 5757 Kirkwood Pl. N., Seattle, WA 98103. Tel. 206/526-0540.

The **Phasor** ($179) is a music, sound, and speech synthesizer card for the Apple IIe and II+ with 12 simultaneous sound channels and four noise generators. It is compatible with the Mockingboard, ALF Music Card, Symphonix, and the Super Music Synthesizer. Music, sound effects, and text to speech software is included. Applied Engineering, P.O. Box 798, Carrollton, TX 75006. Tel. 214/241-6060.

ELECTRONIC PERCUSSION

Drummers who want to stand out in a crowd should check out the **Rhythm Stick** ($895), a remote MIDI percussion controller with eight triggers and two "slap" sensors. Europa Technology, 1638 W. Washington Blvd., Venice, CA 90291. Tel. 213/392-4985.

All **Linn sound chips** have been bought by Gand Music & Sound, 780 Frontage Rd., Northfield, IL 60093. Tel. 312/446-4263.

SIGNAL PROCESSORS

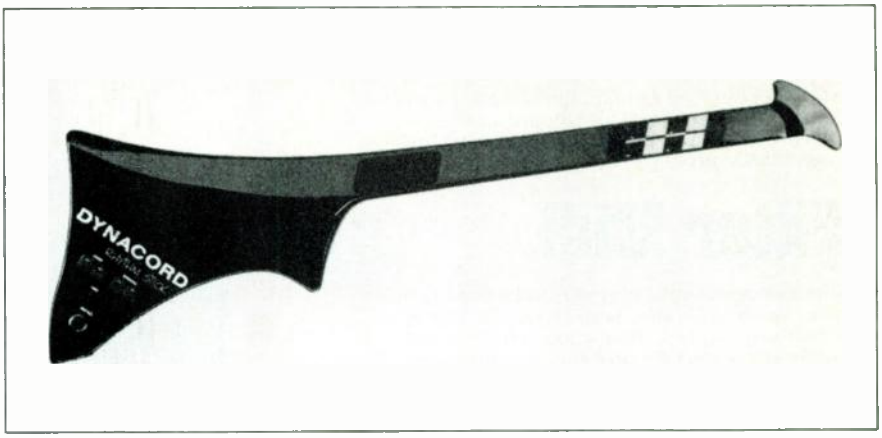
The **SDR1000** stereo digital reverb features 16 bit linear conversion and hall, room, gate, reverse, dual delay, auto panning, and dual reverb programs; these form the basis for 30 factory presets and 70 user-programmable programs, all of which are recallable via MIDI. A free demonstration flexi-disk is available from Hoshino, P.O. Box 886, Bensalem, PA 19020. Tel. 215/638-8670.

SOFTWARE

The **Kawai K3** digital synthesizer now has a computer support system (designed by **Hybrid Arts**) that comprises Wave Table Editor/Patch Editor/Patch Librarian software, MIDItrack III sequencer software, Atari 130XE computer, Atari disk drive, and MIDImate interface. The K3 itself includes a 61 note keyboard with velocity and pressure response, and stores sampled digital waveforms as the primary sound source for its various patches. Kawai America, 24200 S. Vermont, Harbor City, CA 90710. Tel. 213/534-2350.

The **348 Track PC MIDI Recorder** features 32 sequences (48 tracks per sequence), song pointer, live track muting/unmuting, real time tempo changes, and individual MIDI event editing. Requires minimum 256K memory and Syntech IBM PC interface. Other new products include **Omnibus** (system exclusive storage; requires Apple II+/e), and for the C-64/128 or Apple II+/e, **Universal Drum Librarian** (TR-707, RX-11, TOM), **JXPRESS** JX-8P editor librarian, and CZ Master editor/librarian. Syntech, 5699 Kanan Rd., Agoura, CA 91301. Tel. 818/704-8509.

CZ Sounds ($29.95) includes 128 sounds for Casio CZ synths. Available in data sheets or on disk in Commodore Dr. T, Data Dumpstor, or CZ-Rider format. **DX100 Sounds** ($29.95) provides 120 sounds for the DX100, DX21, and DX27. Data cassette or disk in Commodore Data Dumpstor or Sex-Ed format. **Poly 800 Sounds** ($14.95) has 64 sounds available on data sheets or cassette. Demo tapes ($3.95) are available for all packages listed above and include two free sample patches. Leister Productions, 806 S. Market St., Mechanicsburg, PA 17055.



Dynacord Rhythm Stick

Aug 6-10, San Francisco, CA: The San Francisco Synthesizer Ensemble, a quartet of award-winning electronic musicians, will play in concert at the Theatre Artaud (450 Florida, San Francisco). The 90-minute performance combines music and computer technology to produce a synthesis of music and images. For ticket and performance information, call 415/621-7797.

Kawai K3 digital synthesizer and Hybrid Arts computer support package

STANDS

The **Speaker Bin Scaffolds** support speakers above the ground and are available in heights of five or six feet in two diameters of tubing. The **Keyboard Expansion Rack** includes a single tier for keyboards and removable rack section for rack mount devices. F.M. Tubecraft Support Systems, 1121-26 Lincoln Ave., Holbrook, NY 11741. Tel. 516/567-8588.

The **RM-2 Rack Mount Stand** supports up to 1,000 pounds of rack mount devices, weighs 23 lbs., and is 47.5 inches high. The **RM-3** is a half-size model that is 22 inches high and weighs 14 lbs. Both models are tilted for easy visibility. The **RS-1** rack mount shelf holds small effects and other non-rack-mountable devices. Solid Support Industries, 2453 Chico St., South El Monte, CA 91733. Tel. 818/579-6063.

OTHER NEWS

Premier synthesist and EM author **Larry Fast** is the head of A&R for the Audion Recording Company, a new label devoted exclusively to state-of-the-art electronic instrumental music and marketed by JEM records. Artists include **Don Slepian, Neil Nappe, Emerald Web, Roger Powell**, and others. The initial release will be a sampler LP. JEM, 3619 Kennedy Rd., P.O. Box 078, South Plainfield, NJ 07080. Tel. 201/753-6100.

Mills College has released the **Unique Music Anthology**, a three record set with compositions by Lou Harrison, Terry Riley, Luciano Berio, Dave Brubeck, David Rosenboom, Robert Ashley, Anthony Braxton, David Behrman, Elinor Armer, Steve Reich, Maggi Payne, Darius Milhaud, Pauline Oliveros, Anthony Gnazzo, Katrina Krimsky, Larry Polansky, Pandit Pran Nath, Janice Giteck, Robert Sheff, Ramon Sender, and Morton Subotnick for $18 (make check payable to Mills College). *Centennial Album*, c/o Mills College, Office of the President, Oakland, CA 94613.

Church Music Systems provides hardware, software, and training for musical systems designed for church music. A system includes one or more electronic keyboards, amplification, and computer. Church Music Systems, 134 Palo Verde Ave., Palm Springs, CA 92264. Tel. 619/323-4940 collect. A newsletter is also available.

All prices are suggested retail prices, as supplied by the manufacturers. All prices and specifications are subject to change without notice. EM

Concealed within your computer is the ability to make music—with more flexibility, and ease of use, than virtually any conventional musical instrument.

Making Your Micro Musical

BY CRAIG ANDERTON

If you have a computer, you have a musical instrument. This instrument is a natural for computer enthusiasts, since "playing" it involves many of the same techniques as doing any task with a computer. Just as some people are drawn to guitar, piano, voice, drums, or whatever as "their" instrument, a generation of computerphiles are being drawn to the computer as "their" instrument. It's not hard to turn a computer into a music machine—and the possibilities are, well, ear-boggling.

INSTANT SOUND

Most computers can make music "right out of the box"—in other words, there is a sound-generating integrated circuit within the machine itself, and you need nothing further (except for a program that lets you access the chip) to make music. Capabilities vary from machine to machine, from simple sound effects generators (these are more suited to making video game sound effects than music, though) to sophisticated chips that rival the capabilities of some synthesizers. The Commodore-64/128 has a very capable sound chip, as does the Amiga. Atari's ST has a fairly primitive sound chip, but under the proper software control it becomes much more useful; the Macintosh and IBM PC also include simple sound generators.

Internal sound chips are mostly useful for "getting your feet wet" in computer music with minimal expense. However, computer music really starts to blossom when the computer controls advanced musical instruments such as keyboard synthesizers, since these are inherently more sophisticated, cleaner-sounding, and more versatile than internal sound chips; and this brings us to . . .

MIDI!

Numerous standards have evolved in the computer world to make life easier for computer owners. For example, the RS-232 standard allows for the interconnection of printers and terminals from various manufacturers to various computers, and there are also standards for how to transmit data over the phone lines. The purpose of a standard is to allow for unambiguous transfer of data between pieces of equipment made by different manufacturers.

Fortunately for electronic musicians, there is now a standard format for musical data interchange called MIDI (musical instrument digital interface). This is a serial transmission protocol that is sort of the equivalent of "music notation for computers." For centuries, if one human wanted to communicate musical thoughts to another human, it involved writing out symbols (music notation) on a piece of paper. Standard music notation can specify a note's pitch, duration (quarter note, eighth note, 16th note, etc.), the tempo of the song, a rough indication of dynamics (crescendo and decrescendo), and there are even some labor-saving symbols (such as the "repeat" symbol, which saves having to re-write a repetitive part). Music notation, however, cannot capture many elements of the musical performance, such as where vibrato is to be added, or specific tonal qualities of a melodic line.

While conventional music notation works fine for humans, wave a piece of sheet music in front of a computer and you won't get much of a response because computers want everything expressed in their own digital language. And that's what MIDI does.

As you play a MIDI keyboard, serial data that correlates to your playing flows out the MIDI Out connector, in real time, at a rate of 31,250 Baud. For example, if you play a C#, a piece of data that says "C# has just been played" emanates from the MIDI Out connector. But MIDI expresses much more about your playing than just notes; MIDI can also indicate dynamics, the extent of pitch-bending (i.e. a technique popularized by guitarists that involves an upwards or downwards pitch shift), whether modulation (vibrato, tremolo, etc.) is being added or a sustain pedal is being pressed, and more. This data, which corresponds to your performance, flows out the MIDI Out port in real time as you play a MIDI instrument. Note that you may not even need any keyboard playing ability (although it does help) because with some programs, data can be entered from a QWERTY keyboard and stored into a sequencer program (we'll talk more about sequencers later).

Now that we have this data, what can we do with it? For one thing, we can stuff it into computer memory and have the computer remember the exact order in which the data (notes and such) was played. Then, we can read the MIDI data out of memory, feed it into a synthesizer's MIDI *In* connector, and the synthesizer will read the data and play accordingly—just as if you were playing the keys. However, the data that tells the synth's sound generators what to play comes not from your fingers pressing on keys, but by data coming over the MIDI In port. An appropriate analogy would be that of a player piano—except instead of punching holes in paper to represent notes, we're "punching" data into memory. (And when it comes to editing, it's a lot easier to fool around with RAM than punch holes in paper or tape over existing holes!) Note that most MIDI synths have both MIDI In and Out jacks, so that they can serve either as a "master" device that sends data to other MIDI gear, or a "slave" device that receives data from other MIDI gear and plays according to the data received.

MIDI allows for much more—such as simplified music printing and transcription, and the ability to send and receive data over 16 independent channels—but we know as much as we need to know to continue with our odyssey of turning a computer into a music machine.

THE MIDI INTERFACE

Your computer already has several standard connectors (for example, the DB-25 connector that handles RS-232 data) and other connector ports for disk drive, modem, etc. MIDI also specifies a particular type of connector, a five-pin DIN connector. As of this writing, the Atari 520/1040ST and Yamaha CX5M computers are the only machines that include MIDI connectors right there on the back panel with all the other connectors . . . quite a step towards the legitimization of MIDI, I might add. All other computers require some sort of interface to translate data within the computer into serial MIDI data. Commodore-64/128 MIDI interfaces are typically cartridge-shaped and plug into the game cartridge port; for the IBM PC and Apple II, available MIDI interfaces plug into an unused card slot; for the Mac, the MIDI interface usually plugs into the printer or modem port; and so on. MIDI interfaces are available for just about any computer, including the Apple IIc (J.L. Cooper and Passport), Atari 130XE and 800XL (Hybrid Arts), Radio Shack Color Computer (Intercomp Sound), Radio Shack Model 100 (RMD & Associates) and many others—there are even general-purpose RS-232 to MIDI interfaces (Hinton Electronics, RMD & Associates).

A MIDI interface is not too complex or expensive, and generally consists of a UART and a few other chips. Some interfaces are simpler than others; the Mac and Amiga models seem particularly simple. An interface will cost anywhere from $20 for a "dumb" interface you can make yourself up to hundreds of dollars for an "intelligent" interface that does some MIDI "housekeeping" along with the standard MIDI functions. Typically, though, a MIDI interface will run somewhere between $75 and $150.

There are some *standard* interfaces, in that much software was written with these interfaces in mind. For the IBM PC (and, of course, IBM clones) the Roland MPU-401 intelligent interface is the most popular interface. An alternative IBM interface, the OP-4001 from Octave-Plateau, is also MPU-401 compatible. The MPU-401 works with the Apple II family of computers as well.

There are two main interface standards used with the Commodore-64: the Sequential standard and Passport standard (the latter being the most common; virtually all interfaces are "Passport-compatible," meaning that software written for the Passport interface will run on a Passport-compatible model). Fortunately, the differences between the Sequential and Passport standards are relatively minor, so software will often include a set-up menu where you can specify which interface you are using, and the software will run equally well with either model. Passport also makes a line of interfaces for the Apple II family of computers, and is generally credited with popularizing and standardizing the concept of a MIDI interface. Syntech, Yamaha, Mimetics, Dr. T, Korg, and a bunch of other companies make Passport-compatible interfaces. The Moog Song Producer interface is probably the most powerful model for the C-64; although it has limited compatibility with the existing base of Passport-compatible software, it comes bundled with several programs written specifically for the Song Producer interface.

There is a de facto standard for the Mac as well, and software written for the Opcode, Assimilation, or Musicworks interface will work with any of the other interfaces mentioned. A fourth interface, made by Southworth and used with their Total Music program, will run most, if not all, software written for the other interfaces; however, Southworth's software will run only on the Total Music interface. As mentioned earlier, Yamaha and Atari ST computers already have a MIDI interface built-in so no further interface box is required.

One fine point with Mac interfaces concerns a difference between the original Mac design and that of the new Mac Plus. The former provided power at its expansion connector; the latter does not. The Total Music interface includes its own power supply and is therefore immune to this engineering change, but other, older interfaces require a modification to work with the Mac Plus. Make sure that any interface you contemplate buying will work with the Mac you have.

Although software may be compatible with interfaces made by a wide variety of manufacturers, it's important to note that there are differences between interfaces. The most significant is that some interfaces include provision for sync-to-tape, meaning that the interface can accept an external timing reference. This feature is vital when working in a multi-track recording environment, and I recommend that any interface you buy include sync-to-tape.

Now that we've added a MIDI interface, our computer can speak MIDI to any of the MIDI-equipped devices on the market (and that is a *huge* number—just about every keyboard synthesizer on the market has MIDI). But there's more; you can even drive MIDI lighting controllers, audio mixers, and signal processors... and surely more MIDI-compatible devices will appear in the months ahead. Now all that remains is to choose some software that will enable us to actually use MIDI for something useful and fun.

In Part 2, we'll look at the various categories of MIDI software and the different tasks they perform. Stay tuned!

EM

Fast Forward

MASTER TRACKS PC™ for your IBM.*

Now there's a faster, easier and more productive way to make music with your IBM PC—and it comes from Passport. The same people who brought you the best-selling MIDI sequencers for the Apple and Commodore computers.

Passport's new MASTER TRACKS PC is a phrase-oriented, multi-track MIDI sequencer, step editor, song composer and performance software package designed especially for you and your IBM PC.

Incorporating a new, proprietary technology called Dynamic Phrase Allocation,™ MASTER TRACKS PC gives you a free-form approach to composing music.

You can shape and mold your ideas at your own pace. Combine or extract phrases of any length. Then assemble them in any order you want. At any time. As often as you wish. The result: complete creative freedom.

MASTER TRACKS PC is truly revolutionary! No other IBM sequencing program allows you to compose music as quickly or as fluidly.

It's feature packed too. With pop-up menus that let you work quickly and easily. With file windows that let you save, load and erase patterns, songs and song lists. With selectable on-screen colors that let you customize the program. It even has help screens that guide you through the system.

MASTER TRACKS PC is an integral part of today's music technology. It supports the popular MPU-401 MIDI Interface and all MIDI equipped instruments.

MASTER TRACKS PC provides access to the world of SMPTE through MIDI Song Pointer. Now you can easily incorporate MIDI technology into audio and video productions.

Quite simply, MASTER TRACKS PC was designed to be the best IBM sequencer on the market. Its 4th generation design gives you speed, power and flexibility.

Without compromise.

Pattern Window

- Each pattern is a free-form, 64-track scratch pad.
- Create and save any part of a pattern as a phrase, solo, bass line, song section, etc.
- Playback or record from any point in a pattern in real time.
- Record all MIDI events on 16 channels capturing every nuance of your performance.
- Programmable Punch In and Out.
- Mute tracks for mixdown to tape.

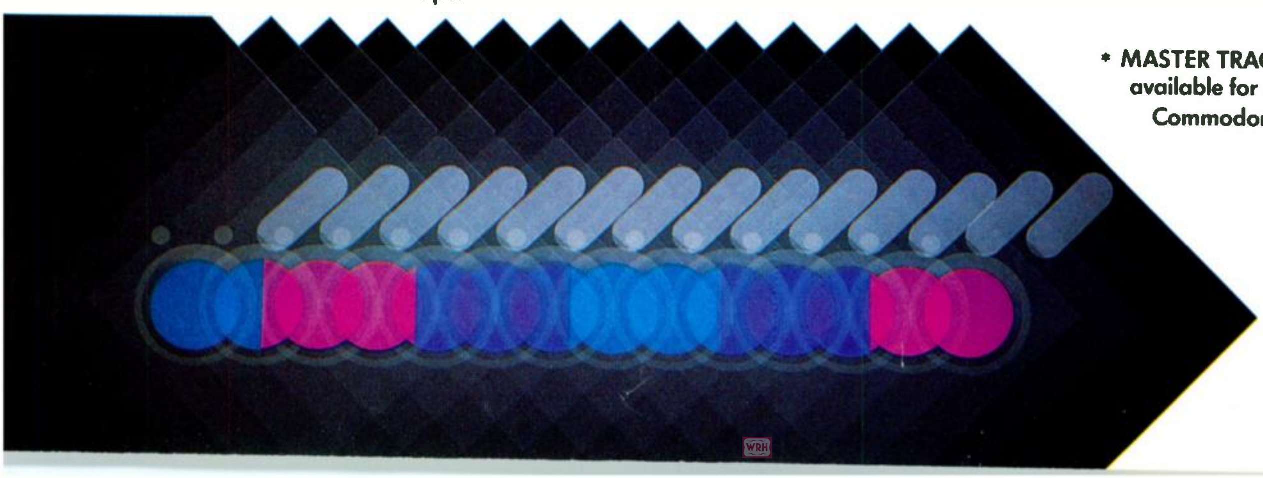
Edit Window

- Error Correct, Merge, Delete, Transpose, Repeat and Copy tracks.
- Change velocity scale for all notes within a track.
- Graphically edit track data on variable resolution grid for visual access to all MIDI events.
- Create new tracks in step time by step entering MIDI events.
- Quickly correct errors in any note, measure or phrase.

Song Window

- Versatile pattern editor lets you combine multiple patterns to create songs, phrases, drum parts, etc.
- Up to 64 repeating patterns can be used in each song in any order.
- Change tempo instantly or gradually at each step.
- Songs can be converted into patterns for use in other songs.
- Create lists of song files for automatic playback of sets.

Color monitor courtesy of AT&T.



* MASTER TRACKS Versions also available for Apple and Commodore computers.

with Passport.

More industry standard MIDI Interfaces.

That's right. Passport continues to set the standard in MIDI Interface technology. With our industry standard MIDI Interfaces you can turn your computer into a complete MIDI studio, creating extraordinary music systems that are incredibly affordable.

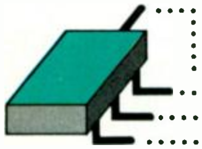
Our interfaces continue to provide serious musicians everywhere with the vital communication link to MIDI equipped instruments, drum machines and personal computers.

Plus each interface from Passport comes with TWO FREE MIDI CABLES and a comprehensive user's manual.

For all your applications—and computers—Passport's got what you need. We have more industry-compatible interfaces than anyone else. Anywhere.

See for yourself.

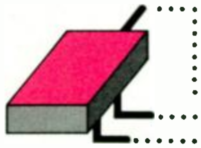
Passport MPU-401 for IBM PC



An "intelligent" MIDI Interface that's 100% MPU-401 compatible.

- MIDI in, MIDI out
- DIN sync out
- FSK tape sync in and out
- Metronome out

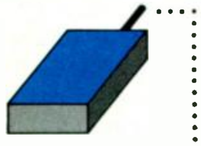
Passport MIDI Interface™ w/Tape Sync



The industry standard for Apple and Commodore computers.

- MIDI in, MIDI out
- DIN sync out
- Tape sync in and out

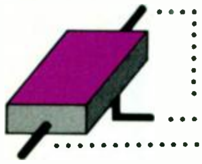
Passport MIDI Interface™ for Apple Macintosh™



Compatible with Assimilation™ and Opcode™ Interfaces and works on the new Macintosh™ Plus.

- MIDI in, MIDI out

Passport MIDI PRO Interface™



An "intelligent" MIDI Interface for Apple IIc. No computer modification required.

- MIDI in, MIDI out
- Clock sync in and out
- Tape sync in and out.

All Passport interfaces are field tested and proven reliable. They're made with the highest quality components and are fully compatible with MIDI.

Passport MIDI Voice Librarians.™

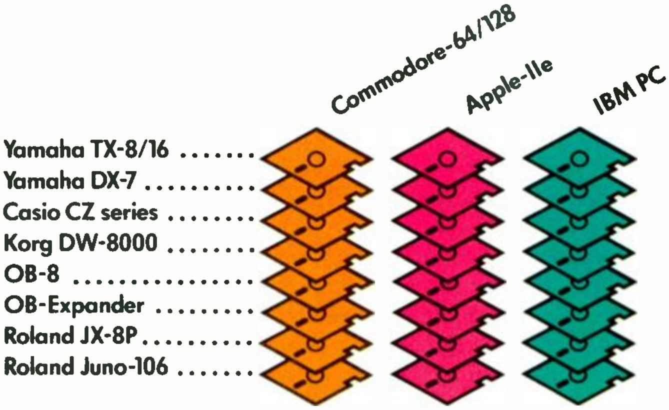
With Passport's MIDI Voice Librarians you get instant access to more than 200 of the latest sounds on disk.

If you make an intelligent comparison, you'll find features in the Passport Librarians that make them the obvious choice.

For instance, you can save and load your favorite presets and banks of sounds on diskette. Create your own custom banks of sounds by organizing and rearranging presets. Monitor your sounds in memory without disturbing the resident presets in your synthesizer.

That's not all. Our Librarians are compatible with DX Pro,™ DX Heaven,™ Data 7™ and a multitude of other popular voicing programs. Preset editors are also available for select synthesizers.

Without a doubt, Passport's MIDI Voice Librarians are the lowest-cost alternative to expensive RAM cartridges or unreliable cassette tape recorders.



Move forward with Passport.

After years of success with MIDI music software and interfaces, Passport is still pushing the limits. We are the first company to bring you MIDI Interfaces and MIDI music software. Others come and go, but Passport continues to set the pace.

We're dedicated exclusively to bringing you the best. In quality, dependability, innovation and support. Our established customer service and update policies assure you of our commitment to your personal and professional growth and success.

We're moving forward and we invite you to join us.

For full specifications on Passport's complete product line, see your authorized Passport dealer. Or contact us direct: Passport Designs, Inc., 625 Miramontes Street, Half Moon Bay, California USA 94019. Phone: 415/726-0280. Fax: 415/726-2254.

PASSPORT

The Speed of Sound.

The Rodney Dangerfield of computers gets some new hardware, new software, and. . .some new-found respect.

The Little Computer that Could: Yamaha's CX5M

BY TONY THOMAS

The concept of a dedicated music computer is an intriguing one. Just about every amateur or professional musician would jump at the opportunity to pick up a reasonably-priced computer that could make routine musical tasks less complicated and monotonous. Although placing composition, sequencing, voice and music editing, score printing, and even sound production under computer control is not a new concept, Yamaha is the first company to optimize an existing computer system to perform musical tasks and offer it as an integrated "music computer" at a price well within the reach of most musicians.

The CX5M is a modest computer when evaluated by just about any standard, yet it can make music much more manageable (and more fun) without the necessity of getting entangled in the intricacies of more powerful and sophisticated computers. With its built-in MIDI jacks and internal "DX-type" four-operator FM sound module, this computer has the jump over machines that require interfaces to get them ticking to the beat, and therefore, has much to offer the "MIDI-fied" musician.

The little CX5M has been beaten, bruised and battered in the music press for some time now since it is, in essence, an 8-bit computer with 32K of memory in a world full of 16-bit/megabyte memory muscle machines that seem like giants by comparison. Many of the criticisms initially leveled at the CX5M were well taken; at the time of its introduction over a year ago, only a few useful programs existed for it (there wasn't even a real time sequencer). Another glaring early defect was that it was impossible to play the computer's internal FM voices from a MIDI keyboard, thereby precluding the CX5M's obvious use as a multi-timbral sound expander module. (Arghh!!)

Finally, the fact that it utilized the new and unestablished (in this country) MSX operating system didn't seem to contribute anything to its popularity. Still, many people (including this writer) thought the idea of a musical computer was so good that they invested in the system anyway, simply based upon its untapped potential, even though much software promised at its release was in the "vaporware" category and would be available "Real Soon Now."

Frustration and disappointment wiped out any initial glee the first CX5M owners may have experienced because, while our friends with Apples and Commodores could sequence tracks in real time and edit them with a great deal of ease, those of us struggling along with the CX5M had to resign ourselves to the tedious task of programming songs in step time using musical notation via the FM Music Composer Program. As a result of these initial frustrations, the thought of turning the CX5M into a high-tech doorstop or a futuristic frisbee entered into more than a few minds. . . including this one.

Since that time, however, Yamaha has hunkered down, done its marketing homework, and accepted the challenge of rectifying the CX's initial design flaws and software shortages, thus giving it a new lease on life (and allowing me to sleep better at night). I'm happy to say that this ugly duckling is finally turning into a swan.

MSX UNDER THE MICROSCOPE

The MSX standard was developed, oddly enough, by two watch importers from New York, Harry Fox and Alex Weiss. Realizing that the watch industry was not quite as profitable as it once was due to the proliferation of cheap electronic im-

CX5M Facts at a Glance

List price: $469.
Processor: Z-80A, 8-bit operation, 3.57954 MHz clock.
Memory: 32K RAM, 32K ROM, 16K Video RAM.
Interfaces: MIDI In, MIDI Out, Yamaha keyboard input, two joystick ports (one useable for optional mouse), Centronics-compatible printer port, CRT monitor port (optional RF modulator available), ROM cartridge port, RAM cartridge/disk drive port, side expansion port, stereo audio output jacks, cassette interface.
Graphics: 256 × 192 pixel resolution; 32 × 24 character alphanumeric display.

CX5M Music Computer with music software and YK-10 accessory keyboard

Tony Thomas has been involved in advertising, broadcasting, recording, and publishing for the past ten years. He is presently the managing director of Target Communications International, a full-service advertising agency, broadcast production company, and MIDI-based recording studio.

63 Sound effects. From the obviously necessary to the quite bizarre. Pre-packaged and MIDI selectable. Single and multi-tapped delays... with filtering and reverb ambient outputs: Perfect for vocals and instruments. Stereo effects that get attention. Reverb effects that don't exist on the plane of current reverb understanding. Interested? We call it

MIDIFEX

MIDIFEX. And to get the effect of MIDIFEX without one, you'd need a very capable digital reverb, a few dozen digital delays, a couple of parametric equalizers and quite a few extra channels on your mixing board. By presetting the variables, we programmed the MIDIFEX to produce these effects at instant recall...at the front panel or through MIDI patch change. The polished sound You could work hours to get is now here. And like MIDIVERB...The best part is the price...

...suggested list price

$399.00

ALESIS

STUDIO ELECTRONICS

ports, they decided to develop a computer that could be produced in quantity overseas for about $30 and sold in this country for about $100 retail. Their initial design called for an 8-bit architecture utilizing the then-popular Zilog Z-80 microprocessor (which was the basis of the CP/M and TRS-DOS operating systems), a Texas Instruments TMS-9918 video chip, General Instruments AY-3-8910 sound chip, between 8 and 16K of RAM (Random Access Memory), 32K of ROM (Read Only Memory), and a number of ports and slots for expansion ROM, cassette, joysticks, monitor and other peripherals... in other words, a consumer computer.

"The little CX5M has been beaten, bruised and battered in the music press for some time now"

Fox and Weiss called upon the talents of Microsoft, the software giant from Bellevue, Washington, to develop an operating system for their proposed creation. (Microsoft had authored IBM's MS-DOS operating system and almost every BASIC language interpreter running on microcomputers today, including Applesoft BASIC, IBM BASIC and TRS BASIC.) Then Microsoft VP, Kazuhiko "Kaye" Nishi, who also was the father of the Kyocera line of lap portable computers (marketed in this country as the Tandy Model 100, NEC PC-8201A and Olivetti M-10), took the project on. His changes to the proposed design allowed it to access up to 256K of RAM, work with disk drives, and support an 80 column display assuming the correct peripherals were added. A disk operating system, known as "MSX-DOS" and modeled closely after IBM's 16-bit MS-DOS Version 1.25, was also developed for the computer; that operating system will reportedly support 8 inch, 5.25 inch, and 3.5 inch drives. These additions increased manufacturing costs up to at least $80 per unit, but added significantly to the computer's capabilities.

The resulting computer, dubbed the "Spectravideo" (now defunct), incorporated a proprietary 8-bit Microsoft-designed operating system known as "MSX." This operating system was licensed in mid-1983 by Nishi and Microsoft to a consortium of Japanese manufacturers including Sony, Pioneer, Panasonic, Hitachi, Toshiba, and last but not least, Yamaha. Since that time, the MSX operating system (which is now owned by ASCII Japan) has become the de-facto standard for all Japanese 8-bit computers and has also become a major force in Europe... much as the Apple, Commodore and Atari systems have dominated the market in this country.

MSX Misfires

Even though the MSX operating system offered several price vs. performance advantages over existing 8-bit systems, its introduction coincided with the home computer shakeout that devastated several of the largest computer manufacturers including Texas Instruments, Atari, Timex-Sinclair, Commodore and Coleco. The fallout of this shakeout even sent computer mega-marketer IBM reeling, which resulted in that company pulling its ill-fated PC Jr. out of the running. At the same time, 8-bit computer systems were making their last stand, with the market share of Digital Research's once-almighty CP/M system evaporating because of the immediate acceptance of such 8/16-bit computers as the IBM-PC and 16/32-bit computers like the Macintosh.

Although the MSX system was an overnight success in Japan (between 150,000 and 500,000 units sold in 1984 alone), the Japanese companies decided to hold off on releasing MSX computers in the U.S. because of the uncertain future of the home computer industry. Also, the Japanese clearly did not want to go head to head with the mighty Commodore-64, a computer whose U.S. sales alone surpassed the two million mark by that time, and which offered more internal RAM at a heavily discounted price of less than $200.

Enter Yamaha

Spurred on by the popularity of such products as the DX7, Yamaha launched its music computer program which was (and still is) based exclusively on the MSX operating system. When the CX5M was introduced at the Winter NAMM show in 1985, its market stance was unclear at best. Strangely enough, although the CX5M was manufactured with built-in MIDI In and Out jacks, its velocity-sensitive internal FM digital sounds could only be accessed through a non-velocity sensitive and non-MIDI four-octave (YK-01 or 10) keyboard. In addition, the user interface for sound generation more closely resembled the Yamaha "Portasound" series of consumer keyboards (complete with cheezy "auto-rhythm" and "one finger chords") than it did a professional synthesizer.

To complicate matters still further, the initial software offering consisted of the *FM Music Composer I* (YRM-101) which made multi-timbral use of the internal voices only through step-time programming using traditional notation. The *FM Voicing Program* (YRM-102), although very powerful in many respects, could not be used with a MIDI keyboard. The *FM Music Macro Program I* (YRM-104), which allowed the internal voices to be used in the context of a BASIC program, was clearly designed for hackers and hobbyists and not for musicians. Of the four programs, the *DX7 Voicing Program I* (YRM-103) was clearly the best program of the lot and was among the best of its genre since it made excellent use of the CX5's graphic capabilities. Although these programs displayed some degree of the CX5's potential, they were woefully inadequate for meeting the perceived and actual everyday needs of the average working musician and composer. Thankfully, all of the above-mentioned programs are being or have been updated recently to eliminate their intial shortcomings.

Under The Hood

The CX5M comprises a Z-80A microprocessor operating at a clock speed of 3.57954 MHz, 32K of RAM, 16K of VRAM (Video RAM), 32K of ROM (Read-Only Memory), a TMS 9929A CRT (Cathode Ray Tube, i.e., video monitor) controller chip, a 1200/2400 Baud FSK (Frequency Shift Keying) cassette interface, and an SFG eight octave/eight note polyphonic FM sound module (SFG-01 in older units, SFG-05 in newer ones). The latter resides in an expansion port underneath the computer and utilizes the proprietary Yamaha-2151 FM Voice Generation LSI (Large Scale Integration, meaning that they stuffed lots of elements on a single chip).

For video, CX5M offers 256 × 192 pixel resolution and a 32 × 24 character alphanumeric display. Slots and ports include MIDI In and Out jacks, Yamaha keyboard input, two joystick ports (port 1 is also used for the optional mouse), a Centronics compatible printer port (as used with many printers), CRT monitor port (an optional RF modulator allows the computer to be used with an ordinary TV set), ROM cartridge port, RAM cartridge/disk drive port, side expansion port, and stereo audio output jacks. Soft-

ware is generally available on ROM cartridges that plug into the computer and all Yamaha programs for the CX5M cost less than $60 at this writing. Software can also be loaded from disk or cassette if available in that format.

To Yamaha's credit, they picked up the pieces of the CX5M, shrugged off their initial marketing missteps, and pushed forward—even after the tirades of the trade press had turned many musicians, dealers and even owners against the system. Several complications made their job much more difficult, however. Alleged bugs in the YRM-301 MIDI Recorder slowed its release until October 1985, and FCC radio interference tests likewise impeded the introduction of the two disk drives (FD-03 single-sided, FD-05 double-sided) and the improved SFG-05 FM sound module, which can be used multi-timbrally from any MIDI keyboard or sequencer. In addition, it seems that Yamaha was counting on the introduction of other MSX computers in this country to help take up the slack of providing additional hardware and software support for the CX5M, and when no such support was forthcoming, Yamaha was left holding the bag.

CX5M Software and Hardware

By winter 1986, Yamaha was showing the refurbished and fortified CX5M complete with some new hardware and a number of new software packages. Some of the system additions include:

SFG-05 Sound Module Although Yamaha reportedly had planned to allow the sound module to function as a MIDI expander, problems with software operating speed initially precluded such applications. The problem was solved by revamping the sound module software and upping the memory capacity of its internal ROM, thus allowing it to function as a multi-timbral, stereo, MIDI FM Expander. Up to four different sounds with two note polyphony can be accessed at any given time on up to four different MIDI channels. Split and layer capability is also available, with the various sounds assignable to either or both stereo outputs using the FM Voicing Program. There are 46 internal presets with velocity-sensitivity residing in the internal ROM, and an additional 48 user voices can be loaded from cassette, disk or RAM cartridge (these can be stored *concurrently* with the internal voices in the on-board computer RAM). Voices can be transposed and detuned; tremolo and vibrato can be added. Simply adding the SFG-05 makes the

"To Yamaha's credit, they picked up the pieces of the CX5M, shrugged off their initial marketing missteps, and pushed forward"

CX5M worth every penny of its list price, even if it is just used as a multi-timbral MIDI expander. New CX5M's are reportedly being retrofitted with the SFG-05 and present owners can have their units updated for a nominal fee ($55 compared to $120 for a new unit). Unfortunately, don't count on using the SFG-05 for sizzling leads since the module won't receive such continuous controller data as pitch bend, mod wheel, or breath controller (although it does receive portamento on/off, sustain pedal on/off and MIDI volume). However, it does fill out the sound of one or two other synths quite nicely, especially when used in its multi-timbral mode.

