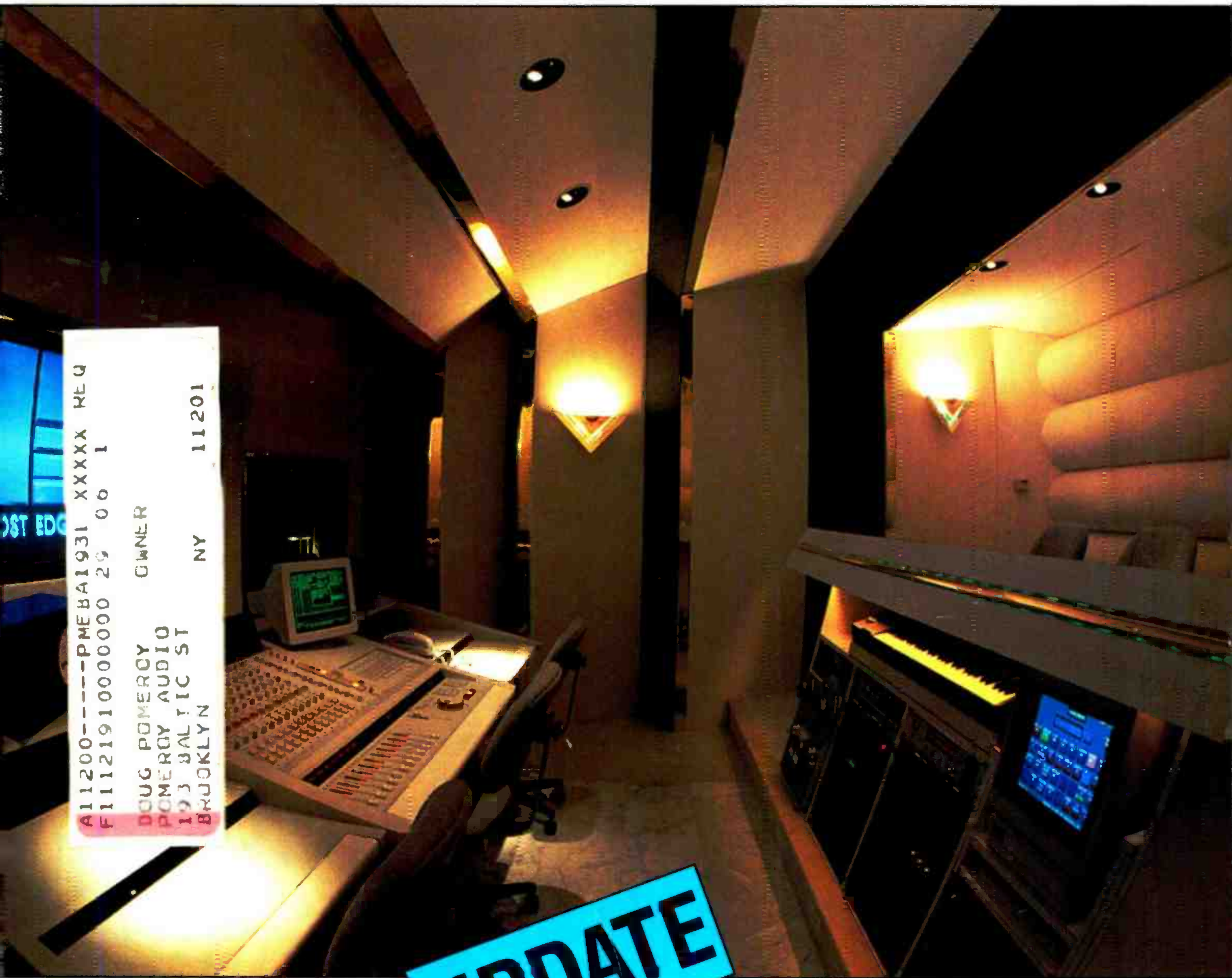
May 1992 Recording ■ Engineering ■ Production $4.00

R-E-P™

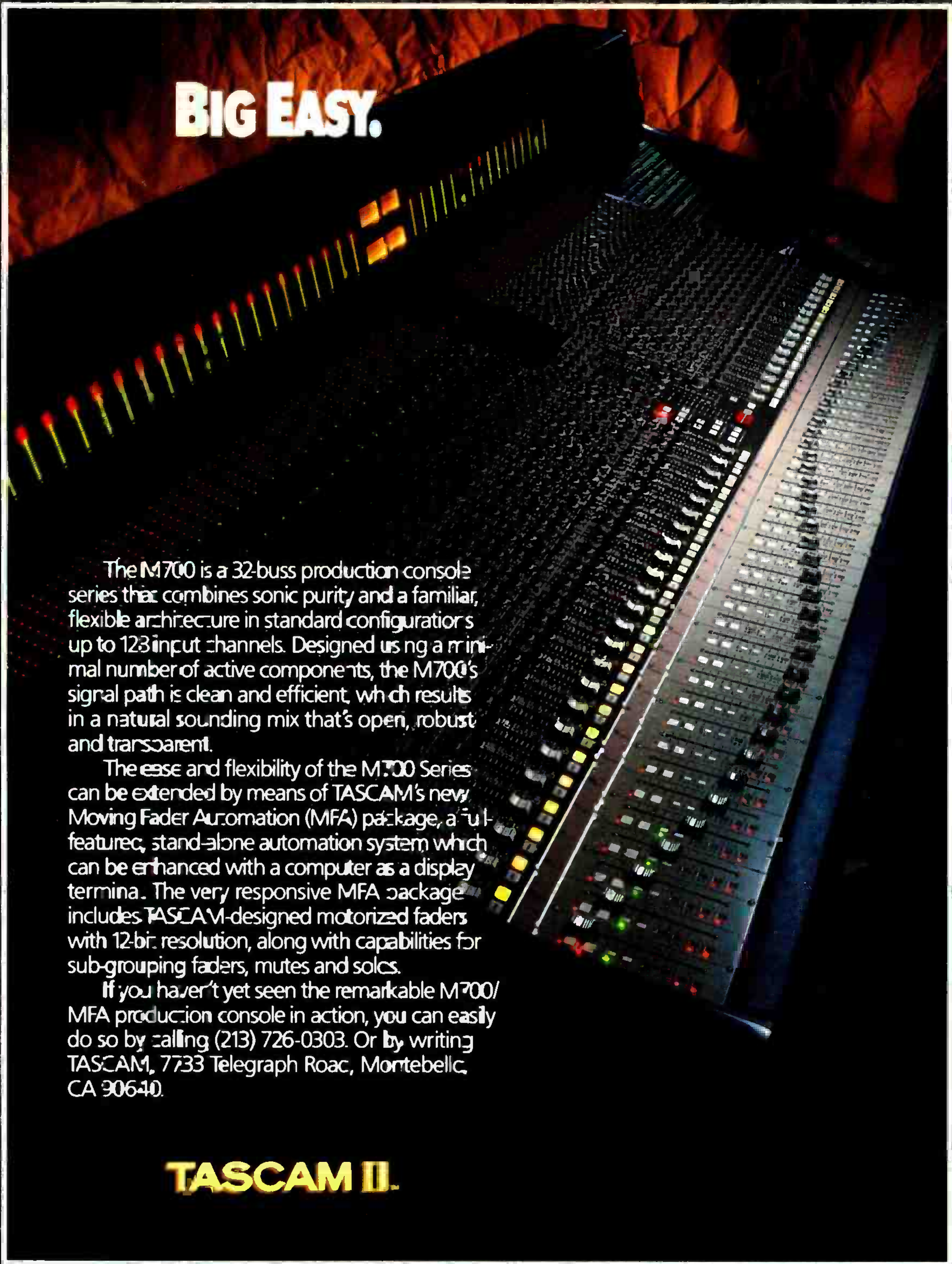
The Pro Audio Applications Magazine

DESKTOP PRODUCTION



UPDATE

An INTERTEC Publication


BIG EASY.
The M700 is a 32-buss production console series that combines sonic purity and a familiar, flexible architecture in standard configurations up to 128 input channels. Designed using a minimal number of active components, the M700's signal path is clean and efficient, which results in a natural sounding mix that's open, robust and transparent.
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If you haven't yet seen the remarkable M700/MFA production console in action, you can easily do so by calling (213) 726-0303. Or by writing TASCAM, 7733 Telegraph Road, Montebello, CA 90640.
TASCAM

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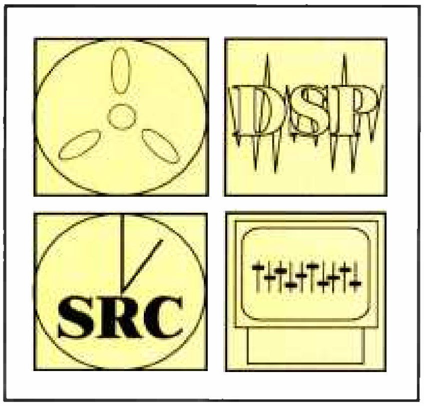

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On the Cover

Lexicon Opus console installed at Post Edge located in Hollywood, Florida.

Audio Production

Sound Reinforcement

Departments

R•E•P and **Live Sound! and Touring Technology** now have a presence on the **CompuServe Information Service** network! Submit article ideas, create Letters-to-the-Editor, download articles from our On-Line library or have a "live" conversation with our staff and contributors on the **R•E•P: On-Line Forum.** To receive a free **CompuServe** membership and sign-up kit, courtesy of **R•E•P,** dial **1-800-524-3388** in the **U.S.** or **Canada** and ask for representative **#232.** Once on the network, type **GO REPMAG** at any prompt to navigate directly to the **MIDI VENDER FORUM, SECTION 14.** Or reach the Editor's Mailbox at **75300,3141** by typing **GO MAIL** at any prompt. See you On-Line!

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The New Dyaxis. Now providing automated real-time EQ and Level Control for our 2 and 4 output virtual multitrack systems. Other advanced features include support of a 1 gigabyte hard drive for up to 10 hours of stereo recording, new full magneto optical capability and full sample frequency conversion.

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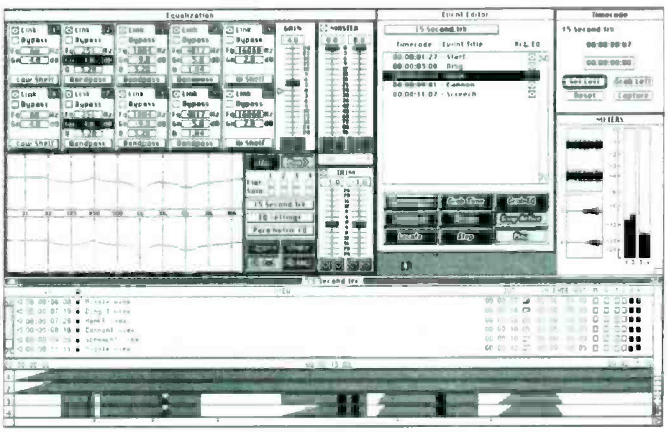

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STUDER

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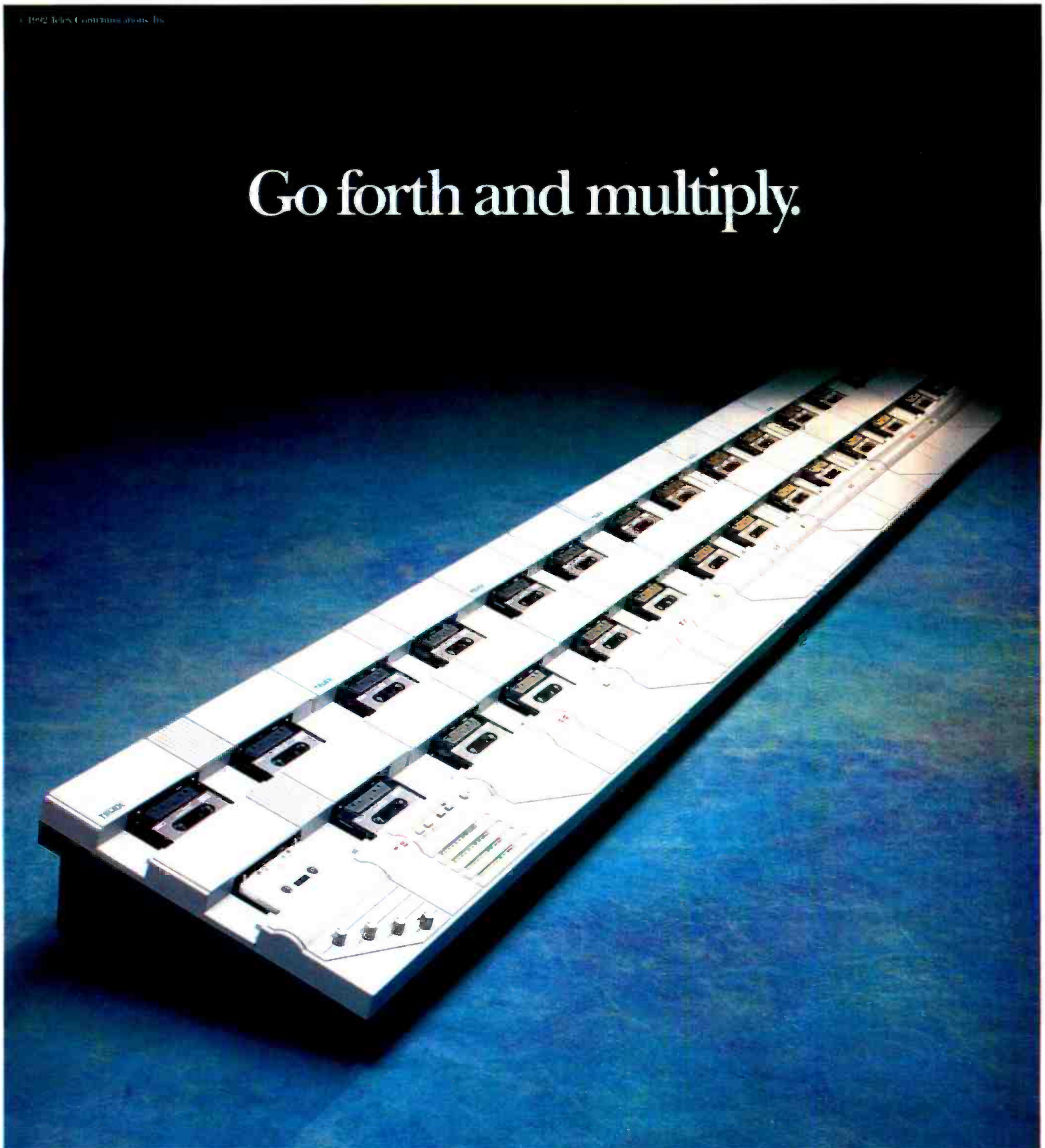
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R-E-P TM

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R•E•P is an applications-based publication targeted at professional individuals and companies active in the commercial business of studio and field recording, audio for video, live sound production and related fields. Editorial content includes descriptions and demonstrations of audio production techniques, new products, equipment application, maintenance and audio environment design.

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From the Top

Virtual Production Defined

One year ago last March, R•E•P magazine presented a very full and complete presentation on digital production workstations. We spent a fair amount of time pulling our hair out over what to call the creatures. Tapeless studios? Random access editors? Hard disk recorders? Audio workstations?

A year later it turns out that all these titles apply, or conversely, none of these titles apply. The reason is, a computer doesn't care what type of interface (meaning CRT screen, video monitor, loudspeaker, QWERTY keyboard, synthesizer, console top) the bits are going to end up addressing, and therefore what process it's actually working on. Bits are bits. Only the clock speeds change to confuse the innocent. The nature of the look and feel of the device, and therefore its function and application traits, are as much a marketing spec and a code cruncher's interpretation as anything.

What am I saying? Basically, all of these devices, assuming similarly able hardware support, are capable of having the identical interface, identical screen appearance, same feature set and function list. Will they? Of course not, primarily because it's a niche marketing world. Companies, if they're smart, will build devices that are job specific, matching function and price to capabilities.

Some people will only need to assemble, sequence and edit 2-track masters. For them, a QWERTY keyboard, mouse, Mac Classic, simple digital I/O, and a 660 meg drive, and they're in biz. Four kilobucks, computer included. Done deal.

The next guy might be doing lots of recording, using a tonnage of tracks, with no real editing, but lots of slip-to-sync assembly prior to linear tape layback. For him, different platform, different spec, different system. More bucks, too.

An audio post house may need the mondo profundo version, due to the ever shifting needs of their revolving client and producer list. Every single date may introduce a whole new shopping list of requirements, functions limited only to the imagination of the operators and the flexibility of the software/hardware configuration. Many man-hours and lots of proprietary hardware research, development and design goes into the multi-processor boxes, reflected in the cost of the systems.

But if you think about it, how much does a console, sync system, digital multitrack recorder and computer controller, all with the same extremely high audio performance specs, cost in entirety? More, I'd wager. And you'd never get the flexibility of features.

Which brings up a very interesting point. When are the manufacturers going to start telling the end users (who, due to the economy, conflicting storage and file sharing protocols, high prices, etc., are *not* rushing to buy systems by the tens of thousands) exactly what their systems do and don't do, *by design*. Not to harangue, but there's a little marketing maturity that needs to enter into the picture here.

Not every production situation requires sample to sample SMPTE sync lock, or full resolve. Radio commercials, for example, might not need sync at all, or a short video stinger might only require trigger start sync lock. Should the customer pay for it if he or she doesn't need it? Yet almost every workstation manufacturer proudly proclaims full sync capabilities, preferring not to discuss the finer details of trigger start (freewheel) versus full resolve, or chase master/slave capabilities. The ideal seems simple to me: provide what the customer needs, at multiple levels, either as options or as different levels of packaged systems.

The problem of windows-behind-windows has been addressed on many recent version revs by simplified direct access tricks, whether via power-user keys or more intelligent interfaces (I'm thinking of the Studer Dyaxis Lite controller and the J.L. Cooper interfaces, which although not ideal, are a big step in the right direction).

Several companies have taken up the Lexicon Opus and DAR Soundstation approach by developing full analog-type console work surfaces. Face it, humans are tactile as well as visual, and we like to punch buttons and slide faders. Mousing a fader on a vertical screen doesn't provide the resolution and feel of a finger on the trigger.

In the near future, expect to see a number of well thought out, nicely realized console surfaces as front ends to the higher-end production systems. The presence of these interfaces alone will go a long way to making disk-based production the standard recording and editing medium in the land of audio. ■

Mike Joseph
Editor

Letters

Retro Info.

From: Nigel Toates, national field service manager, Siemens Audio, Inc., Bethel, CT.

We at Siemens Audio read with interest the very informative article, "Neve Retro" by John La Grou in your February 1992 issue. We do, however, take issue with one point in the article. The writer says that the Neve West Coast, East Coast and U.K. offices could not help him locate a copy of a schematic for a Neve 1063 module. ("Nobody had ever heard of it," he writes.)

We can't understand why our regional offices (or the U.K., for that matter) would not have referred the writer's request to the corporate office here in Bethel, CT. We get four or five calls each week for similar information.

However, enclosed you will find a copy of the schematic for a 1063 module, first built in 1969. Schematics are also available from this office for all vintage Neve consoles.

Editor: R•E•P highly recommends that anyone modifying or refurbishing Neve consoles or modules make use of Siemen's excellent service provisions. They do, indeed, have most schematics and much in the way of components in stock. Some custom modules may not be supported by full documentation, as might be expected. The contact information is:

Siemens Audio, Inc.
Neve and AMS Professional Audio Products
7 Parklawn Drive
Bethel, CT 06801
Tel: 203-744-6230
Fax: 203-792-7863

Turn It On or Leave It Off?

From: Walter E. Sear, Sear Sound, NY.

That ugly and never-ending question has again reared its head here at Sear Sound, and I was wondering if anyone has done the definitive research on that nasty question: *Should we leave the studio equipment turned on all of the time, or should we turn it off at the end of the work day?*

I was wondering whether some of your readers might have some information about this, rather than the endless conjecture and hear-say that I have heard for the past 30 years.

We at Sear Sound have been turning everything off with seemingly no adverse effects. Back in the days of vacuum tubes (which we are still living with), the thermal shock to the filaments as they were turned on was a good reason for leaving everything on. On our vacuum tube console, we have a Variac on the filament supply. We bring up the filaments in the morning and turn them down at night. Being all Nuvistor and vacuum tube circuitry, I have never had to replace a tube because the filament burned out. In fact, I would estimate that 85% of the original tubes are still in place and working after 21 years.

Our Neve console and Studer machines are turned on and off as needed, and I would say that the two transistors which have popped over three years were the normal defect rate in a board full of discrete transistors. Because we live in an age of regulated power supplies, we shouldn't be too concerned about a voltage surge past the power supply. The supplies have never been down for maintenance.

In this year where soaring costs are confronted by falling demand, is it wise to save a few bucks on the electric bill? We have found that the savings, including air conditioning, are significant, especially in New York where we pay the highest rates in the country.

If some of your readers could let me know their opinion, it would be helpful. Incidentally, we turn things on well in advance of sessions so that everything is at operating temperature when it's time to hit the red button.

Hot Juice Response

From: Al D. Forbes, Alpha Sound and Light, Charlotte, NC.

I am writing in response to Robert Wilber's letter in the February 1992 R•E•P issue titled "Hot Juice," regarding balanced power distribution. He claims to be a licensed electrical inspector, but it seems he has overlooked a very important point in the concept of balanced power.

Balanced power is nothing new. All household clothes dryers and window air conditioners which are 240V use a balanced power source, the only difference being the voltage — 240V instead of 120V. In these types of appliances, all motors, heating elements, etc., are 240V devices, connected between two 120V opposing phase hots to yield 240V. The third wire is an Earth Ground, connected to the chassis for safety. Its function is to trip the breaker if either hot wire shorts to chassis, which would be hazardous if the chassis were not grounded.

The balanced power scheme that Martin Glasband proposes in his September 1991 R•E•P article is safe so long as what is normally the neutral (current carrying conductor) and ground conductor are not interchanged. In Mr. Wilber's example of an ordinary desk lamp (or any other 120V unit with a 3-wire grounded line cord), with balanced power the chassis will be grounded as always, but the difference is that 50V would be on each Current Carrying conductor instead of a 120V hot and a 0V neutral.

While it is true that in a standard 120V circuit, neutral and ground are bonded together at the source (breaker panels and service entrance), this is *never* done at the recepticle or inside the appliance. Neutrals and grounds are always kept separate after leaving the panel. This is perhaps the source of Mr. Wilber's confusion.

Balanced power does require a double-pole, common trip breaker as well as a solid chassis ground for safety, but I see no reason why standard NEMA-configured wiring devices cannot be safely used with balanced distribution, so long as neither current-carrying conductors are grounded.

We are considering converting to balanced power distribution for the FOH equipment in our concert sound system, but the balancing transformers required for amp racks would be cost prohibitive.

More, More, More

From: Steve Ett, New York.

Just a note to let you know your magazine looks great. Personally I read it cover to cover. You folks are working real hard — keep up the good work. I do have a couple of suggestions:

1) Print more letters to the Editor, to hear the voice of the people.

2) More Fresh Tracks reviews, with more comments from participating engineers.

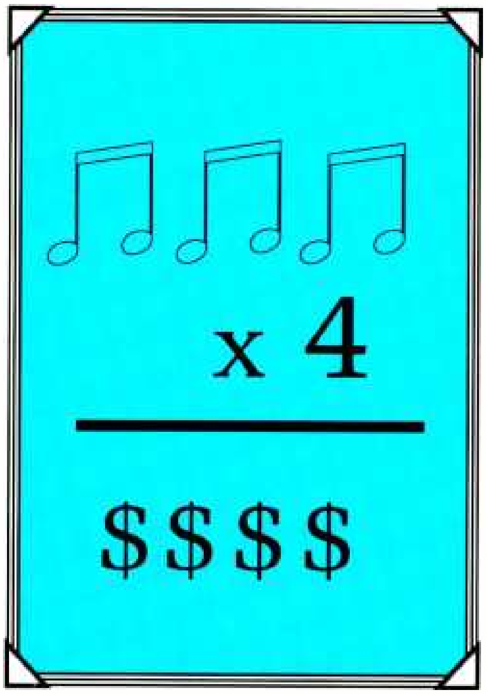
Keep up the good work! ■

Send letters to R•E•P Box 12901, Overland Park, KS 66282-2901; or fax 913-967-1905. Letters must be signed and may be edited for length and clarity.

Now moving to digital is just sound economics.

SONY

WHAT THE FEDS THINK



The Copyright Office has released Copyright Implications of Digital Audio Transmission Services. This 2-volume study assesses the impact of the fledgling digital audio transmission (DAT) industry, provides an explanation of what DAT is, and, based on studies of analog taping and other pertinent materials, makes predictions about digital taping and the potential effect on U.S. copyright owners.

At the earn-a-living level, the report explores avenues to compensate copyright owners for royalties lost on their works. It also discusses copyright protection of sound recordings in selected foreign countries and legislative policy implications regarding digital audio transmission services. The report includes Copyright Office recommendations for the future protection and compensation of copyright owners and their works in the U.S.

In a nutshell, the report concludes that it is currently impossible to make an accurate long-range assessment regarding the potential harm of the burgeoning digital transmission industry.

This 793-page report, stock number 030-002-00172-1, is available for $34 per set. To order, make prepayment to Superintendent of Documents, P.O. Box 371954, Pittsburgh, PA 15250-7954 or visit your nearest U.S. Government bookstore.

WHAT THE FEDS DO

Congress continues to move on the Audio Home Recording Act of 1991. This bill will tax digital audio recorders and digital audio media, create a Federal bureaucracy to distribute tax revenue to foreign-owned record companies, and restrict digital audio decks so digital-to-digital copies can't be made.

The bill does not distinguish between digital media used for computer data storage and digital media used for audio recording. Just as consumer DAT and 8mm video tape have become the standard high-capacity backup media for workstations/minis, it is likely that writable optical digital audio disks will replace today's magnetic floppy disks. Sony and Matsushita own two of the largest record companies in the U.S. and would net the most revenue from this legislation. In theory, every time American users of IBM PCs make a backup (if this bill becomes law), they will be paying a tax to overseas corporations.

The effective end result will be that:

1) Performers would be forced to pay a tape tax just to be able to record their own performances.

2) Citizens would have to pay $5,000 for a "professional" digital audio recorder in order to use the equipment for editing.

3) Any American company (large or small) wanting to compete with Japanese digital audio manufacturers will be forced to hire a lawyer to process the copious legal paperwork concerning tech specs and administrative decisions.

To date, C-SPAN and most newspapers haven't covered this issue. For those concerned, there is still time to make/send multiple calls and letters to Congressman and ignite some legislative thought. This bill is known as S-1623 in the Senate and HR-3204 in the House.

As R•E•P goes to press, this legislation is on the Senate schedule and could be immediately called up for a vote by Senate Majority Leader Mitchell. In the House, the bill is still bouncing through the committee and sub-committee stage.

This one is important. Don't drop the ball! You can communicate with your elected representative at these addresses:

Your Senator
The Capitol
Washington, DC 20510

Your Representative
The Capitol
Washington, DC 20515

PEOPLE

Symetrix has added two engineers **Christopher Hoskin** and **Allen Goldstein**, to coordinate their expanded operations into the digital marketplace ... Meyer Sound has announced the appointment of **Tom Divird** as president and CEO ... New England Digital recently named **George "Gus" E.R. Kinnear II** as vice chairman and **Brian N. Hamel** as vice-president, finance and chief financial officer ... **Jacquelynn Herbrock** has been appointed director of product development for Audio-Technica's professional and consumer divisions; in a related move, **Kenneth Satz** will assume Ms. Herbrock's previous duties as product manager; also at AT, **Darius Bossinas** has been promoted to manager of graphic services ... **Jeffrey J. Pallin** has been promoted to the position of manager of marketing and sales for TOA's commercial and PM&E Professional Music and Entertainment product group. In other TOA activity, PM&E manager **Jonathan Parker** has relocated to Atlanta to serve as general manager of TOA's new East Coast office ... **Gene Czerwinski,** CEO of the MAMA Foundation (Musical Archives, Musical Archives) has announced the following appointments: **Ellen S. Cohn** to general manager, **Andrew Duncan** to chief software engineer, and **Doug Evans** to chief engineer ... Veteran composer/producer **Kelley Bryarly** has joined the staff of the Music Annex Recording Studio in Menlo Park, CA ... Samson Technologies has appointed **Ron Milione** to a design and research and development position within their engineering group ... The Toy Specialists added **John Kayne** as its technical manager ... AudioTechniques has named **Davis Schecterson** to the position of product manager for Tubetech/ Lydkraft ... **Gina Romani** was promoted to production coordinator at Sound Techniques ... Audiomation Systems has promoted **Helen Stevenson** to sales office supervisor ... **Nu-Nu Whiting** and **Dutch Michaels** have joined forces to re-launch the 'Music Bank' backline and production hire services in South London ... City Spark Studios in Kansas City has added **Eric Elwell** as chief engineer and studio manager. ■

trend watch

Hey comrade, over here:

In response to widespread and glamorized rumors reported from the U.S. pro audio media in *Pro Sound News, Home and Studio Recording* and *Radio World* (though none in R•E•P), world-class transducer fabricator/ manufacturer Neumann recently issued a statement of clarification concerning the so-called "East German Neumann" condenser mics imported by Gotham, New York.

The communique reads: "Examination of the construction, circuits, components and quality of the GeFell microphones reveal that they bear very little resemblance to the Neumann line of microphones."

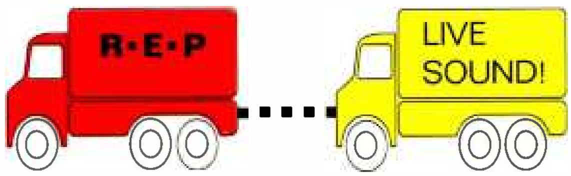
Based on this statement, we must assume that the addition of Mikrofon Bau GeFell to the global marketplace has resulted in a one-world merger of questionable communist technology, Yankee propaganda and multiple pallets of neo-Neumanns (a.k.a. Microtech GeFell) being off-loaded at U.S. ports.

As in most things audio-Caveat Emptor! ■

Good Vibrations: (For when I'm 64)

Australian scientists have developed a gizmo nicknamed the "tickle talker." Roughly the equivalent of braille for the deaf, the shirt-pocket-sized unit uses a microphone and a DSP chip to decode sounds that are hardest to interpret by lip-reading. The information is then transferred to the user via ring-like bands on the fingers of one hand. Differentiation between similar "looking" sounds, such as the consonants "b" and "p," is communicated by a localized tingling sensation. For many of us this could be just in the nick of time. ■

R•E•P and LIVE SOUND! have moved!



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9800 Metcalf
Overland Park, KS 66212-2215

Our new telephone numbers are:
Vox: (913) 967-1300
Fax: (913) 967-1905

"Call the thing 'digital' and some fool will want to buy it."

—Comment from one senior broadcast engineer to a colleague getting a workstation demo in the audio pavillion at the 1992 NAB convention.

Random Access

STUDIO UPDATE

Name/Location	Details
NORTHEAST	
BearTracks Recording Studio/ Suffern, NY	Added a second Studer A820-24 multitrack tape recorder with built-in Dolby SR.
Marathon Recording/NY	Upgraded from their Neve V Series console to a VR60 with Flying Faders Automation.
SOUTHEAST	
AudioImage/Richmond, VA	Acquired the Otari ProDisk 464 digital audio recorder/editor.
In Your Ear Music And Recording Services/Richmond, VA	Installed a New England Digital Post-Pro direct-to-disk recorder in their recently opened studio B.
MIDWEST	
Advertising Giant Leo Burnett/Chicago	Took delivery of two Spectral 8-track digital audio editing systems and purchased a Digidesign Pro Tools 4-track digital audio editing system for Macintosh.
Skyview/Chicago	Took delivery of a Sound Tracs Megas 24-input console.
Editech Post Productions/ Omaha, NB	Installed a Yamaha DMR8 digital audio workstation.
Cedill Records/Chicago	Purchased the first Sonic Solutions Sonic Station in the Midwest.
SOUTHWEST	
Pedernales "Cut 'N Putt"/ Spicewood, TX	Has been completely renovated and re-equipped with the Tascam M-700 40-input multitrack recording console with Moving Fader Automation, plus the DA-800 DASH format digital and ATR-80 analog 24-track recorders.
NORTHERN CALIFORNIA	
Tambling Sound/San Jose	Latest upgrade includes the Dolby XP Frame that provides 24 tracks of Dolby SR.
SMP Audio Post/San Francisco	Expanded into the audio-for-video workstation market by adding the Digidesign Pro Tools and an 11' × 11' sound room.
NORTHWEST	
The Center/Spokane, WA	Installed a TAC Scorpion II mixing console with 32-channel, 16-track recording capability.
Great Britain	
Air Studios/Hampstead, North London	Added a Neve 72-channel VRP Legend console fitted with Flying Faders and Recall and ordered one of the first SSL G Series multi-format production systems.
Abbey Road Studios/London	Installed the CEDAR Restoration and Production System.
Japan	
Sound Design Studio/Tokyo	Took delivery of a 72-input Focusrite Studio console with full GML Moving Fader automation.
DESIGNERS	
Walters-Storyk Design Group/NY	Completed the design and construction of a pair of identical control/recording and vocal booths (for NED workstations) at the JSM Music complex in Manhattan.

NEWS NOTES

A **Dyaxis** users group, known as the Dyaxis SIG (special interest group) is now active on PAN (Performing Artists Network) to create an electronic clearing-house of Dyaxis user information. **Studer** has made special arrangements for normal sign-up fees to be waived when Dyaxis users join the network via Studer's PAN messenger software.

Manley has been given permission to produce the famous Pultec PEQ1 valve equalizer. The Pultec EQ has gained the reputation among many engineers as being the finest available, regardless of changes in technology. Manley has retained the original design plus added some extra frequencies.

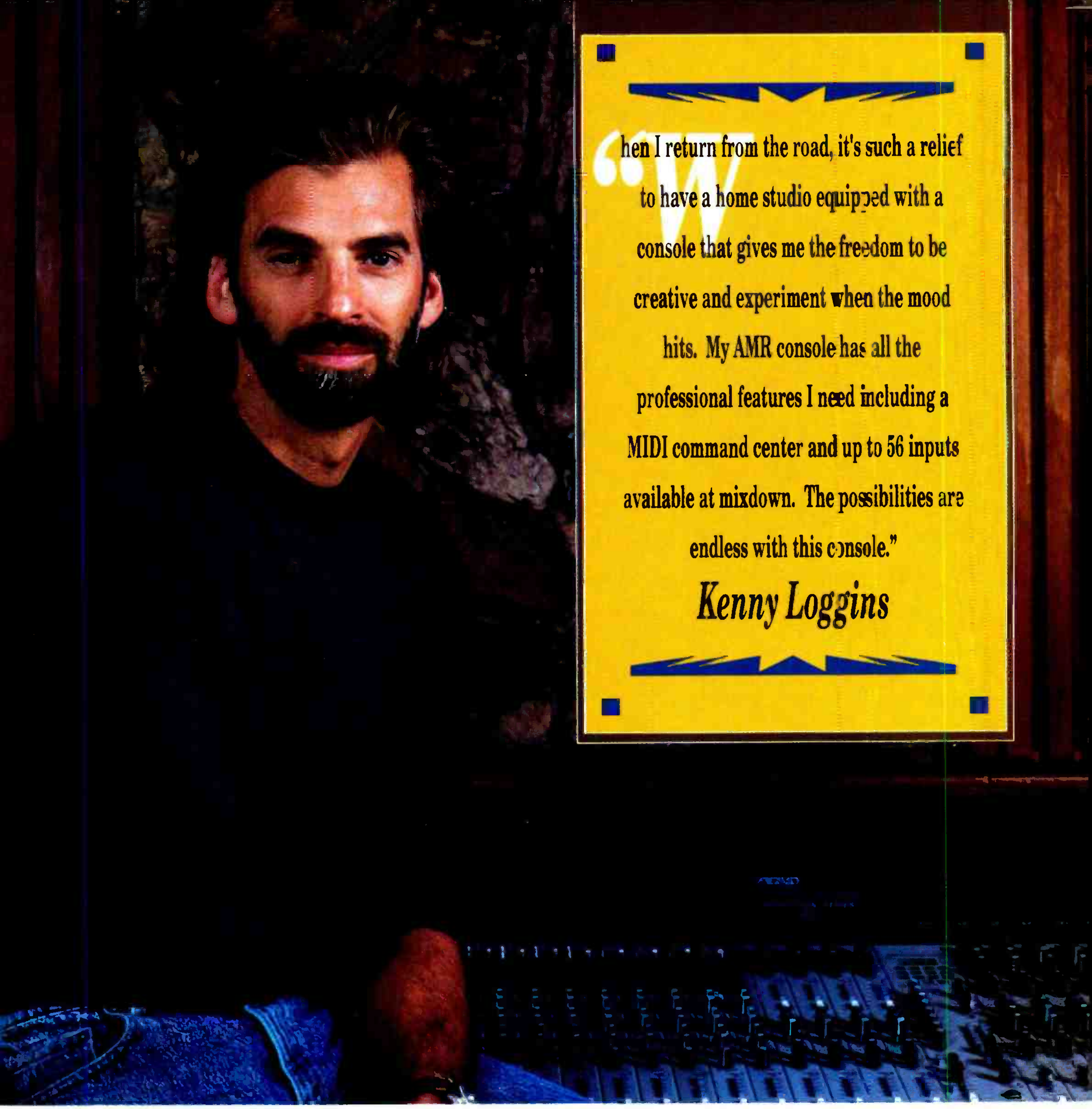
Dr. Peter D'Antonio, CEO of RPG Diffusors and Mr. Donald A. Hoffend Jr., CEO of Hoffend & Sons, Inc. announced that the two companies have allied to distribute the **D'Antonio Performance Signature Series** of products that employ the VAMPS (Variable Acoustics Modular Performance Shell) approach to acoustic control.