MSX Mouse The CX5M's *MU-01 MSX Mouse*, which works with many of the new CX5M programs including the MIDI Recorder, RX-Editor and some of the revised versions of earlier programs, greatly speeds and simplifies many tasks. It also reduces the need for the musician to concentrate on two keyboards (computer and music) at once. It is a virtual "must" for any musician using this computer.

MSX Printer The Yamaha *PN-101* dot-matrix printer provides screen dumps for many of the programs, including all of the voice editors. It also facilitates the use of MSX-compatible graphics, database, and word processing programs.

Teleword™ This program, contained in a cartridge that fits into the side expansion slot usually occupied by the SFG sound module, gives the CX5M some new and interesting capabilities. First of all, it has a basic word processor program with such features as global search and replace, cut and paste, and 16K memory buffer. Secondly, it also contains a 300/1200 Baud modem with tone and pulse auto-dialing and re-dialing, "phone book" data base, and battery backed RAM. Reportedly, with the optional enhancer cartridge this program will send and receive MIDI patch and sequence data over regular dial-up phone lines. Considering the cost of 1200 Baud modems and word processing programs, it is well worth the price (under $400).

Disk Drives Yamaha now offers two disk drives, the *FD-03* single-sided 3.5 inch unit with a built-in controller and the *FD-05* double-sided drive. The latter requires an external disk controller but gives you about twice as much storage capacity (720K—about 300 pages of typed text). At about $350 and $500 respectively, these disk drives are more than just a bit pricey (especially considering that you can buy a competitor's computer *and* disk drive for less). In fact, they may make you fall in love with your $50 data cassette recorder all over again.

MIDI Recorder The long-awaited *MIDI Recorder* real time sequencer for the CX5M (YRM-301) holds up really well in comparison to other programs running on 8-bit computer systems. Its user interface resembles a multi-track tape recorder and it has many useable features including automated punch in/out; autolocator; note-by-note editing of note on, note off, velocity, and pitch; aftertouch filter; step mode; quantization; and track merge, copy, and delete options. Storage media include RAM cartridge, cassette, and disk; the program supports MSX and Epson compatible printers. Although its memory capacity is limited to about 3,500 notes (with velocity), its chain mode allows you to assemble a composition using short segments, with the ability to turn tracks on and off, repeat sections, transpose, and speed up or slow down various parts of the song. Its billing as only a 4-track sequencer is also misleading, since there are four banks of four tracks (only one bank at a time can be accessed in chain mode, however) and each track can contain up to 16 MIDI channels of information in multi-channel mode. The only flaw I encountered with this package is in the quantize function: it has the tendency to chop notes if your playing is a little ragged (in fairness, this is also true of many other low-cost sequencers). Quantization is non-destructive, however, since the time-corrected version must be copied to another track, so you can experiment with a number of different quantization values. All things considered, the program works great, and for less than $60 retail, I feel it's a real bargain.

The RX-Editor The *RX-Editor* package (YRM-302) is specifically designed to work with the Yamaha RX series of drum machines (RX11, RX15, RX21), but will work with any MIDI drum machine that receives MIDI note information. For a complete description and explanation of the RX Editor's features, see my review in the July 1986 issue of **EM.**

II Series Software The first four programs available for the CX5M have since been revised and re-introduced. These include the *FM Music Composer II* (YRM-501), *FM Voicing II* (YRM-502), *DX7/TX7/TF1 Voicing II* (YRM-304), and the *FM Music Macro II* (YRM-504). The FM Music Composer now works with the mouse and disk, has a number of pop-up menus, works with any MIDI keyboard, supports both MSX and Epson compatible printers and is, in general, much easier to use. The FM Voicing II now can utilize disk storage in addition to cassette and cartridge, works with a MIDI keyboard, and supports both Epson and MSX printer standards. The DX/TX Voicing II program supports Epson and MSX printers, uses mouse and disk, and now has a MIDI mix function, thereby allowing it to program the TX7 and TF1 (in the TX-rack) sound modules in addition to the DX7.

DX21 and 9 Voicing Programs Yamaha has also provided voice editing/librarian software for the four operator DX9 (YRM-105) and DX21/27/100 (YRM-305) series of synths so that you can edit and catalog your collection of Yamaha patches. These programs are great if you have an all Yamaha system (a highly unlikely possibility), but if you have a number of synths by a number of manufacturers, you will find that no Yamaha or third-party generic librarians exist for the CX5M. (The Teleword Enhancer will support reception of data from synths that don't need to receive a MIDI send request.) High on my wish list is a generic librarian program with system-exclusive send request capability, and a program to scale down DX7 patches to work on the four-operator FM synths. Meanwhile, you may have to consider buying another computer to keep track of your CZ patches...

Music Education A few programs were introduced recently by Yamaha Music Foundation, a division of Nippon Gakki (Yamaha's parent corporation). They include the *Keyboard Chord Master* (CMW-31), *Keyboard Progression Master* (CMW-32) and *Guitar Progression Master* (CMW-33). In general, these programs seemed to be very helpful and useful in further developing your understanding of basic music theory at a very low cost, although many traditional music educators may not be enthralled with their use of sharps as enharmonic equivalents of flats. In addition, the YMF has released a *Graphic Artist* program (GAR-01) that allows you to manipulate the computer's impressive graphics capabilities via keyboard or mouse to produce true "computer art."

MSX Appeal

The MSX operating system, while not necessarily state-of-the-art, does represent an improvement over other 8-bit systems with more limited expansion and graphics capabilites. The CX5M's low cost, small size, and relatively fast operating speed make it an appealing alternative to more expensive and bulky computer systems. The main frustration you will encounter is that, since it does several functions well and it is not multi-tasking, you will have to decide which function is most important to you at any given time. For example, if you want to use its MIDI Recorder for sequencing, you can't use the internal voices (you might want to keep an inexpensive hardware sequencer around in which to dump your completed sequences so that you can use the internal sounds whenever you want).

The CX5M's biggest limitation is still the memory storage capacity. In the world of multi-megabyte machines, its paltry 32K makes the CX5M look puny indeed (although Commodore-64s don't offer much more *useable* memory). I've been told that a new 128K computer called the CX7 (which utilizes the revised "MSX-2" operating system) is already out in Japan, leading to speculation that a more powerful version CX computer may soon appear in this country. Yamaha sources deny that the either the CX7 or an expanded version of the CX5 will be imported into this country any time in the near future. However, according to Dave Edwards of NYRAC (see sidebar), the CX5M is capable of addressing at least one megabyte of RAM, and his company already sells a 32K expander that, unfortunately, won't work with Yamaha software. Some expansions may eventually surface, however, since at least one third-party company is hard at work on a 64K cartridge that *will* work with Yamaha programs. Even if such expansions don't take place, the CX5M and MSX are by no means dead...this little computer is still very much capable of breathing new life into your musical compositions, particularly if you use a lot of Yamaha equipment.

CX5M Resources

The CX5M world is bound together by a few selfless individuals and organizations who provide support and help to CX5M and MSX system owners. They are:

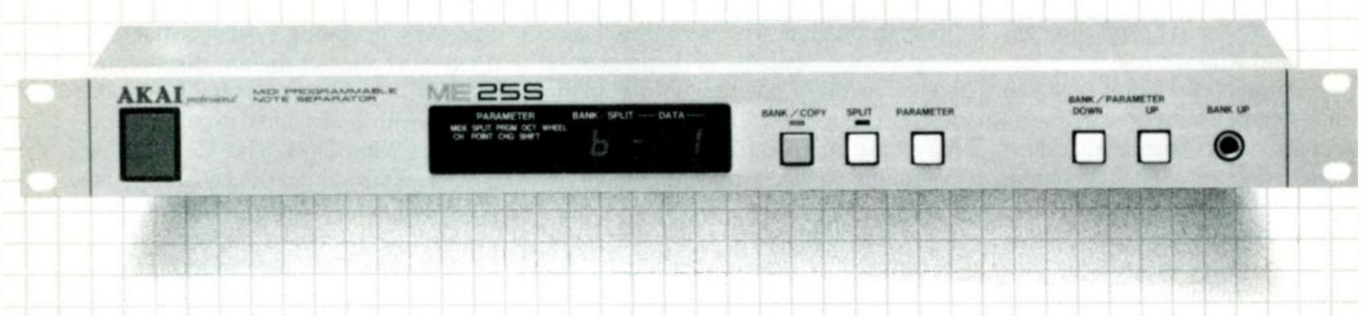
CX5-US Users Group Address: 5218 Scott Street, Torrance CA 90503. Tel. 213/540-3758. Membership fee: $20 (U.S. funds) per year. Contact: Mike Dwyer. This users group maintains a 1200+ voice library, available upon joining, for the CX5M. A number of public domain games, utilities, and programs are also in the library. Monthly meetings are held the first Tuesday of each month at the Torrance address at 8 p.m. Kevin Bierl, a Telephone Service Representative with Yamaha, is generally present to answer questions. A newsletter is also distributed on an irregular basis by CX5-US.

New York CX5M Users Group Address: 551 Central Avenue (Suite 22B), Cedarhurst NY 11516. Tel. 516/295-1427 or 718/461-8057. The New York group likewise has an extensive program and patch library and holds regular meetings.

NYRAC Address: P.O. Box 210173, Montgomery AL 36121. Tel. 205/277-2048. Contact: Dave Edwards. NYRAC is not a users group per se, but a company that claims to be the only distributor of non-Yamaha MSX software in the United States. Its founder, Dave Edwards, is as far as I can tell, the single most knowledgeable authority on MSX in this country. His periodic newsletter, MSXUSA, is a "must read" for every CX5M or MSX user, simply because of the sheer amount of information and the number of helpful tips contained in each issue. Although many of the programs NYRAC sells are games, there are also a number of utilitarian titles such as spreadsheets, databases, word processors and assemblers. NYRAC also sells most of the books in existence on the MSX operating system, and Dave fields a significant number of questions each week from CX5M and MSX users and dealers.

ATTENTION ALL MIDI USERS

KEYBOARD, COMPUTER, GUITAR-TO-MIDI, SYNTHESIZER, DRUM MACHINES AND SAMPLERS!

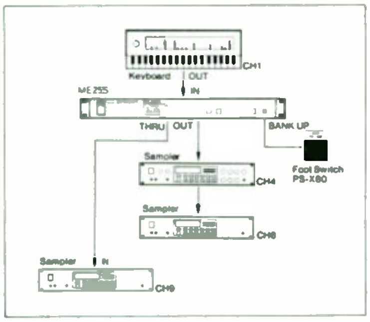


The ME25S Programmable Midi Note Separator is a device that has numerous uses and applications far beyond those that are readily apparent. Its 64 internal memory banks will memorize the MIDI channel, splitpoint, program change, octave shift, and modulation wheel settings. The main function of the ME25S is to convert a non-split keyboard into a multi-split keyboard, but you can use splits for other uses as well. For example, you can use one section of the keyboard to play on, and another section of the keyboard to select programmable effect changes. The ME25S can be used as a sequential program changer, an octave transposer, key overlapper, and numerous other functions limited only by your imagination. Make your instrument the master controller with AKAI's unbelievably versatile rack mount ME-25 Programmable Midi Note Separator.

List: **$199.95**

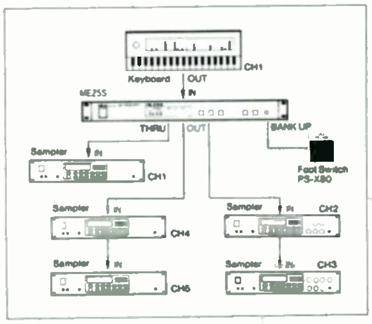
AKAI ME-25S SYSTEM APPLICATION

Using as a MIDI Channel Converter/MIDI Channelizer



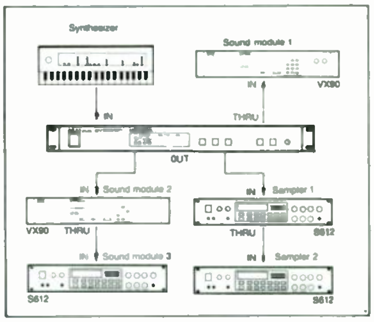
When the ME25S is used, transmission is possible on any channel by using the foot switch, even with a keyboard whose MIDI transmission channel is fixed at CH1

Using as a Keyboard Splitter



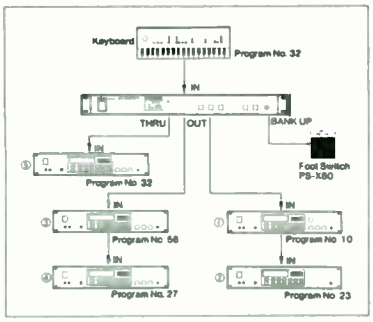
A keyboard not equipped with the split function can be split four ways

Using as a Keyboard Splitter with one Sound Source over Entire Keyboard.



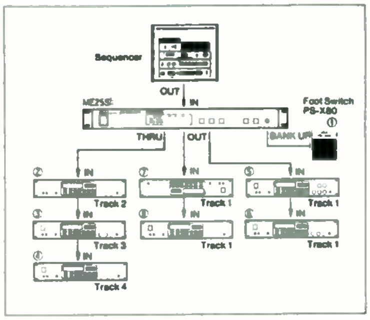
In this example, several external sources (sound modules, samplers) can be played on the keyboard with split voices.

Using as a Program Change Number Presetter



The ME25S can be used to preset the program change numbers, convenient when the keyboard's program change numbers are different from the MIDI delay, reverberator and external sound sources' program change numbers.

Using the Split Function to its Fullest



The split function on the ME25S can be used to run different parts on many external sound sources over a single sequencer track.
The following example fully utilizes the octave shift function.

AKAI *professional*
P.O. Box 2344
Fort Worth, TX 76113
(817) 336-5114

Plug in your EM MIDI interface, type in some software, and you've got a major league MIDI echo program.

Using The EM Interface: A MIDI Echo/Delay

BY TIM DOWTY

How would you like a whole bunch of new sounds for your MIDI synth? How would you like to beef up the sounds you already have by adding a software-controllable dose of ambience or punch? If you've already built the *EM MIDI Interface* (May '86 *EM*), your new palette of sounds won't cost you a cent! The "MIDI Echo/Delay" program shown in Listing 1, along with the EM Interface and a Commodore 64, are all you need to begin exploring the fantastic sonic possibilities offered by MIDI echo. (Please note that the EM interface schematic in the May issue has one error: diode D1 should be reversed so that the anode connects to IC4 pin 2 and the cathode connects to IC4 pin 1.)

This is a full-fledged MIDI delay, complete with changeable delay times, delay velocity, and delay transpose interval. If you're not familiar with what a MIDI delay can do for you, see Craig Anderton's review of the Akai ME10D in the February '86 *EM* and Terry Fryer's "Creating Ambience With MIDI-Controlled Synthesizers" in the May '86 issue.

Design Philosophy

I have tried to make the progam as useful as possible, and also, anticipate lots of different situations where MIDI delay would be used. With that goal in mind, I've sacrificed a degree of user-friendliness to obtain the widest possible range of parameters. Therefore, I suggest that you stick pretty close to the default parameter values, and explore the outer reaches carefully, one parameter at a time.

Tim Dowty has worked as a guitarist with the San Diego-based synth group "Elemental P," as well as done software design for Inter-Ocean Systems. He is currently consulting to various musicians on MIDI and related topics, and has recently introduced a line of low-cost MIDI project kits for electronic musicians.

For the sake of simplicity and space considerations (not to mention programmer burn-out!), only "note on" and "note off" information are delayed. Any other MIDI information presented to the MIDI Echo/Delay is simply ignored. No channel changes or adjustments are made, and channel information is preserved through the delay.

Entering the Program

To use the program, type in the BASIC program given in Listing 1. Typing all that stuff in the DATA statements (lines 5000 through 5032) is a drag, I know, but I've found that having a friend read the lines while I type takes away a lot of the drudgery and makes the entering process more accurate.

"How would you like a whole bunch of new sounds for your MIDI synth? This program gives you a new palette of sounds and won't cost you a cent"

As you type, the program lines that exceed 40 characters will "wrap-around" on your computer screen and thus look different from the way they do in Listing 1. Don't worry; this is normal. Listing 1 is "un-wrapped" on purpose, for the sake of clarity.

After you type in the BASIC program, SAVE it to disk before you RUN it! Certain types of typing mistakes can cause a program to overwrite itself in memory. Hours of work can be destroyed in the twinkling of an eye. Beware!

(Note: If you would like an assembly language listing of this program and have a U.S. postal address, send $3 and a standard letter-sized self-addressed-stamped-envelope with 39 cents of postage to: Xerbitron, P.O. Box 70055, San Diego, CA 92107.)

Operation

Upon RUNning, MIDI Delay first POKEs the machine code appearing in the DATA statements into the C-64's memory (lines 2000-2999). This takes about 30 seconds, so be patient.

Next, the program performs a test on the POKEd code (lines 4000-4999), and prints an error message if things aren't right. If you get the error message, recheck the DATA statements. Chances are there is a typing error somewhere.

If the test checks out, MIDI Echo/Delay immediately starts doing its number with the following default parameters:
✓ Delay time: 50 milliseconds
✓ Transpose interval: +12 semitones
✓ Velocity adjust: 0. If you now hit a key on the C-64 keyboard, you'll be presented with a menu that asks which parameter you want to change. Following the prompts, you can make the change and begin delaying with the new parameter in place. Here are a few comments on each parameter:

Delay Time

The delay time can range from 1 to 3000 milliseconds (0.001 sec. to 3.0 sec.) in one millisecond increments. It's nice to have such a wide range of available delay times, but be careful. The program's input buffer is relatively small, and heavy MIDI activity (such as bombastic glissandos) can over-fill it at longer delay times. If the buffer overflows, the delayed MIDI data quits being a replica of the input, and you're almost guaranteed to experience stuck-on notes and missing data.

A little experimentation at various delay times will give you a feel for just how

Listing 1

```
READY.

1 REM **********************************
2 REM *    EM INTERFACE MIDI DELAY     *
3 REM *                                *
4 REM *           04/27/86             *
5 REM *================================*
6 REM *    (C) 1986 BY TIM DOWTY       *
7 REM * ALL COMMERCIAL RIGHTS RESERVED*
8 REM **********************************
10 C$=CHR$(147):M$="MIDI DELAY BY TIM DOWTY":R$="(REQUIRES EM INTERFACE CARD)"
20 L$="LOADING MACHINE CODE. . ."
30 D$="[D]ELAY CHANGE":T$="[T]RANSPOSE CHANGE"
40 V$="[V]ELOCITY ADJUST":Q$="[Q]UIT"
100 GOSUB 1000:PRINT TAB(10);L$:GOSUB 2000:GOSUB 4000: IF CSM%=771 THEN 120
110 GOSUB 1000:PRINT " CHECKSUM ERROR! CHECK DATA STATEMENTS":PRINT:END
120 GOTO 800
130 GOSUB 1000
140 DT=(PEEK(49162)*256+PEEK(49161))/20
150 GOSUB 1000:PRINT TAB(5);D$;TAB(30);"(";DT;")"
160 S$="+":T=PEEK(49163):IF T>231 THEN T=T-256:S$=""
170 PRINT:PRINT TAB(5);T$;TAB(30);"(";S$;T;")"
180 S$="+":V=PEEK(49164):IF V>128 THEN V=V-256:S$=""
190 PRINT:PRINT TAB(5);V$;TAB(30);"(";S$;V;")"
200 PRINT:PRINT TAB(5);Q$
210 PRINT:PRINT:PRINT"YOUR CHOICE [?]";
220 GET A$:IF A$="" THEN 220
230 IF A$="D" THEN 500
240 IF A$="T" THEN 600
250 IF A$="V" THEN 700
260 IF A$="Q" THEN 999
270 GOTO 150
500 GOSUB 1000
510 PRINT"NEW DELAY TIME":PRINT
520 INPUT"IN MILLISECONDS (1 TO 3000) ";D
530 D=D*20
540 DH=INT(D/256):DL=D-(DH*256)
550 POKE 49161,DL:POKE 49162,DH
599 GOTO 800
600 GOSUB 1000
610 PRINT"NEW TRANSPOSE INTERVAL":PRINT:INPUT"IN SEMITONES (-24 TO +24) ";T
620 IF T<0 THEN T=256+T
630 POKE 49163,T
699 GOTO 800
700 GOSUB 1000
710 PRINT"NEW VELOCITY ADJUSTMENT":PRINT:INPUT"(-127 TO +127) ";V
720 IF V<0 THEN V=256+V
730 POKE 49164,V
799 GOTO 800
800 GOSUB 1000
810 PRINT TAB(9);CHR$(18);"DELAY ON";CHR$(146)
820 PRINT:PRINT TAB(5);"HIT A KEY TO RETURN TO MENU"
830 SYS SADRS
899 GOTO 130
999 PRINT C$:END
1000 REM ******************************
1001 REM * SUB CLEARS SCREEN, PRINTS  *
1002 REM * SIGN-ON MESSAGE & POSITIONS*
1003 REM * CURSOR.                    *
1004 REM ******************************
1010 PRINT C$;TAB(8);M$:PRINT:PRINT TAB(5);R$
1020 FOR I=0 TO 9:PRINT:NEXT I
1999 RETURN
```

```
2000 REM ******************************
2001 REM * SUB POKES MACHINE LANGUAGE *
2002 REM * CODE INTO RAM.             *
2003 REM ******************************
2010 SADRS=49152:EADRS=49679
2020 FOR I=SADRS TO EADRS
2030 READ OP$:GOSUB 3000
2040 POKE I,OP
2050 NEXT I
2999 RETURN
3000 REM ******************************
3001 REM * SUB CONVERTS STRING IN DATA*
3002 REM * STATEMENT TO 'POKE'ABLE    *
3003 REM * DECIMAL NUMBER             *
3004 REM ******************************
3010 OH$=LEFT$(OP$,1):OL$=RIGHT$(OP$,1)
3020 OH=VAL(OH$):IF OH=0 AND OH$<>"0" THEN OH=ASC(OH$)-55
3030 OL=VAL(OL$):IF OL=0 AND OL$<>"0" THEN OL=ASC(OL$)-55
3040 OP=OH*16+OL
3999 RETURN
4000 REM ******************************
4001 REM * SUB CALCULATES CHECKSUM OF *
4002 REM * POKED MACHINE CODE         *
4003 REM ******************************
4010 CSM%=0
4020 FOR I=SADRS TO EADRS
4030 CSM%=CSM%+PEEK(I)
4040 I=I+1
4050 CSM%=CSM%-PEEK(I)
4060 NEXT I
4999 RETURN
5000 DATA 4C,0F,C0,00,00,00,00,00,00,E8,03,0C,00,00,00,20
5001 DATA 2B,C0,20,9F,FF,20,E4,FF,C9,00,F0,F6,20,D8,C1,60
5002 DATA A9,03,8D,00,DE,A9,95,8D,00,DE,60,78,20,3D,C0,20
5003 DATA 52,C0,20,6C,C0,20,83,C0,20,20,C0,58,60,A9,7F,8D
5004 DATA 0D,DC,A9,31,8D,04,DC,A9,00,8D,05,DC,A9,11,8D,0E
5005 DATA DC,60,A9,7F,8D,0D,DD,A9,31,8D,04,DD,A9,00,8D,05
5006 DATA DD,A9,13,8D,0E,DD,A9,82,8D,0D,DD,60,AD,14,03,8D
5007 DATA 10,C2,AD,15,03,8D,11,C2,A9,9A,8D,14,03,A9,C0,8D
5008 DATA 15,03,60,AD,18,03,8D,12,C2,AD,19,03,8D,13,C2,A9
5009 DATA CB,8D,18,03,A9,C0,8D,19,03,60,AD,02,DE,4A,B0,21
5010 DATA 4A,90,27,AD,06,C0,D0,07,A9,95,8D,00,DE,30,18,AC
5011 DATA 07,C0,EE,07,C0,CE,06,C0,B9,14,C3,8D,01,DE,4C,C7
5012 DATA C0,AD,03,DE,20,07,C1,4C,7E,EA,00,48,8A,48,98,48
5013 DATA AD,0D,DD,20,AA,C1,C9,FF,F0,06,20,B7,C1,4C,D3,C0
5014 DATA AD,03,C0,F0,1C,20,AA,C1,8D,06,DD,20,AA,C1,8D,07
5015 DATA DD,A9,80,8D,0D,C0,A9,59,8D,0F,DD,68,A8,68,AA,68
5016 DATA 40,4E,0D,C0,4C,FB,C0,48,30,34,AD,0E,C0,30,1C,F0
5017 DATA 1C,2C,0E,C0,70,03,20,5C,C1,CE,0E,C0,AD,0E,C0,29
5018 DATA 3F,8D,0E,C0,68,20,0B,C2,4C,9D,C1,68,60,A9,01,8D
5019 DATA 0E,C0,68,20,FA,C1,20,9D,C1,A9,FF,4C,9D,C1,29,F0
5020 DATA C9,90,F0,0B,C9,80,F0,07,A9,80,8D,0E,C0,30,DC,20
5021 DATA 5C,C1,68,20,9D,C1,A9,41,8D,0E,C0,60,2C,0D,C0,30
5022 DATA 18,AD,09,C0,8D,06,DD,AD,0A,C0,8D,07,DD,A9,59,8D
5023 DATA 0F,DD,A9,80,8D,0D,C0,30,16,AD,06,DC,49,FF,C9,FF
5024 DATA D0,02,29,FE,20,9D,C1,AD,07,DC,49,FF,20,9D,C1,A9
5025 DATA FF,8D,06,DC,8D,07,DC,A9,59,8D,0F,DC,60,AC,05,C0
5026 DATA EE,05,C0,EE,03,C0,99,14,C2,60,AE,04,C0,EE,04,C0
5027 DATA CE,03,C0,BD,14,C2,60,48,AD,02,DE,29,02,D0,14,AE
5028 DATA 08,C0,68,9D,14,C3,EE,08,C0,EE,06,C0,A9,B5,8D,00
5029 DATA DE,30,04,68,8D,01,DE,60,78,AD,10,C2,8D,14,03,AD
5030 DATA 11,C2,8D,15,03,AD,12,C2,8D,18,03,AD,13,C2,8D,19
5031 DATA 03,A9,03,8D,00,DE,20,84,FF,60,F0,0E,18,6D,0C,C0
5032 DATA 30,06,D0,06,A9,01,D0,02,A9,7F,60,18,6D,0B,C0,60
```

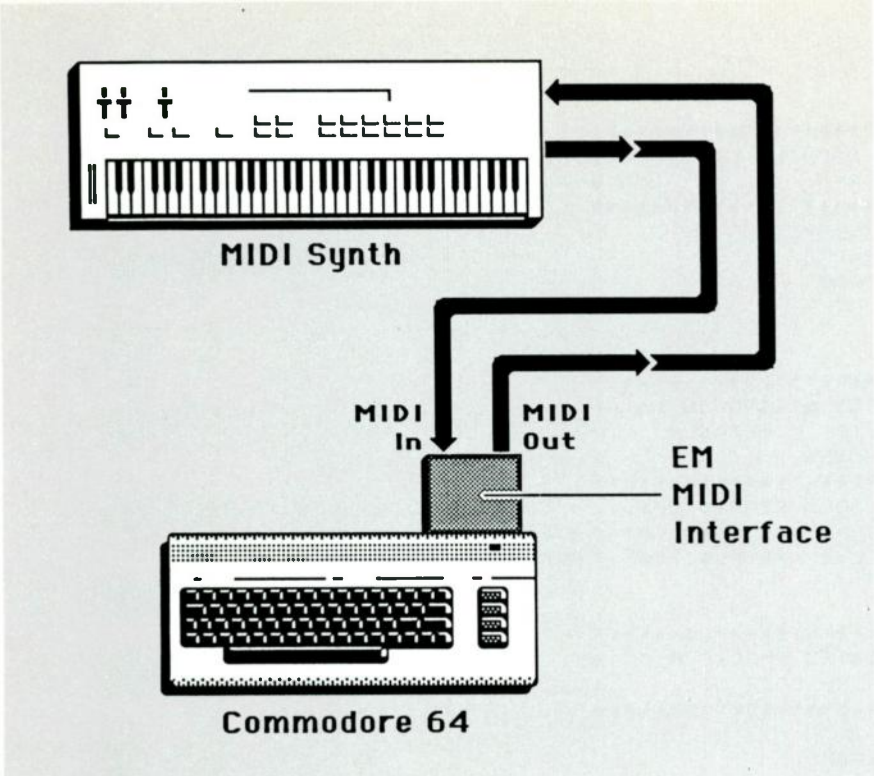


Fig. 1 *Using MIDI delay with a single synthesizer*

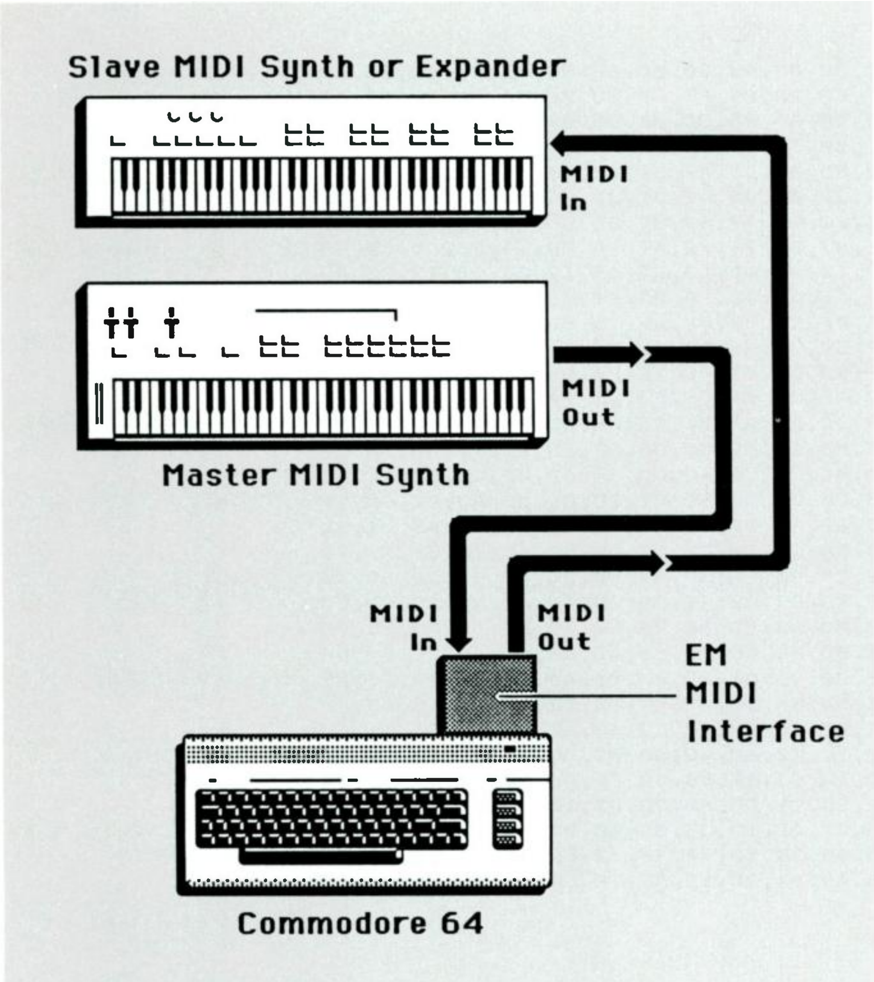


Fig. 2 *Creating echoes with a different timbre with a second synth*

much data the buffer can hold. At shorter delay times, buffer overflow is never a problem.

TRANSPOSE

The transpose interval can range from -24 to +24 semitones. Be aware that big transpositions can take you out of the reception range of your synth. In certain situations this can be desirable, but generally this will cause unpredictable results.

VELOCITY ADJUSTMENT

The velocity adjustment works by adding the selected adjustment to the received velocity. Range checking is performed in the machine code so that velocities of zero are never changed and adjusted velocities never exceed 127 ($7F hex).

The adjustment can range from -127 to +127, but large negative adjustments will cause unexpected results if they yield velocities less than zero.

USING MIDI DELAY

Fig. 1 shows how to use MIDI delay effects with a single synth. Thin-sounding MIDI synths really come alive with delay times in the 1-50 millisecond range. Transposing the delayed sound up or down an octave fattens the sound even further. Delay times of one second or so add ethereal echoes (and adding in some transposition can sound great).

> *"Thin-sounding MIDI synths really come alive with delay times in the 1-50 millisecond range"*

Try some non-octave transpositions. At short delay times, they add realism to bell and percussion sounds, and can add interest to ho-hum factory patches as well.

In **Fig. 2**, the addition of a second synth or expander module makes it possible to select both the original and delayed timbres.

And no matter what you do, don't forget to use MIDI delay with your drum machine! Slapback echoes and flams can have perfect fidelity, and longer echo times work just fine for polyrhythmic effects.

These are just some of the possibilities of MIDI delay; as you experiment, you will surely find more. So plug in your interface and start typing! Before too long, you'll be exploring a whole new world of special effects. EM

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Is MIDI-controlled audio mixing in your future? Here are the basics, as well as a real-world example of how MIDI can be adapted to uses beyond those originally conceived for the spec.

MIDI Muting and Mixing for the Masses

BY J.L. COOPER

The first priority for the MIDI protocol was to interface synthesizers, sequencers, and drum machines—equipment made by the various companies that helped shaped the MIDI specification. Yet MIDI has grown beyond those limits, and has now found its way into the control of lighting, signal processors (reverb, DDLs, etc.), and even guitar amps. MIDI is currently on the verge of entering another field of musical electronics: audio mixers in general, and mixer automation in specific. In fact, at least four manufacturers now tie MIDI to audio mixers in one way or another.

There are two basic ways to marry MIDI control to a mixer. The simplest form of control takes "snapshots" of settings, much like the way a programmable analog synthesizer takes a "snapshot" of the front panel control settings. The second method produces an effect more like taking a motion picture, where multiple still frames blend together to give the illusion of continuous variation. Let's start with the snapshot approach.

SNAPSHOTS

Fig. 1 is a block diagram of how MIDI might control a typical stage mixer. All the usual "controlling" elements of a mixer are present: faders, switches, and knobs. However, these do not control the audio directly, but instead provide data on their positions to the internal microprocessor. In turn, the microprocessor connects to elements such as Voltage Controlled Amplifiers (VCAs); these are analog circuits that can create gain changes, just like pots and sliders. VCAs can control gain settings, equalizer settings, pan (stereo placement) settings and the like. The microprocessor also needs to drive some kind of switching element (relays and FETs are most common) in order to control signal routings. So far, all we have really done is insert a microprocessor into the normal flow of control in our mixer.

The microprocessor also connects to some amount of Random Access Memory (RAM), in which we can save the data that represents the various pot and switch settings. In fact, we can usually save entire "sets" of settings, each of which we would call a "program." (Naturally, we would want this RAM to have battery backup so that stored control settings would be retained even if the power was removed.)

Just as you step through different synth programs as you play different parts of a song, each program can step through snapshots of how controls are set at various points in time. In the simplest form of snapshot control, there is no concept of having a fader slowly move up, or having a pan move from left to right. If we wanted to do this, we would take a series of snapshots, each just barely changed from its predecessor. However, if the changes are small and there were many snapshots, this would use up a lot of program locations.

We could easily add "movement" by adding a parameter in the program that would tell the microprocessor how fast it should change ("segue") from one snapshot to another; for example, if one snapshot had a fader all the way up, and the next had it all the way down, we could define the amount of time to fade out via the "segue rate." A more flexible version might have a separate segue rate associated with the more important controls (i.e. pan and reverb send as well as master level).