TOA, has opened its new Atlanta (East Coast) office. The new address is: TOA, Spalding Woods Corporate Office Park, 3850 Holcomb Bridge Road, Suite 145, Norcross, GA 30092.

Klay Anderson has announced the opening of **Klay-Anderson Audio, Inc.** a pro audio sales and service company in Salt Lake City, Utah.

PUBLICATIONS

Sypha's Second Edition of The Tapeless Directory outlines system specifications of 84 various tapeless audio systems. This publication provides a global review of this marketplace and defines basic terminology generic to the workstation environment. An allied study called "A review of the digital audio workstation market" was also recently released by Sypha. This "review" details existing market shares, trends and historical realities of workstation technology.
For more information about The Tapeless Directory contact: Sypha, 216A Gipsy Road, London SE27 9RB UK, 044 81 761 1042, Fax: 044 81 761 8279 ■



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Production Series™ Recording Consoles

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Circle (8) on Rapid Facts Card

Fresh Tracks

By Dan Levitin

Brahms: "Serenade No. 2 for Small Orchestra; Variations on a Theme by Haydn; Hungarian Dances London Symphony Orchestra/Michael Tilson Thomas"

Label: Sony
Produced by: David Mottley
Engineered by: Michael Sheady
Tape editor: Christian Meincken
Recorded at: Abbey Road Studio No. 1 (London)

Comments: David Mottley and Michael Sheady are one of our favorite teams for recording classical music. [See R•E•P November, 1990 for a review of their Zubin Mehta Bartók recording.] The pair create quiet, well-balanced recordings with a strong sense of both imaging and cohesion, and a silky top end with no harsh digital artifacts.

Of special interest: Listen to the oboe in "Serenade No. 2" as it trades the solo position with the clarinet. The two instruments flow into one another seamlessly. The French horns in the Adagio non troppo of the same piece are also interesting as they climb out above the ensemble. Those who wonder why Abbey Road has its international reputation should hear this recording. The ambiences and frequency range match those of the best concert hall recording. ■

Enya: "Shepherd Moons"
Label: Reprise
Produced by: Nicky Ryan
Engineered by: Nicky Ryan
Mixed by: Gregg Jackman
Recorded at: Aigle Studios
Mixed at: Sarm West

Comments: Enya and Nicky Ryan push multitrack vocal layering beyond conventional limits of reasonableness and set down firmly – and gently – in a soundscape of their own making. Combining the power and beauty of 16th century religious choirs with a hint of modern rock pulses, Enya and Ryan have sculpted what may be the most intriguing and beautiful album of the year.

Of special interest: The recording is rich in reverbs and modern synthesized sounds with a perpetual ear to the classical past. On many cuts, the vocals are run through multiple-tapped delays feeding large hall reverbs, while being simultaneously flanged. These effects (along with the massive layering and doubling) in the hands of anyone else might have decayed into gimmickry. "Shepherd Moons" is as far from gimmicks as possible, using the studio to achieve a vision, not as a substitute for the lack of one. ■

The Cowboy Junkies: "Black Eyed Man"

Label: RCA
Produced by: Michael Timmins
Engineered by: Bob Doidge, John Oliveira
Mixed by: Tom Henderson
Recorded at: Grant Avenue Studios (Hamilton, Canada)
Mastered by: Peter J. Moore (MDI)

Comments: Michael Timmins, the primary songwriter, guitarist and producer, avoids using any flashy production tricks. Instead, he keeps things simple and understated, restraining himself to judicious use of reverb to sweeten the vocals. His production identity comes through most clearly in his interesting choices of instrumentation.

Of special interest: The instruments are well recorded and not exaggerated — no huge drum sounds here, nor any faux arena-sized guitars. The overall tone of the album is rather dark, but despite this lack of crispness, the instruments still retain a good degree of separation and individuality. Indeed, the dark atmosphere complements the overall melancholy, lonesome mood of the album and Margot Timmins' rich and angelic vocals. — Reviewed by Victor Barclay ■

Dan Levitin is R•E•P's music production editor and teaches music production at Stanford University.

The Sugarcubes:
"Stick Around For Joy"

Label: Elektra
Producer: Paul Fox
Engineer: Ed Thacker
Mixing engineers: Paul Fox, Ed Thacker, The Sugarcubes
Recorded at: Bearsville (New York) and Summa (Los Angeles)
Mastered by: Stephen Marcussen at Precision Lacquer

Comments: The 'cubes opt for more mainstream arrangements and slicker recording on this third release. Their songs still seem to come from somewhere in our musical future, similar to the experience one might have listening to the B-52s in 1945. These days Björk sounds less like she is climaxing when she sings than on 1988's on "Life's Too Good" but she is still electrifying.

Of special interest: Drum flanges open up "Lucky Night" spilling into spaghetti-western guitars. The drum ambiences, different from previous 'cubes recordings, don't sound ultra-modern, but maybe this is the sound that will be hip in 2015 as people try and "cop that 1988 'verb." On many tunes, two different snare drums are employed on alternate beats. On "Happy Nurse," Björk sounds not unlike a previous Thacker-mixed diva, Debora Iyall of Romeo Void. At times, such as on "I'm Hungry," the drum kit sounds disembodied from itself, with the ride cymbal sounding as if it's from an entirely different instrument than the kick and snare; not a complaint but a compliment. The overall other-wordliness is what you might expect from an engineer whose initials imply familiarity with visits from extra-terrestrials. ■

FOCUS:

ED THACKER, Engineer, "The Sugarcubes"

R•E•P: What's going on with the snare drums?
ET: Depending on the kind of beat Siggi was playing, we would use two different snares and tune them differently and he could develop rhythms that way. On two-thirds of the songs he is playing both snares at the same time. Our whole approach was to get a lot of contrast out of the different snares. On some songs we used a piccolo and a full size, on others they were both full size. On "Gold" and "Lucky Night" we took the snares off the snare drums entirely. People have remarked about the snare reverb and a lot of that was how I miked the large room at Bearsville. The room has a 40-foot ceiling and it's a huge room. We put 87s in the balcony in the back and a couple of 49s behind the drummer, about 10 feet back and maybe 10 feet up. Then I compressed the room mics with a Fairchild.

R•E•P: Did you record the guitars with chorus?
ET: Yes, that was all happening at the amp. The idea was to get as severe as possible when we used effects and to be as plain as possible when we didn't. Thor played through a stereo setup and had a Marshall on one side and a Peavey on the other. He had pretty strong ideas about the parts, and then Paul and I would have ideas, too. We were trying for big contrasts on this record. I miked his cabinets using a 57 and a tube 67 placed together on one speaker.

R•E•P: Do you have them come in at funny angles?
ET: Sometimes. It varies. I move the mic around looking for the right place. I don't waste a lot of time looking for every position on every speaker.

R•E•P: How long did the mixes take?
ET: Each mix took approximately a day. We like to start around lunchtime and then finish up each mix at lunchtime the next day. Because of the Fairchild compression in the room, I didn't need to use a lot of digital reverb. I used the Sony DRE-2000 for snare drum — that's my favorite these days — and for her vocal, depending on the sound, we'd use some 480 or Rev 7. I used the 480 more for the room and smaller reverbs and the Rev 7 for the larger verbs.

On the 480 I like the jazz hall and music clubs, and the large wood room. I modify them slightly, but generally, just the reverb time and pre-delay. On the Rev 7 I really like the vocal plate. A lot of the sounds on the record were created when we recorded them. There are the odd effects here and there and in most cases the mix is where we controlled the amount of them. But mostly it was on tape.

R•E•P: Björk sounds a little more controlled than usual.
ET: I think it's kind of the nature of their writing this time – everything is a little more accessible. It's still a little quirky, but the combination of their ideas and our ideas were to make it a little more accessible. The songs are a little more structured.

R•E•P: How did you record the bass drum?
ET: We kept both heads on it and built a kind of cage for the mic inside the bass drum — a 421 — and then I put a FET 47 outside. I usually like D112s as well inside a bass drum.

R•E•P: Did you record this 24-track?
ET: Well, it started out that way! But, you know how it goes. It's 48-track analog, using Studer A800s. We mixed at Summa to a great ATR at 30ips. Mixed to Scotch 996, no SR. ■

Fresh Tracks

Jody Watley:
"Affairs of the Heart"

Label: MCA
Producers: Andre Cymone, David Morales, Michael J. Powell, Jody Watley, Jon Nettlesbey and Terry Coffey
Engineered by: Bobby Brooks, Hugo Dwyer, John Poppo, DaveSussman, David Ward, Dan Marnien, Wolfgang Aichholz, Norman Whitfield
Mixed by: Alan Meyerson, Ken Kessie, David Morales, John Poppo, Barney Perkins
Recorded at: Soundcastle, Can-Am, The Enterprise, The Loft, Studio Masters, Encore (Los Angeles); Quad, Soundtrack, Electric Lady (New York); Vanguard (Oak Park, MI); Granny's House (Reno, NV)
Mastered by: Brian Gardner at Bernie Grundman Mastering (Los Angeles)

Comments: The flash in the production of these tracks is in how well thought-out the arrangements are. "Affairs of the Heart" is primarily machine tracks, with occasional subtle enhancements, such as Dean Parks' understated acoustic guitar on the title track. Watley's layered vocals are smooth and supple, washing over the rhythm tracks as leads and countermelody backgrounds.

Of special interest: The tight and intricate rhythm arrangements on the up-tempo tunes such as "Strange Way," "I Want You" and "Call On Me" would seem impossible to mix without resulting in a cacophonous clutter, but Meyerson and Kessie pull these off deftly. The vocals are on the up-front side of things, but do blend well. I listened to "I Want You" before leaving the house this morning, and I haven't been able to get the chorus out of my head since then. ■

FOCUS:

ALAN MEYERSON, KEN KESSIE, Mixing Engineers, Jody Watley

R•E•P: What's your approach to mixing, do you start with the rhythm tracks and work your way up?
KK: I definately start with the drums first, get the groove first.
AM: Most of the time I do that – it depends on the song. I did it for "I Want You" and "Affairs" because there was so much emphasis in the rhythmic structure of the song.

For "I Want You" I ended up using a bunch of delays on the bass to give it the groove that it has. There isn't a whole lot of reverb on those tracks. They were pretty busy and usually when I get tracks that are that filled up, I like to keep reverbs to a minimum. Sound Castle has a great live room, so when I mixed I pumped the drums back out into the room speakers and brought them back in for echo.

R•E•P: How much EQ-ing did you do at mix time?
AM: A whole lot. With those kind of tracks you have free reign to mess around with them as much as you care to, to create new stuff using EQ.

R•E•P: Do you add new parts when you're mixing?
AM: I do tend to run my Akai MPC60 live and add parts, but I didn't in this case. I did add some drum sounds using my Akai S1000 – I added a couple of kicks and on "I Want You" I added two snares and a clap and used them in different parts of the song. On both of the songs I did quite a bit of arranging in terms of dropping things out and doing breakdowns. Like the breakdown in "I Want You," I created that and added the horn thing by doing a sharp mute after the attack and adding a repeated 1/8 note delay.

Andre's tracks are so unique and so good to begin with that you try your hardest to make sure he gets his ideas across. He doesn't do things by the books, the chord changes and bass parts are very unusual and so you don't want to mess with them too much. You just try to highlight them and bring out the best of the songs. I mixed to 30ips at +5, using the new Ampex 499.

R•E•P: How did Andre decide who would mix which tunes?
KK: The ones I worked on are more the funkier, groove-type tunes. I think I've got a handle on street-type grooves and I get good vocals.

R•E•P: How did you get the vocals so clear?
KK: I hate to take credit for that – the stuff was actually recorded really well by Andre and his engineer Bobby Brooks. They cut Dolby SR so everything sounded really good off tape. You could put the keyboards low in the mix and they would cut through – everything had a lot of clarity. Believe it or not, the first 24 tracks are all percussion (on the first machine) and the second 24 were the vocals and everything else.

R•E•P: What kind of reverb did you use on the vocals?
KK: It was a combination of Rev 5 and SRV2000. I'm not actually sure if it was the SRV, but if I say that, maybe when I try to sell mine I'll get more for it!

R•E•P: On "Always" you pan so that everything is in its own space.
KK: Yeah, I like to move everything a little bit so they don't pile up. Sometimes I actually move things around in time, but I didn't on this album because there were so many percussion parts. I do that more on the new EnVogue record. Motion in rhythm parts I find sometimes is very destructive to a groove. Q-sound often has the same problem. A lot of people blow good grooves by auto panning key rhythm parts into outer space or Q-sounding them into left field. Kimsey said this: Sometimes when the Stones made stuff too stereo they would lose the groove. ■

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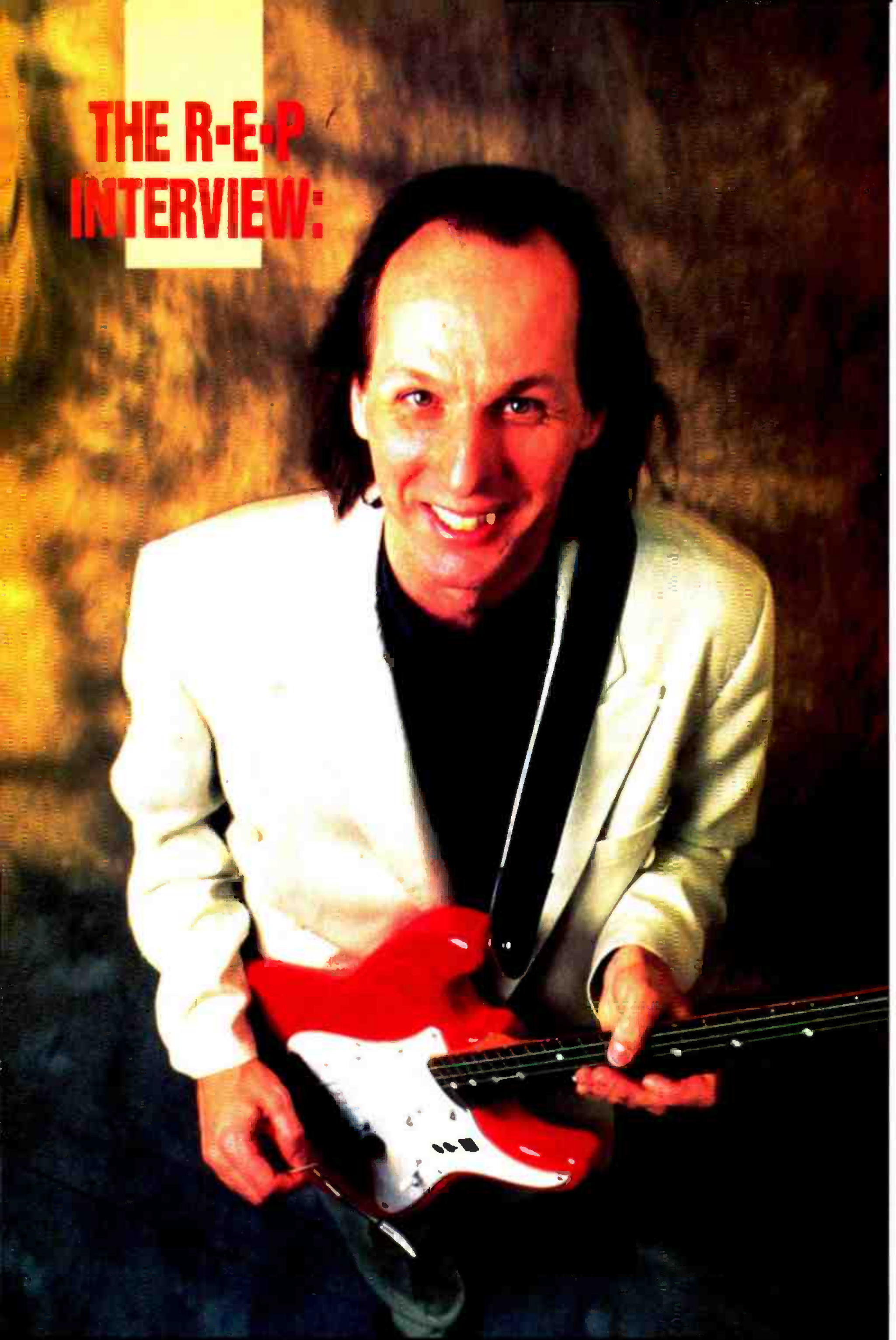

I'm learning how to use that old "less is more" theory.

ADRIAN BELEW

By Dan Levitin

Producer/guitarist Adrian Belew is perhaps best known for his work with David Bowie, King Crimson and The Bears. Liner notes detectives will also remember his distinctive sound on Paul Simon's "Graceland," including lead guitar on "You Can Call Me Al." Belew's third solo album for Atlantic, "Inner Revolution" (see sidebar) was released in March. Belew took time out to discuss his ideas about production, and to recall his studio influences. He also spills a hitherto well-guarded secret about how he sequences (chooses the order for) songs for an album.

R•E•P: What's that sound on the solo on "This is What I Believe In?"
AB: The GR50 Roland guitar synth. It's one of the sounds in there that I developed; I liken it to Stevie Wonder's harmonica playing. Throughout the verses I wrote a program that goes along with the guitar — it's on their sound that they call "Nylon Guitar." I wrote some harmonies so that when you're playing the verse figure, harmonies come out. It sounds somewhat like the interlocking guitar figure that Robert Fripp and I did in King Crimson. In the section you're asking about, I'm playing the same exact sound, but I'm not playing any notes, I'm just playing chica-chica-chica.

R•E•P: What about that weird sound opening up "Standing In The Shadows?"
AB: That's two guitars playing in stereo — the type of harmonic playing where you run your fingers up and down the string, but because there are two of them, the harmonics are not meant to match and it gives you kind of a Jews' Harp-type sound. Immediately following that there is a grinding sound — I put a drill against the guitar pickups and we used that to open the verses.

R•E•P: The guitars all sound like you're using solid state amplification.
AB: Right. I rarely run through amps when I'm recording live these days. I prefer most often to just go through my effects and then direct into the board and then effect it in different ways. Sometimes I use an amp, and when I do it's one of those little practice amps, a Roland MG-10. It's smaller than most ghetto blasters and can actually sound like a monstrous size amp. If there's a sound I want that's clean I might go direct into the board and use the studio outboard delays, but most often I go through whatever specific effects I want before the board.

Dan Levitin is R•E•P's music production editor and he teaches Music Technology at Stanford University.

R•E•P: The production reminded me of some of the old ELO records. I wondered if you were thinking, "Well, people are saying the '70s are coming back. Let me show 'em what I can do with '60s-style production."

AB: I think what happened for me was, from my point of view, I went back to trying to write good, solid simple songs and then orchestrate them in an interesting but not overdone manner. I liked Jeff Lynne's records too. I can't say they influenced me because they were second-generation George Martin records.

When I wrote "Big Blue Sun" I thought, this is a chance for me to write for a string quartet and put it in and that's something I've been dying to do. It's an actual string quartet made up of school teachers. I don't know how to actually write music, technically on paper, so I spent a lot of time with a friend where I would sing the part and he would write it down. He wrote some beautiful-looking computerized charts and they played them.

On the other hand, you have a song such as "Inner Revolution" that is just bass, guitar, drums and vocals. I wasn't thinking so much of the '70s the way you are, because *I* think of the '70s as disco.

R•E•P: Have you thought of producing other artists?

AB: I'm producing the premier rock band from Mexico, Caifanes [on BMG records - Ed.] It's a very pleasant surprise for me. They sell platinum in Mexico — they're the biggest rock band there. They're influenced by Frank Zappa, King Crimson and myself. So for five weeks we're at Royal. I've learned a whole lot and I think they've learned even more.

They're a very good band, very nice to work with, and it's teaching me a lot about being in the producer's seat. I feel I'm a natural for it 'cause I grew up listening to records as a producer would listen. I always tried to figure out why the parts were that way, what kind of reverb that was, why they used that instrument ...

I would try to figure out on guitar what the saxophone player played and what the drummer played. People often said, when the Beatles' CDs came out, "I'm hearing things I've never heard before," but that didn't happen to me because I had heard all those little things that had been in there. My business manager, Stan Hertzman, says I have 24- track ears. Of course now it would be 32-track ears.

I produced the second album by the Elvis Brothers, eight or nine years ago, and then they were dropped by CBS. I understand that producing someone else is a whole different thing, and that's what you get credibility for.

R•E•P: What do you try to do as a producer, with Caifanes specifically?

AB: I'm trying to inspire the best performances of their music, to record them and produce them in a manner that adds something unique, so that it ends up being their music and their band. And it sounds better because there's someone there like me who says, "No, this part really might sound better if you wait to bring it in until the third verse."

A SELECT ADRIAN BELEW DISCOGRAPHY

Adrian Belew:	The Lone Rhino; Twang Bar King; Desire Caught By the Tail; Mr. Music Head; Young Lions; Inner Revolution
The Bears:	The Bears; Rise and Shine
King Crimson:	Discipline; Beat; Three of a Perfect Pair; Compact King Crimson
Talking Heads:	The Name of This Band is Talking Heads; Remain in Light
David Byrne:	Songs from the Broadway Production of the Catherine Wheel
Jerry Harrison:	The Red and The Black
Tom Tom Club:	Tom Tom Club
David Bowie:	Stage; Lodger
Frank Zappa:	Sheik Yerbouti
Herbie Hancock:	Magic Windows
Jean Michael Jarre:	Zoolook
Riuichi Sakamoto:	Left-Handed Dream
Peter Wolf:	Lights Out
Laurie Anderson:	Mister Heartbreak; Home of the Brave
Paul Simon:	Graceland ■

R•E•P: How do you record?

AB: We record digitally, but we also use some of the Neve module preamps that are more old style analog preamps and have a tendency to warm up the sound. We do that with guitar and vocals.

R•E•P: Do you record at your own home studio?

AB: It isn't my studio, it's a nearby studio that's privately owned. It's about 10 minutes away from where I live and it's all in a very nice, clean, quiet resort area where I live, Lake Geneva. In fact, I live on the lake. The studio has an 80-channel SSL, two 32-track Mitsubishis and two 24-track Studers. So I mix and match between analog and digital, but mostly it's digital.

We also used some of the latest computer stuff, such as (Digidesign's) Sound Tools and (Opcode's) Studio Vision for certain editing, sampling and moving parts around. This particular record is one of the more natural recordings I've done in the sense that there isn't a lot of techno-wizardly apparent in it, because that isn't what I felt the songs needed.

I do have a home studio that I've been recently putting together. It started out with Sound Tools, a Mac and a 16-track. I used it in this particular instance just for demo-ing. I have demos of these songs that are just very simple.

R•E•P: What kind of a sound were you going for?

AB: More of a band sound, even though mostly it's being played by me. I wanted to simplify the parts so as often as possible you have the least amount of instruments performing, thereby giving them each their own character, their own space. So instead of adding and adding things, I'd use one single element. Instead of having three guitar parts, I'd have one guitar part doubled. I did all the singing myself.

I'm learning how to use that old "less is more" theory. I like having the voice in the middle and suddenly it springs out to the side. I enjoy double tracking my voice and guitar. I like it because there are so many ways to do it. You can get so precise, or in some cases it's nice if it's loose.

R•E•P: Was there a specific sound or a record you were trying to emulate?

AB: I didn't refer to anyone's records. I'm only trying to make mine sound better and better as I go. Each time I make a record, I've been using the same studio and the same engineers, so that hasn't been a variable. Between the three of us it turns into a lot of unspoken things.

R•E•P: Are there any producers you've worked with who influenced the way you approach production?

AB: Most of the people that I can think of who influenced me would probably be on records I didn't do. Of the people I worked with I would say Brian Eno and Frank Zappa — watching them work was educational for me. One of the first things Frank Zappa did for me was he took me through the mastering process. He was mastering an album set called "Leather," so he invited me along to watch how it was done. In the meantime, he taught me little tricks of the trade, such as how to find the right space between songs. The trick there is counting in the same tempo as the previous song for an even number of bars.

Brian Eno was involved with the first album I'd ever done in the studio - Bowie's "Lodger" — and he had some very unorthodox ways of working in the studio. He taught me composite guitar tracking and that you need to keep an open mind and allow things to happen.

We had something go wrong with some of the modules in the SSL and it starting sounding like the John Lennon "Revolution" distortion sound ... that's why I love the studio.

R•E•P: How did you get involved with Paul Simon?

AB: Roy Halee is another producer I love — I only worked with Paul Simon for four days on "Graceland' and I was amazed with Roy's musical timing and ears. Dickie Landry was working with me and Laurie Anderson at the time and he played some of my music for Paul. Paul called me and said, "I really liked the sounds you're making and they might fit into this idea I have." When he first played the songs for me they had no vocals, they just had the African musicians and they didn't sound very much like Paul Simon records, so I asked him what he wanted and he sang very softly — in a whisper — in my ear while the track went down in the control room. It gave me chills. Then it sounded just like Paul Simon music. I was really thrilled by the music on that album — I was one of the first people to hear it and realize what an enormous thing he had done.

R•E•P: With all the sounds you have on guitars and guitar synthesizers, what percentage of them are sounds you came up with yourself vs. working with electronic and guitar technicians?

AB: I don't consult very much with technicians apart from a period such as right now where I'm having my friend Al Jewer help build me a rack for the road. Jewer has a company called Uptown that produces the Flash units — something that directs traffic for MIDI. So apart from times such as this, when someone's building something for me, it's a case of me sitting in my music room fiddling with knobs and trying to come up with new sounds. They really propel me — I'm thrilled when I come up with some new sound that I haven't heard yet. For instance, I remember when I first got the guitar synthesizer to sound like an Eastern instrument — and when you played it, it had a really different feel to it, the notes played differently and the sustain was different. It was like learning to play another instrument. Or when you find a sound like an explosion or a chicken — even if you can't use it, it becomes part of your motif, part of your materials that you can use later — for instance a day might come where you think "this needs a chicken running across the stereo field right here."

R•E•P: How much background do you have in electronics, chips, etc.?

AB: I have no knowledge of that — I couldn't even solder a wire if I had to. I approach it strictly from a near-sighted artistic angle of just wanting those little boxes to do interesting things for me. Every now and then I have the urge to try something new, so I'll try out some new boxes or something, so that in a sense keeps the thrill alive.

We had something go wrong with some of the modules in the SSL and it starting sounding like the John Lennon "Revolution" distortion sound — great sound — you can't really get it out of any guitar amp that I know of. That's why I love the studio — that's my favorite place in the whole world.

R•E•P: I guess the biggest hit you've had was the surprise success of "Oh Daddy." I tend to think of your work as being avant garde, on the artistic edge of things, and it's always interesting when the avant garde makes a brief appearance into the mainstream.

AB: Up to this point I would say that was really a case of me writing a novelty song that did very well — something I rarely do. And I would think that the video had the most to do with the success of the song. There's something about it; it's charming, and many people of our generation can relate to it. Many people want to be successful, for their family or their friends. Since then it's fueled an old desire of mine to write good songs and get them on the radio and do them in my own manner, so that's been my quest, really since the Bears. I don't think that cancels my usually avant garde tendencies, I think the challenge then is just to fit those ideas into the songs.

R•E•P: Do you have any secrets to spill out to 80,000 readers?

AB: There's one thing, regarding sequencing. My assistant engineer, Dan Harjung, showed me a method for sequencing records that is very interesting. I'm not sure if he invented this or not, but he puts 30 seconds of the beginning of a song and 30 seconds from the ending of a song all in a line on the multitrack machine and he does that with each song. So when you look down on the console you see that faders one and two have a stereo mix of song number one and three and four have song two, and so on. Then we put it into cycle mode so it keeps cycling that 30 seconds of tape, so you can keep hearing the ending of track two going into the beginning of track 10, etc. We've used it for the last four or five albums. It makes sequencing go so much easier. ■

INNER REVOLUTION

Producer/guitarist Adrian Belew has just released his sixth solo album, "Inner Revolution" (Atlantic). The album is marked by strong, melodic pop tunes on which the wildly inventive Belew plays almost all the instruments. Lots of double-tracked instruments and vocals, often split-panned hard left and right, create an open and slightly "retro" sound that stands in contrast to the modern — and sometimes eerie — guitar synthesizer sounds that are Belew's trademark.

His voice spans the range from falsetto to all-out rocking rasp. "Standing In The Shadow" capitalizes on intentional distortion — the lead vocal and the lead guitars sound as though they are overloading the solid-state preamps, a very different sound than the tube distortion or tape distortion most people employ (when they're employing it on purpose). The album's standout tune, "Big Blue Sun," is a beautifully melodic, reflective song that Belew surrounds with more vocal layering combined with an exquisite string quartet. ■

Photographs by Gary Hannabarger, courtesy of Atlantic/Atco Records, Los Angeles.

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DESKTOP PRODUCTION UPDATE

By Anthony McLean
and Rick Schwartz

Comparing similar things *should* be simple. Apples to Apples. Broccoli to Broccoli. That's easy. But, comparing audio workstations is nearly impossible, primarily because no two flavors are really the same. And side-by-side tire kicking isn't fair to anyone. What is important is to define the continuing evolution of these platforms and hopefully establish benchmarks for those considering a workstation purchase. Our goal has been to provide comparative data as much as possible. Still not easy, but the right thing to do.

There were many questions to answer. But the issue mentioned most by potential buyers we talked to was profitability. And profitability is always chained to the "function vs. price issue." But functions and prices aren't always comparable because there are usually alternate ways to perform a single function, for instance, sample rate conversion. The reality is you pay more for proprietary hardware-based systems and less for software-based systems powered by commonly available Macs or PCs. So, comparing 6-figure systems that handle every conceivable audio-post application to 4-channel software-based systems isn't realistic or fair.

In the March '91 R•E•P we covered most basic information about the workstation environment. Relative importance of storage mediums (type and size), processing speeds and many related issues were evaluated. We hear from many of you that our workstation issue is still on your desks.

This year we've reviewed more systems, 23 to be exact, and have included systems upgrades information in each section. In general, this year's update groups workstations according to the maximum number of simultaneously recordable tracks, from stereo through multitrack. Our evaluations were focused on each system's primary application, although ancillary functions are also included.

Quotes about each system, called "On-Line Observations," were obtained from professional operators. The purpose of these quotes is to share some interesting and informative perspectives we received from actual users during our research.

Finally, prices are mentioned, but only to establish a desktop ballpark reference.

R•E•P has labored to establish accurate data. And we know R•E•P readers, the most informed and discriminating professionals in the recording industry, would never buy a system simply because of this data. Please note: We have experienced as many demos as possible, and tried to turn over every rock. However, no endorsements are implied. Also, prices change. This update is reference-only material! End disclaimer.

METHODOLOGY

A solid strategy for considering a purchase is to identify the two or three systems that best meet your needs in your price range and contact the manufacturer or a reputable dealer about options and specific configurations. Also, it is critical to spend time actually using any system you think is right for you, before you buy.

You'll notice it's a world market. Australian, British, Japanese, American, German and Swiss companies are all in competition for your business. From our count, the total number of existing digital systems, of all types, now exceeds 80. From our point of view that also makes it a buyer's market.

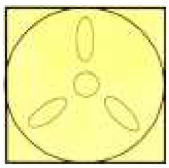
We have grouped workstations from clearly established companies first. These are units that we have been able to view first-hand. A second grouping titled "Too-New/In-Progress" is located in the Cutting Edge section on pages 64 and 65. A listing in that section does not indicate that the products mentioned there are not desirable, but simply that we were unable to develop any dependable evaluations of in-production, on-the-street units...yet.

Let the work begin ...

Anthony McLean is features editor of R•E•P. Rick Schwartz is a contributing editor to R•E•P and director of post-production at the Post Complex, Los Angeles.

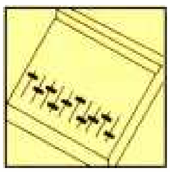
KEY TO ICONS

A



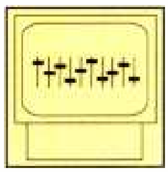
"Virtual" tape deck: A RAM- or disk-based audio recording system that provides all of the capabilities of a 2- or multi-track tape recorder. Number indicates number of discrete tracks.

B



System features hardware interface that is a traditionally styled mixing console, which provides actual faders. Number indicates number or range of available faders/channels.

C



System features an on-screen (virtual) mixing console.

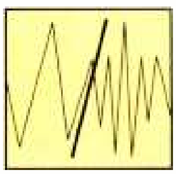
D

System has facilities for digital signal processing, such as equalization.

E

System permits sample rate conversions allowing digital audio to be converted between various sampling rates. (i.e. 44.1kHz samples are converted to 48kHz samples.)

F



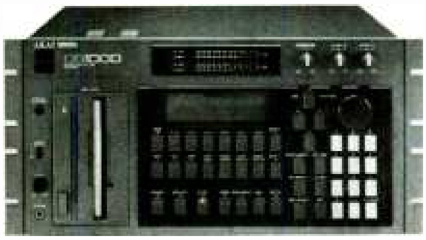
System has time compression/expansion function allowing temporal manipulation of recordings (i.e. 31 second radio spot shortened to 29.5 seconds without affecting pitch).

UPDATE

STEREO SYSTEMS

AKAI DD1000

System Configuration: Two stereo pairs of tracks record to disk, one pair at a time, using MOD (magneto-optical disk) technology. The device records at a nearly universal storage time of 30 minutes of stereo (at 44.1kHz). Additional drives can be added for extended time and the deck can be locked to external LTC or VITC. Although the DD1000 can function as a regular, easy-to-use 2-track, it can



also handle sophisticated editing and signal processing. The optional DL500 trigger interface allows the DD1000 to cue and play cuts much like a standard cart machine. There are two discrete balanced XLR inputs and four discrete balanced XLR outputs for playing the four tracks, as well as digital I/Os.

System Upgrades: Timestretch allows a user to change the speed of audio during playback without changing its pitch. Machine control, using the Sony 9-pin standard, allows the DD1000 to operate as a master or slave. Auto Qlist play puts the DD1000 in a constant play-ready state when recording from a time code source. Users can make up to 50 cuts on-the-fly during recording to auto-assemble a rough EDL. Atari ST users can connect to a DD1000 using MIDI. Macintosh software has been rewritten to support new hardware features and now supports color.

Strengths: Very affordable. The system is self-contained; no extras are needed to operate. It's able to simultaneously play two stereo tracks from a single optical disk and has a built-in jog wheel for smooth scrubbing. Performs real-time time compression/expansion. Up to seven DD1000s can be controlled from single remote, which acts as a master clock for the system. The contents of an entire MOD can be backed up to any DAT machine with digital I/O. Samples can be primed in memory for instant playback, eliminating the delay most disk-based systems have. Sounds can be transferred via a digital bus to an S-1100 sampler for edited effects. Software packages are available for Atari and Mac computers.

Weaknesses: Will not record all four tracks at once. Submixing is done to hear more than two stereo tracks. VITC support requires the DL1000 remote and an optional card. Digital interface does fully support the AES digital I/O. (However, devices that support the Type I standard can be used with an adapter.) Closed architecture limits future expansion. Does not support traditional delete-style edits. Disk directory does not have a sort function.

On-Line Observations: "The DD1000 allows us to quickly move from one client to the next because of its use of removable media. Using the DD1000 with touch screen software is like having 50 digital cart machines loaded and ready to go. By using a DD1000, Akai sampler and expander, a user would have up to 64Mbytes of RAM, 32 voices and six tracks of digital audio from three rack-mount units, all of which are controlled by a central computer. Similar capabilities would cost much more using a dedicated system."

Conclusions: The DD1000 was one of the first systems with support for optical media. Above everything else, removable optical disks allow rapid data retrieval and are incredibly sturdy. The system is highly reliable and among the easiest systems to get running. Macintosh software has made the system less keystroke-intensive. The DD1000 has found a strong niche in broadcasting applications, although the auto-load software is becoming a hit with post-production facilities. By improving its software for the Mac and providing DAT archiving capabilities, the company may have addressed the last roadblock to widespread acceptance.

Bottom Line: System prices start at $14,500.

Circle (101) on Rapid Facts Card

TURTLE BEACH 56K

System Configuration: Turtle Beach seems to hold the position that, for most people, the oncoming wave of low-price digital multitrack makes more sense than the hyper-expensive software/hardware configurations. To that end, a 286- or 386-type PC, Turtle Beach's DSP card, Digital Interface box, and SoundStage editing software deliver a 2-track digital editing system. Outboard A/D and D/A converters are required/recommended allowing the user to choose according to needs and budget. EQ, varispeed, mixing and crossfading are among standard DSP functions. Stereo recordings, or "soundfiles," consisting of "zones," can be stacked for random access playback and triggered by incoming MIDI data or at a user-defined time. Cart player simulation can be triggered via mouse click. Simultaneous playback of multiple zones makes this system suitable for complex effects assembly. LTC can be read and generated.

System Upgrades: None to speak of. The 56K product is essentially mature with the company continuing to iron out its few remaining wrinkles.

Ancillary Markets: CD pre-mastering, sound design.