So far we've described how to automate a mix, but we haven't yet made the connection to MIDI. Actually, this isn't a tough connection to make. The MIDI specification defines a *program change* command, which is typically used to switch programs on a synth. Any one of 128 different programs can be called up by specifying the appropriate MIDI data that calls up one of those programs. However,

Jim Cooper is well-known in the world of equipment modification, design, and manufacture. He is the MIDI columnist for Keyboard *magazine and is currently head of the MMA (MIDI Manufacturer's Association).*

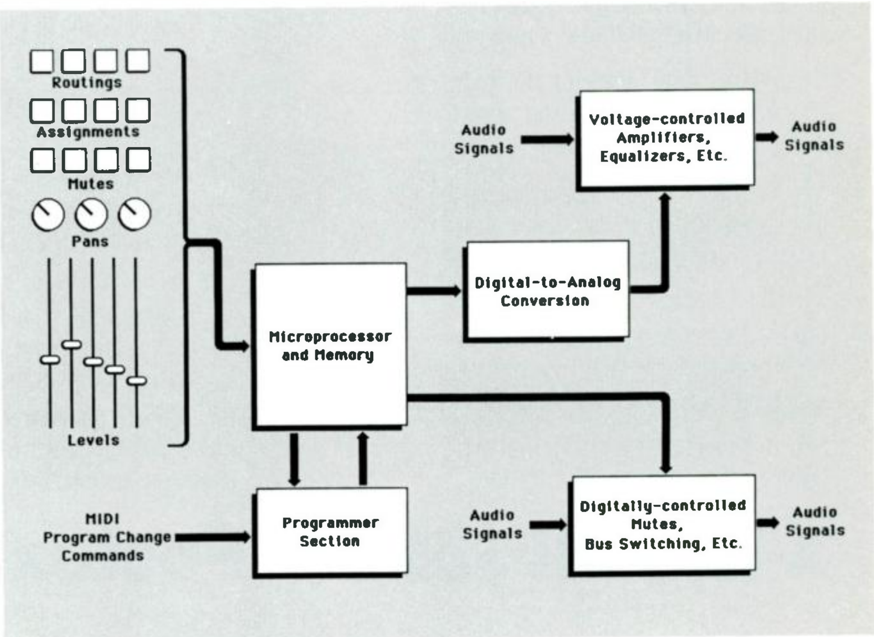

Fig. 1. *MIDI-controlled programmable muting mixer*

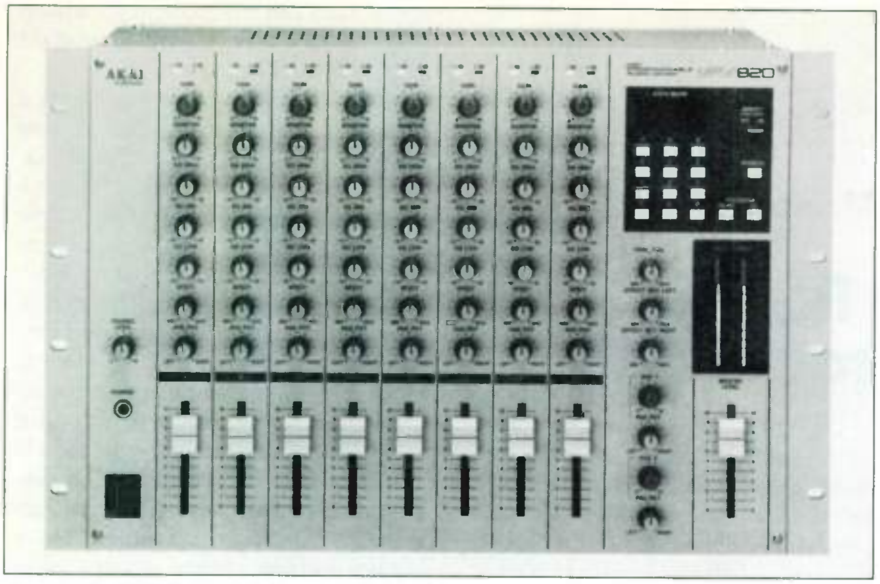

Fig. 2 *Akai MPX 820 MIDI-controlled audio mixer*

program change can just as easily change snapshots on a MIDI mixer. The mixer would have its own MIDI Channel Number, and the controller (whether it be a sequencer, computer, or master keyboard) would send the MIDI Program Change command at the proper moment for the proper snapshot and Presto—new mixer setup!

The AKAI MPX-820 (**Fig. 2**) is the first real MIDI-controlled mixer to appear on the scene. As of this writing, it has not yet hit the streets so some of the details may change a bit. However, the main features are eight input channels with programmable (i.e. memorizable) Level, Pan, Effect Send, EQ Low, EQ Mid, EQ High, and Monitor; two output channels with programmable Effect Rcv (Receive) Left and Right, Aux 1 and 2 Level, Aux 1 and 2 Pan; programmable fade time from 40 milliseconds to 15 seconds; and 99 programs.

Moving Pictures

Fig. 3 shows a different approach to MIDI-fied mixing. Here we have the same basic inputs (sliders, knobs, switches) which you set, and the same microprocessor-driven outputs (VCAs, relays, FETs) that control the audio flow. What is different is that there is no large chunk of RAM holding the settings. Notice that the microprocessor block is divided into two sections: one section converts control set-

—page 65

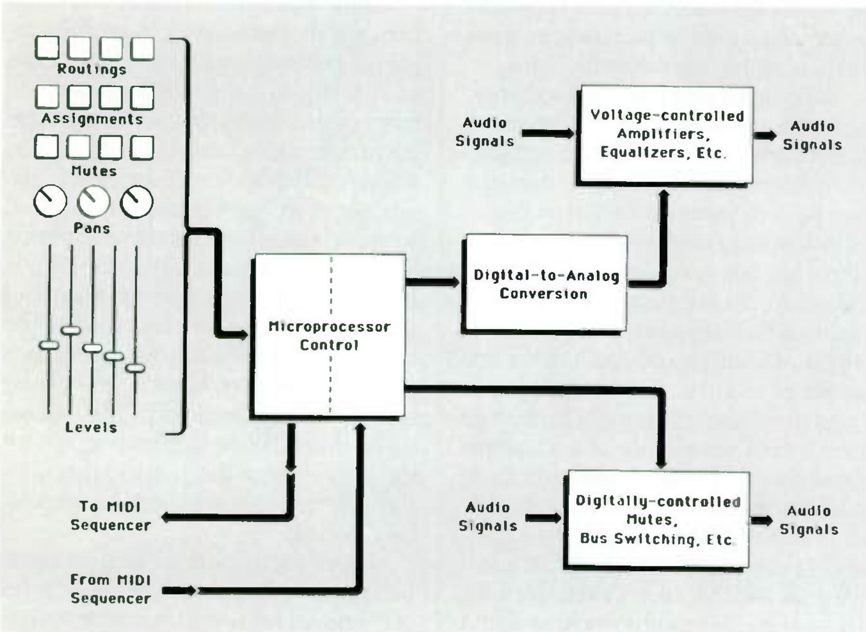

Fig. 3. *MIDI-controlled programmable muting mixer*

Now that you have sounds stored in your computer, here's how to splice, dice, chop, and puree them into something totally new and different.

Making Waves, Part 2: Digital Signal Processing

BY PETER GOTCHER

Part 1 of "Making Waves" (July 1986 EM) covered the use of computers to display sound waveforms and edit digitally sampled sounds. Digital editing allows you to modify sampled sounds with microscopic accuracy using techniques such as "cut and paste" and waveform drawing. This month we'll discuss audio applications of *digital signal processing*—the use of computers to process digital sound.

Digital editing is similar to traditional magnetic tape splicing; pieces of a recorded sound can be cut, rearranged, and reconnected to form a new sound. However, digital editing can be performed with a typical accuracy of 1/50,000th of a second! No razor blade-wielding recording engineer can match that tolerance. Best of all, with a digital display of the sound waveform, edit points can be chosen that "match" perfectly, thus creating a seamless edit. Clearly, digital editing is more precise than "analog" editing (tape splicing).

Digital Signal Processing Basics

Digital signal processing (DSP for short) offers similar advantages over analog signal processing. Let's look at the basic differences between DSP and analog processing. Analog processing equipment (such as equalizers, compressors, flangers, etc.) use electronic circuitry to modify an analog audio signal (i.e., a signal that consists of a continuously varying voltage). Analog electronic circuitry often requires calibration, and can perform erratically as components deteriorate or fail, calibration goes out, etc. Each piece of analog processing equipment in the audio chain also adds some amount of noise.

Peter Gotcher is president of Digidesign, *best known for pioneering visual editing for personal computers with the* Sound Designer *series of software for the Emulator II, Prophet-2000, and Ensoniq Mirage.*

Digital signal processing uses mathematical formulas (called *algorithms*) to modify the actual digital data of a sampled sound. These algorithms are computer software programs—they never "wear out" or suffer from calibration error. In addition, well-designed algorithms for most processing tasks add less noise to the audio signal than their analog equivalents.

To better understand how these algorithms process digital sound, let's quickly review a few basic concepts about how computers store sound. The analog audio voltage is sampled (measured) thousands of times each second, and each individual sample is stored as a digital "word." One digital word can consist of any number of "bits" (typically 8, 12 or 16). A bit is either a "0" or a "1"—the more bits in the word, the greater the accuracy of the measurement. Greater accuracy means less distortion and more dynamic range. At any rate, each of these digital words (samples) is a measurement of the sound's amplitude (loudness) at an instant in time. Running the resulting "stream" of these samples through a nifty electronic device called a *digital-to-analog* converter reproduces the original audio signal.

Suppose we want to somehow change, or process, our sound digitally. The sound data is stored as numbers in the computer's memory, therefore, any processing must be done using mathematical formulas. A simple example would be an algorithm for mixing two sounds—sound "A" and sound "B." In this case, our algorithm might tell the computer to "take the first sample of sound A and add it to the first sample of sound B. Repeat this process for each subsequent sample in the two sounds (add sample two of A to sample two of B, etc.)." The result will be the same as mixing the two sounds using a conventional mixing board, but with the advantages of digital processing such as minimal noise and distortion. (You might wonder what happens if the resulting sound has a greater amplitude at times than our digital word can represent; to get around this problem, we might "scale" the sound's amplitude by dividing it in half.)

If digital mixing is so easy with a computer, why aren't digital mixing boards available? First, a few digital mixers *are* available, but their prices are astronomical (in excess of $500,000 for a Neve digital console). This is due to the vast amount of memory and computing power required to mix multiple channels of digital audio in *real time*. The term "real time processing" means that the computer can process (in this case mix) the digital audio as fast as it enters (and exits) the computer, without causing any delays in the audio playback. If the computer cannot process the digital audio fast enough to maintain the flow of input to output, it is a *non-real time* system—it must calculate the mix at a slower rate, then play back the results. Digital reverb units, delay lines and pitch shifters are real time processors. Current software/personal computer systems that perform digital signal processing are typically non-real time systems. However, the price of computing power is rapidly falling—real time digital signal processing on personal computers is not far away.

Real time systems have one major advantage in that you can listen to changes while you make them (such as increasing delay time, adjusting pitch shift, etc.). With non-real time systems, the sound must be recalculated by the computer each time you want to hear the effect of any change in the processing. For example, with a non-real time digital equalizer you cannot "twist the knob" until it sounds right.

Some algorithms do not benefit much from real time processing. A good example is a *crossfade looping* algorithm for use with a digital sampling keyboard. As many owners of sampling keyboards have learned through painstaking experience, it is very difficult to create a good, consistent loop in some complex sampled sounds (human voices and horn or string sections are notorious offenders). This is because the timbres (tonal qualities) of these sounds change very rapidly. The crossfade looping algorithm performs a smooth transition between dissimilar sounds (as opposed to a "butt splice" where the end of the loop butts up against the beginning of the loop) so that even difficult-to-loop sounds can be successfully looped.

Crossfade looping is based on a simple crossfade algorithm that simply "fades out" one sound (or section of a sound) while "fading in" another sound or sound section. The crossfade looping algorithm consists of a pattern of multiple crossfades that use sound data from both inside and outside the looped section of the sound. This pattern of crossfades essentially fades in sound data from the loop end area at the loop start, and vice versa. As a result, the loop start and loop end points have similar waveforms, so the loop sounds "smoother."

DIGITAL EQUALIZATION

Ready for a more complicated algorithm? Let's look at digital equalization (digital filtering), more specifically a *low-pass* filter. A low-pass filter rolls off (attenuates) all frequencies above a certain "corner" frequency—it allows the *low* frequencies to *pass,* hence the name. If we look at a sound waveform displayed on a computer screen, we notice that lower frequencies take longer to go through their repeating (periodic) patterns, whereas high frequencies change very quickly (see **Fig. 1**). To remove the high frequencies from the sound, we must simply slow down the rate at which we allow the waveform to change. One algorithm used to achieve this "slowing" of the waveform works by averaging the amplitudes of each successive sample with the previous sample. If the next sample changes by a large amount, it is probably a high frequency with a short period. By averaging its value with the previous sample, we "slow" the waveform's *rate of change* and remove the high frequencies.

This is a very simple digital filtering algorithm. Algorithms for parametric type filters with variable center frequency, boost/cut amount, and bandwidth are far more complex. Algorithms for compression and sample rate conversion are also complicated. Mixing and simple crossfading (crossfade looping is more complex) are relatively simple algorithms.

THE FAST FOURIER TRANSFORM

In the early 19th Century, a French mathematician named Jean Baptiste Joseph Fourier developed the theory that complex sound waveforms were actually composed of many simultaneous simple waveforms (sine waves) at different frequencies and relative amplitudes (levels). His theory has proven correct, and a special type of digital signal analysis program, the Fast Fourier Transform (FFT) can be used to divide a digital sound into many separate frequencies. The envelope (changes in amplitude as time passes) of each frequency can be displayed on a pseudo-three dimensional surface. The result is a "mesh" surface that shows the change in amplitude of each frequency (the spectral content) as time passes. You may have seen pictures of this type of display in advertisements for computer music systems—it typically looks a bit like the surface of the moon. The FFT display is the true "fingerprint" of the sound. It reveals a great deal more information about the nature of a sound because it provides frequency information that the simpler two dimensional time and amplitude waveform display does not.

Fig. 2 shows an FFT display of the

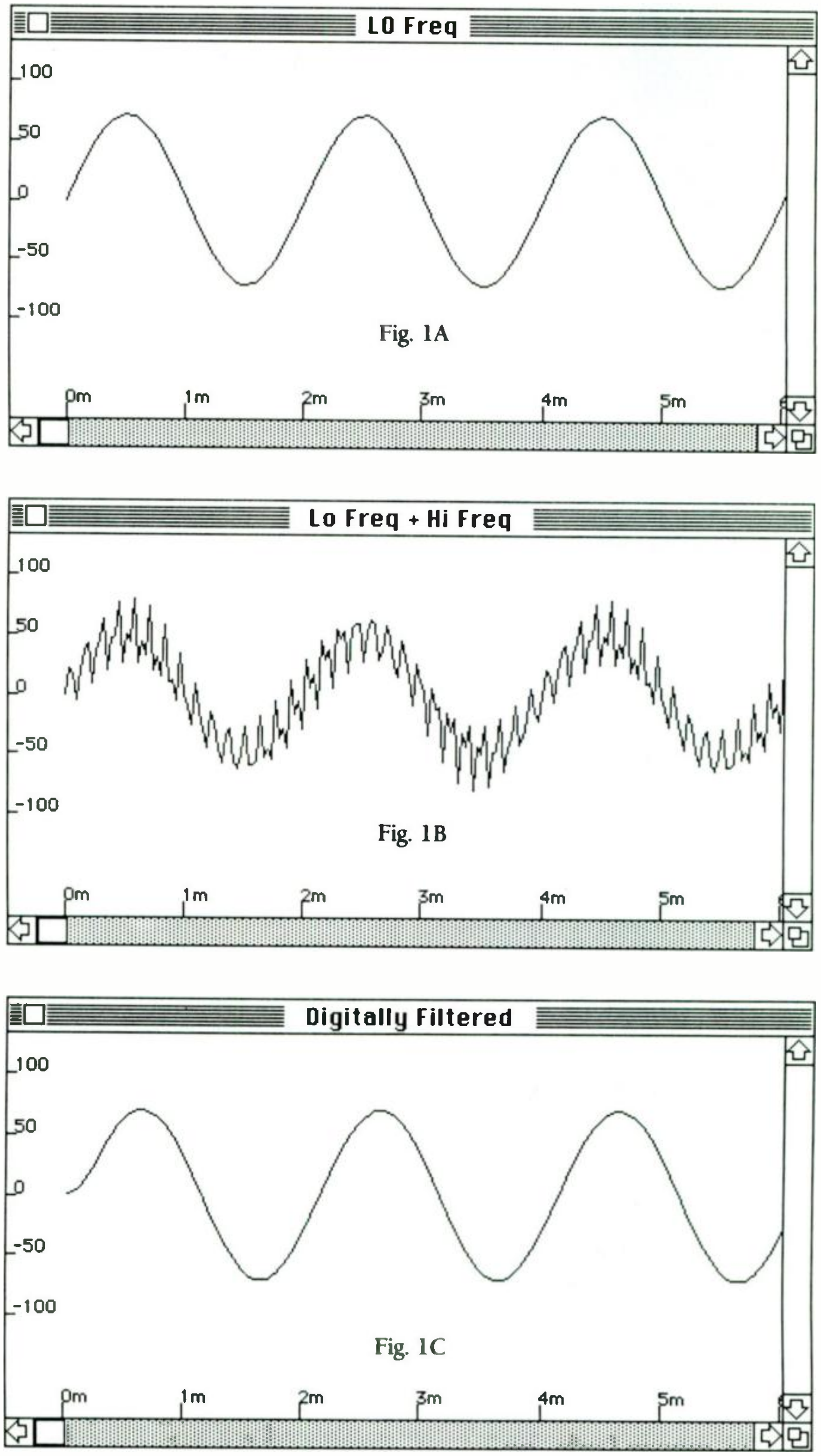

Fig. 1A *shows a low frequency waveform,* **1B** *shows the same waveform with added high frequencies;* **1C** *shows the waveform in* **1B** *with high frequencies removed via low-pass filtering. The horizontal axis is calibrated in milliseconds and the vertical axis in relative amplitude units.*

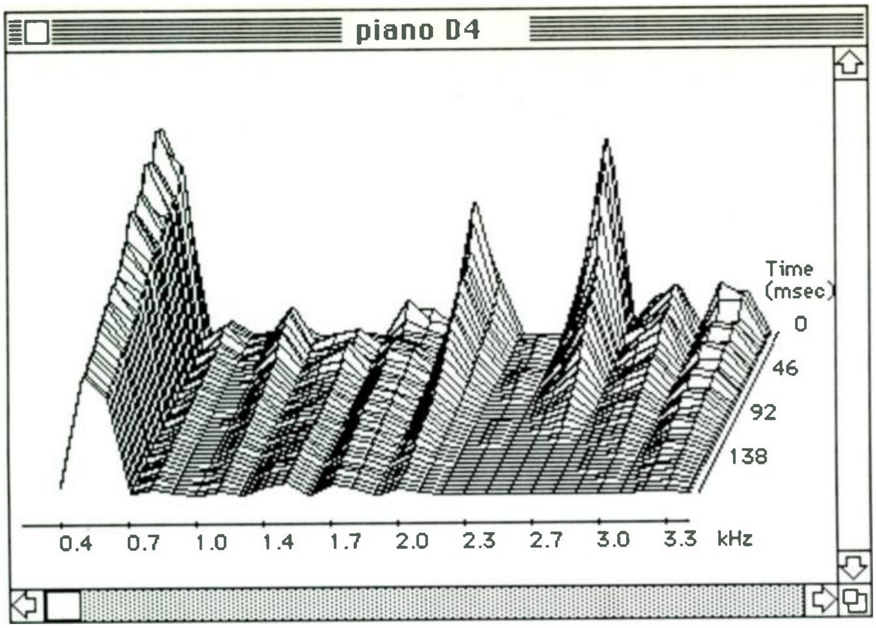

Fig. 2 *FFT display of the attack of a single piano note.*

first 150 milliseconds (0.15 seconds) of a piano note. *Frequencies* are displayed from the left (lower frequencies) to the right (higher frequencies), and *time* is displayed from the "back" of the 3-D display (the sound's start or *attack*) to the front of the display. The height of the "peaks" indicate the amplitude or loudness of each frequency, at each point in time. If we read the frequency values labeled along the front of the display, we can see that the piano note has a strong *fundamental frequency* at about 400 Hertz (each frequency slice is about 100 Hz), and weaker frequencies called *overtones* or *harmonics* at higher frequencies. Notice that the higher frequencies die away (*decay*) more quickly than the lower frequencies. This gives the piano its characteristic bright attack, that quickly gives way to a duller "sustain." With a little experience, you can learn to relate the patterns you see in the FFT display to the actual sound you hear.

Signal Analysis/Resynthesis

However, the best is yet to come. Mr. Fourier's theory also states that any complex sound can be recreated (resynthesized) from an appropriate set of sine waves using another DSP program, the Inverse Fast Fourier Transform (IFFT). By the way, older Fourier transform programs were too slow for practical uses—hence we call the modern versions "fast." If we combine the FFT and the IFFT, we have the basic tools for a very powerful sound processing technique: analysis/resynthesis.

Analysis/Resynthesis is a general digital processing technique that has many specific uses. Let's consider a few examples. Perhaps we have sampled a piano note, and the sample is perfect except for some extraneous noise from an air conditioner in the room. With normal sound editing techniques (digital or analog), it would be impossible to remove the offending noise because it occurs at the same time as the piano note. We may be able to minimize the noise using an equalizer, but the equalization will certainly affect our piano sound. Using FFT analysis, we can *see* the exact frequency of the noise, reduce the amplitude of that frequency, then use the IFFT to resynthesize the sound—minus the air conditioner. Of course, no technique is perfect. The accuracy of the resynthesized sound (how closely it resembles the original) is limited by the resolution (number of separate frequency bands) of the FFT/IFFT. Unfortunately, the more bands the FFT has, the longer it takes to compute the results. The quality of FFT analysis/resynthesis is determined by the available computing power, and the user's patience.

Another interesting application of analysis/resynthesis is a DSP program called the *Phase Vocoder*. In general, vocoders divide sounds into separate frequencies, then process and combine the individual frequencies to create new sounds. Analog vocoders that use banks of narrow bandpass filters to separate the sound into different frequencies have been available for many years (they are typically used to "combine" human voice with instrument sounds). These are called *channel vocoders* because of their multiple frequency channels (bands).

The phase vocoder is based on FFT analysis, which offers two primary advantages over analog vocoders—the FFT remembers the phase of each frequency, and it is practical to have many more frequency bands. Because the phase vocoder retains phase information, it is capable of separating the *frequency* components of a sound from the *time* components. As a result, the length of a sound can be stretched or shrunk without altering the pitch, and pitch can be raised or lowered without shortening or lengthening the playback time (as conventional samplers do). The program can also mix or merge the frequency and/or time components of several sounds, creating some very unusual, interesting results. The phase vocoder is a very powerful tool that will soon become available to musicians in computer music systems. Alas, it is based on rather complex ideas that are too esoteric to present here; I recommend you do a little extra reading on the subject if you are technically inclined (see bibliography).

And Now For The Future...

Many of the digital signal processing tools and techniques I have described (including digital mixing, equalization, crossfade looping, etc.) are available now in relatively low cost systems composed of a sampling instrument, a personal computer and software. Other techniques (such as analysis/resynthesis and the phase vocoder) are available only in very expensive, dedicated computer music systems, or they have been developed in universities and are not commercially available. Fortunately, the price of computing power is falling, the creative needs of musicians are increasing, and digital audio is emerging as the medium of the future for making and listening to music. Just as today's electronic musicians must be familiar with analog recording and processing techniques, tomorrow's musicians will need to understand digital recording, editing and processing of sound. EM

Suggested Reading

Moderately technical:
Foundations of Computer Music, ed. by Curtis Roads and John Strawn, MIT press

Very technical (advanced mathematics):
Digital Signal Processing, Alan V. Oppenheim and Ronald W. Schafer, Prentice-Hall Inc.

Theory and Application of Digital Signal Processing, Lawrence Rabiner and Bernard Gold, Prentice-Hall Inc.

You can successfully perform many upgrades and minor repairs yourself —and here's the information you need to get started.

Service Clinic: DIY Service

BY ALAN GARY CAMPBELL

Many equipment users would like to service their own gear in order to save money, expedite emergency repairs, and learn more about the inner workings of electronic music devices. But, which repairs can you perform simply and safely, what tools and equipment will you require, and where can you find the needed technical information?

GETTING STARTED

Before you can service your own gear, you must be willing to invest the time, energy, and money required to learn about musical electronics. Taking on service that's beyond your abilities, without getting proper assistance, can lead to major repair bills, equipment damage, and even personal injury and liability—*don't do it!* Do-it-yourself (DIY) service as a one-shot, "let's poke around to see if there's a loose wire" affair is inadvisable. But, you have to start somewhere, and you'll have to persevere to expand your abilities. I remember my first "major" repair job: a Model D minimoog. I got about halfway through it and *freaked*—how was I ever going to get this thing back together again? Well, I did complete the repair, and I even went on to learn a bit more about electronic music service! So can you. Always proceed with care and caution, but don't let fear of the unknown prevent you from opening up that first unit.

SAFETY FIRST!

Many electronic music devices contain hazardous AC voltages. *ALWAYS UNPLUG LINE-POWERED EQUIPMENT FROM THE AC RECEPTACLE BEFORE YOU DISASSEMBLE OR SERVICE SUCH EQUIPMENT.* Of course, it is often necessary to have equipment powered-up while disassembled, in order to perform certain service procedures; however, such repairs should not be attempted until you have gained considerable experience with electronics troubleshooting and safety procedures. For example, some Korg and Roland synths have multiple, exposed line-voltage terminals in the power supply sections. A technician would tape up these terminals or cover the entire section with a towel or other insulating material, to avoid inadvertent line-voltage contact while servicing other parts of the synth. If you were not aware of the shock hazard, and were working inside such gear, you could be seriously injured or even electrocuted. You can't afford to be unprepared or careless around line voltages. Never work on line-powered equipment when you're tired, upset, or ill. Don't take chances if you are not aware of proper safety procedures. *Practice safety at all times.*

Alan Gary Campbell is owner of Musitech™, an electronic music consulting firm. Musitech is highly involved in electronic music service and modifications, and has been cited by Moog Electronics, Inc. as one of its top ten authorized service centers worldwide. Campbell is also a synthesist and technical writer.

"...don't let fear of the unknown prevent you from opening up that first unit"

INFORMATION SOURCES

You don't need a college degree to do electronic music service; in fact, the hands-on skill you need is the sort of knowledge you usually acquire on your own. The MIX Bookshelf offers two virtually indispensable references for the beginner: *Electronics Projects for Musicians* (by EM editor Craig Anderton); and *The Heil Guide to Concert Sound.* Your local Radio Shack has several excellent titles (stock numbers are in parenthesis), including: *Getting Started in Electronics* (276-5003); *Dictionary of Electronics Terms* (62-1391); *Basic Electronics Technology* (62-1394); *Using Your Meter* (62-2039); *Building Speaker Enclosures* (62-2309); and the Archer *Semiconductor Reference Guide* (276-4009). Also noteworthy are technical articles in EM (and back issues of *Polyphony*), *Modern Electronics*, and *Radio-Electronics*, to name but a few.

The most important technical reference for repair work is the equipment *service manual.* Unfortunately, only a few manufacturers provide service documentation and/or schematics with their gear; most seem to feel that the end user rarely, if ever, reads or retains even the owner's manual, much less the service manual—which is an expensive document to produce and may contain proprietary information. It is imperative that you acquire a service manual for your gear before attempting self-service. Take down the model name, model number, and serial number of the equipment in question, then contact the service department of the equipment manufacturer and request ordering information. For older gear and items that are no longer manufactured, contact one of the various independent clearing houses for parts and service data, such as Music Dealer Service (ARP, Fender parts/manuals), or Magic Music Machine (Oberheim OB-Xa, older synths). See DataBank for addresses.

Studying the materials recommended above won't make you an instant expert, but it will help you to understand the DIY and service articles in EM, and you'll be well on your way to service self-sufficiency.

TOOLS AND EQUIPMENT

Quite a bit of DIY service can be accomplished with only a Volt-Ohmmeter (VOM), some soldering equipment, and a selection of screwdrivers, pliers, and miscellaneous tools. You'll need a 20,000 Ohms/Volt VOM that can read DC volts, DC current, AC volts, and resistance functions (Radio Shack 22-201, or similar); a

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Do smaller labels stand a chance of success? What's the relationship between the charts and the record industry? Find out the answers in this month's installment.

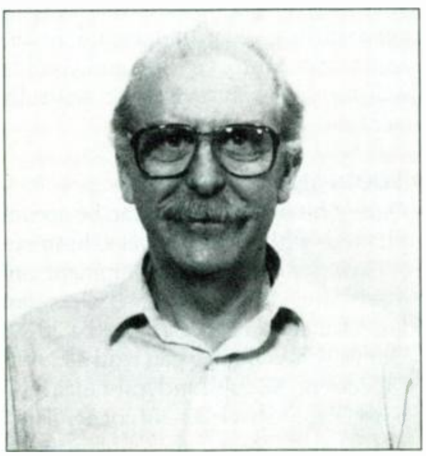
Inside the Record Business Part 2: Smaller Labels and the Print Media

BY AUGIE BLUME

Last month, we talked about how life works at the major record companies. Now let's turn to the moderate sized labels, those who have chosen to become affiliated with the majors, and then discuss the even smaller independent labels.

THE LABEL

Each moderate sized label has a somewhat specialized musical focus that supplements the overall marketing objectives of the major label. They have considerable autonomy in the choice of music they record and release. Their staffs are necessarily more compact, with fewer people to perform the various job functions. They are forced to depend in large measure on the major label for the support of their promotion and marketing departments. The majors are constantly looking for moderate sized labels who have shown themselves to be musically aware of the marketplace, and aggressive in seeking their share of that market. For example, when a label like Windham Hill emerges on the scene, develops a new music market over a period of time and turns it into a profitable one thanks to sound business practices, a label like A & M naturally becomes more than interested.



When a label affiliates with one of the majors they benefit with increased airplay and sales. On the other hand they can be hurt by getting caught in a priority crunch with the major, when their record, although commercially viable, may have other records ahead of it in terms of promotional priorities, and then it dies on the vine due to lack of support by the major.

THE CHARTS

Each record has a life of its own and has a crucial number of weeks during which it can be effectively promoted in order to keep it alive in the eyes and ears of the radio programmers and the trade charts.

The national music trade charts play a vital role in this process. If a record does not appear to sustain a continuing upward movement on the charts it can be the kiss of death for that record. Radio programmers want to be safe and most are not prone to taking chances on records that appear marginal. There are a variety of charts. The one that radio programmers have placed most confidence in is *Radio and Records,* a weekly trade publication in Los Angeles. It has a "Back Page" with four categories, Contemporary Hit Radio (CHR) with 40 slots, Black/Urban with another 40 positions, Album Oriented Radio (AOR) with 30 slots, and Adult Contemporary with another 30 positions. These charts are compiled from reports from those radio stations that report their "adds" (records they've added to their playlist) and chart movement to *Radio and Records* each week. It is all based on airplay. The premise is that if enough stations add a given record in a relatively brief period of time it gets called a "hit." Their charts do not reflect any sales information whatsoever.

"The premise is that if enough stations add a given record in a relatively brief period of time it gets called a 'hit'"

Radio programmers look each week to that Back Page in *R & R* for records that have achieved a "Breaker" status, indicating they have shown sharp upward movement and look like sure shots. Sometimes, despite their appearance on the chart, records only reach a "mid-chart" status, meaning they do not have the strength to move further up the chart. Some records "stiff" early and fade quickly. Don't forget that the major labels bring considerable promotional pressure to bear on radio programmers, by a variety of means, to keep their most promising records alive. Some of these methods are questionable at best, but after all the radio and record industries have a symbiotic relationship that goes back many years. At times it's almost like a love-hate affair.

While we're on the subject of weekly music charts we should mention the other key trade publications. *Billboard,* in Los Angeles, has charts which reflect each type of music. They are compiled from a combination of radio station airplay reports and sales information from reporting stores. Their charts are used primarily by record stores and the big chain outlets to determine what new records they should consider buying, as well as a gauge to know when to begin thinking about returning unsold products to the manufacturer.

Cashbox magazine in Los Angeles has its charts, but their readers are almost

Augie Blume is a widely-respected expert in the field of record promotion. In 1969, he was named "National Promotion Man of the Year" by The Gavin Report *and in 1978, the readers of* Billboard *awarded him "Man of the Year Award for Independent Promotion." His company, Augie Blume & Associates, has helped gain airplay for Charlie Daniels, ELO, Jefferson Starship, Heart, Holly Near, Eddie Money, the Grateful Dead, and many others. He continues to work with his wife Nancy (herself a well-known personality in the field of record promotion) and maintains a national data base of record labels, producers, radio stations, publications, etc.*

entirely the juke box operators market. Another trade magazine, *The Gavin Report*, is based in San Francisco. It focuses entirely on the music selection process of radio programmers. Their weekly charts are quite similar to *Radio and Records*, in that they receive reports from subscribing stations, and they contain no reflection of sales information. *Album Network* is also based in Los Angeles and their focus is entirely AOR radio. Then to round out the pack we have *Friday Morning Quarterback* (covers Top 40, CHR, some Black/Urban), *The Hard Report* (rock and AOR), *Bobby Poe's Pop Music Survey* (Top 40, some Black/Urban), *The Brenneman Review* (Top 40, Adult Contemporary Pop), *Black Radio Exclusive* (Black/Urban), and *Behind The Scenes* (small market Top 40, some Black/Urban). Each has its own musical focus and chart methodology for picking the "hits" of the day.

The national trade publications tend to be dominated by the advertising dollar of the major labels. These publications dutifully print their press release handouts, and they have a consequent fear to honestly report on the actual state of the industry.

This can't help but raise the question in the minds of some if there could possibly be a connection between that advertising and the charts. Of course, the trades would be the first to disown any such correlation. But like it or not, all of these charts are 95 percent dominated by the major label's releases. It doesn't leave much room for anyone else. Recently I saw a card that had a marvelous quote, so

> *"How boring it would be if all the songs in the forest came from the top ten birds"*

I'll pass it along for what it's worth, "How boring it would be if all the songs in the forest came from the top ten birds." I don't know who thought of this, but I'd have to say Amen.

A Piece of the Pie

Let's return to the record companies. There are approximately 1,900 other record labels who scramble hard for their share of the 15 percent that remains of the pre-recorded music retail dollar. These labels are very small in terms of numbers of employees, and most often they are very focused on very specific genres of music. They get into the record business because of their genuine love for a certain kind of music and they want to help keep that music alive within their limited means. We must point out that running a small label takes a lot of hard work, long hours, and much perseverence and patience.

These labels fall into every type of category—folk, jazz, blues, ethnic, rap, reggae, "new age," dance, classical, women's music, minimal music, and "new music," a phrase that attempts to classify adventuresome rock and other new musical forms that are out of the mainstream.

Most of the small labels are underfunded, and once a record is ready to be released, they are left with very little money to promote and market their record. Jazz radio is very open and supportive, their people are really into the music, and they aren't locked into the charts and the major labels hype. Small labels get their airplay primarily from college radio and listener supported radio; these stations don't have to be afraid to take chances, and are into new music and artists because they love it too. The airplay small labels receive amounts to one or two plays once a week if they are fortunate enough to get into playlist rotation at all, rather than the two or more plays a day they would receive at the larger market commercial stations. Relatively small numbers of people listen to college and listener supported stations. Another limiting factor is the typically low powered signal and small broadcast area. As a result, it takes small labels much longer to build any kind of audience recognition for their artists.

DISTRIBUTION

Distribution is one other major difficulty these small labels face. Most of them are handled by independent distributors who are either undercapitalized or shortsighted (or both). These distributors have a small, low paid sales staff, and, more often than not, employ no promotion people. They are catalog sales oriented, and are generally not equipped to aggressively promote and market a label's records. They list each new record in a monthly new release sheet that goes out to all their retail accounts, and if they get an order they fill it, but that depends on whether the record is in stock, and if the account is on credit hold.