Strengths: Turtle Beach is working on the concept of appropriate scale and appropriate technology. The 56K offers maximum affordability. For many people, in fact most people, two tracks of random access audio

and editing are enough.

Weaknesses: Some small bugs still exist. And anyone who wants to grow to more than two workstation tracks must look elsewhere.

On-Line Observations: "To me it is the most transparent machine I've heard to date. It doesn't force you to do DSP all the time." "I used to feel I had to have a scrub function. I haven't used scrub in six months." "We all know where the few remaining bugs are and work around them, because it sounds so good." "We use it as a digital scratchpad for album sequencing and use the flexible playlist functions to stack CDPQ sheets. One record company called in every day for a week with a new sequence list. Then they came in and auditioned four sequences in one day."

Conclusions: Near the lowest possible price point in the workstation universe.

Bottom Line: System prices start at $1,995 (excluding computer and hard disk).

Circle (102) on Rapid Facts Card

4-CHANNEL SYSTEMS

MICRO TECHNOLOGY UNLIMITED

MICROSOUND DAW

System Configuration: The Microsound Workstation is designed to work on any IBM PC-type platform (AT-type through 486), supports Windows 3.1, and is available as a discrete 4-channel, 4-track version, or a 2-track, 2-channel recorder. As many as 38

virtual tracks can be compiled, overlapped and internally mixed before output. In record-mode, on-screen VU meters, tape counters and bar graphs confirm and communicate recording status. Simulsyncing overdubs with the Microsound emulates traditional

UPDATE

multitracking. By using standard MS-DOS partitioned disk drives, sound files can be managed/stored with any other MS-DOS file.

Edits can be made at the waveform level (via zoom-in view) and auditioned in real time. Multiple internal stereo tracks can be simultaneously triggered for cue list assembly. And the Microsound can emulate a broadcast cart machine, with manual spot cuing (from the keyboard) while prior/post spots remain visible. Unbalanced or balanced analog I/Os are available, as are optional digital I/Os and an optional time compression/expansion function. An LTC reader/generator is also available.

System Upgrades: Microeditor functions are on constant user-feedback driven upgrades. Upgrades to existing software (Version 2.0) include Cut and Boost, which permit real-time amplitude changes of a zone within a segment, effectively allowing the engineer to "duck" music under dialogue.

Skip and Delay permit skipping samples within the zone as if deleted. Using Snap-to-Grid, segments can now be positioned by starting/trailing edge, or selected Flag/Zone point within the segment and moved to start playing at a new time. Marker flags can now be set on the fly. Record and play can be triggered by: SMPTE, MTC, MIDI Note and Mouse. The optional Microsync board allows Sync-Lock Resolve and SMPTE generation. Future enhancements include an anticipated summer 1992 release of a parametric EQ option.

Ancillary Markets: Music recording/editing, radio mastering, CD pre-mastering, sound design and computer music synthesis.

Strengths: More than 300 audio segments can be used in a single mix. The price point here, especially considering the cheap availability of PC platforms, is impressive.

Weaknesses: The fact that this system does not grow beyond 4-track will eliminate many music production applications. The look and feel of the interface seems to bother die-hard Mac types. But what's new?

On-Line Observations: "The company has provided excellent support ... we're making money with (Microsound)." "I just completed a project in two days (with the MTU System) that would have taken at least two weeks in linear (traditional) editing."

Conclusions: The company is aggressive and innovative. The overall speed of most Microsound functions indicates the company has a grip on writing high-efficiency software. Small details, such as immediate access to deleted files regardless of edit length, are their forte. In general this system is a good-bang-for-the-buck-type product.

Bottom Line: Systems start at $3,690 (excluding computer and hard drive).

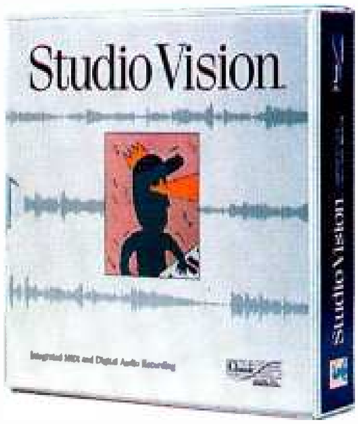
Circle (103) on Rapid Facts Card

OPCODE STUDIO VISION

System Configuration: The original Studio Vision was an upgrade path from Opcode's Vision software program. It offered 2-track playback linked with comprehensive MIDI sequencing, editing and automated mixing from the Macintosh platform. Combined with Digidesign SoundTools, Studio Vision opened up the entry level of workstation ownership.

System Upgrades: For the first time the system can playback up to four tracks using two AudioMedia cards or ProTools hardware. Up to 16 tracks can be stored per song. Although only four tracks can be played at once, you can select two audio tracks and mix them together along with dynamic changes. OMS support is a time saver for users with large MIDI setups.

Strengths: One of the most affordable 4-track hard disk recorders on the market. Software is easy to learn and easy to use. Includes powerful automated editing features, such as strip silence that cuts up a track



based on a preset level threshold. Audio and MIDI tracks may be dynamically automated with level changes and pans. Sound files can be edited using Sound Designer II or Alchemy software. One of the few companies with after-hours phone support.

Weaknesses: Older Mac processors may get bogged down when locking automated digital audio tracks to time code. Has almost no internal DSP capability. Program must export sound files to SoundDesigner II for advanced waveform editing, time-expansion or sample rate conversion. System is dependent on Digidesign hardware to record, and for house sync and machine control.

On-Line Observations: "The audio part of this system has really seemed to come together. Studio Vision is perfect for the way I work, because most of my time is spent sequencing MIDI tracks which need audio recorded on top of them. The MIDI sequencer was not an afterthought, like many other hard disk systems."

Conclusions: Studio Vision is ideal for project studios or musicians who need to record and edit a few digital tracks, but don't want to learn a new software. The company has many loyal users.

Bottom Line: System prices start at $10,000.

Circle (104) on Rapid Facts Card

MULTICHANNEL SYSTEMS

AKG DSE7000

System Configuration: AKG's DSE7000 is a RAM-based recording, editing and playback system intended for the broadcast production market. Eight digital audio tracks can be mixed to stereo internally. The DSE7000 emulates the combo board of a radio production studio with an onboard 10-channel digital mixer, and an invisible-to-the-engineer 386-type PC. Computer chops are not required and the computer keyboard is only used for text entry of project titles. The interface completely emulates the broadcast production environment with square plastic Start, Stop Record, Fast-Forward, Cue and even dedicated Help, Undo buttons. Recording times are limited to 4.4 track-minutes at 32kHz sampling and expandable to 17.6 track-minutes with optional RAM cards. There is a 15kHz upper bandwidth of sampled material.

System Upgrades: An optional data/DAT archiving system has been added.

Strengths: By design, radio pros can handle the DSE7000 more easily than most workstations. The RAM-based system delivers essentially instant production access to the systems memory. (Hard disks are included for safeties and data backup.)

Weaknesses: Not the workstation for everybody. The DSE7000 is especially strong, although somewhat pricey for broadcast because of its high-tech transparency.

On-Line Observations: "It is absolutely the only machine that everyone here (East Coast radio station) can use. We have a production staff ranging from 17 to 66 years old. They fight for time on it." "It keeps the clients out of our hair because we can get them in and out so quickly."

Conclusions: For tidy radio production and short track learning curves the DSE7000 works very well. This unit capitalizes on broadcast production requirements for spe-

cific, less comprehensive workstation features. Also excels in ad agency environment.

Bottom Line: System prices range from $27,500 to $37,500 (includes computer).

Circle (105) on Rapid Facts Card

AMS AUDIOFILE PLUS

16

System Configuration: AMS was among the first hard-disk recording systems; the current version, AudioFile PLUS, consists of a "controller surface" and two outboard rackmounts, one for the hard disks, and one for processor cards and analog/digital I/Os. The AudioFile PLUS can interface digitally with the company's Logic One or Logic Two digital mixing consoles for hands-on digital mixing. The Controller Surface is the user interface with a small, rack-mountable panel holding a video display, a QWERTY keyboard, two shuttle/jog/value wheels, a floppy drive and various buttons.

The transputer-based AudioFile PLUS easily handles basic 2-track production. The base unit can function as an 8-track digital recording system or 16-track recording system, both configurable for eight inputs. The

base unit offers up to two track-hours of mono recording (at 44.1kHz).

DSP functions include time compression/expansion, simulated reel-rocking, varispeed and crossfades. The AudioFile PLUS can be controlled on-line by Sony BVE or Ampex VPR-3 editors, read/generate LTC time code and read VITC. The "Jukebox" mode simulates eight cart machines. All 16-channel AudioFile PLUS systems can be paired with fully automated AMS Logic 1 consoles to provide AudioFile with digital mixing, EQ, gating, limiting and compression/expansion. Future plans include segment-based mixing.

System Upgrades: AudioFile enhancements include the ability to read CMX-type EDL diskettes directly and to control where and how source audio is recorded. AudioFile can now sense gaps of five seconds or more (since the previous cue) and use external machine control to fast forward to the next edit point.

An optional buffer card allows events to be placed closer than 18 frames together in a track. Improved software is much more stable than previous versions. New EDL auto-conform software dramatically reduces assembly time. The system will read most popular EDL formats, automatically load all source elements and conform them. Other enhancements now available to all multi-input PLUS users are an MOD, an Exabyte backup and cue-audition from the tape.

SPECTRA, the newest AudioFile upgrade, also provides on-line MOD recording/editing. A new slim-design control surface permits lap-top operation of the control surface. And a color LCD active matrix and VGA-resolution control surface screen take AudioFile beyond "green-screen" limitations. (A standard color VGA can serve as an extension monitor.)

Ancillary markets: Music editing, CD premastering, and broadcast applications.

Strengths: The system is very fast. One of the few systems that records eight tracks at once. Scrubbing is smooth and feels like analog. Time compression sounds good. Powerful auto-conform software loads an EDL and automatically records desired sections from a source tape. The AudioFile also works with Gefen's auto-load software to automatically record sound effects from CD. Edits can be made with single subframe accuracy. Disks are not tied to tracks; any sound will appear from any output. Punch-in and punch-outs are totally non-destructive. Programmed record-in times can be saved to a floppy and loaded pre-session. Optional ADR software available.

Weaknesses: Fairly expensive. Limited DSP and mixing capacity. Has no dynamic automation capabilities. Proprietary computer system may limit expansion. Current (under development) lack of on-screen waveform display inhibits some applications. Users must define and mark edit points by scrubbing. Small screen at desk may bother some users. But larger screen for client and/or operators is an easy add-on. System lacks an all-digital signal path.

On-Line Observations: "I find the AudioFile very intuitive because it closely parallels the way analog systems work. It also has the cleanest scrubbing of all systems I have heard." "We spent 18 months, we looked at everything. Nothing even came close."

Conclusions: The system is fast and easy to learn. With many units in the field, it has proved itself in the advertising, film and post-production industries. With one of the largest feature sets on the market the AMS system offers controls 'post' editors need. The adaptable card-cage design has allowed the system to remain competitive for more than five years. Most important, the 500+ worldwide users make money from their AudioFile investment.

Bottom Line: System prices start at $95,000.

Circle (106) on Rapid Facts Card

AUGAN 408 OMX

32 SRC

System Configuration: The 408 OMX is a stand-alone, rack-mount proprietary unit configured to emulate a 64-track ATR with four recording heads and eight playback heads. Augan, a relatively small Dutch company, developed VLSIs to control flow and processing. The 408 OMX stores all audio and session data directly to two MODs. Operator interface is reminiscent of a conventional multitrack recorder with a separate, lightweight control panel with dedicated keys, a shuttle jog knob and a small back-lit alphanumeric LC display.

A large electroluminescent panel (similar to those used in laptop computers) and a computer monitor can both display track sheets, libraries and other data. Most often, the EL panel is used by the operator and a large computer monitor is positioned for client viewing.

Editing is "virtual," meaning no audio is actually moved, modified or erased until a Clean-Up command is executed. Audio segments that are copied or moved to an empty track area are always faded-in and faded-out, using preset default fade times. Fades can then be lengthened or shortened, and the material trimmed.

A routing/mixing matrix is interposed between eight playback tracks and eight physical outputs. Any track can be patched to any output, its levels changed and mixed with any other track to create a mono or stereo sub-mix without using an external mixer.

Output and track levels can be changed in real time, using the control panel's knob or MIDI. MIDI-in,-out and -thru ports permit the OMX to interface with keyboards, sequencers and automated mixers. A Gate function allows threshold level to be set so that only audio material will actually be recorded, while quiet areas will be shown as "digital black" or "silence" on the track sheet display and will occupy zero disk space. Recording can take place in the background while editing occurs in the foreground so material can be edited as soon as it is recorded.

The 408 OMX has eight 1,000-bin libraries for storing audio segments, effects, jingles, dialogue and musical elements. These bins are identified by name, length and output, not time code position. Audio can be put into and removed from any bin by using the copy or move commands. Copying a sound segment into a library takes up no additional disk space, although libraries can be hard-copied from session to session if necessary. Library segments can be instantly triggered from the system's keyboard or from a MIDI or computer keyboard with up to 64 library

bins key-assigned. An RS-232 port allows communication with printers and computers. A SCSI port permits an additional MOD and digital output allows several OMXs to be linked. RS-422 provides external video machine control.

Ancillary Markets: Music recording.

Strengths: A powerful, mid-price device whose home may be destined for larger commercial production and broadcast facili-

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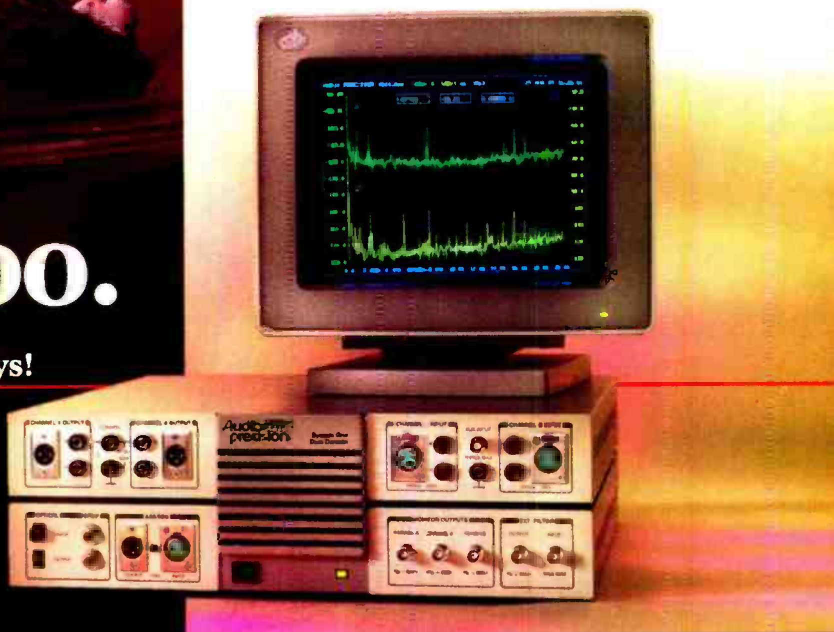

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Weaknesses: It doesn't feature many of the software bells and whistles that put many engineers in their comfort zone. The system's graphic display verges on austere. In the near term, it will be hard for potential North American users to experience the device. Distribution is sparse.

On-Line Observations: "The Augan was new, so we had doubts, but it was the best available option as a centerpiece recorder for our multiroom facility." "The OMX is conceived to meet actual studio needs, rather than match the competition's features."

Conclusions: Some startup problems will exist as the company brings the product fully into the North American market later this year. But Augan has a small, dedicated European base. As witnessed in a 1992 NAB product demo, the machine is positioned by its features/benefits to profile far away from most other systems. See one if you can.

Bottom Line: System prices start at approximately $45,000.

Circle (107) on Rapid Facts Card

DAR SOUNDSTATION SIGMA

System Configuration: The original SoundStation II (no SoundStation I was made) is an elegant piece. The keyboard-sized Control Console is a striking jog/shuttle wheel and dedicated controls package. Equally impressive, the DAR-authored software allows you to touchscreen select and manipulate desired audio events.

This disk-based system is available in 4-, 8- or 16-track models, with up to 16 channels of

analog XLR balanced I/Os. A separate processor and disk rack holds hard disks, CPU and I/Os. Optional DSP functions include time compression/expansion and EQ. The DAR audio building block is called a "segment." The building process stores segments in directories and constructs a finished product by copying, editing and moving segments to a production "reel."

DAR hangs its hat on the innovative "Wordfit" option, designed to automate dialogue replacement for film/video. Typically, any replacement track is "out of sync" with the original. Wordfit compares a location-recorded track with the studio-recorded replacement track. Wordfit then processes the replacement track to start/end in the same time as the original and conforms the replacement to the original pacing. The system can lock to external LTC and follows the Sony P2 format.

System Upgrades: The relatively new SoundStation Sigma is DAR's flagship. Sophisticated varispeed, real-life crossfades and a feature called Segment-Based-EQ are major Sigma enhancements. In particular, the Segment-Based-EQ allows an engineer to create EQ templates that can be stored, copied and moved as needed to minimize the nasty re-invent-the-curve redundancies of traditional post-production methods. A complete digital mixing package has been added to complete the Sigma's internal control of recorded tracks. Future upgrades include the ability to rock video and have all 16 audio tracks follow the rock frame-by-frame.

Ancillary Markets: Advertising production.

Strengths: Lots of "little things" such as a strong librarian function that eases the workstation file management bottleneck. Though small, SoundStation's onboard editing screen (and associated icons) reinforces the system's refined approach to workstation interface. Also, Wordfit is an option any audio-for-film/video facility must consider.

Weaknesses: To date, SoundStation has been undersupported in the U.S. In the other "little things" department, some people who have seen SoundStation demos are distracted that the "tape" emulation (as viewed on the playback sequence screen) spools from right-to-left. A minor point.

On-Line Observations: "We've been booking 20-25 hours a week at competitive rates. It is a money-maker." "You have a better chance of them (DAR) hearing your operational requests than from a big company." "I had lots of demos, this is the warmest and best-sounding system I've heard."

Conclusions: A lot of power to this package. And if the soul of a workstation is its interface, all the SoundStations are saints. We expect special features such as Wordfit and Segment-Based-EQ to really solve problems and save time. You pay for both performance and elegance here. Worthy of consideration for any fast-paced production environment.

Bottom Line: System prices start at approximately $85,000.

Circle (108 on Rapid Facts Card

DIGIDESIGN PRO TOOLS

System Configuration: One of the few systems that visually represents audio and MIDI data in the same window. Included are effects sends and DSP, such as parametric EQ. The on-screen mixer features static and dynamic automation capabilities. Crossfades are hard disk-based. Currently in version 1.1, Digidesign recommends a Mac IIci, IIfx or Quadra with 8Mbytes of RAM, System 7 and at least a 600Mbyte hard disk.