Smaller labels tend to have great difficulty in getting their records or press material reviewed in most of the trades. *Billboard* has recently become more open to giving them space. But on the other side of the coin, there are quite a few specialty magazines in every field of music who are ready and willing to support the music, anxious for a chance to review a new record or act, and will print most well-prepared press releases.

> *"Redwoods Records (a small women's label) has sold over one million records in the past ten years without the help of major label distribution"*

Some small labels, if they are really aggressive, do quite well on this level of print media. Some work well with college press, and a handful get their acts and records reviewed in their local daily and weekly newspapers, mainly due to hard work and persistent effort.

One of the most difficult problems confronting the small labels centers around getting paid by their distributors. The main reason is that their distributors are the last in line to be paid by the retailers. After all, the retailer must pay the major labels first, since keeping a pipeline open to the hits of the day is necessary to stay in business. Small labels have to expect a 90- to 120-day wait to receive payment. This presents a perennial cashflow problem to small businesses and keeps them off balance when trying to plan new releases. These small companies are placed in the position of playing "catch-up" all the time.

The bright ray of hope on the horizon is that the number of small labels that are able to stay alive, and the list of new companies coming on the scene each week, keeps growing steadily. Little by little they are learning how to keep it together. Hey, talk about hope, how many people know that Redwood Records (a small women's label in Oakland, California) has sold over one million records in the past ten years without the help of major label distribution. Rounder Records, in Cambridge, Massachusetts has grown to over 400 records in their catalog, fueled in part by the sales success of George Thorogood and The Destroyers. A large measure of the reason why smaller labels succeed can be attributed to their having realistic expectations, as well as having acts that are out working most of the time and who do their own limited promotion. It all adds up. Of course, we can't leave out something that Dr. John once said about "being in the right place at the right time."

If there are close to 2000 active record companies, there must be another 5000 groups who put out records on their own label. They face the same problems as other small labels, but even more so. Some of these bands are good enough both musically and as performers to develop a sizeable local audience, and eventually sell well enough to attract the attention of a local label that is better equipped to promote and market their record. Some wind up getting a deal with a major. Eddie and The Tide from the Bay Area, who are now on Atlantic Records, is a good example and proof positive that it can happen.

PAYOLA?

There is a need to touch briefly on the fact that federal investigations have turned up substantial evidence to link organized crime with a role in the music business. The links between music and crime are generally kept well out of sight. They have their hands in a variety of activities, they control some record companies, some distributors, some publishing companies, some pressing plants and tape duplication facilities, and hard as it may be to believe, they control some recording artists. They sell their share of records and tapes, millions of dollars worth of cocaine and other controlled substances, and are often into loan sharking and blackmailing people. They own some night clubs, and like very much to be around entertainment and well known entertainers. All of these operations serve as a means to launder money and make it appear to be legitimate. It's always a good idea to be sure with whom you are doing business, and not be overwhelmed by fast times, fast talk, and fast money. Always ask for references and check with others in the business. If it smells fishy, you don't have to get hooked.

However, most of the people in the music/record business are ethical, hard working folks, all trying to do the best they can. There are many hurdles, as you can already see, but they are not insurmountable. Half the battle rests in having your music and your performing act together. The other half lies in having your head together enough to develop a good business sense, and to formulate realistic goals. It will help if you use both halves of your brain, left and right, the logical and the intuitive. And then ask that all important question . . . what does my heart tell me? EM

(The above is excerpted with permission from an upcoming book by Augie Blume on the music business.)

A new way to mix? No, it's the oldest way to mix. . .but it can be incredibly effective in the new world of synthesizers and drum machines.

An Acoustic Mixer

BY TERRY FRYER

During the heyday of Sequential Circuits' Prophet 5, my tech and I decided that a stereo mixer for the five voices would be a useful device. After constructing the mixer and using it for several months on sessions and live performances, I mentioned to Dr. David Luce (Moog Music) how the use of the five voices in a stereo image enhanced the instrument's sound. He agreed, then asked if I had ever considered the possibility of an acoustic mixer rather than an electronic one. I said "no," and then "huh?"

*If you watch TV, you've heard **Terry Fryer's** work in commercials for Levi's, United Airlines, McDonald's, and many other national advertisers. Terry is currently a partner in Colnot/Fryer Music, Inc., a Chicago-based music and sound production company.*

ELECTRONIC MIXING

Any electronic instrument with more than one oscillator or voice needs a way to combine the various sound sources, which usually involves an electronic mixer. There's nothing mystical about this part of things. The output of an ideal mixer is simply the algebraic sum of the input signals, as determined by their individual level settings.

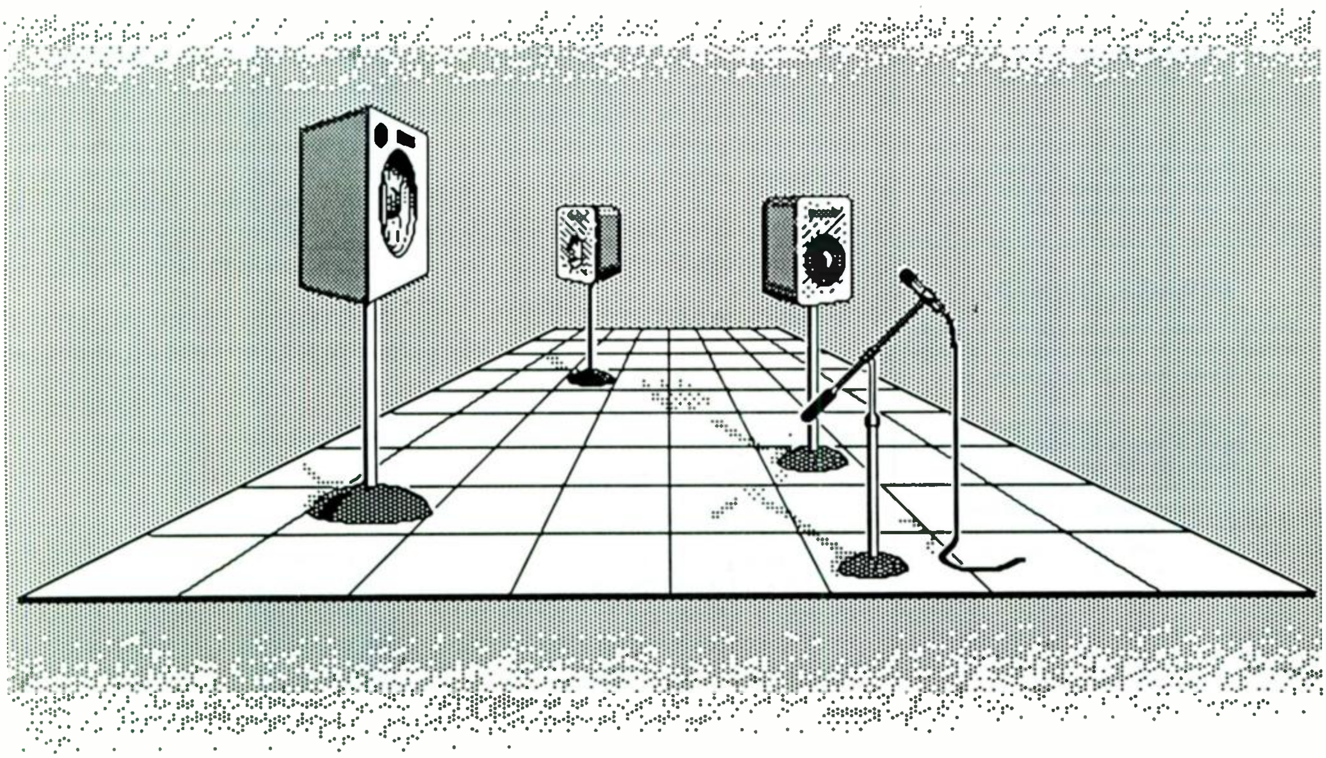
Panning, or the ability to place a sound within the stereo field, is also a common technique. An electronic mixer has a knob called a *pan pot*. Depending on the knob's position, a proportionate amount of the signal goes to either the left or right side of the stereo bus, thus placing a sound at any point from the listener's extreme left to extreme right.

ACOUSTIC MIXING

In a natural environment, things are a little more complex. Suppose you are sitting in a small room listening to a violinist. Part of what you hear comes directly from the sound source (violin) to your ears. Another portion of the sound bounces off the room's boundaries, as well as objects within the room. All of these sounds, direct and reflected, interact with each other to produce an acoustic environment. Due to the intensity and phase relationships of these various signals, it is possible to make accurate predictions about your position in the room, the sound source's position in the room, the shape of the room, and whether the room is "live" or "dead."

Let's add another instrument. . .say, piano. The two sounds now interact to produce nodes and antinodes in the sound. For the most part, this type of interference will enhance the sound of the two instruments.

Our definition of an ideal electronic mixer includes the requirement that no distortion or alteration of the sound occurs. In the example of the piano and violin, a number of very important interactions were introduced that gave us information about the environment. An electronic mixer is not capable of introducing these types of spatial cues. The technique of panning relies upon simple level changes between the left and right speakers rather than the rich interplay of



reflective surfaces that occurs in a live environment. The use of artificial reverberation devices can in part replicate these spatial cues, but there is a much easier and inexpensive method.

HERE'S HOW TO DO IT

Find a room that sounds good for the type of music being played; experiment with putting different amounts of furniture in the room, as well as moving some rugs in and out. The only requirement is that the sound is pleasing to your ears—even a very small bathroom can work for some situations.

Next, assemble enough speakers and amplifiers so that there is one speaker and one amplifier for each voice of the synthesizer. The speakers needn't be incredibly expensive (I initially used five Auratones for the Prophet 5 setup). Sometimes I used a combination of speakers. For amplifiers, my tech went out and got five, ten Watt headphone amplifiers. These amps don't have to be super-powerful, but if you have the budget, a little upgrading at this point does help a lot.

The next step is to route each voice of your instrument to a speaker, put some microphones in the room, and start to play. We decided to be very scientific about our Prophet 5 setup, so we devised a grid system that made it possible to duplicate a particular setup. The floor was marked with masking tape in a standard X-Y grid. The speakers were numbered and then mounted on microphone stands, with markings scratched into the stand. With this system, it was possible to notate exactly where a speaker was in the room. The X-Y grid identified the floor position and the markings on the microphone stand located it vertically. The direction of the speaker was notated by compass settings. A typical speaker description looked like this:

Speaker 1
Position—15 × 27
Height—8
Direction—NNW

The same grid, height, and direction notations were used for a pair of microphones that monitored and recorded the sound. Some mention should be made here about coincidental microphone techniques. Much of the work that I do ends up on radio and television as commercials. At the present time, these are mostly broadcast in monaural. This means that I must be concerned with mono compatibility of the stereo signal. When using normal X-Y microphone techniques, some great stereo effects turn to mush or disappear when turned into mono. Try experimenting with the microphone setup and alternate stereo recording setups.

Speaker and microphone placement have a great effect upon the sound of the mixing system. Don't be afraid to try setups where certain speakers are facing the wall. . .or make a circle around the microphones, with the speakers facing towards the walls.

A couple of interesting techniques were developed while using this setup. The most interesting was the use of EQ (equalization) before the signal was sent to the speakers. In this setting, the EQ sounded much more "musical" since the room acoustics tended to smooth out whatever anomalies were introduced by the EQ. A little EQ was necessary to take out some undesireable room characteristics, but this was usually minimal.

The second technique involved changing the reverberant characteristics of the room. We found that pieces of furniture, rugs, wall hangings, drapes, and reflective pieces of painted plywood all contributed to a widely varied acoustic environment. This part of the process was a little more difficult to document, and involved a lot more listening and guessing.

The acoustic mixing technique is not

> *"It. . .helps to do this when your neighbors are not home"*

confined to synthesizers alone. A great use of this technique is to provide ambience as well as mixing for drum machines. Mixing direct sound and acoustically mixed sound produces some incredibly realistic sounding drum sets. For this purpose, it helps to have a bit heftier speaker and amplifier than the Prophet 5 setup. It also helps to do this when your neighbors are not home.

A blend of acoustic and electronic mixing will help bring a realism to your tracks that is not possible with electronic mixing alone. It also provides an economic alternative to costly artificial reverberation devices. A few speakers, amplifiers, and microphones will cost a lot less than the newest top-of-the-line digital reverberators. Not only that, the realism of the system will exceed that of any presently available digital reverberators. Besides, it's fun to walk into the room, close your eyes, and use your ears to adjust the sound. EM

Need a few extra MIDI cables? It's easy to make your own, and you'll save money while you're at it.

Making MIDI Spec Cables

BY JACK ORMAN

MIDI, the TTL compatible 31,250 Baud asynchronous serial data link, binds together today's electronic music gear. But it can only be as accurate and reliable as the cable that carries the data from unit to unit. While a commercially-available, good quality MIDI cable will cost close to 20 dollars, you can get the necessary supplies at an electronics store and make your own cables for a fraction of the cost.

MIDI's serial transmission method simplifies the cable and connector requirements. The MIDI 1.0 specification calls for low capacitance, twisted pair, shielded wire. I use Belden No. 9271 cable for several reasons. Its capacitance is quite low (12.2 pF/foot), it is flexible, low resistance (12.0 Ohm/thousand feet) and the foil covering provides 100 percent shielding. In addition, this wire is easy to find and reasonably priced; in fact, a 100-foot spool can be bought for the price of a single pre-made ten-foot cable. Other wire manufacturers make suitable cable, but the Belden is my first choice.

Each end of the cable should have a 180 degree, 5-pin DIN plug. Tops in quality are the shielded metal, high reliability plugs with built-in strain reliefs. For me, the extra cost of the metal plug is justified by the shielding and extra strength provided; however, a suitable plastic version can be bought for about 40 percent less if keeping the price to a minimum is important.

THE MIDI CABLE CONNECTION

Fig. 1 shows how to connect the wire to the plug. Only three of the five pins are used by MIDI, with the middle pin (pin 2) grounded to the shield (via the shield wire) at both ends. Similarly, one of the tinned copper inner conductors connects pin 4 of one plug to pin 4 of the other, and the remaining inner conductor joins pin 5 on both plugs. Don't cross-connect these conductors, as this will "lock-up" your unit until the defective cable is disconnected.

Of course, proper soldering technique is vital to making a good MIDI cable. Texts on basic electronics, available at most libraries, should be of some assistance; be sure to use the right solder (60/40 rosin core), a fine tip soldering iron, and apply the right amount of heat—enough to completely melt the solder, but not so much as to overheat (and possibly deform) the plug or cable insulation.

MIDI CABLE TESTER

Now that you've made some cables, a simple cable tester would certainly come in handy (**Fig. 2**). This one uses two

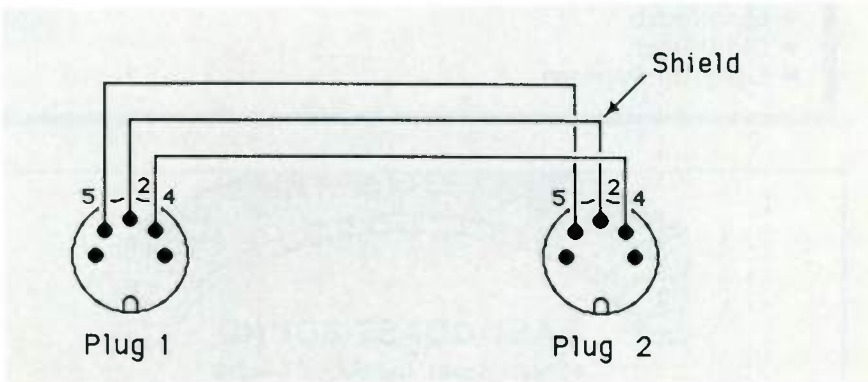


Fig. 1 *MIDI Cable*

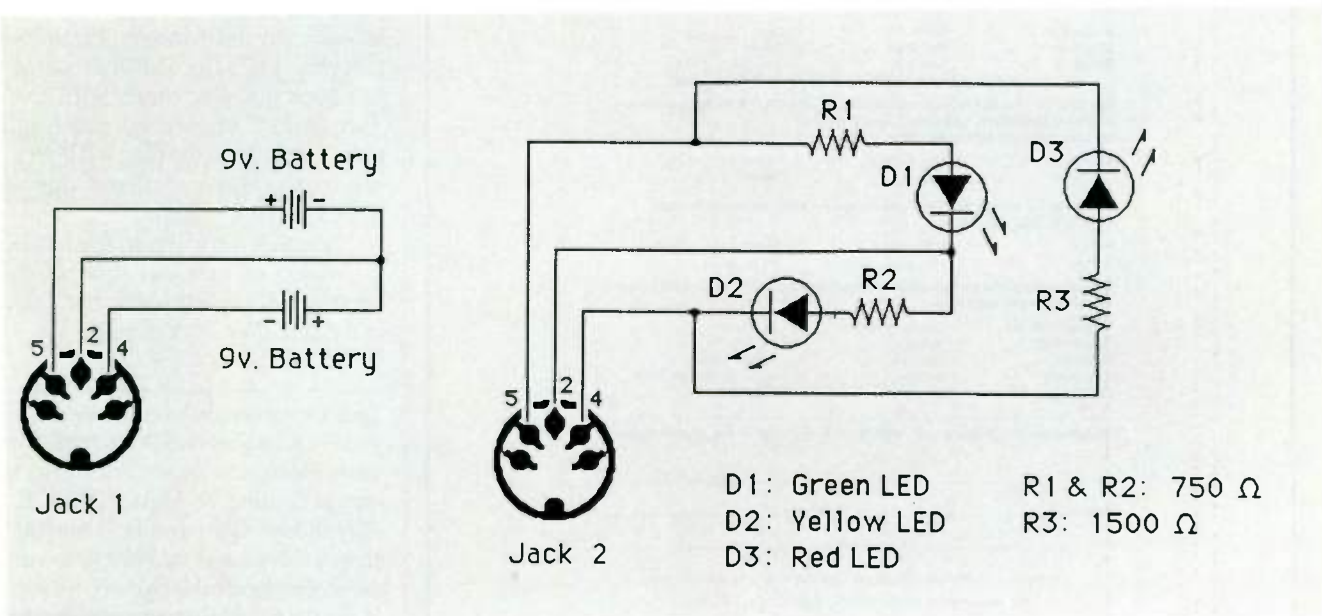


Fig. 2 *MIDI Cable Tester*

5-pin DIN jacks, most likely mounted in their own enclosure. One of the DIN jacks connects to a pair of 9 Volt batteries and the other jack connects to three LEDs as shown. Plug one end of the MIDI cable into one jack and the other end into the other jack. The MIDI cable should now transfer the voltage from the batteries to the LEDs; the green LED will be lit if there is continuity between pin 5 on both plugs, and the yellow will be lit if there is continuity between pin 4 on both plugs. If either is not lit, then the respective wire in the cable is not making connection. If the red LED is on and both of the others are off, the wires to pins 4 and 5 are cross-connected.

Don't leave a cable connected to the tester except when testing as this draws unnecessary power from the batteries. Due to the intermittent use of a device like this, the 9 Volt cells will last almost as long as their shelf life (a year or more).

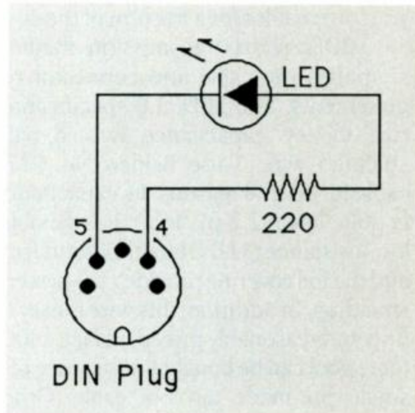


Fig. 3 *MIDI Data Tester*

MIDI DATA DETECTOR

Since the soldering iron is hot anyway, let's wire a resistor and an LED to a 5-pin DIN plug (**Fig. 3**) for a MIDI data detector. Just stick this plug into a MIDI Out (or Thru jack, if applicable) and begin to play. The LED on the plug will light up as the data is being sent, and turn off if there's no data present.

Now that you've gotten this far, there's no reason to ever run short of cables again. And you can even test them to make sure they work all right! EM

Jack Orman attended Memphis State University and has worked in electronics for ten years. He has written for Cinemagic, Modern Recording & Music, DEVICE, *and* Polyphony. *At present, he is working as a technical director in the Video Access department of his local cable channel. He is available on the Delphi electronic mail network via user name JAO.*

Looking for a lead/rhythm switch? A way to drive your choice of amp inputs or effects? How about a remote footswitch? These are just some applications for this pro-quality audio switcher.

Remote Control A-B Box

BY THOMAS FIGUEIREDO

Recently I was asked by the lead guitarist of a popular Northern California band to design a remote controlled A-B box. This player plugs his guitar into a preamp with two outputs; the "lead" channel goes into various effects, then into one input of an A-B box. The "rhythm" channel goes directly from the preamp output into the other channel of the A-B box.

When he performs live, he runs all around the stage, yet whenever he wanted to switch modes with on his old setup, he had to return to his pedalboard—which hindered his stage presence. We considered radio control, but this opens up a potential Pandora's box of problems. Finally, I came up with the idea of placing footswitches at different places onstage, and also realized that the only way to use multiple switches, and be able to switch channels without regard for the setting of the other switches in the chain, was to use momentary contact switches. By shorting a switch to a ground momentarily, we can derive a logic pulse to control our A-B box—and remember, this can be from any point onstage.

How It Works: Logic Section

The footswitches connect to the A-B box through standard ¼-inch phone jacks. You may use any two-conductor cord, shielded or not. The footswitches control the input of IC1, one-quarter of a 4093B quad Schmitt trigger. D1 and D2 protect the input from any input overvoltage conditions. R1 ties the input to ground; this prevents static electricity damage to the IC from the open circuit condition that normally exists in this circuit.

Thomas Figueiredo has been a working electronic guitarist for the last 12 years and has been studying electronics and music for the past 16 years. He does custom electronic modifications and repair work for himself and other musicians in the San Francisco Bay Area.

"I couldn't have come close to achieving these low noise levels using standard op amps and switches"

Note that all footswitches are paralleled. S1 could be part of the main unit, or eliminated if you wanted only remote switches. Also note that the remote switch connects to two jacks; this allows for "daisy-chaining" switch assemblies together if you need multiple switches.

Shorting any footswitch to ground discharges C2, thus pulling IC1's input to ground. R2 and C2 slow down the circuit's reaction time, which allows the switch to settle to a stable state (mechanical switch contacts tend to "bounce" when first closed) and prevents false triggering. When the switch opens again, C2 charges through R2, and the input returns to a high state, ready to be triggered again. The NAND gate is configured as an inverter, so whenever the input of IC1 goes low, the output goes high, which gives us a nice clean pulse when a switch is closed.

IC1's output feeds the clock input of IC2, one-half of a 4013B dual "D" flip-flop. IC2 provides a toggle function; every time the IC2's input goes high, the output changes state. This is exactly what we need to drive the rest of our circuit.

IC2's output goes to IC3, a 4049B hex inverting buffer. This chip performs two functions. IC2's output needs to be manipulated, by a series of inversions, to drive the indicator LEDs in proper sequence. The 4049B is a high current output device, which gives us enough current to drive the LEDs as well as the TTL compatible inputs of the next device.

The Heart Of The Circuit

The GAP-01, which most of us are about to meet for the first time, was developed a couple of years ago by Precision Monolithics Inc. and is billed as a "general-purpose analog signal processing subsystem" (has a nice ring to it, doesn't it?). This little gem saves us lots of time and trouble, and helps keep the parts count—and noise level—way down.

The chip consists of two differential input transconductance amplifiers, two low-glitch current mode switches, an output buffer amplifier, and a precision comparator (the latter is not used in this circuit). Both transconductance amplifier outputs are switched by current mode switches into the voltage follower output buffer, thus providing two digitally selectable signal paths through the device... perfect for an A-B box.

The transconductance amplifier inputs are high impedance, so we can plug low level sources (such as electric guitars) directly into the A-B box without the need for a preamp. IC4's inputs can withstand ±24 volts without damage, thus eliminating the need for noisy input resistors.

The output buffer can handle signals up to ±11 volts, with 0.01% distortion. We don't need any output resistor or capacitor because the buffer amp output has virtually no offset (typically 3 to 6 mV) and is short-circuit protected. Since our A-B box can go all the way down to DC, we can use it for control voltages as well as audio signals.

C1, which must be polystyrene, is the compensation cap for the buffer amp. Any value between 500 pF and 1 nF can be used, with higher values lowering the slew rate of the amp.

The Audio Section

The two channel inputs couple directly into the positive inputs of the two transconductance amplifiers. The inverting inputs connect to the buffer amp output to complete the unity gain feedback.

If you look at the GAP-01's internal diagram, you'll notice that the logic inputs are inverted with respect to each other. So, we just connect them together and apply the output of the 4049B to both inputs. One channel will always be on, and the other will always be off. An internal timing circuit guarantees that the switches turn off faster then they turn on, so they are never both on at the same time.

Construction Tips

I built my prototype on perf board using IC sockets, and TO-42 terminals to hold things together. These terminals are available through Vector, and are also sold at

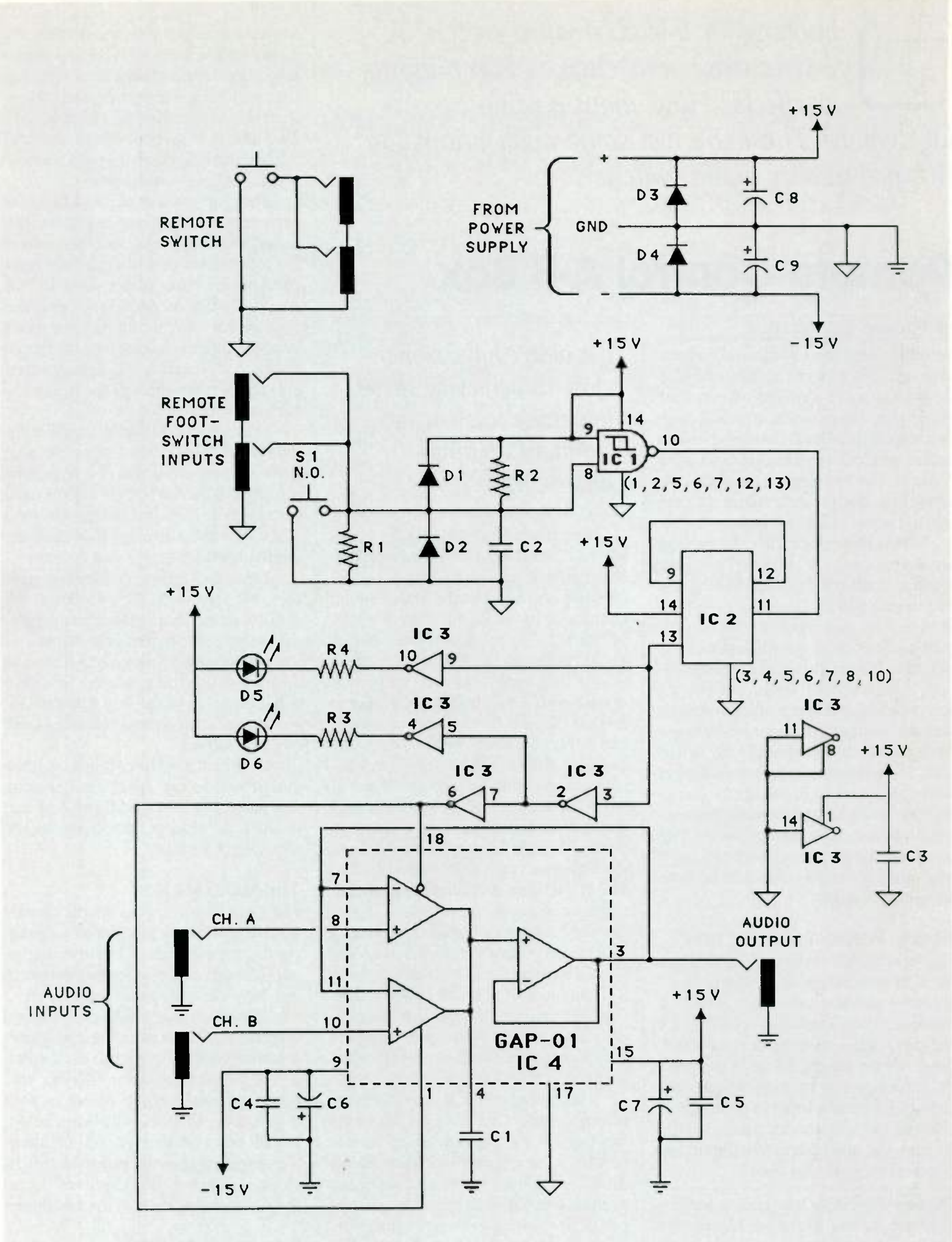


Fig. 1: *Remote Control A-B Box*

Radio Shack.

It's a good idea to keep the CMOS logic ICs away from the audio inputs, as well as IC4 pin 4 (the compensation capacitor), as these are high impedance inputs. Care must be taken to keep the logic ground points (shown by triangles) separate from the analog ground points (shown with the conventional ground symbol); connect these two ground buses only at the point where the power supply enters the board. Pin 17 of IC4 is the ground for the internal logic, and should also be connected directly to this point. (Therefore, you will have at least three connections to this ground point.) The reason for separate ground paths is to prevent logic switching noise from entering the audio circuits through the ground paths. We also want to keep the power buses separated. I ran wires from the power supply inputs on the board directly to the power supply pins of IC4, and bypassed them to ground right at the socket pins with C4, C5, C6, and C7. Then I ran a positive power bus common to the three CMOS ICs. I bypassed this bus with C3 where power entered the first of these chips. Careful layout and power supply bypassing results in a virtually noiseless signal.

All audio inputs, and the output, must be hooked up with shielded cable. If you are using a metal enclosure, ground the shields at one end only, and ground the entire enclosure with one wire to ground. The importance of single point grounding, which is what we've been discussing, cannot be overstressed.

Checkout

Before I actually try out a new project, I apply power to the unit before plugging the ICs in their sockets. Then I take my VOM and check to see that power is present at the proper pins. I check the rest of the pins to make sure that power isn't getting to where it shouldn't be. This saves much cursing, and check writing when the smoke starts pouring from a spot on the board where I made a wiring error. Naturally, this always seems to occur on the most expensive chip.

Circuit Evaluation

The A-B box is remarkably free of any kind of noise, hiss, or buzz. In fact, this is one of the quietest preamp devices I have ever tried, let alone built. P.M.I. did a fine job designing the GAP-01; I couldn't have come close to achieving these low noise levels using standard op amps and switches. This is truly a hi-fi, professional unit.

Packaging and Powering

You will need a well-regulated and filtered power supply. The unit will operate with supplies as low as ±9 volts, however ±15 volts offers more headroom.

I mounted my prototype in a floor box, but you can use a rack panel or similar enclosure. You can add as many switch jacks as you want. I put two jacks on each switch box, so you can plug several switch boxes into each other, and string them across the stage. You can even add a little switch in your guitar, and run a stereo cord to your pedalboard, to control effects from your guitar.

Finding Parts

All of the parts in this device, with the exception of the GAP-01, are available through any mail order electronics house, and most well-stocked electronics stores. Make sure that you get 4000B series CMOS ICs, as the B suffix means that the parts are guaranteed to operate up to 18 Volts.

The GAP-01 is hard to locate in small quantities and fairly expensive. I bought a stock wholesale; I'm offering the chips in single quantities for $11.95 each (please include $1.00 for postage and handling). A spec sheet is included. All orders should be sent to "Electric Ear," P.O. Box 2749, San Anselmo, CA 94960. EM

Parts List

Resistors *(¼ Watt, 5% tolerance)*	
R1	*1 Meg*
R2	*270K*
R3, R4	*470 Ohm*
Capacitors	
C1	*500 pF polystyrene*
C2	*100 nF mylar*
C3-C5	*100 nF ceramic disc*
C6-C9	*10 µF tantalum or electrolytic*
Semiconductors	
D1-D4	*1N4001*
D5	*Green LED*
D6	*Red LED*
IC1	*4093B CMOS quad Schmitt-trigger*
IC2	*4013B CMOS dual "D" flip-flop*
IC3	*4049B CMOS hex inverting buffer*
IC4	*GAP-01 Analog signal processing subsystem*

Miscellaneous

Jacks, case, switches, wire, perf board and terminals or printed circuit board, etc.

THE INTERFACE

Volume 1, Number 5 — PASSPORT DESIGNS, INC. — August, 1986

Editor's Note

This month's issue of **The Interface** is primarily dedicated to providing you with answers to your questions on Polywriter and the MIDI PRO Interface which are covered in our **In, Out & Thru** column.

In addition, we've given you yet another great chance to expand your collection of Passport MIDI software. Look for more details on this exciting opportunity inside.

Finally, we'd like to remind you that we're looking for Passport product reviews or applications-oriented articles from our users' group. That's you! We realize that it will take some work on your part. But if your article is accepted, don't be surprised when you receive your FREE GIFT to show our appreciation. So please send in your ideas or call us at any time. We are eager to hear from you.

In, Out & Thru

By Jay Lee

Polywriter

Most questions we've received lately are on Polywriter and what it takes to generate a printout of your files. As stated in your manual (you do read your manual, right?), Polywriter needs to have specific parameters met before it will work properly. These parameters are listed below.

1. First, you'll need a printer interface card that is capable of printing graphics and doing a page 2 screen dump. Page 2 is where the notation graphics reside. For parallel printers we recommend the PKASO/U card by Interactive Structures or the Grappler+ card by Orange Micro. Note that the default print commands in Polywriter are for the Grappler+. If you have a serial printer such as the Apple Imagewriter, you will need either the Serial Grappler by Orange Micro or the SeriAll card by Practical Peripherals. The Super Serial card provided with most Apples is a text only card and will not print Polywriter graphics. You can purchase the cards mentioned above from your local computer dealer.

2. Next you'll need a printer that is capable of doing graphics and that is compatible with your computer and printer interface card. We have had good experience with Epson printers, although there are many fine printers on the market.

3. Once the hardware is taken care of, all connections must be secure and the proper dip switch settings and print commands must be used. One improperly set dip switch can cause your system to stop functioning!

Below are dip switch settings and print commands. Note that when you push Control I [CTRL I], you will not see it on the screen. This is normal. Also, remember that your printer interface card goes into slot #1.

DIP SWITCH SETTINGS FOR GRAPPLER+ AND SERIAL GRAPPLER

PRINTER	SWITCH # 1	2	3	4
EPSON & C. ITOH 8510	OFF	ON	ON	ON
OKIDATA	OFF	OFF	ON	ON
APPLE DOT MATRIX	OFF	OFF	ON	OFF

(Note: the PKASO/U card does not use dip switches.)

PRINT COMMANDS:

GRAPPLER —

STANDARD DUMP	[CTRL]IG2
ENHANCE	[CTRL]IG2E
DOUBLE SIZE	[CTRL]IG2D

PKASO/U - [CTRL] II, (A), (B)H— please note that "A" and "B" are variables. The "A" represents the print height factor and the "B" represents the print width factor. Changing the values of "A" & "B" will change the appearance of your printout. Example: Try each of these commands and note that the size of the printout (2) is increased in height and width.

1. [CTRL] II,1,2H
2. [CTRL] II,2,3,H

MIDI PRO Interface

Another question raised recently regards interface errors. We've checked out some cases and have found that the problem was computer-based. Apple IIc computers manufactured prior to January 1985 have a problem with the modem port, the same one used to connect the MIDI PRO Interface. This problem could potentially affect the performance of the interface. We were told that the problem was associated with crystal frequencies and has been corrected in all IIc's with serial numbers greater than D51000. If your Apple's serial number is less than D51001, you have the modem frequency problem and your authorized Apple dealer should repair it free of charge.

Till next time . . . may all your tracks be keepers!

For more information, call or write Passport Designs, Inc., 625 Miramontes Street, Half Moon Bay, CA 94019, 415/726-0280.

In his last CZ-101 article, Alan "blew the lid off" CZ programming secrets. This time, he takes the lid off and shows how to add a CV input, mod wheel, octave transpose switch, and much more.