System Upgrades: The new System Accelerator increases processing speed and allows expansion (in groups of four). When fully implemented, new enhancements for audio-post users will include a spot mode for faster on- or off-line spotting. A new grid mode allows the user to nudge a sound by a predetermined amount and to quantize edits by selected increments. Start and end times for each region are now screen-displayed and voices are now color-coded for clarity.

Strengths: The modularity of the system allows a user to start small and expand as needed. Huge original "Sound Tools" user-base. More affordable than dedicated systems. Intuitive graphic editing allows regions to be easily trimmed, expanded or moved. Supports 64 virtual tracks. Open architecture ensures future growth. RAM-based sampling capabilities can be added using Sample Cell card. Extensive sound effects and music libraries are available on CD-ROM. A hard-

ware control surface is available from J.L. Cooper, with eight faders, scrub wheel and assignable soft controls. Pro Tools hardware works with Studio Vision, Mark-of-the-Unicorn's Digital Performer and Q-Sheet A/V.

Weaknesses: System requires a very fast Mac CPU to function well. System Accelerator is required to expand beyond four channels. Without the System Accelerator, screen updates take too long, making on-screen metering and moving faders almost useless. No more than four tracks can be simultaneously heard in a basic 4-channel system. Voice-allocation system is unacceptable for some applications. MIDI sequencing and editing abilities are very limited. Memory requirements of ProDeck and ProEdit have caused some user problems when trying to run both applications on an 8Mbyte machine using System 7.0. Interleaved stereo files are no longer supported, requiring them to be split before they can be used. Does not support traditional EDL window.

On-Line Observations: "Pro Tools is the only system that features multitrack recording, powerful DSP, MIDI sequencing and automated mixing at a reasonable cost." "Hard disk editing (such as Pro Tools) is not the place you want to do mixing."

Ancillary Markets: Project/mid-size studios for MIDI and multitrack linked recording, spot production and sound design.

Conclusion: Pro Tools shows great promise, but is still young compared to most products in this survey. Currently, the system requires the use of two applications, which creates multiple problems for some users. The system is ideal for project studios that use digital tracks with MIDI.

Bottom Line: $5,995 for basic 4-channel configuration (without Mac and hard disk).

Circle (109) on Rapid Facts Card

DOREMI LABS DAWN

48

System Configuration: Independent of large company support, the Mac-based DAWN has survived, during the ramp-up years of the workstation industry. One of the most affordable 8-track systems on the market, this system works with MOD drives and records at 32, 44.1 and 48kHz. Events are shown using a scrolling track sheet or EDL-style cue list.
System Upgrades: The Digital DAWN features eight analog inputs and four stereo digital inputs. The new DAWN chassis includes digital I/O on all eight tracks, a new sync board with built-in LTC reader/generator and a programmable word clock I/O. Future developments include a digitally controlled mixer, produced in conjunction with Pacific Recorders and Engineering.

Strengths: Dawn is expandable to 48 tracks (in groups of eight). The system supports AIFF standard sound files that can be edited using Alchemy or Sound Designer II software. Machine control card is standard with DAWN software and cues can be triggered using a MIDI keyboard. Variable speed playback is

possible. Software updates are delivered by modem. The new built-in synchronization module resolves to house sync or external SMPTE. A dedicated hardware control surface was shown at the 1992 NAB convention and is now shipping.

Weaknesses: Currently, the system offers very limited DSP, lacks digital EQ and has no internal mixing or automation capabilities. The anticipated summer delivery of a DSP card and digitally controlled mixer should solve most of these problems.

On-Line Observations: "The open-environment was critical for us. Using off-the-shelf Ethernet cards, our networking capability rivals more expensive, dedicated systems. We routinely transfer sound files between multiple locations using Timbuktu software for the Mac."

Conclusions: On the market for more than four years, DAWN is popular with many film and post houses as an editorial tool. The simple Mac-based interface, open architecture and accessible price have gained DAWN acceptance among owner-operators seeking an affordable, high-performance start-up.

Bottom Line: A DAWN Workstation Nucleus, including two inputs/eight outputs and a sync-board, starts at $16,500.

Circle (110) on Rapid Facts Card

KORG SOUNDLINK

8

System Configuration: Physically, the proprietary hardware and user interface are reminiscent of a Lexicon Opus. SoundLink permits simultaneous recording of up to eight tracks. The digital mixer section features eight channel faders, one stereo master fader and a MIDI master volume fader, each with continuous real-time automation; plus eight automated channel mutes. Other mixer functions that can be snapshot-automated include panning, aux 1/2 send levels and internal effects balance. Snapshot automation of equalization settings is also provided for the high/low EQ bands (shelving-type) and the midrange (parametric).

Automation modes include off, write, read and update. The update function incorporates thresholds to control the update process for fader position changes. Automation data is edited in a similar manner to audio editing via move, erase and copy functions. Channel settings for the noise gate and high-pass filter are saved with each session.

Three types of reverb (hall, room and plate) are provided with variable parameters while all channels use the processor send as a reverb balance control. Reverb parameters are also snapshot automated. A compressor/limiter, resident to the L/R mix bus, can also be snapshot automation controlled. Time compression/expansion and crossfades are implemented.

The console features soft/hard keys, alphanumeric keys, transport controls, jog wheel and faders and the system provides a 16-track MIDI sequencer. Tempo maps can be created, overdubbing and step recording is provided and track bouncing can merge data from two different tracks. The basic SoundLink is shipped with a rack-mounted 670Mbyte hard drive and one 2.3Gbyte 8mm Exabyte back-up. The system will follow/lock to LTC and VITC sources plus lock to House Sync, Digital Word Clock, MTC and MIDI clock. RS-422 provides external video machine control.

System Upgrades: It is a newly delivered system, not widely circulated yet. MOD support is under development.

Ancillary markets: Music recording and mastering for radio, CD pre-mastering.

Strengths: The SoundLink has a strong user interface offered in a turnkey package. Looks good, sounds fine, support from major company seems assured. Benefit-to-price seems good.

Weaknesses: Too new to tell. Units are shipping and seem to work. To some, the on-board screen/display may seem small. One improvement would be to allow the machine to interface with a third-party computer so it could operate within a Windows (or Mac)-type graphic environment.

On-Line Observations: "The user interface and hardware controller have a very natural feel. It is nice having a workstation with hardware controller; it makes more sense." "The SoundLink niche seems to us to be a low-price (comparatively) tapeless tape machine. The rear panel is incredibly well thought out. Machine control is a major plus, because all you really need is a SoundLink, a professional VTR and speakers."

Conclusions: An attractive and well-considered entry into this market. The package seems to make sense and work. With only beta test users to speak with, hard-edged opinions about the unit are limited.

Bottom Line: $37,000 and up.

Circle (111) on Rapid Facts Card

LEXICON OPUS

8

System Configuration: The 12-channel, 8-track Opus offers faders, pan pots, mute and solo switches, patch points, four aux sends with pre-post/on-off selection switches, EQ and polarity invert on each channel. Analog I/O and digital I/O can run simultaneously. The digital mixing console employs a 24-bit-wide audio data bus. The OPUS/E digital editing system offers standard OPUS recording/editing features without the console user interface. OPUS/E contains a built-in 8x2 digital monitor mixer with analog or digital outputs. A virtual disk environment permits unlimited simultaneous crossfades and each "reel" in the OPUS architecture can handle up to 160 automated mix revisions. OPUS slaves to external time code (NDF, DF, EBU,

FILM rates) and can control a variety of serial interface video decks. OPUS also resolves to house sync or external word clock data.

System Upgrades: Current software (Version 3.0) is quicker in function implementation, provides remote machine control, CPEX-time compression/expansion pitch-shifting and Opus console automation for all faders, mutes, switches and rotary pots. Track level editing allows automation and audio data to be simultaneously edited. Version 3.0 also permits control of video recorders from the Opus. Future upgrades (Version 3.1) include support of a 2.5Gbyte hard disk, a color monitor and expanded machine control.

UPDATE

Strengths: A high performance platform, the Opus boasts high fidelity, serious automation and was the overall winner in R•E•P's 1992 workstation issue.

Weaknesses: When voicing-to-picture, it is tedious to backup six or seven takes. But the company has an ADR package in development intended to eliminate this bottleneck. Like other integrated, stand-alone systems, you may be paying for more than you need.

On-Line Observations: "We tested every major system ... we chose the Opus based on its ability to record/edit and mix in the digital domain. (All of the manufacturers) were excellent in certain areas but Opus was the best in terms of audio-for-video post." "Any client who uses it doesn't want to go back to the analog room. Once you get out of the linear thinking, it's hard to go back."

Conclusions: The company's ability to continuously upgrade and Opus' strong track record makes it a reasonable choice for those capable of a 6-figure purchase.

Bottom Line: $150,000 (configurable).

Circle (112) on Rapid Facts Card

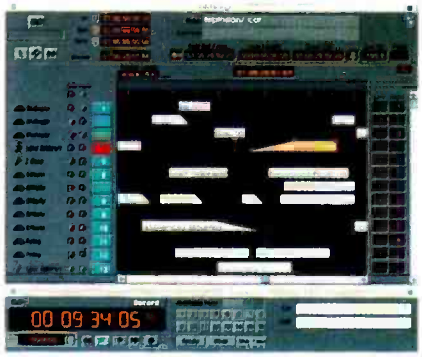
NEW ENGLAND DIGITAL

POSTPRO

System Configuration: The NED system has grown continuously for nearly eight years since the seminal Synclavier made most people aware of the meaning of direct-to-disk. Among other features, the original system has a large selection of music and SFX libraries available on 2Gbyte optical platters and 1Gbyte MODs. The DAT backup system operates at 8x real-time and holds 1.2Gbytes per tape.

System Upgrades: NED unveiled a new 16-channel version of the DSP mixer in late 1991, which is its first product to employ the Multi-Arc hardware platform. The EditView software lays out a track visually, so you can fade-in or fade-out on a sound using the mouse. Cues can be positioned by simply dragging them where you want them on the screen. TransferMation librarian software quickly stores and retrieves sounds from disk or RAM and now includes machine control. CMX Autoconform software loads and conforms source audio to an EDL and now supports rehearse and looping functions for ADR applications. AudiMation software provides up to 16 channels of digital mixing with 5-band parametric EQ. Using AudiMation, up to eight snapshots can be saved to disk. In addition, dynamic mixer moves can be recorded by MIDI sequencer.

Strengths: Fast and powerful. Large customer user-base. Hardware supports sampling rates of up to 100kHz. EditView software displays RAM and disk events with EDL or graphic-based playlists and has excellent color support. Librarian window keeps track of off-line volumes. Database performs keyword searches by name and category. Optional CMX Autoconform software records selected events via machine control, will



read head/tail trims and imports event names. Optional SoundDroid software includes spotting notes, cue sheet printing and a database. Current software version is highly reliable.

Weaknesses: Overdue DSP option is expensive and required for digital EQ. DSP option cannot be expanded beyond 16-track capability. Current software has no integrated dynamic automation capabilities.

Macintosh networking capabilities are disabled when EditView software is used. Library searches can be slow. EditView lacks a global ripple function. Multi-user systems not yet demonstrated.

On-Line Observations: "I think the PostPro is a superior system because of its speed. As fast as you can click on cut, it makes the cut. In addition, EditView software has made the system much easier to use."

Conclusions: One demonstration of EditView software will make anyone a believer in the power of visual editing. NED has always been a leader with many firsts under their belt. The company has always traversed the high-end market and can be expected to continue to do so.

Bottom Line: Entry level systems are approximately $70,750.

Circle (113) on Rapid Facts Card

OTARI PD464

64

System Configuration: ProDisk 64 was the first system on the market to match the storage capabilities of a 24-track (analog) recorder and is still the only system that can be expanded to 64 discrete tracks (even in the field). Users can punch-in on multiple tracks at once. Phase-accurate sample lock eliminates phase cancellation when mixing tracks to stereo. All editing is real-time.

System Upgrades: Otari's acquisition of Digital Dynamics has resulted in improved R&D, marketing and technical support. New ProDisk Guide software (Version 4.0) has a graphic (more user-friendly) interface with more color support and a CMX-EDL autoconform feature. ProDisk also now offers removable disk packs and an off-line backup station. In addition, DSP, mixing and dynamics processing can be automated using sequencing software that runs in the background. The newly released CB-158 Hardware Control Panel provides dedicated transport and edit function controls as an alternative to mouse and keypad commands. New system controller cards incorporate improvements, such as VITC and external videosync. A software release is scheduled to add time compression/expansion and pitch shift.

Strengths: One of the only systems that will simultaneously record more than 24 tracks and is designed for complex production with a high volume of track lengths and widths. The Guide graphic interface allows the user to move tracks by dragging them. Has movable transport palette and on-screen meter bridge. Cues can be auditioned from the sound library and placed on screen. Software supports drag and drop of time code addresses. Superior technical support.

Weaknesses: Currently has limited DSP support (only levels and pans) and no digital EQ. Timeline/shuttlebar is too small to show large numbers of edits. System is not hard to learn, but has some unusual logic. Display of waveforms is sometimes slow. Machine control currently supports fixed slow-speed jogging only. Current EDL does not display fade in/out times.

On-Line Observations: "The company is very hungry for feedback and very responsive. Its strength is that it did not try to be everything to all people. They decided to build a really good multitrack engine and that's what they have done."

Conclusions: The ProDisk system has pushed their envelope with visual editing capabilities, individual level control of every track with real faders, color support and more. And Otari has demonstrated a comprehensive commitment to the workstation environment. Anyone who needs to do serious multitrack production in the digital domain should test drive a PD464.

Bottom Line: $31,950 for a 4-I/O 4-track system.

Circle (114) on Rapid Facts Card

ROLAND DM-80

32

System Configuration: Expandable to 32 tracks, the Roland DM-80 is a slightly overdue, highly anticipated hybrid that defies easy classification. If the maximum six external drives are used, the system can deliver 12 hours per four tracks of on-line audio.

The DM-80-8 uses two disks to provide eight channels and is also available as the DM-80-4, a 4-channel system that cuts I/Os and recording time in half. Multiple DM-80-8 and DM-80-4 units can be linked via the optional DM-80-S Macintosh Track Manager which emulates a tape recorder via software.

The DM-80-R remote has function keys, track select buttons, transport controls, jog wheel, LCD readout and an input for an optional keyboard hardware controller. The DM-80-F hardware mixer interface features faders, pan pots and EQ controls. Automated mixing, via MIDI, includes level, panning and 2-band EQ per track.

System Upgrades: One improvement under development is a software resolver (anticipated for late '92), which is designed to allow the DM-80 system to slave to any source of nonsynchronous, varispeed time code.

Ancillary Markets: Audio post for video/film, music editing, sound design, dialogue editing and broadcast production.

Strengths: Roland's DM-80 attempts to be all things to all people. You can get in relatively cheap and theoretically go really big with it. In the Roland tradition, the DM-80's hands-on feel is both user-friendly (non-intimidating) and intuitive.

Weaknesses: The small onboard screen could be a problem. You can't view more than four tracks simultaneously without the Macintosh software package.

On-Line Observations: "Roland spent a lot of time working the bugs out. And the size of the company and their ability to hang with (the product) makes me secure in my decision." "We were especially interested in growing with this system."

Conclusions: The user universe is not yet large enough to know exactly where Roland fits into the big picture. But modular design and relatively inexpensive entry level price combine with Roland's sales reach to make the DM-80 a lock for significant market penetration. It feels like a home/project studio device. It even looks like most other Roland devices and that means even the least technically oriented end-users should feel at ease with it. Great place to start. No one else offers a modular system in such an easy-to-assemble package. Rates high on the price-vs.-benefit continuum.

Bottom Line: Starts at under $6,995.

Circle (115) on Rapid Facts Card

SONIC SOLUTIONS SONIC SYSTEM

24

System Configuration: Sonic Systems start with a 2-track "Sonic Mini Editor" and can expand to 24-tracks. Any Mac II can serve as the system's front-end, although the faster, the better. Control is via mouse, keyboard and the Mac's video display. NoNoise Version 2 can digitally find and remove unwanted recorded noise, including hiss, pops and clicks. DSP options include real-time EQ, varispeed and reverb.

System Upgrades: The new SonicStation includes two channels of digital I/O. Boot time has been decreased. Segment editing includes variable crossfades, movement and deletion of sounds (with 32 levels of undo), level adjustments and precise zoom in/out. The system is capable of real-time playback of any eight channels of the 64 virtual tracks displayed in a segment window. All edits, level adjustments, shelving and presence filters can be auditioned in real time. Optional A/D and D/A converters can be added and the company now manufactures its own converters. The SonicStation is expandable with four tracks per card. Options include multitrack editing, automated mixing, machine control, project management and NoNoise. The new PQ editing option combined with the Sony/START Lab CD printer allows the SonicStation to record reference, broadcast and archiving CDs plus the new PreMaster CDs for direct glass mastering.

Sonic System's waveform editing now uses 33MHz DSP chips to accelerate DSP operations and allow the system to perform real-time stereo No-Noise. The system can now record audio at 48kHz but doesn't offer analog I/Os. Multiformat digital I/Os are standard. Sync options include an LTC reader/generator and external Ampex VPR-3 and Sony P2 control. Time compression and expansion is scheduled for mid-June release (Version 1.4). Announced upgrades (fall 1992) will allow 12-channel playback with expanded memory and full MIDI support.

Strengths: The Sonic System is the only workstation with serious sonic restoration capabilities. Flexible pricing allows users to get onboard at reasonable costs. The system has extensive disk pre-mastering/mastering capabilities and on-screen mixing with full automation. Extensive DSP, filtering capabilities, real-time dynamics control and a disk-based fade editor are included. Background on/off-loading to hard disk allows users to foreground edit while loading or storing program material. Software can find the same waveform in different pieces of audio by syncing to waveform. Open architecture allows expansion as each card can handle seven SCSI devices. Fiber optic networks can be established to simultaneously transfer up to 100 audio channels.

Weaknesses: The system still lacks the ability to fully emulate a true-multitrack machine. Dynamic limiting/compression are less smooth than in analog systems. The learning curve is hindered by cluttered menus, which are sometimes confusing, and some terminology that is not intuitive.

On-Line Observations: "The new hardware, running on a Mac Quadra, is five to 10 times faster than my old Mac II system. Operations which used to take three to five seconds now occur instantly. For laying up and sliding tracks in a multitrack environment, I can't imagine a more powerful system."

Conclusions: Sonic Solution's waveform editing has a loyal following in music post-production. A more streamlined user interface would shorten the learning curve and would probably attract more users. Full moving fader automation improves the system.

Bottom Line: The SonicStation starts around $5,000. Depending on the Mac and hard disk size complete configurations range from $11,000 to $15,000. 2-channel No-Noise system prices start at $8,750 (excluding computer and hard drive).