CZ Mods

BY ALAN GARY CAMPBELL

There is another dimension. A place where outraged techs chastise screwdriver-wielding consumers. . .where authorized service centers hand you outrageous bills. . . signpost up ahead. . .you're about to enter *The Warranty-voiding Zone!*

Seriously, though, performing the modifications described in this article **will void your CZ's warranty.** Proceed at your own risk. The mods are presented in approximate order of difficulty; while some of them are fairly simple, do *not* attempt to do the work unless you have the proper tools and skills. Work slowly and carefully, and observe all cautions as stated. If you're new to this sort of thing, I can't think of a better reference than Craig Anderton's *Electronic Projects for Musicians* (1980, Music Sales; available from Mix Bookshelf). While the CZ-101 is the focus of this article, most of the information applies to the CZ-1000 as well.

Ted Grier, National Service Manager for Casio, Inc., reports that modifications performed by Casio Authorized Service Centers will not void instrument warranties. However, Authorized Service Centers are not obligated to perform modifications at the customer's request; this is for the customer and service center to work out on an individual basis. Contact the Authorized Service Center nearest you for an estimate.

DISASSEMBLY

CAUTION: Before disassembling, store the Internal memory patches to disk or cartridge, or write them down on a patch sheet. While these mods should not affect the memory contents, make a back-up for insurance. Also, the CZ's circuit boards contain integrated circuits that can be damaged by static electricity. If you are working in a high-static environment, correct this problem before opening the CZ's case.

> *". . .most of (this) information applies to the CZ-1000 as well"*

To disassemble, turn off power and disconnect all power and audio cables. Remove the RAM cartridge, if installed. Turn the CZ-101 over on a soft, clean surface (i.e., a bath towel on a tabletop), with the front (keyboard) of the unit toward you.

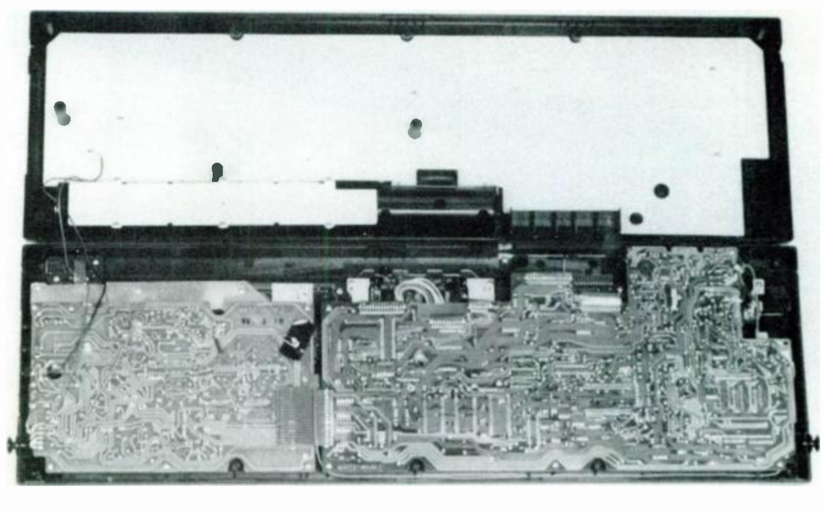
Remove the battery compartment cover. Take out the first D-cell only. Remove the two Phillips screws in the battery compartment. Reinstall the D-cell *immediately.* If the battery is removed for more than ten minutes, the Internal memory contents may be erased.

Remove the 11 remaining Phillips screws that attach the bottom cover. *Gently* lift up the bottom cover; pry up around the edges with your fingertips. Take care not to strain the connecting leads. Lay the cover down behind the now-open CZ-101 (**Fig. 1**). Collect the 13 Phillips screws so that they won't be lost. The unit is now disassembled enough to perform these mods.

(Note: Since most of the mods involve the top "half" of the instrument, you may want to disconnect the two sections to allow easier handling. Make a reference sketch of the battery connection points; then desolder the red and black wires that attach the MA2M circuit board in the top section to the battery holder contacts in the bottom section. Be careful not to overheat the pads during soldering—they could lift from the board.)

PITCH-BENDER MOD

The CZ's spring-loaded pitch-bend wheel is not very good for those "Jan Hammer" bends: the tactile feedback is all wrong since the spring works against you once you're into the bend. Also, the wheel can be used to transpose pitch—but, due to the spring-loading, it won't stay transposed. This mod removes the spring and adds a center detent (a la Moog or Prophet pitch-bend wheel), that consists of a plastic cone that fits into a countersunk spot on the side of the wheel. When pitch-bending, you can feel the detent, which makes it easy to return to the center (zero-bend) position. (Note: the components required for this mod are



ALL PHOTOS BY BILL KINNAMAN

Fig. 1 *The disassembled CZ-101*

Alan Gary Campbell is owner of Musitech™, an electronic music consulting firm. Musitech is highly involved in electronic music service and modifications, and has been cited by Moog Electronics, Inc. as one of its top ten authorized service centers worldwide.

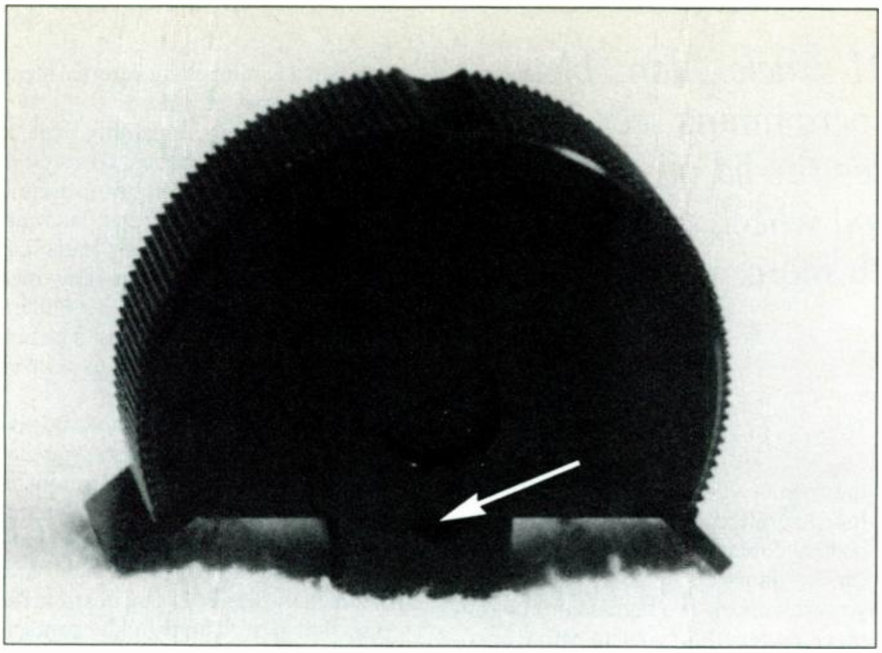


Fig. 2 *The modified pitch bend wheel; the arrow points to the detent*

available via mail order; see sidebar. You'll probably want to obtain the components before you proceed.)

To access the wheel, remove the two Phillips screws that hold the pitch-bend wheel assembly and lift it out. *Do not* strain the connecting leads. *Gently* pull the pitch-bend wheel off of the potentiometer shaft. Pry off the return spring retention clip with a flat-bladed screwdriver; work your way around the circumference of the clip, prying up a little at a time. Remove the return spring.

Lightly center-punch the closed side of the wheel, 9/32-inch (7mm) up from the bottom, along the centerline. Place the wheel in a vise. Using a 1/4-inch (6mm) standard bit chucked in a hand drill (a hand drill is easier to control than a power drill), drill a small, shallow "countersink" in the wheel (**Fig. 2**). Just barely drill into the surface; the countersunk area should be no more than 1/8—3/16-inch (3-5mm) in diameter. The "dent" should be just big enough to receive the tip of the detent cone.

The smooth surface of the wheel can be hard to grip, especially with sweaty hands. While the wheel is in the vise, you might want to cut a notch in the center (refer again to **Fig. 2**), to make the wheel easier to grab. Use a 1/4 × 6-inch (6 × 150mm) round file to remove an approximate semicircle of material (don't overdo it; compare your work to the photo). Be careful not to cut the notch at an angle, or off-center. It's helpful to first cut a shallow groove with a triangular file, along the indented centerline of the wheel, to act as a guide for the round file. Use some matte black or matte olive drab paint to cover the filed area of the notch; allow to dry overnight.

Epoxy the detent cone to a "bracket" fabricated from a piece of self-adhesive hard foam rubber (see sidebar). Replace the wheel on to the potentiometer shaft. Re-install the pitch-bend assembly. Make sure that the wheel is *centered*. Align the detent cone with the countersunk spot on the wheel, and stick the foam in place.

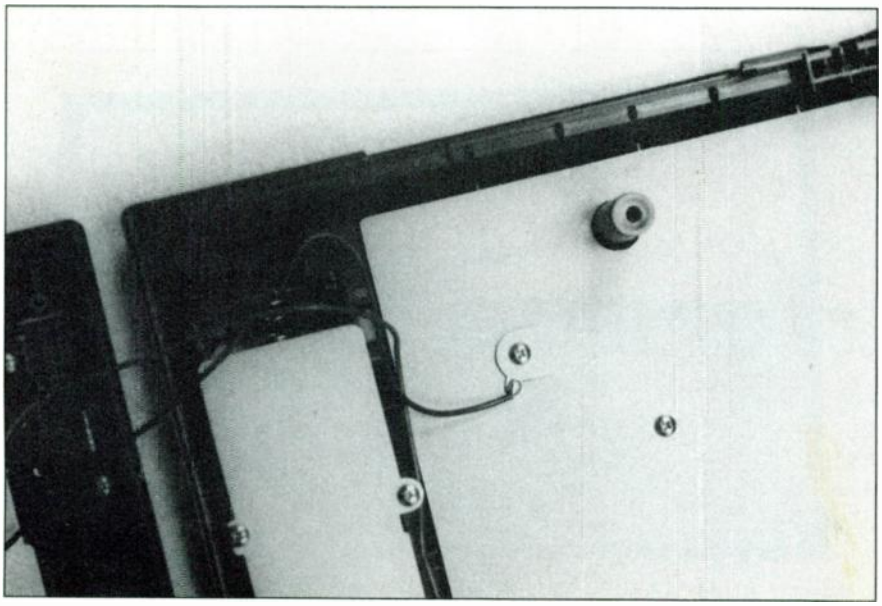


Fig. 3 *Locating the battery connector's ground lug*

9V Battery Mod

When the CZ is powered by an AC adapter, the six bulky D-cells can be replaced with a single 9V rectangular battery. If the proper adapter (Casio AD-5 or equivalent) is used, the batteries simply "back up" the Internal memory when power is off. This requires only about 0.01 mA, which the 9V battery can provide for virtually its entire shelf-life. If the 9V battery is inadvertently used to power the entire instrument, it will be depleted quickly, but the Automatic Power Off (APO) circuit will protect the 101's Internal memory contents from erasure.

Prepare an 18-inch (460mm) length of 20 or 22 gauge hookup wire; strip 3/16-inch (5mm) of insulation and tin the wire. Have a 1N4001 rectifier diode, 9V battery connector, and fresh 9V alkaline battery ready to install.

Locate the green ground wire connected to the CZ's *left* (negative) battery contact (**Fig. 3**). Unscrew the lug that attaches the ground wire to the shield inside the bottom cover. Attach and solder one end of the 18-inch wire to the lug. Re-install the lug.

Locate the *right* battery contact, identified by the red, positive lead attached. Attach and solder the *cathode* (banded end) of the 1N4001 diode to the contact (**Fig. 4**). Don't apply heat to the connection for any longer than is necessary. Do not inadvertently detach the red lead. (Note: if you do not intend to install the MIDI/power cable mod, the diode may be deleted, but if the MIDI/power cable mod is added, the diode *must* be added to prevent reverse-current flow between the batteries and power supply.)

Now read over the rest of this section to familiarize yourself with the procedure. After the batteries are removed you must complete the mod within ten minutes to avoid erasing the Internal memory data.

Tilt up the bottom cover to access the battery well. Remove the six D-cells from the battery compartment. Feed the 9V battery connector's red lead down through the slot adjacent to the right battery contact (**Fig. 5**). Lay down the cover. Attach and solder the red lead to the anode of the 1N4001 diode (refer back to **Fig. 5**).

Feed the 18-inch wire through the opening adjacent to the left battery contact, and out into the battery compartment. Temporarily place the bottom cover back on the instrument. Gently lift the wire up out of the compartment; splice and solder the wire to the *black* (negative) lead of the 9V battery connector (**Fig. 5**). Insulate the splice with electrical tape, Liquid Tape, or pre-loaded heatshrink tubing. *Immediately* plug the 9V battery into the connector.

Mount a standard 9V battery holder in the battery compartment (**Fig. 5**) using silicone sealer; don't let the holder block the two screw holes. Wipe away any excess material. Apply some silicone sealer to the 1N4001 mounting location as well. Let the silicone sealer cure overnight, then insert the battery into the holder. For a less permanent installation, mount the battery holder using self-adhesive velcro strips.

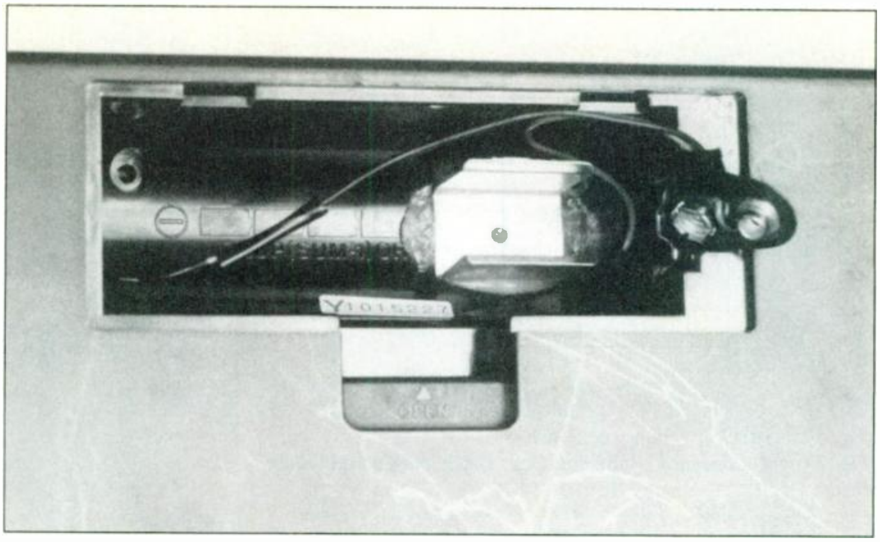


Fig. 5 *Adapting the CZ-101 for 9V transistor radio battery backup*

Octave Transpose Switch

The CZ-101's pitch-bend circuit lends itself to a simple Octave Transpose Switch mod that will shift pitch up or down an octave. With this mod, the keyboard range is extended even beyond that available under MIDI control, and can produce piercing highs and subterranean lows.

Fig. 6 shows the schematic. The CZ's µPD7811 microprocessor has an internal A/D (Analog to Digital) converter that "looks" at the pitch-bend pot. The +5V supply (Vdd) sets the A/D reference, and drives the pot, which acts as a linear voltage divider. The A/D input digitizes the wiper voltage; voltages near zero (ground) correspond to a maximum bend flat, voltages near +5V correspond to a maximum bend sharp, and voltages near +2.5V (± approximately 0.2V) produce zero pitch-bend. An SPDT center-off toggle switch can short the A/D input to ground or to +5V, thus producing the electrical equivalent of a maximum flat or sharp bend, respectively. Setting the Bend Amount at "12" transposes pitch down or up one octave. When thrown, the Octave Transpose Switch disables both internal and MIDI pitch-bend functions; but in the center "off" position, the pitch-bend operates normally.

You can install a suitable miniature toggle switch on the rear panel of the upper case, in the blank area behind the pitch-bend assembly. There is just enough room to mount the switch and a 1/8-inch (3 mm) miniature phone jack for the Control Voltage Input, described in the next section (see **Figs. 8** and **9**). Alternately, the switch could be mounted on the side of the case, in the space just forward of the PC-board-support adjacent to the pitch-bend wheel; this leaves room on the rear panel for a full-size 1/4-inch (6 mm) jack, or a separate CV Input on/off switch.

If you're going to install the Control Voltage Input and Octave Transpose Switch, it's much simpler to perform the mods concurrently. In the interest of brevity, we'll take this approach. The centerline for the mounting holes is 1/2-inch (13mm) to the left of the case alignment tab located behind the pitch-bend assembly; **Fig. 9** shows the case in a partially open state, to more clearly indicate the dimensional relationship between the mounting centerline and the tab. The jack and switch mount 7/32-inch (6mm) and 3/4-inch (19mm), respectively, from the *bottom* edge of the upper case half.

Mark the mounting hole locations. Lightly center-punch and drill the hole; deburring is probably unnecessary. Solder a 3.5 inch (90mm) length of 20 to 22 gauge hook-up wire to each of the three switch terminals. Mount the switch; don't overtighten the mounting nut—you might crack the CZ's case. Position the long axis horizontally to allow adequate space for the jack; see **Fig. 8**.

If you're going to install the Control Voltage Input, skip to the next section. If

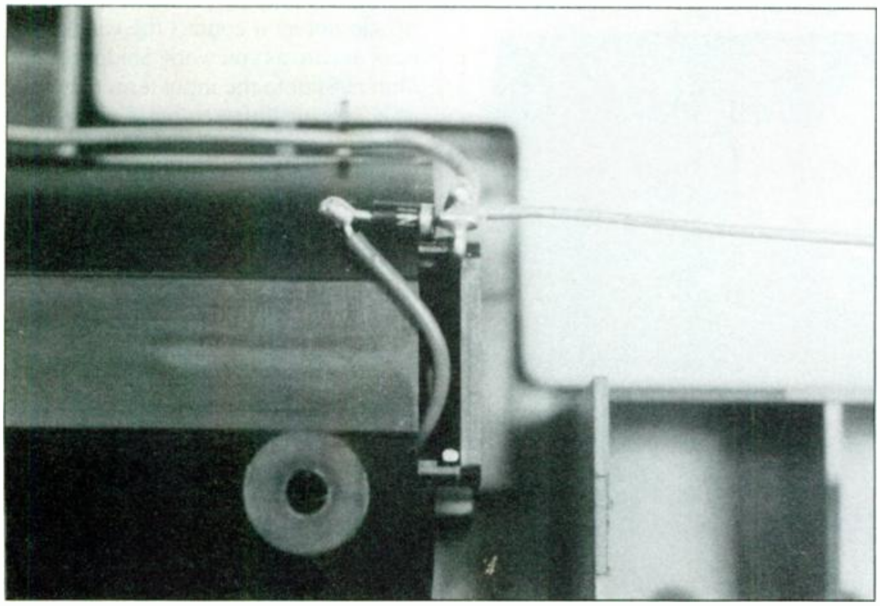
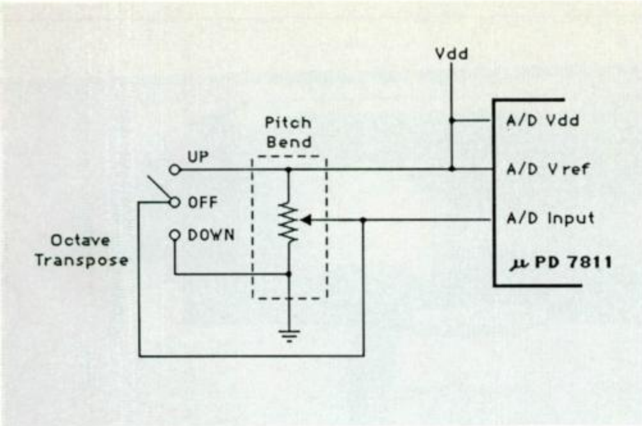


Fig. 4 *Installing the reverse-current protection diode*



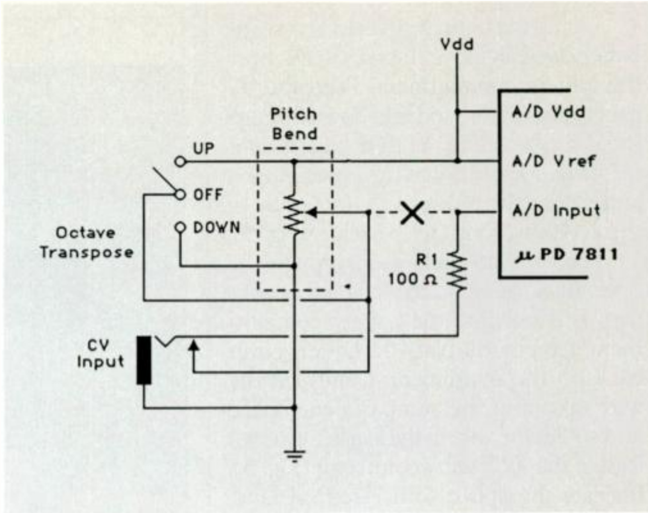


Fig. 6 *(above) Octave Transpose Switch*
Fig. 7 *(right) Combined Octave Transpose Switch and Control Voltage Input*

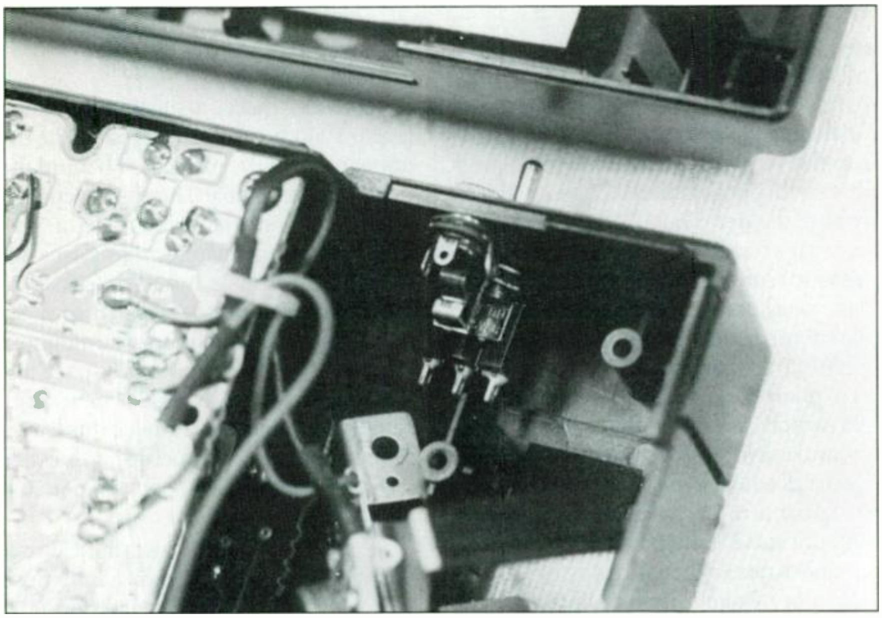


Fig. 8 *Mounting the Octave Transpose Switch and CV Input Jack*

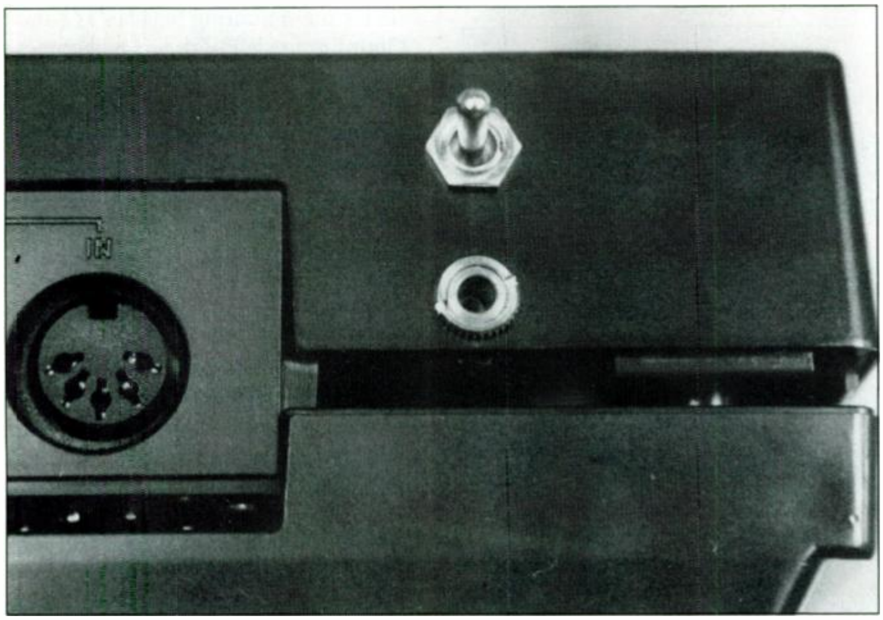


Fig. 9 *Detail of CV Input Jack and Octave Transpose Switch*

not, take out the pitch-bend assembly, and flip it over to access the pot terminals. Tack-solder the three switch leads to the appropriate pitch-bend terminals. Use a small soldering iron tip; take care that the iron doesn't contact the wiring harness or the instrument case.

Control Voltage Input

If the pitch-bend pot is disconnected from the input, the A/D converter can process an external control voltage (CV). Fig. 7 shows the CV Input circuit (and Octave Transpose Switch, for clarity); a switching jack disconnects the pot. CAUTION: *Do not apply conventional, 0 to 10V control voltages to the CV input. Limit the control voltage range to 0-5V to avoid damage to the A/D converter.* With conventional CVs, use the CV processor described below.

Install the mini-phone jack as described above. Use a small soldering iron tip; do not let it contact the wiring harness or case as you work. Solder the 100 ohm resistor to the input terminal of the jack. Take out the pitch-bend assembly to access the pot terminals. Desolder the center (white) lead from the pot wiper. Splice and solder a length of hook-up wire to the lead; insulate the splice; solder the unconnected end of the hook-up wire to the resistor. Solder a length of hook-up wire from the jack's switching terminal to the pot wiper. The "center" lead from the Octave Transpose Switch can attach to this point also, as indicated. Solder another length of hook-up wire from the jack ground to the Octave Transpose Switch ground or to the pot ground. Attach and solder the remaining two leads from the switch to the appropriate pot terminals. Double-check your wiring, then re-install the pitch-bend assembly.

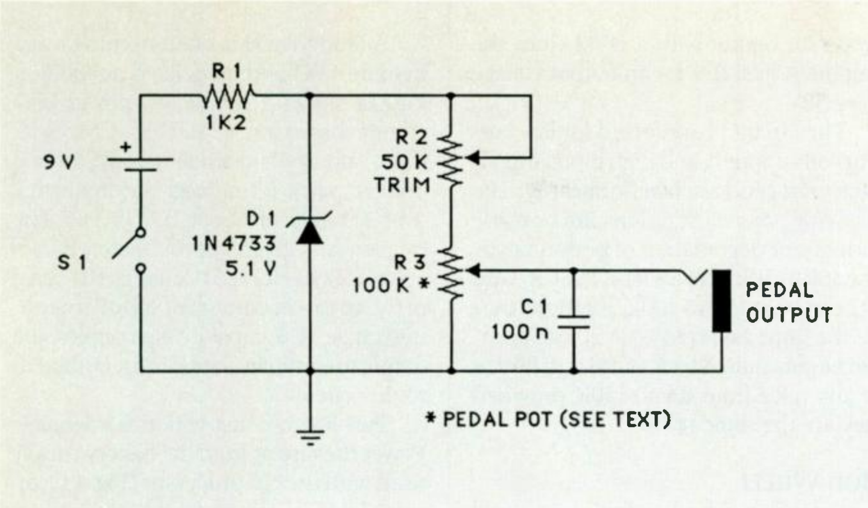


Fig. 10 *Control Voltage Pedal*

CV Pedal

Fig. 10 shows a simple CV pedal circuit that outputs the 0-5V range required by the CZ's A/D converter, and incorporates safeguards against overvoltage and reverse-polarity (the CV processor is not required with the CV pedal). When connected to the CV input, the pedal will transpose pitch down when pulled back and up when pushed forward, with continuous pitch changes in-between. Note that this pedal is suitable for use with any electronic music equipment requiring a stable 0-5V CV.

Zener D1 provides a regulated reference voltage and protects against reverse polarity during battery installation; R1 limits the zener current. The pedal pot (R3), in conjunction with range trim R2, forms a linear voltage divider. C1, a simple low-pass filter, reduces output noise caused by pot "chatter."

Pedal Pot	R2
1M	500k
500k	200k
200k	100k
100k	50k
50k	20k

Table 1

Build the circuit inside a conventional volume pedal (De Armond model 1600, or similar). All components can be mounted "point-to-point" across the pot and jack terminals (the input jack is not used functionally, but provides a convenient terminal point for wiring). R2 should equal 40 percent to 50 percent of the pedal pot's value; see Table 1. You might want to replace the output jack with a switching jack to avoid having to drill a mounting hole for S1. In this case, the battery (-) lead would go to the ring connection on the output jack instead of to the switch, which would be eliminated. Plugging a mono cord into the stereo output jack automatically turns on the battery; unplug the cord to turn off.

C1's value is not critical—use a 10n to 1μF disc, polystyrene, or tantalum cap. A battery holder can be epoxied in place at a convenient location inside the pedal.

To calibrate the pedal, monitor the output voltage with a DVM (Digital Volt Meter). Turn the pedal on and check for proper polarity. Make sure that the output goes to ground (0.0V with the pedal all the way back; adjust the pedal pot position if necessary). With the pedal pressed fully forward, adjust R2 for a pedal output of +5.20V.

CV Processor

The CZ's pitch bend circuit is referenced to a baseline of +2.5V (i.e., +2.5V equals "zero bend"); voltages from +2.5V to +5.0V bend pitch sharp, and voltages from +2.5V to 0.0V bend pitch flat. Conventional control voltages, however, are referenced to a baseline of *zero volts*, so that positive voltages bend pitch sharp, while negative voltages bend pitch flat. To use the CV input with conventional CVs, the voltages must be offset by +2.5V, *and* the output range limited to 0-5V to avoid damage to the CZ-101's A/D converter. The CV processor of Fig. 11 provides the offset and range-limiting, as well as mixing and attenuation for multiple CV sources.

Potentiometers R1-R3 set the CV input levels. The 100k pots, in conjunction with the 150k input resistors, provide a nominal impedance of 60k per input. The high impedance insures that the CV sources are only lightly loaded by the processor inputs, thus allowing a single source to drive several destinations, for synchro-sonic or similar applications. Op amp U1A forms a combination unity-gain inverting summer and precision diode (negative voltages are clipped). Zener D3 limits the output range to approximately 5.0V. Op amp U1B forms a second unity-gain inverting summer to re-invert the signal, and provide a constant impedance looking back into the circuit.

The circuit is easily constructed on perf board, and can be housed in a small project box, or in an enclosure with the Mod Wheel and/or other mods. The LM324 quad op amp was selected for its ease of use, low current consumption, and ability to operate from a single 5V supply (we'll make use of this feature later). The two spare sections can be used for the LFO and/or other purposes; tie any unused inputs to ground (refer to the pinout in DataBank).

Calibrate the circuit as follows: Disconnect any input cables. Monitor the processor output with a DVM. Turn the unit on. Adjust R11 for an output voltage of +2.50V.

The circuit is optimized for low current consumption, and high input impedance with good attenuator linearity. The following parts substitutions can be made with minor degradation of performance: Use 10k to 50k pots for R1-R3; R4-R7 and R12 can be as low as 100k, as long as they are the same value (±5%). A 20k trimpot can be substituted for R11. R8 and R9 can be any value from 10k to 150k, provided they are the same (±5%).

Mod Wheel

A modulation wheel is the one control feature that the CZ-101 sorely lacks; even under MIDI control you can only turn the LFO "on" or "off." But once the CV processor is implemented, it's much simpler to add a true Mod Wheel. The Mod Wheel LFO and the CZ's internal LFO may be used simultaneously for complex modulation effects.

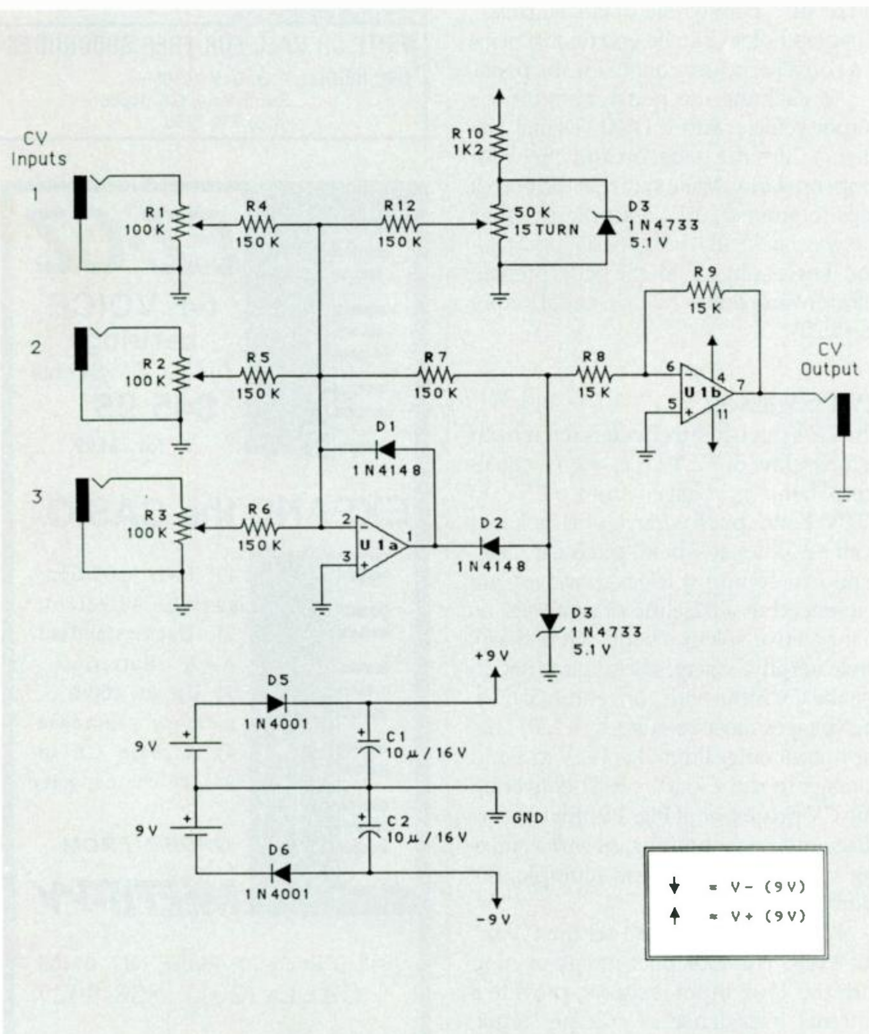


Fig. 11 *Control Voltage Processor*

A Mod Wheel is often nothing more than an LFO and associated attenuator. **Fig. 12** shows a simple low-power LFO circuit, based on a TLC555 CMOS IC timer (do not substitute a conventional bipolar part). R3 adjusts the frequency over a range of about 0.7-35 Hz. The frequency is given approximately by: f = 1.4/((C2)(R1+R2+R3)); change the value of C2 up to a maximum of 100μF to shift the range. C3 capacitively-couples the output to eliminate the DC offset inherent to this circuit.

Perf-board construction is adequate. Power the circuit from the battery supply used with the CV processor (**Fig. 11**); op amp U1C is one of the LM324's spare sections from that circuit. C1 can be a 100n to 10μF ceramic disc or tantalum cap. Use low-leakage caps for C2 and C3. The output attenuator can be a modified volume pedal, panel control, or a Mod Wheel attached to a pot. The Mod Wheel/pot combination could be mounted close to the instrument for easier access; in this case the attenuator would connect to the LFO circuit via a three-conductor cable and stereo jack, and the LFO output would be hard-wired to one of the CV processor inputs.

Advanced experimenters might want to install a modulation circuit *inside* the CZ-101, powered from the CZ's supply. *Note: prerequisites for this version of the mod are a service manual and considerable circuit construction experience.* **Fig. 13** shows the internal LFO/CV Processor circuit. The CZ's pitch-bend pot is disconnected from the internal circuitry to create a "floating" attenuator for use with the LFO (prior installation of the pitch-bender mod is assumed). S1 selects either pitch-bend or modulation control; S2 is the Octave Transpose Switch (turn S2 "off" when the wheel is used for modulation). The op amp forms a simplified CV Processor and replaces the circuit of **Fig. 11**. The indicated LFO is the circuit of **Fig. 12**.