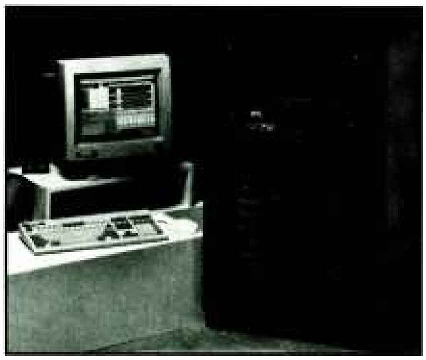
Circle (116) on Rapid Facts Card

SPECTRAL SYNTHESIS DIGITAL AUDIO WORKSTATIONS

System Configuration: The Spectral Studio's cornerstone is the FlyBy digital audio bus, which links multiple hardware components via a Windows-based software program called Studio Tracks. The FlyBy Bus Controller, a single-slot card anchors data transfers between the host PC and any Spectral hardware. The Digital Studio board combines a SCSI drive controller and a DSP chip.

A high-performance IBM PC or compatible (386 or 486) with at least 4 to 8Mbytes of RAM, color monitor (VGA or EGA) and mouse are required. Balanced and unbalanced analog connections are provided by Spectral's external ADDA-2218 18-bit A/D-D/A converter. For audio storage and retrieval, Spectral also supplies SCSI hard drives, ranging from 330 to 670Mbytes.

System Upgrades: Demand for multichannel I/Os to replace the original stereo-only configuration resulted in the release of the AudioEngine Digital Audio Workstation, which integrates digital audio for synthesis, sampling, signal processing and hard disk recording/editing. The new AudioScape software module uses the SynthEngine DSP hardware to perform real-time DSP on any signal present on the FlyBy bus. As each effect is loaded into the DSP engine, a corresponding



"rack" is displayed on the screen. Control knobs and faders emulated on these racks emulate traditional rack-mounted effect controls. Control sequencers can provide automated effect mixes and every control can be executed via MIDI and the Digital Studio subsystem now records stereo or multiple discrete sources, either digital or analog.

The Windows-based software interface StudioTracks (Version 1.4) for the Digital Studio sub-system has many added features, including 10-level Undo. By using the J.L. Cooper CS-10, you can remote control hardware, such as mixers (levels) and tape decks. Scrubbing and jogging can be accessed in the multitrack window. A special Spectral-authored Windows application for Mark-of-the-Unicorn "Video Time Piece" maximizes audio-for-video editing capacity. Up to 16 channels of digital I/O can be added by stacking pairs of Spectral's ADDA 2218 converters. The new MAX-880 rack mount peripheral offers multiple discrete I/O channels.

Ancillary Markets: CD mastering and tapeless mastering for duplication systems.

Strengths: Relatively inexpensive. Users indicate strong company support. There are more than 200 Spectral systems installed globally. High sonic quality for investment.

Weaknesses: Help screens could go further, especially for novices. After-sales training could be enhanced because the learning curve, even for advanced engineers, is steep, especially in multitrack mode. A video (hint, hint) might help, because as one user noted, "It's almost impossible to master something this powerful straight out of a manual. Sometimes it's like driving a Lamborghini and not getting it out of second gear."

On-Line Observations: "We did a large amount of looking. And there are just too many advantages to (Spectral). Because it is IBM-based, I think it is closer to the faster and faster future. Because they use Windows, most users are now ready to accept it. The part I really like is that they quickly integrate user suggestions in software upgrades. I can't say enough about their response."

Conclusions: Spectral's success stems from fusing its digital audio subsystems into high performance turnkey workstations. The company's excellent foothold could expand with less learning curve incline.

Bottom Line: A single 330Mbyte drive, 4-track system starts at $6,985 (with a computer and a DAT machine as a converter). A typical 8-track system package, including A/D-D/A converters and two 330Mbyte drives, comes in at $10,730 (without computer).

Circle (117) on Rapid Facts Card

STUDER EDITECH DYAXIS

(The following System Configuration section deals with the new Dyaxis II. The Dyaxis I system is covered throughout the balance of this section.)

System Configuration: Dyaxis II is a modular multichannel digital recorder using a multitasking CPU, DSP board with eight DSPs and four channels of analog and digital I/O. Dyaxis II systems can be configured with up to 24 input channels and 48 playback tracks.

The system is controlled by the MultiMix software package, which integrates major recording/editing tasks into one interactive window called the "Edit Desk." Five individual window "panes" interact with one other to provide track recording/playback controls, snapshot automations, multichannel waveform editing, text-based editing and EDLs. The accompanying on-screen mixer provides real time, 24-bit digital mixing and 5-band parametric EQ across all channels. Each 4-channel module can playback eight tracks in real time or can combine unlimited tracks (with sample-precise editing) in non-real time. And each 4-channel processor offers edit, gain adjust and crossfade con-

trol. Also included are event-based automation, multiformat digital I/O, internal time code synchronizer and built-in sync with a LTC/VICT reader/generator. Optical media support is provided. 24-hour technical support can be accessed via the PAN network.

System Configeration: Dyaxis I is a 2-channel recorder/editor capable of mixing unlimited virtual tracks and locking to time code with better than single subframe accuracy. Software automatically adds a 'take' suffix to files. The Event Editor provides static EQ and level change automation. EQ settings can be previewed in real time, committed to disk and instantly recalled on any file.

Strengths (Dyaxis I): More affordable than most dedicated systems. DSP includes time compression/expansion, graphic or parametric EQ and tone generation. Multiformat digital I/O locks to almost any digital source and includes a digital effects loop. Entire signal path is digital. Software allows unlimited virtual tracks and supports graphic and EDL playlists on the same screen. Any number of tracks can be auditioned or mixed to a stereo pair. System synchronizer operates as master clock source that puts out SMPTE and MTC and reads SMPTE, VITC or tach. An optional dedicated control surface uses buttons to emulate tape-based editor.

Weaknesses (Dyaxis I): DSP does not currently accelerate mixing or crossfades. Gain changes within a mix window are slow. Elements do not ripple in a playlist. One change in a mix window can cause the whole section to be reprocessed. System sometimes has problems locking to IEC-958 Type II digital sources. System has limited MIDI support (MTC only) and no machine control. Does not perform equal-power crossfades.

On-Line Observations (Dyaxis I): "Other companies are heralding visual editing with the ability to slide elements around on the screen as a new development, but Dyaxis has had it for more than four years. Virtual tracks are another big plus, because no matter how many tracks you have, it is never enough."

Conclusions (Dyaxis I): Dyaxis I is a very powerful editing system that works well in music editing and post-production environments. One of the first systems to edit on a magneto-optical disk, the system was a pioneer in the use of virtual tracks.

Conclusions (Dyaxis II): Dyaxis II is just beginning to ship, but general consensus about demos (in Europe and the U.S.) indicate this system will be a major contender in

UPDATE

the multitrack workstation market. The fact that Dyaxis II's modular expansion architecture can configure as a single large system or several small ones should fortify sales.

Bottom Line: Dyaxis I starts at under $10,000.

Circle (118) on Rapid Facts Card

AUDIO FOR FILM/VIDEO SYSTEMS

EDIFLEX SYSTEMS OPTIFLEX

40

System Configuration: Optiflex offers the flexibility of 35mm mag film with digital sound quality. A Westrex control system provides programmable motion and record/play control of multiple machines, including magnetic film dubbers, projectors, multitrack ATRs and VTRs.

Optiflex uses proprietary hardware called a Sound Engine that is based on the Motorola 68000 32-bit processor with a 68450 DMA coprocessor with a 386 PC serving as the user interface. The system provides linear PCM re-recording to a MOD and the Sound Engine system can record, playback and edit stereo or mono. Windows-based editing for each channel can be filed or stacked according to user application.

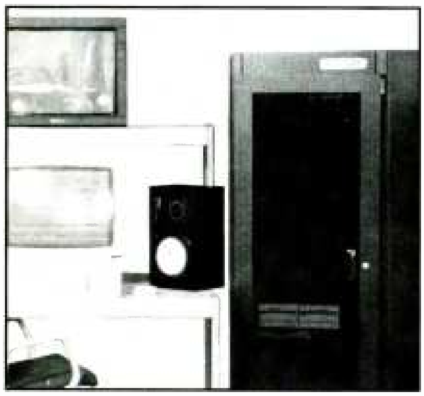
The I/O system is completely separate from the processor section and uses 16-bit converters with Apogee filters. Each MOD holds 55 minutes of digital sound per side. Maximum total recording time for the system is 440 minutes.

For film sound systems, track-by-track offset capability provides for sync adjustments on the mix stage. Alternate tracks can be loaded at will. Unlimited multiple takes can be recorded on interchangeable media. The system can read and conform to CMX I Ampex EDLs. Ediflex's Audiflex and Optiflex systems can be used together to provide multitrack record/playback.

Systems Upgrades: Expandable to 100+ tracks.

Ancillary markets: Theme park systems.

Strengths: Instant rollback and short preroll maximizes time in the recording session.



Weaknesses: The only Optiflex problems uncovered in our research were non-specific but conform-related. (Investigate this if you are considering this system.)

On-line Observations: "Editing (on the Optiflex) is the best. My tracks are so much better and faster that they save money everywhere else in the process. It has great waveform displays. You don't have to scrub to find mistakes in the background."

Conclusions: The Windows-style editing creates relatively quick learning curves. The company's significant inroads in both the audio-for-film and theme park markets places the Optiflex within the "specialty" workstation universe.

Bottom Line: $61,500 for a 4-channel stereo system.

Circle (119) on Rapid Facts Card

ELECTRONIC DIGITAL INNOVATIONS PTY., LTD.

EDI-TRACKER

8

System Configuration: This Australian system is aimed at film editors and uses traditional film sound terminology and functions. Previously called Soundtracker, all EDI-TRACKER recording and editing can be done on the same screen, although there are separate screens for each. Work is organized into jobs, subjects and reels with 99 editing tracks per reel. As many as eight tracks may be accessed at any instant and each cue on any of these eight tracks can have automated level and pan. There is an internal 8x2 mixer.

The system auto-assembles from a CMX-type EDL, and a multi-user system allows up to four different operators with different sync references to share a common resource of disk drives and a sound library. Project management can be organized by the off-line spotting system provided.

Proprietary hardware and user interface plus IBM 386/30MHz PC with 2Mbytes RAM. A 4-channel module can expand the entry level, 4-channel system to eight channels. The multi-user model supports up to six 4-channel modules and four workstations. Each workstation features a pressure-sensitive touch screen with plasma display, dedicated function keys, transport controls and two jog wheels for scrub, level, pan and cursor control. Synchronization is provided for LTC, VITC, wordclock, video reference, film chain quadrature and pilot tone.

System Upgrades: Dynamic equalization and mixing are planned.

Strengths: Although not a workstation for all applications, the EDI-TRACKER is among the slickest looking players in the audio-for-film market. It does what it does well and can perform in a multiworkstation environment.

Weaknesses: EQ functions aren't yet operative, and a color monitor is not supported.

On-Line Observations: "The touchscreen matches traditional technology so I can immediately see where I am." "We have to compete with the world. In the Aussie film industry we recover (gain profit) offshore, plus TV is getting faster and harder than ever, and the EDI-TRACKER price allows us to remain competitive." "For picture/sync, it speaks our language. Our operators successfully used it on the second day. It is a cutting room tool. For the price nothing can beat it."

Conclusions: Due to lack of availability in the U.S. we were unable to reach any hands-on conclusion. The lower-than-most entry price for this system should attract potential buyers for the relatively expensive audio for film/video market.

Bottom Line: $50,000 for system with 3-hour hard drive, MOD and four o/p's.

Circle (120) on Rapid Facts Card

FAIRLIGHT ESP MFX2

24 SRC

System Configuration: The Fairlight MFX2 is a digital 24-track disk-based system using hard and soft control function keys for multitrack editing. Recording and editing take place on the same screen and cues may be edited across all tracks. The MFX2's integrated video transport control and "Audio Freeze Frame" feature provide synchronization of SFX to video and can play one audio frame at normal pitch down to zero speed. Integrated sampling, cue list and music sequencing are available from Fairlight's Series III package.

A proprietary "Turbo SCSI" disk drive interface can provide up to 16 continuous audio tracks per single hard disk. And six 1,200Mbyte hard disk drives may be connected to create more than 20 track hours of storage on-line. An optional MOD allows simultaneous stereo recording/playback and instant sound archiving access. External VTRs can be scrubbed simultaneously with MFX audio. Replay channels can be extended to 16 by 2-channel modules. RAM may be extended to 32Mbytes and 16 cues can be replayed simultaneously. Synchronization includes MTC and FTC.

System Upgrades: The MFX's new EDL Package was jointly developed with software video editing specialist Digiteyes (authors of Shotlister) and can command the MFX2 to capture off-line video edit information. Shotlister can then load EDLs from disk, sort them into reels and then into chronological order. An EDL can then be loaded/conformed simultaneously through a series of edits and formatted into most on-line standards.

Ancillary markets: Music recording/editing and sound design.

Strengths: For the audio-for-video/film post workplace the MFX can manage complex SFX. The zoom-in, zoom-out interface seems intuitive. Sync-to-video is strong and Fairlight's experience in sonic excellence puts

the system in the ball park.

Weaknesses: Although the MFX is in 15 countries it has little North American penetration. "Sound clip" location/organization is "managed" through dedication of certain tracks to certain SFX (i.e. track #2=various small arms fire). Until recently the MFX was incapable of tracking which clip is where without listening/viewing a track until the desired sound rolls by. A text entry search-librarian is either finished or nearly complete. See this function yourself to be sure.

On-Line Observations: "Even first-time users are effective with the MFX2. The learning incline isn't too steep." "We went with (MFX2) because we had already invested in the Fairlight CMI Series III, and wanted to preserve our investment. We have no regrets." "You rarely concern yourself with time code numbers to enter cue points. It's all done visually on the screen."

Conclusions: Historically, Fairlight worked the high ground of sampling technology. Few sampling keyboards ever sounded better. Recently, Fairlight has struggled. But the heritage of those pioneer sampling machines should combine with the high speed editing and strong MFX2 user interface to invigorate Fairlight's market penetration.

Bottom Line: $70,000 and up.

Circle (121) on Rapid Facts Card

SSL SCREENSOUND

System Configuration: ScreenSound is the only system to interface via wireless pen and tablet. One of the first systems to offer machine control, ScreenSound employs an all-digital signal path plus static and dynamic automation. Multiple units can be slaved to one master for additional track capabilities.

System Upgrades: SSL has added a number of enhancements, including autoconform software with EDL import and support for MODs. The system records up to 16 tracks, although only eight can be played back at once.

Strengths: The only system with disk-sharing capabilities and an automated SCSI patchbay that supports up to 16 devices. SoundNet provides central mass storage available to a group of users. Operating system retains the directories of all hard drives and off-line optical libraries. Built-in database performs multiple keyword searches. Disk to disk copies are 7x faster than real-time. As far as we know, the only system that can name a file while recording.

Weaknesses: System is expensive considering relatively limited DSP options are available. Reel names as viewed in Windows can only be five characters long. Only two channels can be recorded simultaneously. SoundNet is a costly option and will not allow two users to access the same disk at once.

On-Line Observations: The system is very easy to use. Once you use it, you can't go back to an analog system." "A great advantage is how clients understand the system interface, and can attend a session without constant explanation."

Conclusions: In ScreenSound, SSL has created a totally digital production system with a one-of-a-kind approach. Powerful machine control and an easy-to-use interface

has helped the ScreenSound achieve significant market penetration in post-production and film environments.

Bottom Line: $89,000 for a 3-hour system.

Circle (122) on Rapid Facts Card

WAVEFRAME AUDIOFRAME

System Configuration: Using a mouse-controlled 386/486 PC and outboard processing card rack, disks and I/Os, the original AudioFrame had five major "building blocks," including SoundProcessor (RAM-based recorder, 16 to 48 tracks); EventProcessor (triggerable EDL); StudioCAD (16x4x2 on-screen virtual mixer with EQ, reverb and other DSP functions); DRM (disk-based recording/editing, with three to eight discrete tracks depending on disk size and use of 16- or 24-bit storage); and Texture (MIDI sequencer). Analog I/O was optional between two to 32 channels of XLR inputs, and eight to 32 discrete XLR outputs.

System Upgrades: WaveFrame recently reconfigured and renamed its AudioFrame/CyberFrame systems. The AudioFrame is now the WaveFrame 1000 (with 10 slots) and CyberFrame has become the WaveFrame 400 (with four spare slots). The highly modular design allows Windows-compatible software from both WaveFrame 1000 and WaveFrame 400 to be integrated within one system.

The new system performs timefitting and the on-screen mixer now includes digital inserts. The 400 supports MODs, and software functions, such as digital filters and screen redraw, now load much faster. Windows 3.0 delivered system improvements, memory limitations are history and macro-software reduced function command redundancies. WaveFrame software also works while other Windows applications are running. Write-once CD capabilities allow users to master straight to a CD recorder. Machine control of the VTR is possible from alphanumeric keyboard.

Ancillary markets: Music production, pre-production and editing, sound design, and CD pre-mastering.

Strengths: Allows manual punch-in capabilities. System architecture allows routing of digital audio from the sampler through the mixer and to the disk recorder via digital audio busing. The system now contains an on-screen mixer, offers more than 200 virtual tracks and generates time code (based on disk recorder location is in a chain) so it can serve as a master. The mixer and playlist are user-customizable as are assignable roving meters with a clip history. There is a built-in database for music and SFX libraries.

Weaknesses: System currently is limited to eight disk-based tracks and suffers from a long boot-time. The 1000 has optical media support. Tighter integration between the sampler and disk recorder would improve performance. The disk recording software still lacks any waveform display (editorial software has this). You still can't have MIDI and disk-based events in an EDL and mixer automation is only available via MIDI.

On-Line Observations: "The software is becoming compatible across the entire product line. Now you can run editorial software

on the 1000 or mixer software on the 400. If you don't need a sampler, you can use a 400 instead of a 1000. Because you can save your mixes, you can always return to any previous state. I'll go back to the mixes from two or three days ago and it sets up everything in the room, including DMP-7 mixer parameters for the synths. Very precise recall."

Conclusions: The Series 1000 is now a contender for any post-production or music application. Lower-priced configurations should increase AudioFrame user-base. The card-cage architecture will simplify upgrades.

Bottom Line: Series 400 systems start at $22,500.

Circle (123) on Rapid Facts Card

The Compellor is the best way to even out levels from the same or different sources. This combination of compressor, leveler and limiter, sounds as if someone is riding faders extremely well — controlling level without any impact on short term dynamics.

Already in use in thousands of production, broadcast and installed audio systems around the world, the Compellor is now even more attractive to audio professionals. The Model 320 Compellor features *dual monaural circuitry* to provide you with two independent channels of the best dynamics control available which can be linked two ways for stereo operation.