The circuit can be constructed on a small piece of perf board and mounted in the blank area along the rear panel, behind the wheel—assuming that the Octave Transpose Switch is mounted on the side of the case. To conserve panel space, the LFO rate control can be a PC-mount trimpot. Use two sections of an LM324 for the op amps in **Figs. 12** and **13**—do *not* substitute other op amps for the LM324. Note that both the TLC555 and the LM324 use the Casio's +5V VDD supply; pin 4 (V+) of the LM324 connects to +Vdd, and pin 11 (Gnd) connects to "G AN" (analog GND; same as pitch-bender

GND). Tap into the +Vdd supply by tack-soldering to a convenient location on the MA1M board.

Double-check the wiring before power-up. Calibrate the circuit as follows: Set S2 to "off"; set S1 to "modulation"; turn the unit on. Monitor the op amp output with a DVM. Adjust R4 for an output of +2.50V.

MIDI/Power Cable Mod

The AC adapter is almost impossible to use when the CZ is played as a strap-on keyboard since the adapter cable is short and fragile. Fortunately, a simple modification allows the MIDI cable to carry not only MIDI data, but DC power too.

Fig. 14 shows the schematic for an interface that transfers the adapter power to the MIDI cable; the indicated MIDI Switch is optional. Construction of the interface is very similar to the "Build a MIDI Switch" project described in the June, 1985 **EM**. Readers desiring comprehensive, step-by-step instructions should refer to that article. If you're going to construct the Regulated Power Supply and/or other mods, you'll probably want to integrate these circuits with the interface, in a single enclosure (see "Circuit Consolidation," below).

"Fortunately, a simple modification allows the MIDI cable to carry not only MIDI data, but DC power too"

The CZ's MIDI Out jack must be adapted to transfer the DC power. Locate the MIDI jacks along the right-rear panel of the CZ. Note the pads where the jacks attach to the PC board. The mod connects to the *leftmost* (MIDI Out) jack. The two uncommitted outermost pads, corresponding to DIN connector pins 1 and 3, are used to transfer the DC power. Prepare 24.5-inch (620 mm) and 9-inch (230 mm) lengths of 20 or 22 gauge hookup wire. Strip and tin the ends. Tack-solder the 24.5-inch wire to the *rightmost* pad (pin 3); tack-solder the 9-inch lead to the *leftmost* pad (pin 1); see **Fig. 15**. Avoid overheating the connections—the pads could lift from the PC board. Make sure that there are no solder bridges between pins.

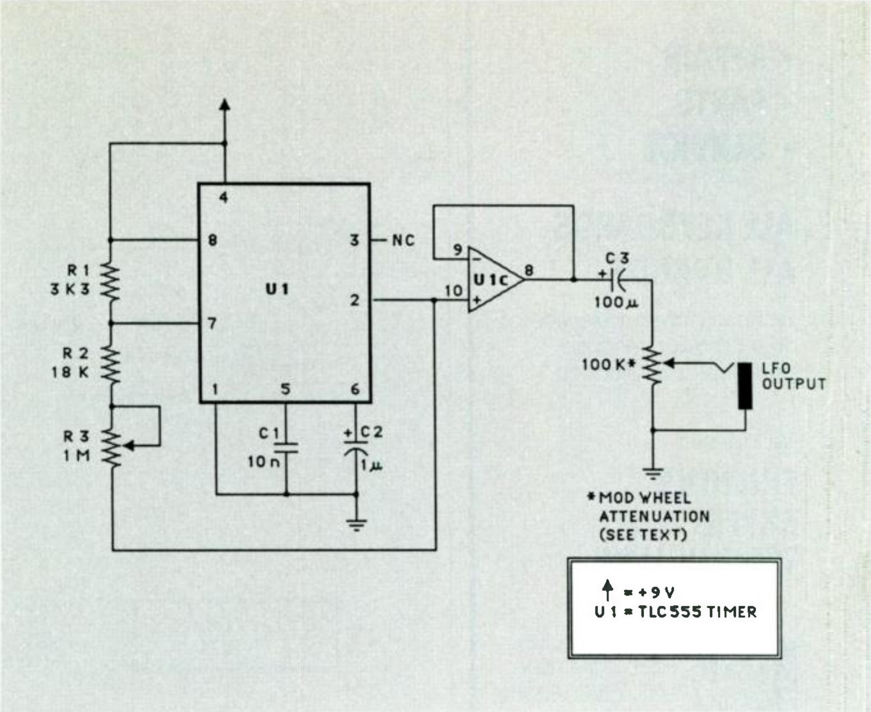


Fig. 12 *LFO/Mod Wheel*

Attach and solder the 24.5-inch wire to the ground lug previously identified in **Fig. 3**. Attach and solder the 9-inch wire to the *right* (positive) battery contact identified in **Fig. 4**. Refer to the "9V Battery

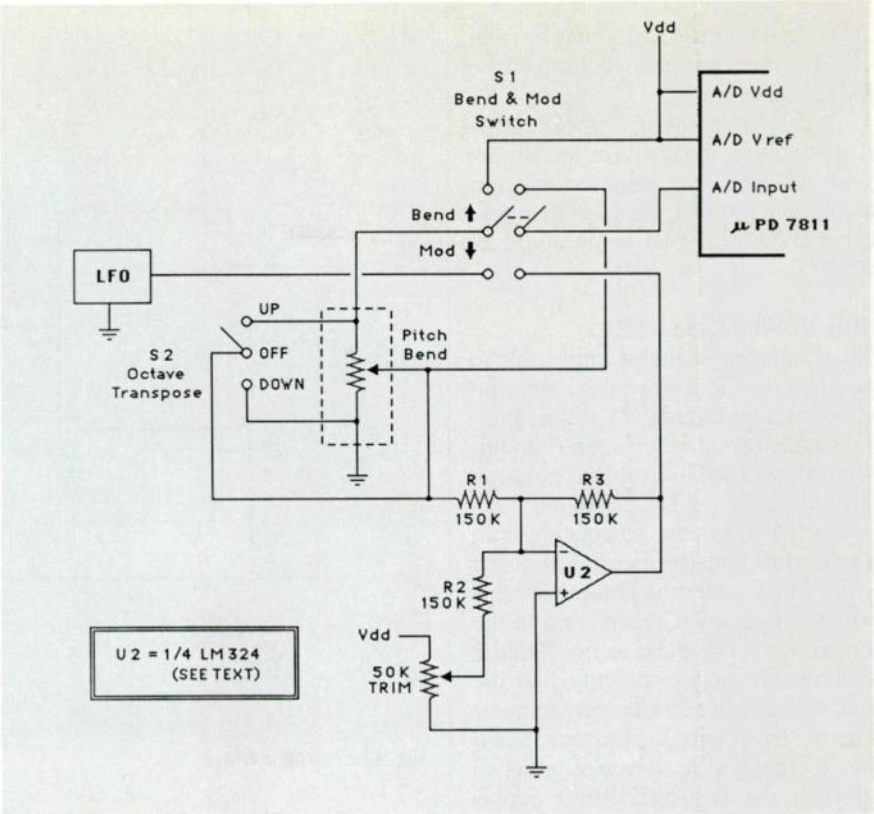


Fig. 13 *LFO Circuit for Internal Installation*

Mod" section for more information.

Note: this modification requires *five-conductor* MIDI cables, so it does not strictly follow the MIDI 1.0 specification. However, the MIDI current loop is reasonably immune to EMI (Electro-Magnetic Interference), so there is no reason not to use the "extra" conductors to transfer DC. If you can't locate suitable five-conductor cables, you can make your own. You'll need a sufficient length of four-conductor, shielded cable, and two 5-pin 180 degree DIN plugs; check with electronics supply houses in your area. Recommended maximum length for MIDI cables is 50 feet (15m).

Regulated Power Supply

The Casio AC adapter is a well-made unit, but, like most, it's susceptible to RFI and "glitches" caused by line transients. For about the same cost you can build a bipolar IC-regulated power supply with onboard surge suppression that will power the CZ-101 *and* mods.

CAUTION: This circuit contains hazardous AC voltages, and requires a good bit of fabrication skill—it is *not* a project

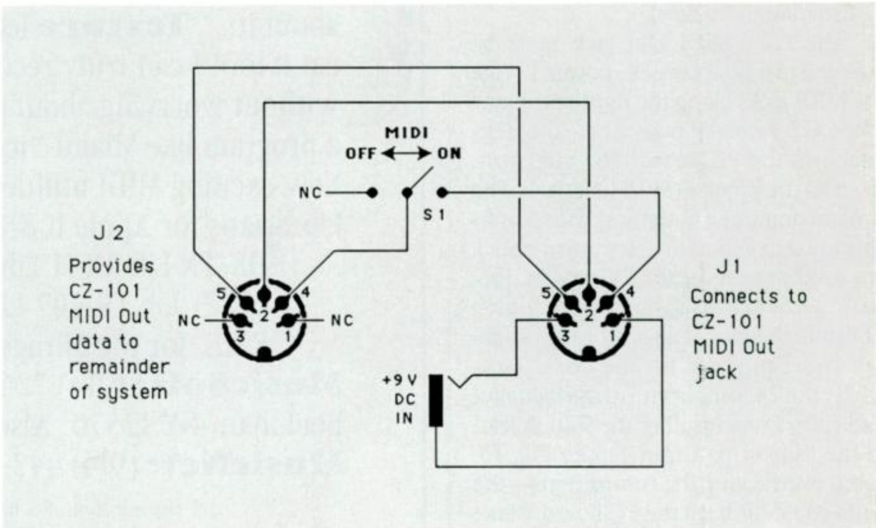


Fig. 14 *MIDI/Power Cable Interface*

for beginners. Even the experienced do-it-yourselfer should treat line-powered projects with the utmost respect: Insulate any exposed AC connections within the project enclosure with electrical tape or silicone sealer. Always double-, triple-, even quadruple-check your wiring before you plug in and power up such projects. Never work on line-powered gear when you are tired or upset. Practice safety at all times.

Fig. 16 shows the supply schematic. The +9V output delivers 1.0 Amp, more than enough current to power the CZ-101 or most any portable synth requiring 9V DC. The -9V output delivers about 200 mA.

Circuit construction is straightforward; use perf board, or even terminal strips and point-to-point wiring. If your application does not require the -9V output, eliminate D2, C2, U2, C4 and C6, and use a 12.6VAC RMS 1.2A transformer (*not* center-tapped) to drive D1. The indicated EMI filter is optional, but improves the supply noise rejection considerably. Use only a three-wire grounded power cord for this project. Operating the power supply without a proper AC ground is unsafe (especially if the supply enclosure is metal), and will render the common-mode varistors and filter elements *useless*. The AC ground should attach to the case, and all system grounds should return to this point also. Mount the caps as close to the voltage regulator ICs as possible. Do not substitute electrolytic or polystyrene caps for the tantalum and ceramic types specified.

The AN7809 and AN7909 voltage regulators are Panasonic ICs, available from Digi-Key (see DataBank). The AN7809 must be adequately heat-sinked. If the regulator gets too hot, it will turn itself off temporarily. This won't hurt anything, but it could be a nuisance. Use a finned heat sink, rated at 10 Watts or more, mounted in a vented enclosure. Apply a liberal amount of heat sink compound between the regulator, mica insulator, and heat sink surface. Before power-up, visually inspect the mica insulator for proper alignment; use an Ohmmeter to test for shorts from IC pins 1 and 3 to ground. No heat sink is required for the AN7909, U2, but a simple clip-on type will increase the available output current. Note that the AN7909 pinout is *different* from the AN7809's.

The power supply's ground return may create a ground loop in conjunction with the ground return of an amplifier connected to the CZ-101. If this occurs, use a three-conductor to two-conductor

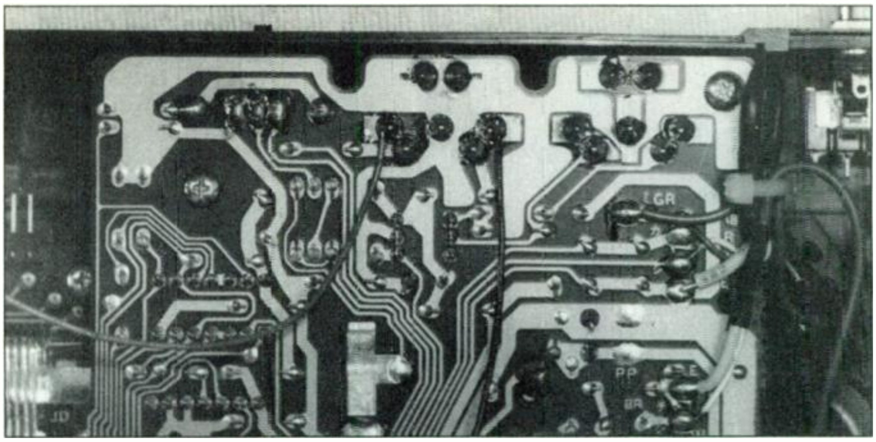


Fig. 15 *Note two wires in center of photo; these connect to the MIDI Out jack*

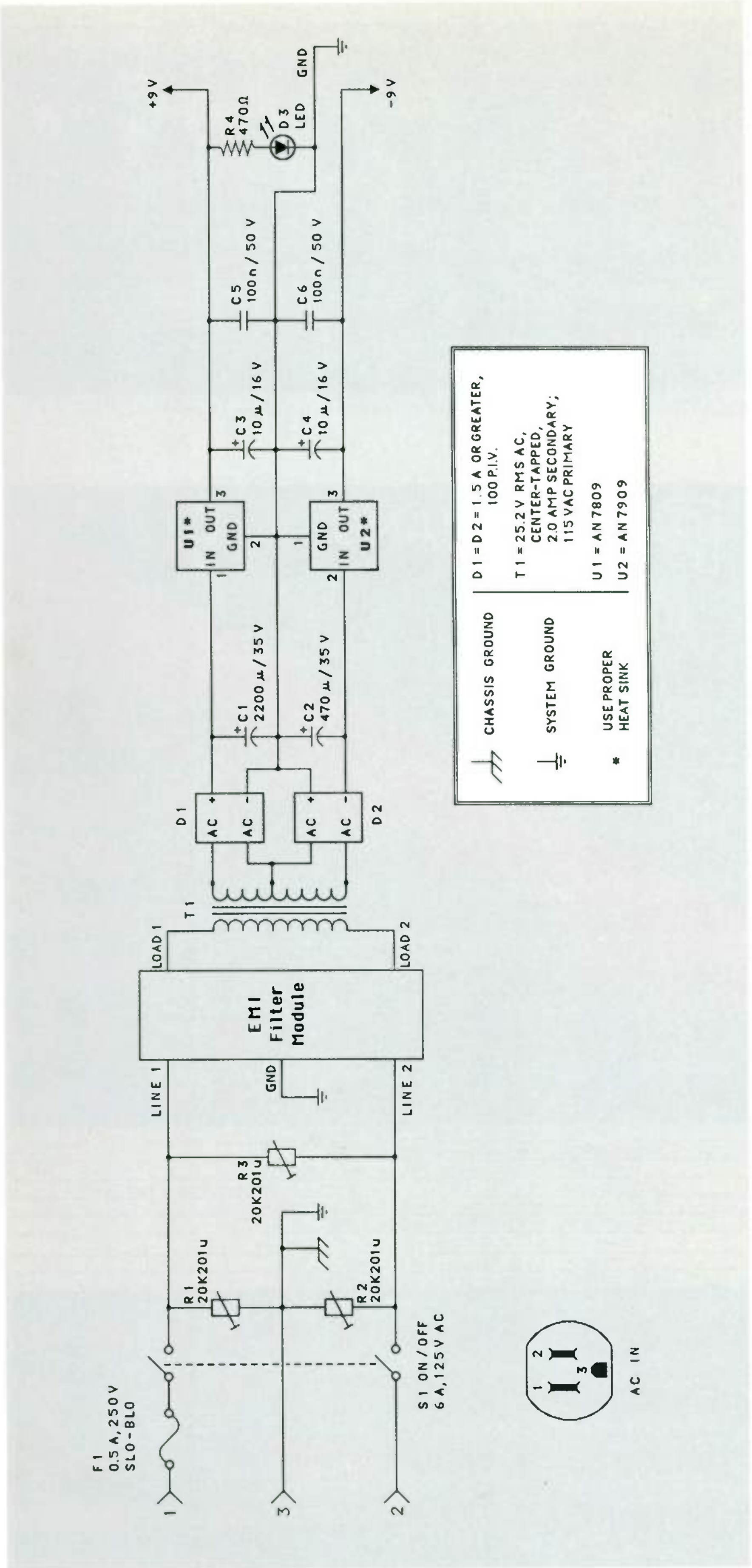


Fig. 16 *Bipolar Power Supply*

AC plug adapter ("three-prong" adapter) to "lift" the amplifier ground. Make sure that the interconnecting audio cable has a functional, low-impedance ground return. Remove the three-prong adapter when the amplifier is not used in conjunction with the regulated power supply.

Circuit Consolidation

The Regulated Power Supply, CV Processor, Mod Wheel circuit, and MIDI/Power Cable Interface can be integrated into a single enclosure; the Power Supply has sufficient reserve capacity to drive additional bipolar circuitry, such as a mixer, delay line, or pedalboard.

The consolidated circuitry can be somewhat simplified by the following changes to the *CV Processor* (**Fig. 11**): omit D4, D5, D6, C1, and C2; change R10 to 47k if R11 is 50k, or to 33k if R10 is 20k.

Reassembly

When you've completed the desired mods, and you're satisfied that everything is working properly, carefully close up the CZ-101. Check that the MIDI/Power Cable Mod wires do not interfere with any case tabs or reinforcements, or the cartridge connector; when the case is closed, the 24.5-inch wire routes over the main circuit boards to the ground lug; the 9-inch wire routes to the right of the battery contact, then back at a 90 degree angle toward the MIDI Out jack. Finally, replace the 13 screws that secure the bottom cover.

More Mods

As much as these modifications do add to the instrument, there are still more possibilities. For instance, the PRESET sounds and part of the operating system are stored in ROM. The ROM contents could be disassembled, and new sounds and instructions coded in a replacement PROM—like an E! for the CZ-101 instead of the DX7. Or, the memory could be expanded by paralleling the data and address buses of the existing RAM ICs with those of the memory expansion RAM, and bank-selecting the desired RAM via the chip enable inputs. A cartridge memory-expansion mod by Scott Morgan was detailed in the July issue of **EM**. If there is sufficient reader interest, we'll cover more CZ memory expansion mods in a future article.

Service Documents

A CZ-101 service manual (with schematics, PC board layouts, alignment/test procedures, and annotated parts lists) is

ALPHA JUNO 1 & 2

FLASH BOARDS

Flashy sounds, flashy features and a flash to operate, the new Alpha JUNO 1 and 2 are two hot keyboards that are sure to move any musician's set-up into the fast lane of performance. Both Alpha JUNOs offer powerful sound synthesis (14 waveform DCO, 8 parameter envelopes) but at the same time offer incredible advances in user-friendly programming and editing features. The key is the JUNO's Alpha Dial. Just whirl it to edit any of the 64 preset patches or to create 64 new program patches—sound programming has never been so easy. Add an M-64 C Memory Cartridge and you have 192 patches at just a touch. The Alpha JUNO's innovative Tone Modify Function lets you quickly modify the many related parameters of Brilliance, Envelope Time, Modulation Rate and Depth—all at the same time with just one touch of the Alpha Dial—a system called Biometrics. Or you can add the optional PG-300 Programmer to make programming even easier. You'll also find the Alpha JUNO's MIDI implementation to be the most sophisticated on the market. Most amazing is that all these features don't carry a flashy price. Which means they're going to move out fast, so you'd better do that too—only to your Roland dealer.

Roland

RolandCorp US, 7200 Dominion Circle, Los Angeles, CA 90040 (213) 685 5141 available from Casio for $10. postpaid. (Service manuals are available for all CZ instruments.) Also, for $50. annually, individuals and service centers may subscribe to Casio's Technical Information Service, which includes monthly Service Bulletins, a selection of service manuals, any new manuals published during the subscription period, and an open account for ordering parts. Contact: Service Department, Casio, Inc., 15 Gardner Road, P. O. Box 1386, Fairfield, NJ 07007.

Acknowledgement: I'd like to thank Mr. Jerry Kovarsky, Product Manager E.M.I., and Mr. Ted Grier, National Service Manager, both of Casio, Inc., for their kind assistance in the preparation of this article. EM

Parts Lists and Parts Availability

Equivalent Radio Shack part numbers are given in parenthesis. Other components are available from Digi-Key; see DataBank.

Pitch-Bender Mod

A detent cone, pre-mounted to a self-adhesive hard foam rubber strip, is available for $3.75 postpaid (Cashier's check or Money Order, U. S. funds only) from: Greene Parts, Dept. EM, 320 Crest Terrace Drive, Chattanooga TN 37404. The detent cone is a Moog Electronics original equipment part; the foam rubber strip is a first-quality industrial grade compound. These components *only* are offered as a service to EM readers; please do *not* order any other mod components from Greene Parts Co.—they are not available.

9V Battery MOD (Fig. 5)

1	1N4001 1.0A 50 PIV rectifier diode (276-1101)
1	9V battery holder (270-326)
1	9V battery connector (270-325)
1	Alkaline 9V rectangular battery, Duracell MN1604 or equivalent (23-553)
Misc.	Hook-up wire; silicone sealer or self-adhesive velcro strips (64-2345)

Octave Transpose Switch (Fig. 6)

S1	SPDT, center "off," mini toggle switch (275-325)
Misc.	Hook-up wire

Control Voltage Input (Fig. 7)

J1	2-conductor closed-circuit mini phone jack (274-253)
R1	100 Ohm ¼W 5% carbon film resistor (271-1311 or 271-012)
Misc.	Hook-up wire

Control Voltage Pedal (Fig. 10)

R1	1k2 ¼W 5% carbon film resistor (271-024)
R2	50k trimpot (271-219), see text
D1	1N4733 5.1V 1W zener diode (276-565)
C1	100n, 16V or greater ceramic disc capacitor (272-135)
Misc.	Potentiometer-type volume pedal (see text); 9V battery holder (270-326); 9V battery connector (270-325); 9V alkaline battery, Duracell MN1604 or equiv. (23-553); On/Off switch or switching jack (see text); hook-up wire

Control Voltage Processor (Fig. 11)

R1-R3	100k linear-taper potentiometer (271-092)
R4-R7, R12	150k ¼W 5% carbon film resistor (271-047)
R8-R9	15k ¼W 5% carbon film resistor (271-1337 or 271-036)
R10	1k2 ¼W 5% carbon film resistor (271-024)
R11	50k multi-turn miniature trimpot (Digi-Key part no. CFG54 or CEG564); may substitute 20k value (271-340), see text
C1-C2	10μ 16V tantalum capacitor (272-1436)
D1-D2	1N4148 silicon switching diode (276-1122)
D3-D4	1N4733 5.1V 1W Zener diode (276-565)
D5-D6	1N4001 1.0A 50 PIV rectifier diodes (276-1101)
U1	LM324 qud low-power op amp (Digi-Key part no. LM324N)
Misc.	Input/output jacks; perf board; hook-up wire; two 9V battery holders (270-326); two 9V battery connectors (270-325); two alkaline 9V batteries, Duracell MN1604 or equiv. (23-553)

LFO/MOD Wheel (Fig. 12)

R1	3k3 ¼W 5% carbon film resistor (271-1328 or 271-028)
R2	18k ¼W 5% carbon film resistor (Digi-Key part no. 18KQ); may substitute 15k and 3k3 in series
R3	1M linear-taper potentiometer (271-211)
C1	10n, 16V or greater ceramic disc capacitor (272-131)
C2	1μ, 16V or greater tantalum capacitor (272-1434)
C3	100μ, 16V low-leakage electrolytic capacitor (Digi-Key part no. P6620)
U1	TLC555 CMOS IC timer (276-1718)
Op amp	¼ LM324 quad low-power op amp (see text)
Misc.	Mod Wheel/attenuator (see text); perf board; hook-up wire; output jack (see text)

LFO Circuit for Internal Mounting (Fig. 13)

S1	DPDT mini toggle switch (275-1546)
S2	SPDT, center "off," mini toggle switch (275-325)
R1-R3	150k ¼W 5% carbon film resistor (271-047)
R4	50k multi-turn miniature trimput (digi-Key part no. CFG54 or CEG54); may substitute 20k value (271-340)
Op amp	¼ LM324 quad low-power op amp (see text)
Misc.	Perf board, hook-up wire

MIDI/Power Cable MOD (Fig. 14)

J1-J2	5-pin 180 degree DIN jack (274-005)
Misc.	DC power jack; enclosure; optional switch S1 (use low-capacitance insulated-bat type)

Regulated Power Supply (Fig. 16)

F1	0.5A 250V SLO BLO fuse (270-1282)
S1	125V, 3A or greater DPDT mini toggle switch (275-1546)
R1-R3	MOV, GE V130LA20A / Panasonic 20K201U or equiv. (276-568)
T1	120V AC rms line transformer, 25.2V 2.0A center-tapped secondary (273-1512)
D1-D2	1.5A 100PIV full-wave bridge rectifier (276-1152)
C1	2200μ, 35V aluminum electrolytic capacitor (272-1020)
C2	470μ, 35V aluminum electrolytic capacitor (272-1018)
U1	AN7809 9V 3-terminal positive voltage regulator, T0-220 case (Digi-Key part no. AN7809)
U2	AN7909 9V 3-terminal negative voltage regulator, T0-220 case (Digi-Key part no. AN7909)
C3-C4	10μ 16V tantalum capacitor (272-1436)
C5-C6	100n 50V ceramic disc capacitor (272-135)
R4	470 ohm ¼W 5% carbon film resistor (271-1317 or 271-019)
D3	Red LED (276-041 or similar)
Misc.	EMI Filter Module (273-103); enclosure; line cord/strain relief; fuseholder; perf board; T0-220 heat sink, 10W or greater (Digi-Key part no. HS114); T0-220 mounting hardware; heat sink compound; output jack

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—from page 35, MIDI MUTING & MIXING

ting changes into common MIDI commands (note on-off, for example) and the other translates these MIDI commands back into the proper form to control the outputs. This approach assumes that an attached sequencer will store any changes as they occur, then play them back later in real time to re-create the mix changes.

In a studio situation, particularly during mixdown, we would have a sync track (preferably SMPTE) on tape that would drive the sequencer's sync input. The basic operation would be as follows:

1. Play back the tape, and set the sequencer so that it is recording on an unused track.
2. Make muting or level changes ("moves") while listening to the mixdown. Data representing these changes get recorded into the sequencer, and since the sequencer is synched to the tape, the changes are synched to what's happening on tape.
3. At the end of the piece, rewind the tape, then restart it and the sequencer.
4. The previous "moves" that were recorded into the sequencer now take place automatically, thus leaving you free to either overdub new moves, or devote your attention to more musical details. The synchronization will be perfect every time, and we have simulated many of the automation features of considerably more expensive automation systems by using MIDI as our storage medium.

Real-World Design Challenges

The first available MIDI automation unit, the J.L. Cooper Electronics MIDI Mute (**Fig. 4**), is a muting accessory that patches into any mixer with insert points. All the muting moves are done on a small remote control unit. As muting changes are made, MIDI Note-On commands are generated on a MIDI channel. As far as the attached sequencer is concerned, it is just getting the type of commands it would expect from any synthesizer. Each of the audio muting tracks has its own MIDI note number; different velocity values indicate whether the mute is on or off. Upon playback, the Note On commands are interpreted by the MIDI Mute and converted into signals that control the internal reed relays, which in turn switch the audio on or off.

Since reed relays can often be used in place of standard switches, they can also initiate many footswitchable functions (such as the loop function of a DDL or effects box in/out). Automated muting even allows for interesting types of gating effects that would be impossible even

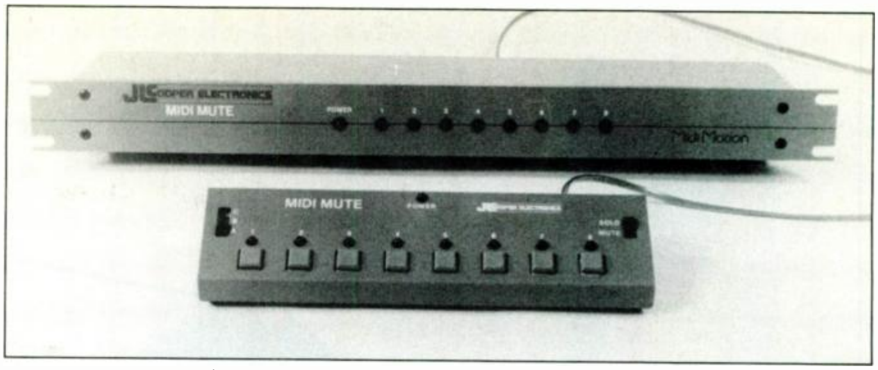


Fig. 4 *J.L. Cooper Electronics MIDI-controlled audio muting system*

with higher-priced automation systems.

Of course, there is often a large gap between theory and practice, and while designing the MIDI Mute several non-obvious aspects of MIDI controlled automation came to light. All had solutions, but required additional features not originally anticipated.

✓ One of the design goals was to allow the engineer to wind back to *any* point in the song to change the muting. This requires a sequencer that can sync directly to SMPTE, or a MIDI sequencer that can sync to SMPTE via a SMPTE-to-MIDI converter (Roland SBX-80, SMPL System, etc.). A MIDI sequencer must also be able to recognize a special MIDI command called Song Position Pointer. This command keeps track of how many 16th notes have elapsed since the beginning of a piece; when the master device issues a Song Position Pointer command, the sequencer knows exactly how many 16th notes it is into the song. Although early sequencers did not usually recognize SMPTE and Song Position Pointer, many newer ones do.

But one of the aspects of MIDI is that there is no real "history" to the MIDI data. That is, information is only sent telling of *changes* of status, such as when a new Note On occurs. That one command tells nothing of what other notes may be currently playing. Now suppose a sequencer jumps to a point in the middle of a song, as specified by the Song Position Pointer command. If notes were turned on before that point and remain held after that point, the sequencer might not recognize that notes were being held and therefore not know which tracks are on or off. This would not be a problem if the sequencer went through its memory from the very start of the piece to check the status of all notes, then sent out those commands as a burst of data as soon as it autolocated to the Song Position Pointer position. Unfortunately, this behavior could not be guaranteed.

To circumvent this situation, we decided on a concept called Auto Mark. With Auto Mark, the MIDI Mute sends out a burst of MIDI note commands every few seconds that reflect the current status of all of the muting tracks. Therefore, the sequencer is never more than a few seconds from a point in the song where the status of all the mutes will be updated.

✓ Even if the sequencer is able to update the track status, an additional problem is how to know the status at the beginning of the song. MIDI Mute has no way of knowing that the song has started, so it can't notify the sequencer of the starting status of the mutes. This is no problem on the first pass, since the engineer will just start with the desired tracks muted. But on playback, no transfer of information from the sequencer will take place until the first change of status. The brute-force solution is to manually reset the muting status just before starting playback.

A slightly more elegant solution is to do a Manual Mark at the very beginning of the song (i.e *just* before the song starts). This sends a burst of note commands to the sequencer (exactly as with Auto Mark) at the beginning of subsequent playbacks.

✓ Most of the time, muting moves can be laid down in one or two passes with no trouble; it is much easier than laying down a synth part and usually will not require editing. But sometimes musical decisions dictate changing the muting moves. For instance, if you decide to switch over from one take to part of a different take, you would need to mute the old part at the switch point and unmute the new one.

Some sequencers let you edit out individual notes and/or change their velocities, but most do not. The challenge, then, is to figure out *how to change the velocities of specific notes* (remember that each audio track has its own note number controlling it, and that different values of velocity specify if the mute is on or off).

Just overdubbing with the new moves won't work, since the sequencer would

leave the old note commands in place, and you would end up with a jumble of mute on and off commands. As an analogy, consider a traditional sequencing situation where you hit a C# and then decide you really wanted it one 16th note earlier. If you just overdub the new strike of the note, the old one would still exist in memory, and would sound one beat later. The solution to our problem was Conditional Echo.

Conditional Echo requires a sequencer that can record on one track while playing back another. The editing procedure would be as follows:

1. Lay down the first pass of muting moves on sequencer track 1.
2. Rewind the tape. Have the sequencer play back track 1 while recording on sequencer track 2.
3. MIDI Mute automatically passes the MIDI data coming in (from the playback of track 1) through to its MIDI output (to be recorded onto track 2). This is called "echoing" in MIDI parlance. So, if you make no changes, track 2's data will be the same as what was on track 1. If you *do* touch one of the Mute track switches, MIDI Mute will stop responding to the MIDI note number associated with that mute track. Therefore, the echoing process will cease, and only moves on that mute switch will go through to the output. Of course, all other MIDI note numbers will echo and respond normally. In this way, we can "tell" the sequencer to essentially erase the undesired track data, and replace it with the new moves. Track 1 would still hold the old version of the moves if you wanted to do a comparison. With a multi-track sequencer, you could retain several different versions of the mute moves, starting with a basic one and making edited versions. Or, have everyone in the band take a crack at their own version of what makes the perfect mix . . .

". . . normal musical sequencers were not designed with studio automation in mind"

Okay, So What's Next?

All of the "tricks" mentioned above, were developed to get around the fact that normal musical sequencers were not designed with studio automation in mind. Although these tricks allow existing sequencers to operate the automation in a very useable fashion, our next project is the design of an easy-to-use sequencer designed specifically for automation. Not only would this free up the musical sequencer, but would allow post-production houses and other non-MIDI-oriented facilities to have an easy-to-use, low cost automation facility. Our goals include SMPTE compatibility (both read and write at either 30 or 25 frames per second), transparency to the user (the unit would "know" when tape was rolling and advise the user of its operating mode), memory saves via either a high-speed tape or via MIDI System Exclusive dump to external disk, and the ability to communicate to a computer via MIDI for expansion to more esoteric options such as splicing, etc. This communication would include the newly proposed MIDI SMPTE Systems Exclusive protocol so that any attached computer (with the proper software) would know what SMPTE frame was being read.

Conclusion

MIDI has found its way into applications far beyond the original synth/sequencer/drum machine usage. Using MIDI for the control of mixers on stage and for automation in the studio is one of those applications that we will probably take for granted in just a couple of years. New systems will probably incorporate MIDI as a matter of course, and retrofit-oriented products (like MIDI Mute) will allow the huge number of non-automated boards to find new life. EM

What a difference a resistor makes . . . specifically, you can get your Roland SBX-80 and Oberheim gear talking to each other.

The SBX-80/Oberheim Connection

BY MARK KOVACH

One of the major advantages of the SBX-80 is its ability to sync any drum machine to tape during recording, then at a later time, use another machine to either replace the original programs with new ones, or overdub the part (or parts) with a different sound. For example, you could program bass and snare on a Linn, and later, even after recording guitar or vocals, use the toms, hi-hat, and claps from a DMX. Well, not quite. You see, Oberheim uses a different pulse width for its ppqn (pulses per quarter note) output and input than most other manufacturers. Why they did this, and why Roland did not compensate for this, is just one more example of the kinds of inter-equipment incompatibility the consumer must endure. However, there is a solution.

The first and easiest solution is also (surprise) the most expensive. For an additional $200, you can purchase one of those boxes that will convert one ppqn or FSK (frequency shift keying) signal to any other signal. Now you need only attach the 96 ppqn sync signal from the SBX-80 output to the appropriate input of your new box, send the 96 ppqn output from the new box to the input of your Oberheim gear and like magic, all drum machines and sequencers will work together properly. If you spend a bit more money and purchase a box with simultaneously available outputs, you can run two or more machines from different manufacturers at the same time. This will allow you to sync Korg, Linn, E-mu, Oberheim, Roland, and most other drum machines, while selecting outputs from each and recording them as if they were one drum machine.

You may find, in some cases, a small delay with the addition of the second box, especially if it is loaded down with multiple instruments. However, this can be overcome by using the SBX-80's SMPTE offset function, a powerful feature that Roland was wise to include. By experimenting with the amount of offset, you can effectively eliminate any delays.

THE MOD

The second solution to the Oberheim sync problem is more difficult but less expensive; it is for those with some electronics background and should not be attempted by those who don't have the necessary experience. You will need to replace R106, a 330 Ohm resistor located on the main printed circuit board, with a 33 Ohm 1/4 Watt resistor. The procedure is painless, and will take about 30 minutes.

". . . I find more satisfaction in doing it myself and spending the money saved on beer"

1. Disconnect power and remove the batteries (be certain to save the current program to tape and verify it first).
2. Remove the front knobs (they pull off), then remove the eight screws from the bottom plate.
3. Remove the two screws securing the sideplates and carefully pry them from the main frame (they are held on with clips).
4. Disconnect the ribbon cables from the circuit board. Label all connectors and their respective mates. I not only make a drawing, but also put different colored electrical tape on each connector half to make life easier.
5. Remove the screws securing the circuit board and slip it out.
6. Carefully de-solder R106. Pre-cut the 33 Ohm resistor leads, insert in R106's place, and solder.
7. This is my favorite step (with apologies to Chilton's): reassemble in reverse order.

Of course, you can always take the SBX-80 to an authorized repair center (or unauthorized, for that matter, since Roland isn't picking up the tab) and have them perform the transplant, but I find more satisfaction in doing it myself and spending the money saved on beer. You can also write to Jim Mothersbaugh at Roland, who informs me that the above engineering change will be incorporated into all newly manufactured SBX-80s, and ask (nicely) whether this modification could be performed under the warranty.

ONE MORE THING...

One other SBX-80 limitation is its inability to commit more than one song at a time to internal memory; the memory is configured so that the user may save tempo, time signature, and other data for up to 30 minutes of only the current song. I feel that a better approach would have been to allow storage of more songs of less time, but since this doesn't lend itself to any kind of simple mod, I suggest that you use file numbers incorporated in the SBX-80 to help you keep track of songs or film cues for a given session as follows.

Before saving information to tape, assign a specific file number to a song or film cue. For example, call song one "File 1," song two "File 2," and so on. Now, you can save each successive song on the same cassette sequentially. If you need to call a previous song, for overdubs or fixing, you can ask the SBX-80 to load a particular file, then start the cassette, and the SBX-80 will find the information for that file and load it.

I hope you find the above tips useful when using the SBX-80. EM

***Mark Kovach** is owner of* The Music Deli *(a New York music house) and has written original music for Ford, Radio Shack, Manhattan Cable TV, and Seagrams. He is also co-owner of* MIDI-Impact, *a SMPTE-based computer/MIDI synthesizer room at Media Sound.*

MIDI, SMPTE, synthesis, and smarts all come together when it's time to glue sounds to images.

Alan Howarth: Secrets of Successful Cinematic Synthesis

BY TONY THOMAS

Alan Howarth may not be one of the best-known synthesists around Los Angeles, but he certainly is one of the busiest. If his name doesn't ring a bell, maybe the list of films that he's worked on in the last six or seven years will help jog your memory: *Star Trek—The Motion Picture, Escape From New York, Raiders of the Lost Ark, Halloween II, Poltergeist, Star Trek II—The Wrath of Khan, The Thing, Christine, The Osterman Weekend, Runaway Train, Star Trek III—The Search for Spock*, and the list goes on and on.

I first met Alan when we both lived in Cleveland a decade ago. He was just another musician with a dream when he helped run Pi Keyboards and Audio, then a small music store specializing in synthesizers. At the time, I was a green engineer at a small 4-track studio called "Terra Tu" that began down the hall from Alan's studio on seedy East 59th Street. There, the only entertainment was provided by the drunks who occasionally stumbled out of the bar downstairs. I lost touch with him until we ran into each other at a recent NARAS synthesizer forum held in Los Angeles.

In 1977, Howarth got the call to be Joe Zawinul's road synth-tech for Weather Report, from which he gained a great deal of experience and self-confidence. Buoyed by his work with the legendary Zawinul, in 1978 Alan ventured to L. A. in search of fame and fortune. After a short stint as the Regional Sales Manager for ARP, through a series of events that can only be described as being at the right place at the right time, he landed the gig as one of the sound designers for *Star Trek—The Motion Picture*, the movie that catapulted his career as a sound designer and film composer. Since then, he has amassed an impressive and ever-growing list of credits.

How does a composer work in the Hollywood environment? What are the mechanics of scoring a film? Is this the kind of career an electronic musician should consider? To find out some of the answers, I met with Alan recently at his spacious Glendale home, which houses his 24-track "Electric Melody Studios." An impressive synth collection includes a Kurzweil, Emulators I & II, E-mu SP-12 drum machine, the new Sequential Prophet VS and 2002, a Linn LM-1 and LinnDrum, a pair of ARP Avatars, a rack mounted Oberheim four-voice, and the Prophet V with which he started. At the time I spoke to him, he was hard at work on John Carpenter's next film, *Big Trouble in Little China*.

EM: What is it like working as a film composer in Hollywood?
AH: At this point, I operate in Hollywood without an agent. I've approached a couple of big agencies and they're aware of me; they're watching what I do this year and they'll see how I'm doing next year. And maybe two years from now, after they see that I'm for real 'cause I'm cranking out stuff, and my phone is already ringing because my name is big enough, maybe then they'll take me on and handle my business for me. But right now, they're not interested.

EM: By then you won't need them!
AH: Right, just when you don't need them, they're willing to help you. So, I have a little press kit I've worked up, I have a videotape of edited segments from the various films I've worked on—sort of a video resume—and that is what I supply to the film producers. Plus, I subscribe to the *Hollywood Reporter* and *Variety*. I look for "Films in the Future" and I read down the list to see if there is anyone that I've worked with in the past. If I know the editor or producer, I'll make a call and get a feel for what's happening. Occasionally, someone will call me based upon the fact that they've seen *Escape from New York* or *Christine* or another film I've worked on and say "I really liked what you did for that show. This is the sound I want for my show. Let's talk."

Once you've made contact, the process is pretty much the same. You have a meeting and they give you the script.

PHOTO ANN SUMMA

Tony Thomas has been involved in advertising, broadcasting, recording, and publishing for the past ten years. He is presently the managing director of Target Communications International, a full-service advertising agency, broadcast production company, and MIDI-based recording studio.

This is a departure from the way it worked a few years ago because, for the most part, they used to shoot the movie and then go looking for a composer. But, in the last two years, composers are picked before the movie is even shot. The casting of the composer has become as important as the casting of the actors and they try, obviously, to get the most talented composer for the least amount of money. If they have a lot of money, then I'm out because they can get Jerry Goldsmith, John Williams or Vangelis. To be honest, I'm not in that league yet. Five years from now, I might be . . . it's a long-range process. I'm getting there, though. This is the first time in my life that I've been booked six months in advance. Up until this point I would do one show, finish that and start the hustle for the next thing, and so on. Film work is sort of seasonal—it's usually dead in June and December. During the gaps, I try to supplement my film work with commercials and TV work.

EM: After you get the gig, where does it usually go from there?

AH: Next you make arrangements to see the movie, when it's ready. If you're in on the very early script stages, you end up sitting around for three months. Finally, when the film is ready, they give you a screening. The first time you see the movie, it's best not to even make notes. Just watch the movie, get a feel for it, and develop an overview of the musical style that would be best for it. By that time, the producers have usually cut in pre-recorded music from records to indicate the general type of overall sound they want. You have to remember that making movie music is the opposite of making records. In records, you're attempting to second-guess the public. When you're making a movie, you need only second-guess one or two people: the producer and the director. If they're happy you're off the hook, at least for the time being.

Then, you go in and, reel by reel, make notes with the producer and director in what is called a "spotting session." Your first big theme statement is in the opening credits. You usually don't have any competition in those opening moments; no one talks and there are no sound effects. After that, you are into the movie, and when the dialog begins, you either underscore the visuals or the music goes out completely and shows up again at the next point of action. So you come up with what are called "spotting notes."

Because I'm working in my own studio, I'm also the equivalent of the music-editor. In more sophisticated situations, there is a music-editor who will take your music cues and cut them into the show. But, because I am working with videotape, I make all the music cues on one reel with spaces in between. All they have to do is to take my reel, which is ten minutes worth of music with spaces, and sync it up to the 2-inch "sync-pop" at the beginning of the reel. The way I work has a certain appeal to the producers because they can hear the work in progress rather than having to wait to hear it on the scoring stage. It has a certain appeal to me because it gives me complete autonomy in my scoring.

> *"The casting of the (movie's) composer has become as important as the casting of the actors"*

EM: How do you use each piece of equipment?

AH: Well, it's obvious what LinnDrums are for. I use the ARP Avatars as guitar synths, but I also use them sometimes for short bass sequences. They're essentially ARP Odysseys and I love the ARP bass sound. I also recently had my old Oberheim Four-Voice rack-mounted and MIDIed and that's my other favorite analog bass sound. I use the Kurzweil for acoustic events. I use the Emulators and the Prophet when I want to create something totally unique.

EM: Take us through the average film-scoring session and tell us how you approach it from the technical standpoint.

AH: I start with the LinnDrum and use it as my master click. Even if there are no drums in the cue, I always lay down my synth parts to the Linn click. The main sequencer I use is the one in the Emulator II. I'm very familiar with it, I know its limitations and I work within them. I write short sequences on the Prophet Polysequencer, little ostinato things that repeat over and over, because it single-steps so easily.

Once I am in the process of scoring, have my click down, and have overdubbed to the point where I have the cue up and running, it's time for the producer or director to come over and see how things are going. At that point, I play what I've done, get some kind of reaction, and make changes accordingly. I used to track the whole movie, and then the week before it was due, have one big mix session. On this movie I'm doing now with John Carpenter, *Big Trouble in Little China*, I've changed that and have begun to mix as we go. Even if we only use half of the mixes I create at that stage, I still have saved four or five days worth of work. Another thing I've been doing is saving reverb settings with each track. I used to record everything dry and add reverb at the mix stage because everyone told me it was taboo to record with reverb. But now I like saving my effects, because when you're ready to dial up the mix, magically, everything is there.

EM: How does using SMPTE help?

AH: It really helps when they make your video dub, you do your thing, and then they change the movie! That's why I now ask for footage and SMPTE to be burned into my video dub. If they've lifted 22 feet out of the movie at a certain spot, then I can change the SMPTE offset on the 4-track (Right-Center-Left-SMPTE) master to compensate for the change. You have to have a Q.Lock, Adams-Smith, BTX-Shadow, or other high-end SMPTE synchronizer if you are going to do that kind of frame-accurate editing in the studio. Of the low-end units, I have heard that the new Fostex unit will do the same thing, though I don't have any hands-on experience.

EM: Before we sign off, I'm curious whether you still do a lot of overdubs on your 24-track even though you have a very complete MIDI setup.

AH: I still overdub. But what I also do is save more keyboards to one track for a MIDI-stacked sound. Before, I may have used five tracks of keyboards, whereas now I'll mix five keyboards down to one track. In a way, MIDI has expanded my 24-track to 36 or 48 tracks, which helps make the sound that much more powerful. I have also been making two- and three-channel stereo mixes of several MIDI-stacked keyboards panned across the stereo field. If you had to take that same sound and spread it over five tracks, you end up with much more tape hiss than if you mix down the MIDIed sound to one, two, or three tracks. So, among the other things that it does, MIDI has also made analog recording more viable. EM

The "one size fits all" concept has traditionally not worked too well, but now it seems that a new generation of signal processors is about to realize that ideal.

SPX90: The First Generic Audio Box

BY KIRK AUSTIN

Outboard effects have come a long way in the last few years, and digital technology has been mostly responsible for these rapid advances in products designed to enhance, modify, or control the audio signal. Digital has definitely become the most popular buzzword of our time. Everything's got to be digital! I even have a digital soldering iron (actually it's a soldering iron with a digital temperature readout). Seriously, though, digital audio devices are able to do more things with more flexibility than their analog predecessors, and the power of digital technology is going to carry us into a new era of audio signal processing.

What's significant about digital audio is that once you convert the audio signal into a digital representation, you can do whatever you want with it in software. Thus, the same *physical* components can reverberate a signal or compress it—the only difference is the data-manipulating program inside the device. The problem with all of this is that programmers don't work for free. As the programs get more complicated, the time and expense required to produce a product becomes greater. However, programming is more or less a one time cost; at first there is a great deal of time devoted to working out the basics of any particular program, but after some experience is gained, things move along pretty smoothly.

THE GENERIC AUDIO BOX

Enter the "generic audio box." Yamaha has implemented this vision of a general purpose audio signal processor in their SPX90. While this product is not "perfect," it sets a new standard for digital signal processing since it can perform so many functions. For those on a limited budget, the SPX90 would be the most economical way for a small studio to gain a great deal of functionality with just one unit. This is particularly true if you are a solo musician/composer who does a lot of overdubbing. While you are laying down the bass track, the SPX90 could serve as a compressor. Then when you record the guitar, switch over to stereo flanging. For vocals, the SPX90 is best utilized as a digital reverb, although you will undoubtedly want to experiment and try other effects with it.

> ## Product Summary
>
> **Type:** Digital Signal Processor capable of reverb, echo, flange, chorus, tremolo, noise gate, compression, pitch change, short sample, pan, parametric EQ, program change via MIDI.
> **Price:** $745.
> **Memory:** Thirty preset effects and 60 user programmable effects.
> **Bandwidth:** 20 Hz to 12 kHz
> **Manufacturer:** Yamaha International Corporation, 6600 Orangethorpe Avenue, Buena Park, CA 90620. Tel. 714/522-9011.

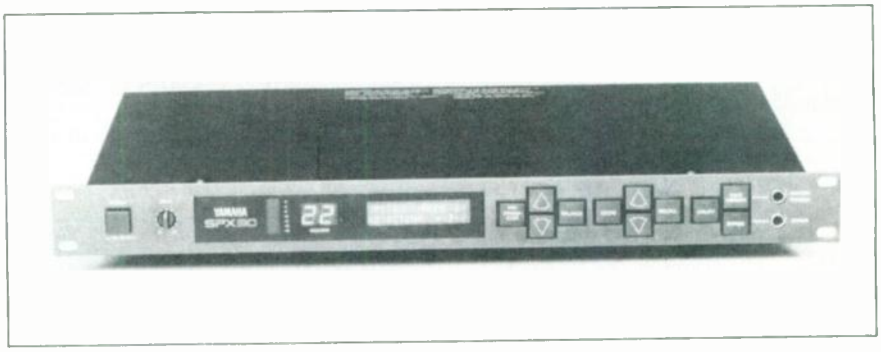
THE MIDI CONNECTION

Just the fact that this piece of equipment does all the things it does is pretty terrific, but the various effects can be recalled via MIDI to produce *sequenced* effect tracks. Imagine being able to switch from reverb, to chorusing, to echo, to vibrato, all on the same track *while the track is being recorded!* I think we will start hearing more of this type of processing on records now that it is available. We're all used to just setting a particular effect on a track and letting it stay that way through the entire song, but MIDI-controlled signal processors like the SPX90 are going to change that forever. . .once artists are given that kind of capability, you can bet that they are going to use it. I think that this is wonderful; it not only gives us more control over the outboard effects, but is also a big step towards making MIDI the powerful standard that I had hoped it could be. The great thing about the MIDI standard is that it is so inexpensive to implement that there is no reason why every piece of gear with a microprocessor shouldn't include it. Once every musical device has a MIDI port, the MIDI recorder (I refuse to call them sequencers anymore—sequencers were crude, limited devices compared to today's MIDI recorders) will become an extremely powerful tool in the creation of music. Maybe it will be even more significant than the multi-track tape recorder.

FEATURES AND CAPABILITIES

Now let's move from generalities to specifics. Of course, there are other multifunction audio boxes (such as the Lexicon PCM70 and Korg DVP-1) but for now the SPX90 is unmatched for versatility. Believe it or not, the following is only a partial list of the many programs available as standard equipment. By the way, the SPX90 has stereo outputs even though it has only a mono input; this makes it very useful for creating stereo signals from mono ones.

—page 76



Kirk Austin designed one of the first stand-alone MIDI keyboard controllers. He is employed as an assembly language programmer for an industrial controls company and creates much software for the Mac. In addition to writing for EM, he also contributes to several Macintosh magazines.

You say you spent every penny you ever had on an Apple Macintosh and have nothing left over for software? Well, help is on the way.

Budget MIDI for the Mac

BY GEARY YELTON

If you want the advantages of MIDI without having to sell your car, there's hope. In this article, we'll examine three programs that can help towards setting up a budget Mac MIDI studio.

MIDI COMPOSER

MIDI Composer, from Assimilation, gives no-frills multi-track emulation for $29 list. Like most sequencers, you can record and play back multiple tracks through any MIDI instrument. Operation is easily mastered by anyone who can handle a 4-track tape recorder.

While the instructions specify Assimilation's MIDI Conductor interface, any MIDI interface that operates at a clock rate of one MegaHertz will do. The on-screen graphics show channel and mode selector buttons, tape transport controls (play, fast forward, rewind, and return to beginning), and a cassette tape with hubs that really turn (see Fig. 1). All operations are performed with the mouse; there are no keyboard equivalents. There's an adjustable-rate metronome (which can also be turned off) but unfortunately, changing the tempo affects only the metronome speed, not the playback rate. Thus, the metronome's only function is as a reference for synchronizing additional tracks by ear.

Only note information is recorded. Once you begin recording, a real time counter indicates the passage of time in seconds as the metronome ticks through the Mac's speaker. Clicking "stop" automatically puts the track into play mode. I noticed that if you stop playback in the middle of a sequence, note off commands are not automatically sent, so it's possible that notes will stick on.

The instructions say you can edit recorded music, but other than being able to easily reassign MIDI playback channels, you can't change a thing. If you want to alter any part of your recording, you must re-record the entire track. And since you can't change the playback tempo, it's impossible to record a part slowly and then play it back faster.

Although there are handy on-line operating instructions available, MIDI Composer is so simple to use I can't imagine anyone needing to see them more than once. Unlike some new software, MIDI Composer operates smoothly and without detectable bugs. It's not ex-

Geary Yelton is a writer, synthesist, and music software consultant. His music has been heard in bars, parks, concert halls, films, and on television. He is the author of The Rock Synthesizer Manual, *from Rock Tech Publications.*

Product Summaries

MIDI Composer
Type: Four-track tape emulation type sequencer.
Price: $29
Hardware requirements: Apple Macintosh, 1 MHz clock rate MIDI interface, MIDI instrument(s).
Main function: Recording of note information.
Manufacturer: MIDI Composer is no longer manufactured by Assimilation, Inc., but is still available in some stores.

Deluxe Music Construction Set
Type: Eight-track sequencer with cut and paste editing.
Price: $49.95
Hardware requirements: Apple Macintosh, 1 MHz clock rate MIDI interface, MIDI instrument(s) optional.
Main functions: Sequencing, music printing.
Manufacturer: Electronic Arts, 2755 Campus Dr., San Mateo, CA 94403. Tel. 415/571-7171.

Concertware+ MIDI
Type: MIDI or non-MIDI eight monophonic track sequencer with editing.
Price: $139.95
Hardware requirements: Apple Macintosh, MIDI interface (500 kHz, 1 MHz, or 2 MHz clock rate), MIDI instrument(s) optional.
Main functions: Sequencing, music printing, automatic transcription, sound construction via Fourier analysis.
Manufacturer: Great Wave Software, P.O. Box 5847, Stanford, CA 94305. Tel. 415/325-2202.

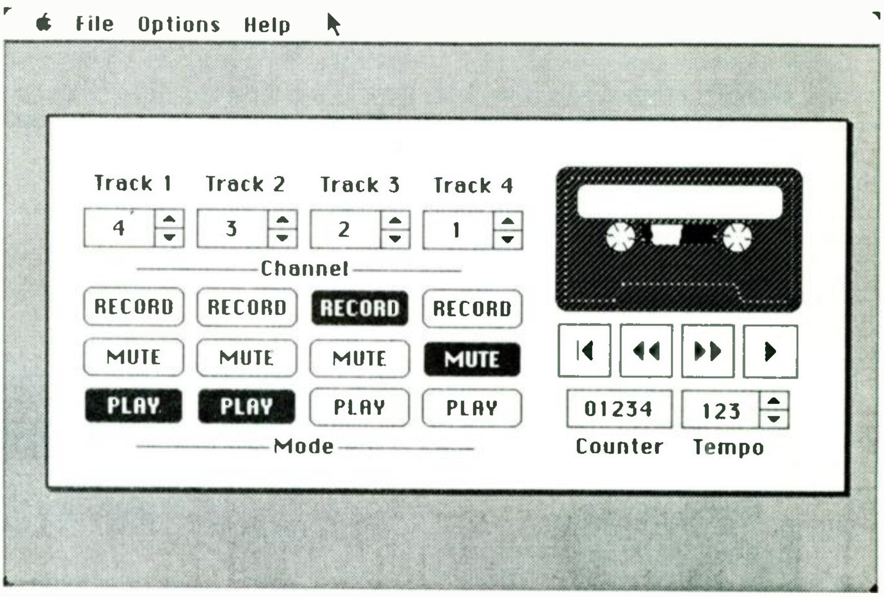


Fig. 1 *MIDI Composer's graphics resemble a cassette transport's controls.*

actly packed with powerful features, but where else can you get a 4-track sequencer at this price?

Deluxe Music Construction Set

Deluxe Music Construction Set, from Electronic Arts, is a real bargain that delivers a lot for the price. DMCS offers high-quality music transcription with features second only to Mark of the Unicorn's Professional Composer (reviewed March '86 *EM*). Professional Composer is admittedly easier to use, but DMCS can play back through MIDI instruments as well as the internal Mac voices, and costs about 1/10th as much.

At first, DMCS seemed complicated to use and not as intuitive as most Mac software. At this price, I hadn't expected such a complex music program. Many operations are not obvious; I had to plow through the rather thorough documentation before I could figure out how to assign MIDI voices, for example. Procedures I expected to find as menu selections were hidden away in the Score Setup window. Only after I read the manual and actually composed a few short pieces did I begin to truly appreciate the program.

Booting DMCS gives three windows: the score window, the symbols palette, and an on-screen musical keyboard called the player piano (see **Fig. 2**). Unlike a true sequencer, DMCS doesn't provide any sort of MIDI input; all music is entered via the mouse (or the mouse and Mac keyboard). You select note or rest values from the symbols palette, and then position symbols on the staff or click the keys on the player piano. Number keys can assign rhythmic values (1 for whole notes, 2 for half notes, etc.) as you enter notes and rests. Triplets and quintuplets are possible. Notes are heard as they are entered. Accidentals may be added to any note, but double sharps and double flats aren't supported. Groups of notes can be tied, slurred, beamed, and bracketed as desired. Two tracks can appear on a single staff, including chords. A maximum of eight polyphonic tracks can be transcribed and played back on any eight MIDI channels. Only four Macintosh voices will sound simultaneously, but you can play Mac voices and MIDI instruments at the same time.

A Score Setup window allows for specifying such details as the number of staves, the space between them, and how many measures appear across the width of the screen and the printed page, as well as the initial tempo, clefs, and the volume of the Mac's internal sound generator. One menu lets you insert key and time signatures, clef, tempo, and instrumental assignment at the beginning of any measure. MIDI program change is supported; although invoking it seems needlessly complicated, it's not hard to get used to. More frustrating is that program changes can only take place on the first beat of any measure.

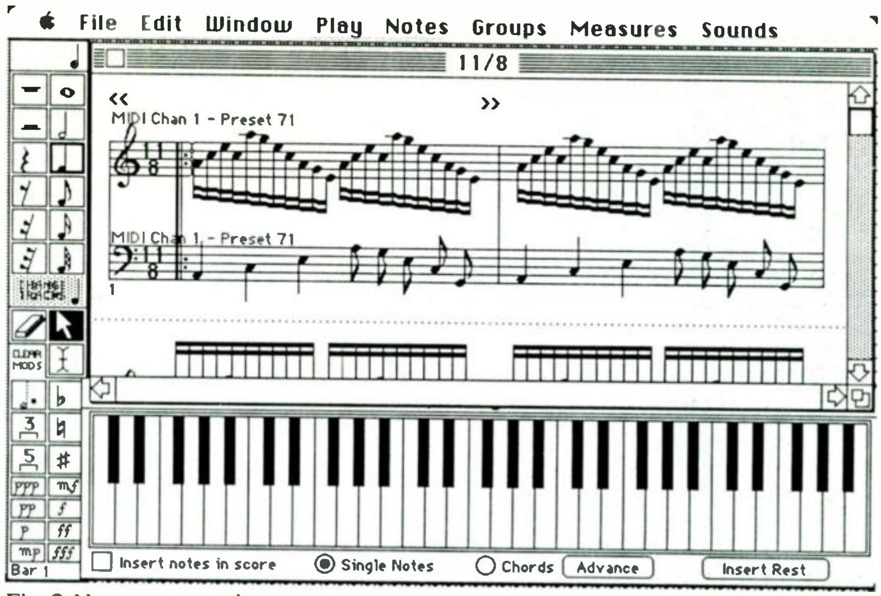


Fig. 2 *Notes are entered into DMCS either by positioning symbols on a grand staff or by clicking on the "player piano" keyboard.*

Lyrics and other text can be placed anywhere in the score, and three special fonts let you type in additional musical symbols and guitar fret notation. As far as I know, the ability to include guitar chord symbols makes DMCS unique among music notation software.

Editing capabilities are impressive. Adhering to the Macintosh interface, selected portions can be cut, copied, and pasted. Symbols are easily repositioned with the mouse, while the backspace key deletes selected notes. Note duration can be easily changed, and dynamic symbols can be placed anywhere—you can even enter MIDI velocity values to correspond to dynamic levels (e.g. ppp can equal a velocity of ten and fff a velocity of 127). Crescendo and diminuendo signs can gradually vary the velocity between two dynamic symbols.

Like MIDI Composer, you need an adapter with a 1 MegaHertz clock rate for MIDI playback. For non-MIDI playback, a total of 27 preset waveforms with ten variations are available. The music scrolls across the screen in proportional notation as it plays, and the keys can be highlighted on the player piano keyboard as if played by tiny invisible hands. However, all this visual activity sometimes taxes the microprocessor, and stalls the playback in progress. Finally, playback can be stopped and the most recently played note highlighted by pressing the space bar. For editing purposes, only a section of the score can be played by indicating its beginning and end.

Deluxe Music Construction Set has a few bugs. I had no trouble getting it to bomb until I learned which operations to avoid. But it also has many features which make it an indispensible tool for any Macintosh musician on a budget. It lets you write and print attractive, musically correct manuscript. It allows MIDI playback, and it doesn't cost an arm and a leg. Interestingly, other software makers recognize the accessibility of DMCS. If you happen to have the latest version of Opcode's MIDIMAC Sequencer, you can print your sequences by saving them as DMCS files. If you have Musicwork's MegaTrack and MIDIWorks, you can convert MegaTrack sequences to DMCS files or vice-versa.

ConcertWare+ MIDI

ConcertWare+ MIDI is a step up from DMCS in terms of price, and in only some ways, performance. It offers functions you won't find at twice the price. The original ConcertWare was the first integrated music software for the Macintosh, and can compose and print scores, construct sounds by Fourier synthesis, and play back compositions using those sounds through the Mac's internal sound generator. The latest version also includes both MIDI input and playback. ConcertWare+ MIDI is the only Mac program

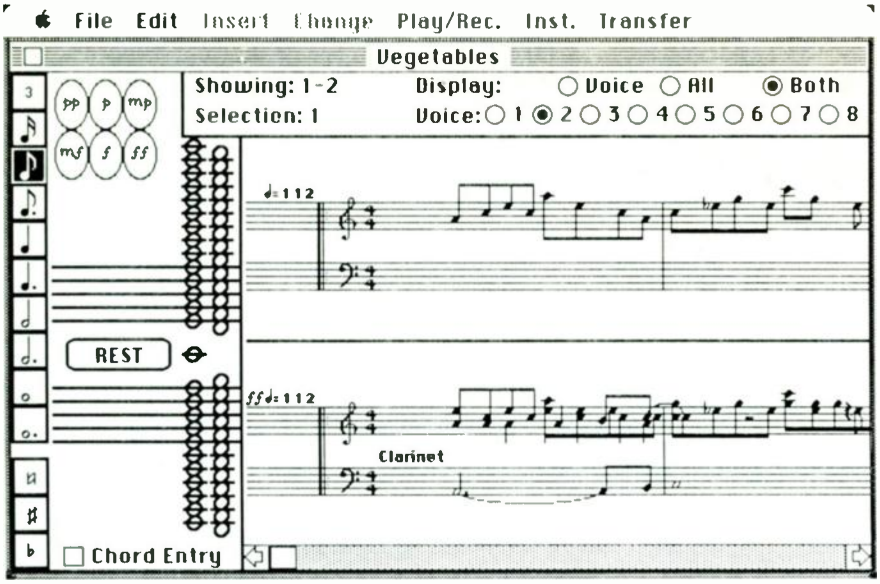


Fig. 3 *ConcertWare+ MIDI Writer allows note entry from the Mac keyboard, from a MIDI keyboard, or with the mouse.*

under $150 that notates what you play on a MIDI keyboard (subject to certain limitations).

ConcertWare+ is available with and without MIDI. Both versions actually comprise three interrelated applications: Writer, Player, and InstrumentMaker. The MIDI version of Writer lets you score monophonic parts on eight staves, using the mouse, the Mac keyboard, the non-MIDI MacNifty keyboard, or any MIDI instrument. The grand staff you're working on sits above another grand staff that shows all the parts (see **Fig. 3**). The shortest duration possible is a 16th note, and triplets are supported. When input is from a MIDI instrument, all notes are automatically quantized.

Opening MIDI Writer summons a MIDI Setup dialog box. The MIDI adapter speed (the rate at which the adapter communicates with the Mac through one of its serial ports) can be adjusted to one of three settings to match the brand of MIDI adapter you're using. I wish all MIDI software for the Macintosh offered this choice. Each of eight voices can be assigned a MIDI channel. Velocity data and patch changes can be filtered out, and the MIDI mode can be changed. In Omni mode, data received on any channel is recorded. In Poly mode, only channel 1 data is recognized, and Mono mode receives on all channels.

MIDI Writer's greatest shortcoming is monophonic input—chords must be built up one track at a time. Notes, velocities, and patch changes are all recorded and displayed in the score. When you choose the Record command, a dialog box appears and a metronome begins ticking. At this point, you may change the recording tempo and determine which voices will be heard during the overdub. Recording begins when you start to play or when you click the Start Recording button. When finished, you may re-record the sequence or insert it into the selected track.

Editing capabilities are similar to other musical Mac programs. Notes and tracks may be transposed, and their durations may be halved, doubled, or converted to triplets. Groups of notes may be slurred or beamed. Repeats and alternate endings are possible. (I noticed a moment's hesitation at the repeat symbol during playback.) Tempo, program, and dynamic changes may be added, and velocity values may be displayed and altered. Marks can be inserted in the score which are automatically located on command. Any or all tracks or selected portions can be played from any point in the score.

Scores can be printed on an ImageWriter or LaserWriter, but the results are not as attractive as music printed with DMCS. Scores can also be saved as MacPaint documents for further visual embellishment. Notation appears either too big or too small on the screen, but three sizes are offered for printing. Eight parts on a single grand staff get a bit crowded, but individual monophonic parts can be printed.

ConcertWare+ MIDI includes preprogrammed public domain songs. Opening up a song file sends you to MIDI Player (**Fig. 4**). Icons representing as many as eight Mac-voice instruments appear on the screen, which you can change and rearrange as desired. Instruments can be assigned to particular synthesizer patches for MIDI playback. MIDI Player displays the song title and additional text, usually concerning the history of the piece, and a proportional notation display rolls by as it's playing. If you're using ConcertWare+ MIDI for strictly MIDI applications, then Player is redundant. You can play your sequences just as well from MIDI Writer, but if you use the Mac-voice instruments, you need Player to hear those voices with their programmed envelopes.

InstrumentMaker has nothing to do with MIDI but lets you design your own instrumental sounds by additive synthesis. You define the amplitudes of the first

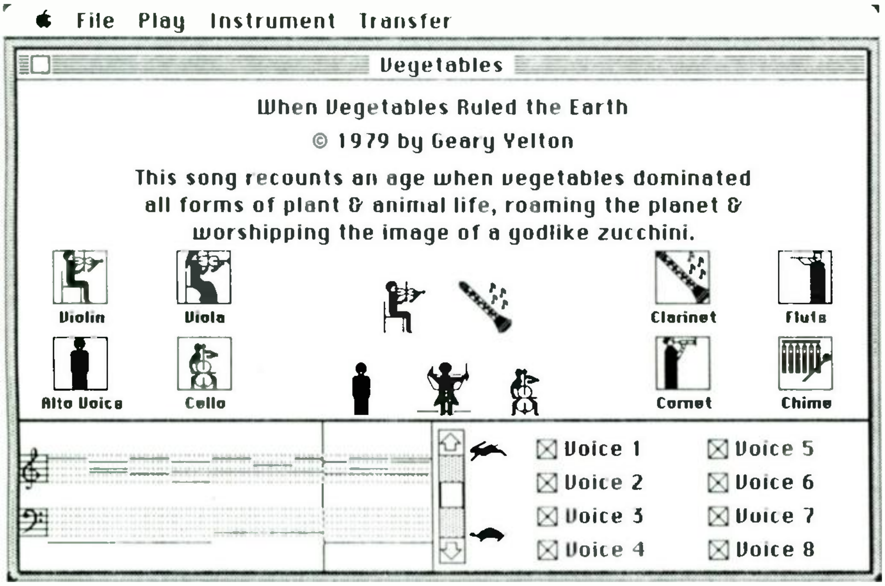


Fig. 4 *MIDI Player lets you assign MIDI program numbers to specific Macintosh voices.*

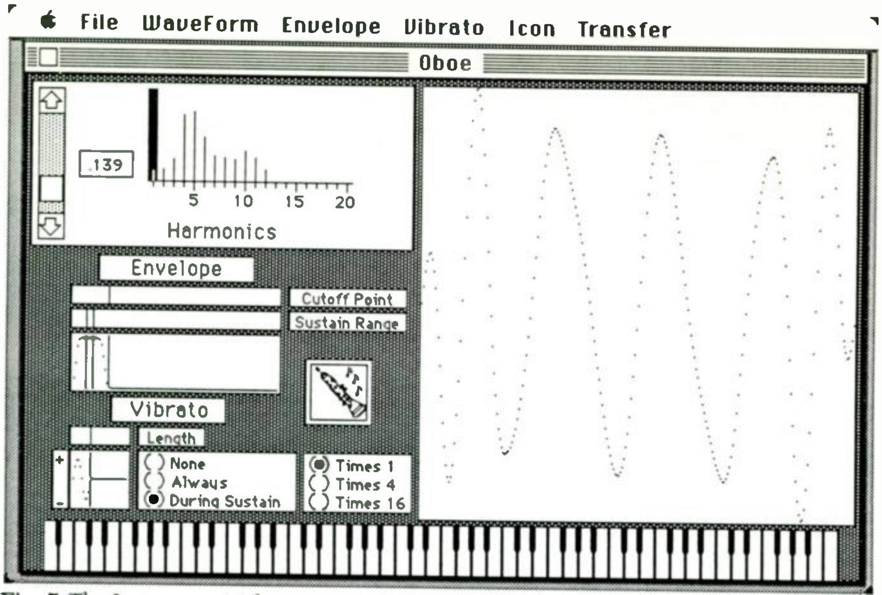


Fig. 5 *The Instrument Maker portion of ConcertWare+ turns your office appliance into an additive synthesizer.*

20 harmonics, then let the program calculate and display the resulting waveform in its window. An alternate method lets you draw waveforms with the mouse. Another window lets you draw the envelope (freehand or in Fat Bits) and set the sustain stage loop points. Waveform and envelope parameters can be transferred from any other instrument file. You can also draw the vibrato waveform (freehand or in Fat Bits) or select from a number of shapes, then specify when vibrato will take place and at what depth. When your new instrumental voice is complete, you can even design an icon to represent it on the MIDI Player screen, and use the Mac keyboard or the on-screen musical keyboard to hear it.

The Macintosh sound generator is hardly professional-sounding, but timbres produced by InstrumentMaker are the very best I've heard from a Macintosh (except for sampled sounds). ConcertWare+ MIDI comes with 40 instrumental sounds already programmed.

Conclusion

For bargain software, you need a bargain MIDI interface. The MacMIDI Star, from Musicworks (18 Haviland, Boston, MA 02115; tel. 617/266-2886), has one MIDI In and three MIDI Outs for $80. Its clock speed is 500 kHz, but it can also run software at 9 or 2 MHz. For those who want to roll your own, see Kirk Austin's plans in the October and November '85 issues of *MacTutor*. If anybody knows of any other MIDI bargains for the Mac (we covered OpCode's MIDIMAC patch librarian line in the March '86 EM), feel free to contact me c/o EM.

Not everyone needs the power offered by software like Performer or Total Music. Not everyone can afford them. But all told, Macintosh users can get on the MIDI bandwagon for less than you might expect. **EM**

—from page 71, SPX90

The **reverb** programs are good, but not as smooth sounding as my favorite digital reverb, the Yamaha REV7 (which also costs, twice as much as the SPX90). The **delay** programs are nice, particularly since you can set different delay times for each output channel; this property can turn a mono input signal into stereo. You can also roll off the high frequencies of the delayed signal to make it sound "less digital." Other delay-type effects include **stereo flanging** (this creates a broad stereo image as the sound spins around, and is great on guitar) and **stereo chorusing**, which is excellent for getting an ensemble sound out of a single instrument. Synthesized strings really fill the soundstage when processed with this program.

For harmonization effects, the **pitch shifting** programs allow you to make slight variations (1/100ths of a semitone) in pitch for a thickening effect, or semitone increments that can be played from a MIDI keyboard. This capability is useful for playing a melody from a single pitched input signal. You can also create an echo that rises (or lowers) in pitch with each repetition, which is really strange.

There's also a primitive sampling capability, the **freeze** program. This allows you to sample an input signal and play it back, which in itself is nice, but you can also shift the playback signal pitch using a MIDI keyboard. While this is no match for a dedicated sampling keyboard, it is good for sounds such as toms and staccato bass parts.

> *"(the) 'one box does all' concept . . . is going to have tremendous impact on how future products are designed"*

Another time-based effect, **delayed vibrato**, introduces vibrato. Adding vibrato to a sound that doesn't usually have vibrato can be pretty effective. Try this one on an acoustic piano sound for a real surprise.

Not all effects are time-based, however. The SPX90 can also be used as a **noise gate** if you need one, although dedicated noise gates are much cheaper. It is interesting to know that it can be done, but I think the other SPX90 effects are more useful. There's also a **compressor** program, which is probably most useful on something like electric bass since the bandwidth is limited to 12 kHz. This limited bandwidth also limits the usefulness of the **parametric EQ** program.

There's even an **auto panner.** That's right, an auto panner. Now your guitar leads can sound like Jimi Hendrix on "House Burning Down." Well, I guess you would have to be a pretty great guitarist also. Practice, practice, practice.

Again, this is only a partial list, but you get the idea. It's a "one box does all" concept that is going to have tremendous impact on how future products are designed. As memory gets cheaper (which it is doing constantly), we will see an implementation of this type that overcomes the limitations found in the SPX-90. The 12 kHz bandwidth is fine for reverb and echo, but not too good for effects like compression and equalization. Also, the maximum delay time per channel is half a second, which could be a bit longer for certain applications. All things considered, though, this is a terrific product for the price. It's pretty astounding to think that a few years ago all you could get for this amount of money was a simple digital delay. My hat's off to Yamaha. **EM**

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If you want to get your MIDI signals mixed up, do it right.

SynHance M1X and M1X+ MIDI Mixers

BY JIM JOHNSON

The SynHance M1X ($295) and M1X+ ($375), from Harmony Systems, are MIDI data processors that merge two incoming MIDI signals into one or two outputs. They also have system exclusive (sysex for short) implementations that allow them to be used as remote controlled MIDI switchers. For this review, I evaluated the M1X+.

Why Merge MIDI?

Sooner or later, anyone with a reasonably large MIDI system runs into the need for a MIDI mixer. In my case it was sooner. When the Oberheim Xpander first came out, I was thrilled at the prospect of being able to play some of the voices from my remote keyboard (a Yamaha KX5) and the rest from my sequencer. However, this necessitated squeezing two MIDI signals into the Xpander's single MIDI In jack. Although putting the sequencer in Record mode let me echo the KX5 data through the sequencer while simultaneously playing back previously recorded tracks, this method never worked very well; doing a quick glissando from one end of the keyboard to the other would often hang up the sequencer for the duration of the gliss, which always caused the rest of the band to glare at me. Clearly, I had to find another solution.

Intelligent Processing

The SynHance MIDI mixers solve this problem by using a dedicated microprocessor to control the flow of data within the mixer from the two inputs to the two outputs, and to eliminate conflicts caused by receiving messages simultaneously on both inputs. Data from each input is stored in a separate buffer until a complete message is received, then the message is sent to the MIDI Out when the output is not busy. The SynHance mixers also employ continuously variable data filtering to prevent embarrassing glitches. As the input buffers start to fill up, the mixer removes an increasing amount of secondary data (pitch bend, aftertouch, etc.) from the data stream to prevent overloading the MIDI bus, as well as the mixer itself. This is a nice touch, since you don't have to make a choice about what data to send and what to filter.

Jim Johnson is a linear integrated circuit test engineer by day; by night he plays synthesizer in the Phoenix synth trio girl:bike:dog. He also writes music software for fun and profit and wears pointed sideburns.

Stop the Presses!

One day before we went to press, Harmony Systems discontinued the M1X, changed the name of the M1X+ to M2Y, and dropped the latter's price to $199.95. Yes, things do move fast in the world of musical electronics . . . —Ed.

Both mixers allow the user to assign either or both inputs to one or both outputs through sysex messages. The M1X+ also has four switches and LEDs on the front panel to set up and display the output assignments, as well as battery backup for the assignments. If you have a sequencer or keyboard that allows you to program your own sysex messages, you can use this feature to change ouptut assignments as you play. The front panel also has two LEDs that show which input is currently receiving data, and indicate errors such as bad data or input buffer overflow.

In addition to the two inputs and outputs, the back panel has two jacks which can be set up as MIDI Thrus, or as duplicates of the two outputs. The jacks are initially set up as Thrus; altering this involves opening up the mixer and changing the position of two jumpers. No soldering or cutting is required. An AC adapter (included) supplies power. Two mixers can also be mounted side by side in a 19-inch rack using the Boss Micro Rack Adapter and two screws that come in the box. By itself, the mixer fits perfectly on top of a Commodore SX64 using the stick-on rubber feet.

The manual is very complete. In fact, if I were nitpicking, I'd say that it's too complete, since each chapter is only about three pages long and includes both an introduction and a summary of the chapter. Still, in this case, too much is definitely better than too little. The manual includes a long section showing how to hook up the mixer in various applications, such as synching a sequencer to a SMPTE or MIDI clock while recording, controlling a single synth from two keyboards, and using a computer-based patch editor along with a master keyboard. The manual also goes into some depth regarding the mixers' sysex features, which include the ability to assign a mixer a unique device number and to request a mixer to send a message regarding its current input/output assignments, as well as the previously mentioned functions. The sysex implementation on these instruments is apparently a subset of an extensive set of commands that Harmony Systems is planning for future products, and which will allow different types of MIDI processors to talk to one another and supposedly do all kinds of neat things. I don't know what these would be, but it is something to think about if you are buying a mixer for a very large studio.

Performance

All this looks good on paper, but the bottom line is how glitch-free the machine is in actual use. To test the mixer, I connected the inputs to my KX5 and an SX64 running Dr. T's Keyboard Controlled Sequencer, and the outputs to the Xpander, two drum machines, and four other synths we use in the band. The sequencer was supplying MIDI clock for the system, as well as sending sysex messages for different instruments, and of course, note data. I used this setup in rehearsal, and also spent considerable time just fooling around with it, trying to make the mixer glitch. I'm happy to say that not once was I able to make the mixer drop any data. I did run into a few annoyances, though. Quite often, when I was done playing a song, I would look at the mixer and notice that the error light was flashing on the input assigned to the KX5, though there had been no audible problem. I think this may have been due to my habit of turning the remote off immediately after each song, to save batteries. I never saw an error on the sequencer input.

All in all, I would say that the SynHance MIDI mixers do their job very well, with a minimum of effort required of the user. The price may seem high for something that performs such a simple function, but remember that the MIDI data format is much more finicky than a simple audio or CV format, and this is reflected in the complexity of the circuitry. I would also recommend spending the extra bucks for the M1X+ if you intend to do any switching, since you have to send a sysex message to the M1X every time you want to change configurations. These devices are sure to justify their cost in any medium to large MIDI setup.

Acknowledgements

Special thanks to Synthony Music in Scottsdale, Arizona, for providing the M1X+ used for this review. EM

—from page 39, SERVICE CLINIC

quality soldering iron (Ungar model 9375, or equivalent); some quality 60/40 *rosin-core* solder (never, ever use acid-core solder with electronics gear); a vacuum desoldering tool (Radio Shack 64-2098); 1/4, 3/16, and 1/8 inch (6, 4.75, and 3 mm) flat-blade, and #0, #1, and #2 Phillips screwdrivers (any quality brand); a selection of miniature flat-blade screwdrivers (Radio Shack 64-1982); small and medium-sized needlenose pliers; small diagonal cutters; slip-joint pliers; 1/4, 5/16, and 1/2 inch (6, 8, and 13 mm) sockets/driver (or nutdrivers); a hex-key set; and one or two "TV" adjustment tools (Radio Shack 64-2223).

Proper soldering techniques are a critical service skill. Ideally one would learn the best techniques one-on-one from a master technician, but often such instruction is unavailable. There are numerous books and even correspondence courses that will show you in excruciating, step-by-step detail how to do everything—*wrong*. (Well, maybe they're not *that* bad.) We'll thoroughly cover soldering, tools and test equipment in future Service Clinics. Meanwhile, use a good soldering iron with a clean tip and quality solder, and *think before you melt!*

Electronic Kits

Once you've acquired some knowledge and skill, you'll want to "get your hands dirty." Building electronic kits can be an excellent introduction to fabrication and troubleshooting techniques—skills that are vital to electronic music service and mods. Kits from larger manufacturers include easy-to-follow instruction manuals that often contain tutorials on circuit theory. Two well-known sources for kits are Heathkit (wide variety of kit types) and PAiA Electronics (synthesizers, audio, computers). Xerbitron carries MIDI kits by EM author Tim Dowty. All three addresses are in DataBank. You can find everything from synthesizers and stereos to robots and radiation detectors, in kit form.

Equipment Cleaning

Equipment cleaning is a good DIY service project; the extent to which equipment can be maintained or restored is little short of amazing. But, key contact, potentiometer, and even case cleaning often require disassembly of the unit. This is where the service manual proves indispensable. Disassembly frequently involves a "trick"—a hidden screw or latch, or some ritualistic aerobic motion required to avoid ripping out a wiring harness—as detailed in the service manual. Thankfully, electronic music gear is not as difficult to get into as is some consumer stuff, e.g. cassette decks.

Pots and key contacts are generally cleaned with 100 percent TF solvent (Freon) in a spray can, often referred to as "no-residue" cleaner. For key contacts TF is sprayed on the contact surfaces and the key is struck repeatedly to "wipe" the contacts. For stubborn contacts, the surface can be burnished with a cotton swab sprayed with TF. TF is sprayed directly into a pot (cleaning can only be accomplished with open-frame pots) and the pot is "worked", i.e., turned through several complete cycles of rotation. Note: some TF-based products also contain a silicone lubricant. Generally these products should only be used after all else has failed, since their overspray tends to make a terrific mess, and most electromechanical components are intended to be self-lubricating, anyway.

"...never overlook the obvious"

To the uninitiated, case cleaning usually amounts to running a vacuum over the outside of the gear, or, worse yet, spraying the wood parts and front panel with Lemon Pledge. The stiff bristles of a vacuum cleaner brush can scratch keytops, LED/LCD display windows, and other plastic or acrylic components, and should be used with care. Vacuum systems are also a source of static electricity, so vacuuming circuit boards is a no-no, lest you zap some $40 CMOS CPU that takes eight months to get from Japan. To dust off delicate components, use some compressed air in a can (Tech Spray No. 1668-15S, or equivalent). Wood parts should be treated periodically with lemon oil or tung oil on a clean cloth (Formby's, or similar); follow the directions and use it sparingly. Never use furniture polish or wax on your gear.

To properly clean front panels, you'll have to remove any knobs and switchcaps, then dismount the circuit boards from behind the panel; otherwise you're limited to just cleaning "around" the pots and switches...a halfway job, at best. Use clean water and a soft, clean cloth—*never* use Windex, Fantastik, or other commercial cleaners or solvents. For major grime, use a mild solution of Ivory dishwashing detergent, or some rubbing alcohol (isopropyl alcohol and water). *Take extreme care not to let any liquid get into the gear.* All-plastic knobs (no metal inserts) and switchcaps—keyboard keycaps, too—can simply be washed in the sink. Tolex and similar covering materials should be cleaned with a mild soap solution, then treated with "Armor-All" or a similar vinyl protectant.

Changing ROMs

With so many sampled sounds and software updates available in EPROMs/PROMs/ROMs, it's important to know the correct way to change them. This is a skill so basic that paying a technician to do it seems almost un-American—but you do have to get it right, or you'll destroy the ROM.

ROMS are big multi-pin ICs that (for the ones you can change, anyway) fit in receptacles called *sockets*. Basically, you have to remove the old one and insert the new one without bending or breaking any pins, and you must also make sure that the new one *isn't installed backwards*. Before removing the old ROM, make a diagram showing the correct direction of installation; most ROMS have an orientation notch on one side, and pin 1 is often indicated by an indentation or a dot. The socket should have some kind of indication for pin 1 as well.

Use a miniature flat-blade screwdriver and pry up each corner *gently*, a little bit at a time, until the old ROM is free. Work slowly and carefully; if you pry up too much on one corner, you'll bend the pins on the opposite side. Don't try to pull the ROM out with your fingers—the socket will "let go" all at once, which means you'll probably bend a lot of pins (maybe even break them off) and probably stick some in your finger. Ouch.

Before you can install the new ROM, you might need to reform the leads slightly; take a pair of needlenose pliers and gently bend each of the two rows of pins toward the centerline of the IC, until the pin spacing matches that of the socket. Then, carefully place the ROM in the *proper orientation* in the socket making sure that all the pins line up with their respective socket holes. Gently push it into place until it "seats," but check at the halfway point to make sure that all pins are correctly seated, and that no pin is bent underneath the IC. Take care not to apply excess or off-center force, or you might fold-under and/or break a pin. Double-check the ROM for correct installation before power-up.

Tuning

Analog and hybrid synths, and other electronic instruments, require periodic tuning. Generally this involves such parameters as VCO range, scale, and high-frequency track adjustments, and requires the aid of a test instrument called a *strobe tuner*. Renting a strobe tuner is cheaper than paying a technician to tune your gear, and you may do a better job, since you have to listen to the results! Strobe tuners are easy to use (make sure you get a manual with the tuner, though), and your equipment service manual will detail tuning procedures. Analog synths often provide access to tuning adjustments without opening the case, and some hybrid synths (e.g., Memorymoog) even provide software tuning routines, so you don't need a strobe tuner. On any tuning job, be sure to let the equipment warm up for at least 30 minutes beforehand, and avoid tuning in hot, cold, humid, or drafty environments. Note: tuning acoustic and electric pianos requires special training, and is not recommended as a DIY service project.

CUSTOM MODS

Installing custom mods in your gear is an excellent way to hone your service skills to a finer edge. Even if you're not to the point of designing your own modifications, there are lots of articles right here in EM that will guide you step-by-step through adding power features to your gear. Mod articles are generally written for people like you by people like you, and provide simple and well-documented instructions to let you get into your gear without getting into trouble.

USE YOUR NOSE

And now we come to the philosophical part, the mystic quest for the proverbial *broken wire* that everyone is looking for when their equipment konks out. . . does it exist? Be *aware*: use all your senses; never overlook the obvious, *including* broken wires, loose connections, bent IC pins, cold solder joints, dead batteries, blown fuses, miss set controls, or operator error. Even if you're not a pro, you can follow a service manual and safely get into your gear. Perform the diagnostic test procedures in the manual. Look carefully for damaged or discolored components, even traces of smoke; and, yes, *use your nose*—fried components have a definite odor. If you find a suspect component, especially an inexpensive part that's socketed, or otherwise easy to remove, then replace it. It may have been the hapless victim of a power surge. If that's all that's wrong with your gear, then you've learned something, and saved an expensive repair bill. . . and that's what DIY service is about. EM

A/D converter: A circuit that converts analog signals into digital data that corresponds to the analog signal's level.

capacitively-coupled: Also called AC-coupled—an interconnection technique between stages in an electronic circuit that blocks direct current (i.e. a constant current flow) but allows alternating current (AC, such as an audio signal) to pass. A series capacitor, which blocks DC but not AC, is ideal for interconnecting stages in this manner and essentially acts like a high pass filter.

chip enable: A terminal found on logic chips that selects whether a chip is active or not.

clipper: A circuit that prevents a voltage from exceeding a particular level by removing (clipping) anything above that level.

CMOS: An acronym for Complementary Metal-Oxide Semiconductor, a type of digital integrated circuit known for extremely low power consumption and the ability to run from a wide variety of power supply voltages.

control voltage: A DC voltage produced by a controller (keyboard, modulation wheel, envelope generator, etc.) and used to modify or control some aspect of a synthesizer's or signal processor's sound (pitch, timbre, level, etc.).

detent: A spot along a control's rotation that tends to "grab" the control, and requires additional force to move the control off the detent. Example: most synthesizer pitch bend wheels have a center detent so that it is easy to locate the zero pitch bend spot.

Digi-Key Corp., P. O. Box 677, Thief River Falls, MN 56701.

E!: A retrofit board for the DX7 that enhances the instrument's performance and adds several extra features such as increased memory. (For more information see the review in the May, 1986 issue of EM.)

Heathkit, Benton Harbor, MI 49022

Magic Music Machine, 1207 Howard Street, San Francisco, CA 94103

heat sink: A passive metal part that conducts heat away from heat-generating parts (i.e. power transistors). It typically contains a finned surface, thus providing a large surface area from which heat can dissipate.

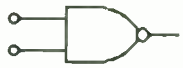
MIDI clock: A piece of MIDI data, sent out 24 times each quarter note, that serves as a master rhythmic timing reference in a MIDI system.

Music Dealer Service, 4700 West Fullerton, Chicago, IL 60639

NAND gate: A digital circuit whose output is a function of the input stated by the following logic table:

In 1	In 2	OUT
0	0	1
0	1	1
1	0	1
1	1	0

Two Input NAND Gate



PAiA Electronics, 1020 W. Wilshire Blvd., Oklahoma City, OK 73116

perf board: A flat, thin piece of phenolic or epoxy-glass material pre-punched with holes to allow for the insertion of electronic components. Once inserted, the components are wired together.

PROM: Acronym for Programmable Read Only Memory. This is an IC similar to ROM (Read Only Memory), however, the PROM may be reprogrammed. Example: some instruments can be updated by removing a PROM and sending it to the manufacturer, who loads new data into the PROM.

QWERTY keyboard: The standard keyboard used with most typewriters and computer terminals, so called because the first six letter keys spell out QWERTY.

RAM: Acronym for Random Access Memory. Data in a RAM IC can be recalled at will, and new data can be written in over old data. Most RAM is volatile, and loses its data when power is interrupted.

ROM: Acronym for Read Only Memory; an unchangeable program permanently written into an IC's memory. Data stored in a ROM IC can only be recalled; unlike RAM, new data cannot be written in over old data.

slew rate (of an op amp): An op amp's output cannot change instantaneously from no output to full output. Slew rate specifies the rate of voltage change per microsecond. Example: Applying an instantaneous voltage change of 0 to 5 Volts at the input of an op amp with a slew rate of 0.5 Volts/microsecond means that it will take ten microseconds for the op amp output to go from 0 to 5 Volts.

shelf life: How long a battery will retain its full charge under no-load conditions (i.e. "sitting on a shelf").

switching jack: Also called Closed-Circuit jack. A mono switching jack's third terminal connects to the "hot" terminal with no plug inserted into the jack, and disconnects from the "hot" terminal when a plug is inserted. A stereo jack may include a switching terminal for both "hot" terminals.

SMPTE clock: A master system timing reference extracted from SMPTE time code.

SMPTE time code: Originally developed by NASA as a means of accurately logging data and later adopted by the Society of Motion Picture and Television Engineers, SMPTE time code labels each frame of a videotape by recording a unique piece of digital data on that frame. For American (NTSC standard) television and video, each second of SMPTE time code is divided into 30 frames (the standard number of frames that pass by in one second of video; the standard frame rate for films is 24 frames per second, and for European television and video, 25 frames per second. Each frame is further divided into 80 subframes, with each subframe being 0.417 milliseconds long. A typical time code location might be 00:10:08:29:(76), which you would read as 00 hours, 10 minutes, 8 seconds, 29 frames, and 76 sub-frames into the tape. The SMPTE time code emanating from a SMPTE generator can be recorded on tape and played back into a SMPTE time code reader, which precisely identifies where you are on the tape. This data not only helps synchronize audio to video but can also synchronize two or more audio recorders together.

synchro-sonic: A musical technique where sound parameters relating to time (LFOs, envelope generators, delay time, etc.) may be synchronized to tempo. This produces a rhythmically cohesive effect that is only possible in electronic music systems.

tack-solder: The process of making a solder connection by pre-tinning the connections to be soldered, then simply heating these connections and adding little, if any, additional solder.

TO-42 terminals: Small terminals suitable for inserting perf board.

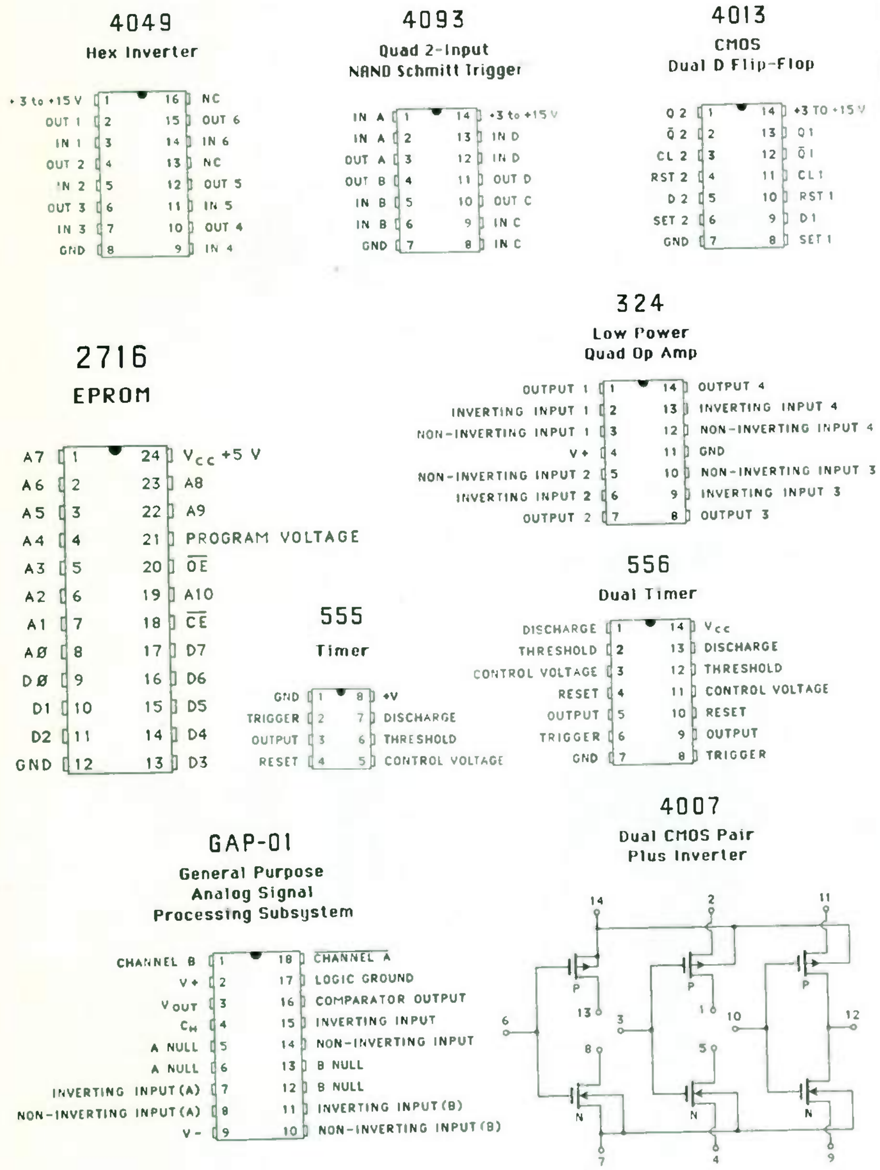
varistor: A component whose resistance depends on the applied voltage. It is principally used to "absorb" high-level voltage transients by offering a low-resistance path to these voltage spikes.

voltage divider: A passive, resistor-based circuit that accepts a full-level voltage at its input and produces an attenuated version of that voltage at its output.

voltage regulator: A device that produces a stable output voltage from a varying input voltage; the input voltage must typically be greater than the desired output voltage.

Xerbitron, P.O. Box 70055, San Diego, CA 92107

Zener diode: A diode with voltage regulating properties.



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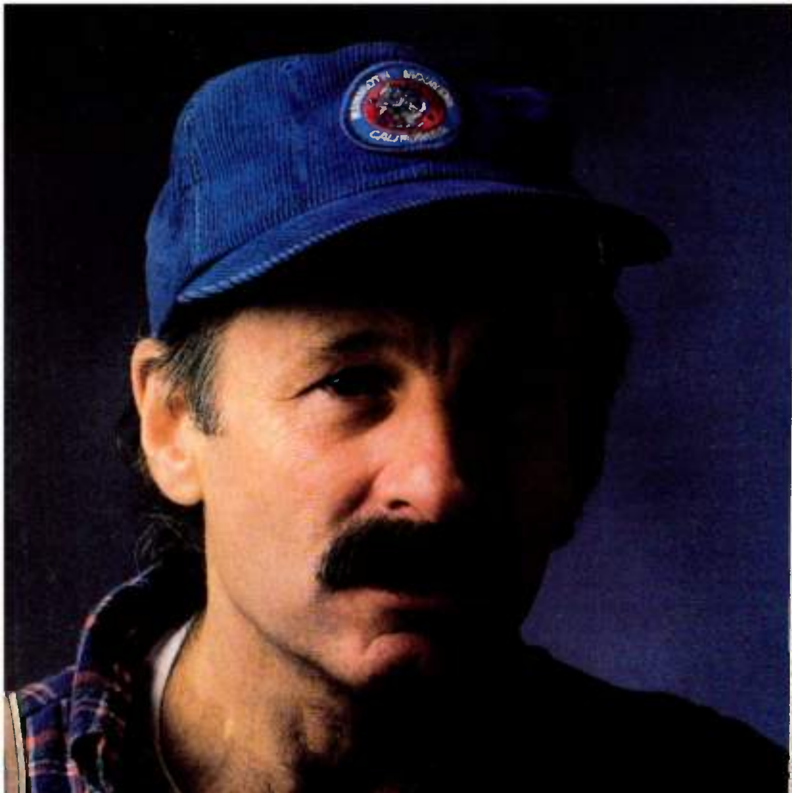
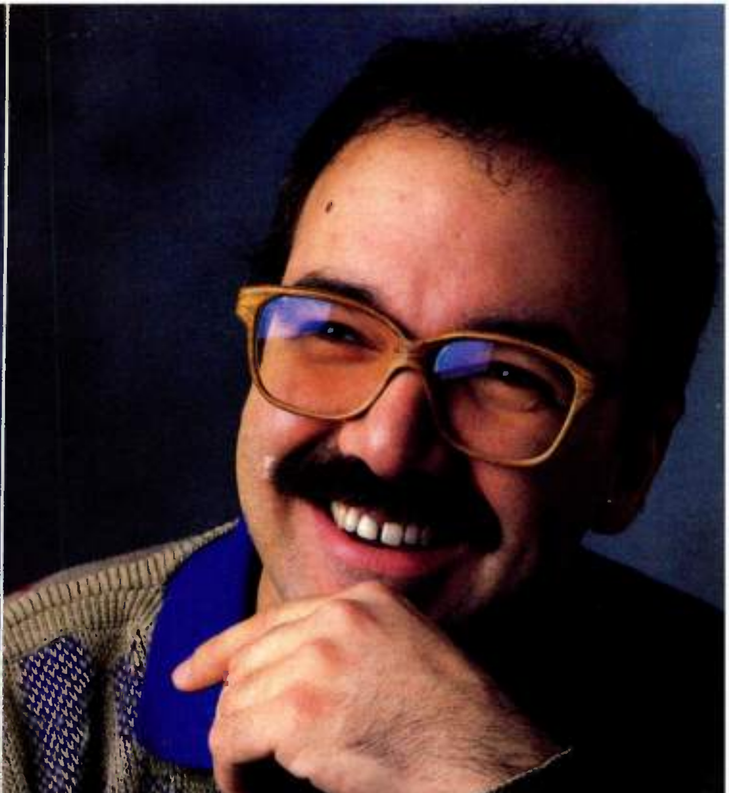


JOE ZAWINUL
keyboardist/composer

"MIDI is a powerful weapon. Now any fool can pull the trigger. In the hands of a real musician, it's the most valuable tool since the invention of synthesizers.

I'm using Korg's new Digital Voice Processor for several tunes on the new Weather Report LP. A great addition!"



PETER ERSKINE
drummer/composer

"I really like the sounds that Korg synths have. For any writing I do, they're the first instruments I turn to. I found a beautiful combination using the DW-8000 with the DW-6000 that became an essential part of "Something I Said," on the next Steps Ahead record. The tune wasn't really complete until I got that sound on there."

ALAN PASQUA
keyboardist/composer

"In L.A., everyone has their own MIDI setup, so you try to make your stuff one step ahead of everybody else. I get tonal textures and timbres on the DW-8000 that I can't get from other synths. I came across a sound on it that got really great response, people heard it and said 'What's *that?*' It's a very powerful tool."

ELLIOTT RANDALL
guitarist/composer

"I'm not a skilled keyboard player, so an effective sequencer that lets me edit is an integral part of my setup. Korg's MIDI stuff makes my work a lot easier. The SQD-1 is a breeze to work with. All the functions are right there in front of you."

What is MIDI really about?

Music. It's that simple. And that hard to engineer into a new instrument. Technology should open up the creative process for more musicians. If it becomes an end in itself, it's only a barrier.

Korg technology works the way you do. It brings you closer to your music. Isn't that the reason for any new instrument?

DVP-1 DIGITAL VOICE PROCESSOR: Multiply single note lines at mixdown with polyphonic pitch shifting. Create lush backing vocal textures with vocal waveform digital synthesis. Plus digital vocoding for unparalleled accuracy. All programmable and MIDI controllable.

SDD-2000 MIDI PROGRAMMABLE SAMPLING DELAY: Generate automated effects mixes with 64 MIDI controllable programs. Plus 4 seconds of mono sampling with MIDI controlled playback.

EX-8000 EXPANDER MODULE: 16 DWGS digital waveform oscillators plus fast analog programming without an external keyboard. Built-in 64 program digital delay. Key Window for splits and zones, six-part envelope generators. MIDI System Exclusive allows full access to program and parameter memory for Memory Expanders and Voice Editing software.

SQD-1 MIDI RECORDER: Controls your total MIDI recording/performance system. Familiar tape recorder-style operation, 16 channels. Real or Step time record, 15,000 note Quick Disk storage, full Edit functions. MIDI, Tape and drum sync connections.

MEX-8000 MEMORY EXPANDER: Store 256 sounds for the DW-8000 and DW-6000, Poly 800 and Poly 800 Mk II, EX-8000, EX-800 and DVP-1 in 4 libraries. Transparent MIDI Thru connection, 2 second data transfer. Plus Real time sequencing capability.

DW-8000 MASTER KEYBOARD: The widest range of velocity- and pressure-sensitive sounds. Send or receive all MIDI data including after touch, pitch bend, modulation, program select, damper and portamento. Variable MIDI Send and Receive channels.

VOICE EDITING SOFTWARE: Edit program parameters in real time with graphic envelope, filter and waveform display. Create, edit and display sequences in real or step time. Store and edit program and sequence libraries. Available for the Poly-800/EX-800, Poly-800 Mk II or DW-8000/EX-8000.

POLY-800 MK II: More professional than ever with a streamlined look and new, more colorful sounds. Integral 64 program digital delay, 1000 note sequencer, programmable EQ.

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TRUTH... OR CONSEQUENCES.

If you haven't heard JBL's new generation of Studio Monitors, you haven't heard the "truth" about your sound.

TRUTH: A lot of monitors "color" their sound. They don't deliver truly flat response. Their technology is full of compromises. Their components are from a variety of sources, and not designed to precisely integrate with each other.
CONSEQUENCES: Bad mixes. Re-mixes. Having to "trash" an entire session. Or worst of all, no mixes because clients simply don't come back.

TRUTH: JBL eliminates these consequences by achieving a new "truth" in sound: JBL's remarkable new 4400 Series. The design, size, and materials have been specifically tailored to each monitor's function. For example, the 2-way 4406 6" Monitor is ideally designed for console or close-in listening. While the 2-way 8" 4408 is ideal for broadcast applications. The 3-way 10" 4410 Monitor captures maximum spatial detail at greater listening distances. And the 3-way 12" 4412 Monitor is mounted with a tight-cluster arrangement for close-in monitoring.
CONSEQUENCES: "Universal" monitors, those not specifically designed for a precise application or environment, invariably compromise technology, with inferior sound the result.

TRUTH: JBL's 4400 Series Studio Monitors achieve a new "truth" in sound with an extended high frequency response that remains effortlessly smooth through the critical 3,000 to 20,000 Hz range. And even extends beyond audibility to 27 kHz, reducing phase shift within the audible band for a more open and natural sound. The 4400 Series' incomparable high end clarity is the result of JBL's use of pure titanium for its unique ribbed-dome tweeter and diamond surround, capable of withstanding forces surpassing a phenomenal 1000 G's.
CONSEQUENCES: When pushed hard, most tweeters simply fail. Transient detail blurs, and the material itself deforms and breaks down. Other materials can't take the stress, and crack under pressure.

TRUTH: The Frequency Dividing Network in each 4400 Series monitor allows optimum transitions between drivers in both amplitude and phase. The precisely calibrated reference controls let you adjust for personal preferences, room variations, and specific equalization.
CONSEQUENCES: When the interaction between drivers is not carefully orchestrated, the results can be edgy, indistinctive, or simply "false" sound.

TRUTH: All 4400 Studio Monitors feature JBL's exclusive Symmetrical Field Geometry magnetic structure, which dramatically reduces second harmonic distortion, and is key in producing the 4400's deep, powerful, clean bass.
CONSEQUENCES: Conventional magnetic structures utilize non-symmetrical magnetic fields, which add significantly to distortion due to a nonlinear pull on the voice coil.

TRUTH: 4400 Series monitors also feature special low diffraction grill frame designs, which reduce time delay distortion. Extra-large voice coils and ultra-rigid cast frames result in both mechanical and thermal stability under heavy professional use.
CONSEQUENCES: For reasons of economics, monitors will often use stamped rather than cast frames, resulting in both mechanical distortion and power compression.

TRUTH: The JBL 4400 Studio Monitor Series captures the full dynamic range, extended high frequency, and precise character of your sound as no other monitors in the business. Experience the 4400 Series Studio Monitors at your JBL dealer's today.
CONSEQUENCES: You'll never know the "truth" until you do.

UBL

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8500 Balboa Boulevard
Northridge, CA 91329