The Compellor uses patented control circuitry to make its processing "invisible" regardless of type of program. It's easy to set up. The Compellor then self adjusts to the dynamics of the input, providing complete dynamics control ... smooth, inaudible gain riding for consistent levels ... all automatically.

Features of the Compellor Model 320 include reference level switching from the rear panel; Leveling Speed switchable from the front panel; Peak Limiter defeatable from the front panel; two remote controllable bypass relays.

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Circle (17) on Rapid Facts Card

TRACKING

The AKAI DD1000 Version 2.0

By Alan Parsons

The Akai DD1000 now features Timestretch and DAT backup of all data.

In September 1990, Akai first released their flagship DD1000 optical disc recorder. Two significant software updates have been released since. Engineer/producer Alan Parsons has been working with the system for a year. Here he updates us on the latest version.

Akai seem rather proud of the fact that they listen to their users, and don't seem afraid to incorporate outside suggestions into their products. Communicating with the Japanese has never been easy — they never let on when they don't understand you. It's a comforting thought that the development team for the DD1000 is U.K.-based and speak real English!

As well as getting a fair amount of use from two DD1000 units in different locations, I have been in regular touch with Akai to discuss updates and dare I say it, the occasional bug. They have packed a wealth of new facilities into the machine in the last year, many of the ideas emanating from their broad base of owners (such as myself).

With stunning originality, the latest update is to be called Version 2.0, though the version I have installed in the machine in front of me as I write is a pre-release.

Alan Parsons is best-known for his work with The Beatles, Pink Floyd, Wings and his own Project.

THE BASIC FACILITIES

For those of you who are not yet familiar with the machine, here is a very brief breakdown of its basic functions:

- 30 minutes of stereo 16-bit recording time at 44.1kHz on each side of a Sony removable magneto-optical disk cartridge, slightly larger than a CD case. 48kHz, 44.056kHz and 32kHz also available. 24-bit internal processing. Each take on disk can be named.
- Stereo balanced XLR inputs and two balanced stereo outputs allowing 2-track record and 4-track playback. AES/EBU in and outs compatible with S/PDIF from most DAT machines. Optical interface, RS422, SCSI, Centronics port for printers, two footswitch inputs and of course MIDI.
- Edit cuts mode, with non-destructive editing of up to 50 cuts for each take. Cuts can be entered on the fly.
- Playsheet mode, that allows instant playback of up to nine takes or cuts by pressing the buttons on a numbered keypad. No more than two takes can sound at once.
- Song mode, allowing takes/cuts to be chained as on a sequencer.
- Cue list mode, a series of takes/cuts, fades and/or MIDI cues that interact with other pages to allow a sequence of audio events to be assembled.
- External synchronization to SMPTE/EBU time code, PAL/NTSC video or digital word clock. The machine will also sync to MIDI time code and start from incoming MIDI clock.
- Time display options in all four SMPTE/EBU formats, feet/frames, sample points or bars beats and clocks.

The unit is 5U high, has a 6-line backlit LCD, several dedicated function buttons and a rotating jog-wheel. An optional full-function remote controller — the DL1000 — or the much less comprehensive and cheaper DL500, intended only as a remote playsheet controller, are now available. Front-end software is also available for the Apple Mac.

Akai's first update, Version 1.3, allowed SCSI connection to the DL1000 remote controller. Akai also specified a jumper wire to be soldered internally on an inaccessible board that entailed the laborious process of taking all the XLR connecters off the back panel to get to it. Luckily, inserting updated software chips is a great deal easier. The pair of software ICs is easily replaced in a couple of minutes after removing the outer shell of the unit.

Version 1.3, as well as ironing out a few bugs, implemented quite a few new features:

- Seamless recording across four disk drives, for longer recording times using additional DD1000 units or external disk drives.
- Various improvements to the Qlist mode including auto Qlist creation after record. This eliminates a rather cumber-

some process of having to compile a list of instructions before anything can be heard on the Qlist page.

- Improvements to edit cuts mode including a spool function, that allows emulation of a tape deck jog-wheel spooling across the heads.
- MIDI synchronization in song and playsheet modes, and improved graphics and displays while playing back.

The manual for version 1.3 consisted of a number of update pages to be inserted in the supplied loose leaf binder included with version 1.0. The result was unfortunately a complete shambles of repeated pages and incorrect cross references — pity. Only with this latest update did I receive a complete corrected version of the 1.3 manual. Now of course, I need a Version 2.0 manual — picky picky! Thankfully, the 69-page update documentation is well written, but it will be tough for new owners having to refer constantly to both documents. With luck the complete 2.0 manual will be ready soon. (An aside here — manufacturers should ensure that their distributors are supplied with as many copies of update documentation as there are units sold to them. We live in a 'soft' age.)

THE LATEST UPDATE

Onward to the new capabilities of Version 2 0. I will also be referring to updates implemented on version 1.3, but won't bother to draw attention to when each update actually occurred.

By pressing a soft key marked INIT, blank templates can now be called up for Qlists, songs, and playsheets, so that a new page of settings can be started from scratch. Until now, it was necessary to copy an existing file, modify each item one by one and then save it.

There is also a new facility to save several main setup files, therefore your different 'projects' (good word that) can have their unique synchronization, digital sampling rates and display parameters saved onto disk.

One complaint in the original version was that entering numerical data was a bit fiddly. A new function displays times with an underline cursor (as opposed to a highlight), that can be moved around a numerical field, and entries can be easily modified with the + and – keys.

Many users found that there was a danger of inadvertently erasing a valuable take on the record page, because the most recent recorded take remained armed for re-recording. A new option on the record screen called auto take name incrementation prevents this from happening. After recording say, take 1, the display shows take 2 as the next item to be recorded. The machine is intelligent enough to know that if for example take 6 existed from an earlier session, it would jump straight from take 5 to take 7. Take titles do not have to be "Takes" of course. "Alan 1" would increment to Alan 2 and so on. All takes, cuts, songs, Qlists, in fact anything, can be renamed on the disk page.

I recently discovered a bug in the record mode. If the time display is set to bars beats and clocks on main setup, and MIDI time code is the record time log source, recording on the DD1000 is impossible. Furthermore, Qlists cannot be played with these settings from any page. The same problem occurs when time display is SMPTE 29.97 ND, SMPTE 30, SMPTE 30 drop and sample points. However, time displays of EBU, SMPTE 24, and feet/frames seem to be fine with MIDI time code start.

There is a new time code option — 29.97 non-drop. If I may quote from the manual, "This is for NTSC users who wish to use a non-drop 30fps style time code for contiguous frame reference but whose time code is synchronized to NTSC." Who am I to question the utter indispensability of such a feature, as a mere mortal living in the United Kingdom, world of 25 frames and PAL?

In version 1.0 it was not an easy matter to compile a Qlist from a newly recorded take. This meant that a lot of button pushing had to take place before a playback of a take could be referenced to external time code or MIDI. By switching the new function 'Qlist after recording' from its default of none, (changed since V1.3 in response to users' requests), to 'create new Qlist,' a cue list is made with the last take placed in the correct time position relative to the external source, if any. The Qlist name defaults to the take name. Two other 'Qlist after recording' options, namely 'add to Qlist and sort' and 'add to end of Qlist' can also be selected. The former will place the recording in the correct time position, and the latter will place it at the end of the Qlist regardless of its time position.

Edit cuts mode has a useful new auditioning function which lets us hear the take up to the start of a cut. Experienced users will also note that the start mark, end mark, and general purpose mark keys now only move the cursor without pasting in the last time reference, when the virtual "tape" is stationary.

Another change that will be appreciated by experienced users is the fact that the delete soft key for Qlist cues and song

entries is no longer a "repeater." Before this update it was too easy to erase a whole bunch of cues without a moment for thought. A prompt now checks with us to make sure that we really want to delete each separate cue.

Playback of takes in disk mode is now possible, which means that we can hear a take before deleting it, for example.

Another new facility since version 1.0 is the activation of RS422 9-pin connections for video and audio visual applications. A VTR equipped with RS422 can be played, rewound, jogged and shuttled from the Akai, and times can be 'grabbed' for spotting effects and lip-syncing. Recently, Fostex audio machines have introduced RS422 control, which could mean some interesting applications. Note that RS422 and MIDI cannot be used at the same time. Finally, every cue in a Qlist can now be routed to one or both outputs.

TIMESTRETCH

There was a time when the very concept of being able to alter the length of a recording without affecting its pitch seemed an impossible dream. Even *Hal* being slowly deactivated and grinding to a halt in perfect tune as he sang "Daisy Daisy" in Kubrick's *2001:A Space Odyssey* back in the '60s failed to bring us a realistic commercial product resulting from the clever but primitive rotating head techniques that were used for the film. Now in the age of digital processing and real time pitch shifting comes another digital miracle in the form of *Timestretch*.

AN UPDATE

For a number of years, studios have had the ability to timestretch (after a fashion) with existing equipment . All they needed to do was slow down or speed up the tape so that it ran for the required time, usually by trial and error, and then use a Harmonizer or similar box to restore the original pitch. Of course, they probably had to put up with the inherent problems exhibited by all pitch shifters.

Broadcasters are going to love this ...

Although not the first with the technique, Akai have experience with 'proper' timestretching on their S900/1000 Series MIDI samplers. With the DD1000's much greater storage capability, however, we can get into some serious time warps. Rather like regular pitch shifting, perfection is 'hard to achieve,' but provided the amount of Timestretch or "Timeshrink" (my word, not Akai's) is not too great, very acceptable results can be obtained.

There are 18 different preset modes for the timestretch function, each named according to its suitability for a particular application; e.g. speech, dance, perc, rock, etc. Each mode has three variations giving a total of 54 different settings.

Timestretch involves a copying process, leaving the original intact. There is a real time 'check' algorithm that allows us to verify the new length, which lets us hear a rough approximation of the effect that will be produced in real time, and a 'commit' algorithm of higher quality which takes slightly longer than real time. An advantage of having plenty of storage is that several versions with different settings can be committed for comparison.

Alan Parsons from a different angle, at home, at work.

We are given the choice of adjusting the amount of Timestretch with either a fixed new start time, or a fixed new end time. The timestretched length is displayed along with the start and end times of both the old and new versions, so it is very easy to adjust for the required amount of time change. A unique facility not found on other devices is the possibility to alter the timestretch factor *and hear the result as the take plays.* A new take name is assigned for the timestretched version, and the commit soft key starts the process.

Timestretch can be applied to Qlists, songs and playsheets. If bars beats clocks is selected as the time display, stretch factors can be calculated according to tempo. Remix engineers will find this useful for matching tempos between two or more sources.

EDITING DURING RECORD

Believe it or not, this means exactly what it says. While the recording to disk is actually in progress, pressing the end mark key at appropriate moments will create 'block edits' through the take. Each time this key is pressed, the display moves on to the next cut number. For example Cut 2 is automatically butted up to Cut 1 and Cut 3 to Cut 2 and so on, unless a new start time for the cut is specified by pressing the start mark key. This means the most rapid removal of unwanted material I have ever seen. No sooner has the recording been finished, a Qlist can be created automatically to play the cuts in order and we can hear our edited version. Broadcasters are going to love this — imagine removing coughs and page turning during a performance! Of course on the fly edits can be a bit hit and miss at the best of times, and you have to have your wits about you to get a series of them right in one pass, but they are easily trimmed up afterward using the normal edit cuts page facilities. Up to 50 cuts are possible for each take.

Think of it this way — Supposing I record someone counting from one to ten. As he is counting with the DD1000 in record, I could press the start mark key after "one" and end mark after "two," (Cut 1). I then press the start mark after "three" and the end mark after "four," (Cut 2). In a similar manner I could mark the start and end of each alternate number until the final "ten" making a total of five cuts. I stop recording and with auto QList set to on, I press play and out comes "two, four, six, eight, ten" — utterly brilliant.

... imagine removing coughs and page turning during performance! ...

Occasional clicks were heard on butt joints. I tried to cure these with the crossfade facilities, but didn't have any luck. Akai are aware of the problem, and will presumably be addressing it. Akai promise that a similar facility will be implemented soon for the edit cuts mode, so that we can do the same with material already recorded.

SEQUENCE MODE IN PLAYSHEET

Again, for those unfamiliar with the DD1000, the normal mode of Playsheet gives us instant play of nine different takes/cuts from the numerical keypad "on the buttons" as it were. Broadcasters use this as the equivalent of nine 'cart' machines. The 0 key calls up the next playsheet in a compiled list. The new Sequence mode is a real winner. It is activated on the setup page in playsheet mode. Assuming the playsheet has a number of takes/cuts assigned to the buttons, we can make a series of keypresses representing a "running order" of different takes as a playback sequence. As soon as the first one has finished, the next one starts, until the end of the programmed sequence has been reached. The duration of a sequence can range anywhere from a few seconds to several hours if long takes are repeated. Items can be added at any time during playback without interrupting the sequence.

I found that for sequencing an album, the gaps between songs sometimes might not be long enough. Then I had the idea to record a take of digital silence which I trimmed to last exactly 1 second, and assigned this take, which I called "gap," to a key on the playsheet. Presto — for a 3-second space between songs all I had to do was press the gap key three times; for five seconds, five times and so on. It's a far cry from my days as a second engineer when a sequencing session meant sticking in little bits of leader tape until the spaces between songs sounded right.

SAVING DATA TO DAT

It was always possible to back up audio from the DD1000 via the digital outputs, but now a welcome archiving facility for *all* the data has been added. Although the manual states the intention is for DAT archiving, there is no reason why backups should not be made to 1610/1630 or any other format supporting the AES/EBU digital interface.

... of course on the fly edits can be a bit hit and miss at the best of times ...

We are given the choice of archiving the "whole disk" (meaning one side of the disk, not both sides), "all takes," "all songs," "all Qlists" or "all playsheets." Other archiving possibilities also exist, such as Qlists or songs with takes, or just single takes on their own.

Takes are sent to the AES/EBU outputs as regular digital audio, and can be monitored during the archiving process through the analog outputs, but *all other data, e.g. edit and setup information, is also sent to the analog outputs*. It is extremely nasty-sounding, and could easily damage speakers (and ears) if inadvertently played too loud. It can be distinctly heard as a series of three bursts of data. (Each byte in a file is sent three times, and the file itself is also sent three times, giving a nine-fold redundancy of data.) The audio itself is only output once.

There are three data rates available for archiving that coincide with the available sampling rates of 48kHz, 44.1kHz, 44.056kHz and 32kHz. Recordings made at 44.1kHz and archived at 48kHz therefore take a little less than real time to be backed up. As would be expected, a comprehensive set of restore functions gets the data from tape back into the DD1000.

One of the reasons I bought the DD1000 in the first place was that it was a self-contained unit requiring a bare minimum of setup. In spite of quite a bit of forward selling by manufacturers and distributors of Mac- and PC-based hard disk recorders and editing software, I am still convinced that I made the right decision for my own purposes. As well as using the unit extensively in the studio as a mixing medium with rapid access to different songs and for re-structuring demos, I have been using it for the recording of technical and musical material for a forthcoming test LP for CD and DCC.

Since the end of 1990, I have had a unit in daily use as an effects source for a stage show in Vienna called *Freudiana* in which I am involved. Of course the March AES in Vienna was a perfect opportunity for you all to go and see the show (plug plug) and see and hear the DD1000 in use. All effects and pre-recorded music loops are keyed from playsheets that are called up by MIDI program changes sent from 'scene' commands on the theater's main console. A MIDI note sent from the unit within a cue fires up a laser machine for the finale.

I have found its MIDI capabilities useful, but not without problems. One is a bug that prevents incremental changes in BPM settings having any effect on the MIDI song position until a round number of BPM has been reached. In other words 123.0BPM gives the same song position figures as 123.9BPM.

Although the DD1000 recognizes MIDI clock and song position pointers from sequencers, it does not actually *chase* MIDI clock. It senses its presence, but once started will not respond to any tempo changes or variation in the clock rate. This means that syncing to analog tape with a SMPTE to MIDI box will cause a sync drift, because no analog tape transport running wild is stable enough to perfectly reproduce a recorded item without causing it to drift slightly from its original duration.

... and you have to have your wits about you to get a series of them right in one pass.

The same problem occurs with MIDI time code — once the DD1000 has located the take or cue to the received time reference, it plays at exactly the original recorded speed and is not tracked by the incoming MIDI time code. I have drawn these difficulties to Akai's attention. Unfortunately, the ability to chase MIDI clock and MIDI time code would necessitate hardware that the DD1000 does not have available. Their recommendation for syncing to analog tape is to use SMPTE/EBU time code to run the whole system, in which case the SMPTE bit clock is faithfully followed and works just fine, but SMPTE is usually not available at the early stages of a MIDI-based production. Digital tape formats, and analog systems *locked* to a stable SMPTE/EBU time code, should not give any problems driving the DD1000 via MIDI.

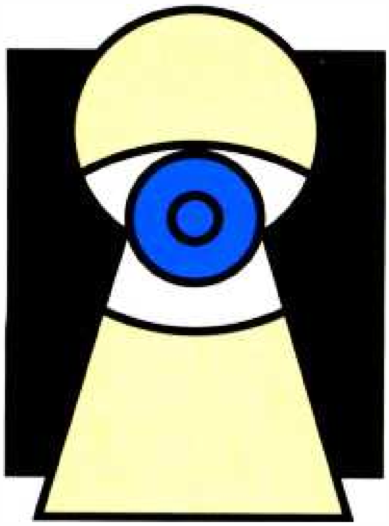
I do not advise syncing sequencers or drum machines to the Akai with MIDI clock. MIDI time code works much better. Different manufacturers have different algorithms to calculate a song pointer at a particular tempo, so the DD1000 might locate to a slightly different point than the sequencer when started in mid-sequence.

DIRTY BUGS

A few weeks ago I had a couple of annoying drop-out problems when I was playing back old material. I delved into the manual to look for a trouble shooting section. I found a paragraph about disk care that read, "Read/write errors may result from stains or marks on the surface of the optical disk." Following the instructions, for the first time I carefully opened the disk shutter to look at the disk inside — Yuk!! I had got into the bad habit of leaving the optical disk in the drive overnight. In fact there was a disk inside for about three weeks at Christmas. This disk was *thick* with a grimy layer of dust. The disk shutter is open inside the machine, and being stationary for any extended period, it is more susceptible to dirt accumulation. The story has a happy ending — I have had no further error problems since carefully cleaning my disks as the manual instructed, but the moral of the tale is 'disks don't like dust,' so always replace them in their plastic cases after each use. It's also a good practice to clean the optical drive lens with a special Sony MOA-L55 cleaning cartridge from time to time. I think Akai should supply one with the machine.

If you are going to be getting a lot of use from the DD1000, you may want to think about forking out the extra amount for the DL1000 remote controller. Unless the main unit is at eye level, it can be pretty tiring working with it for long periods of time. The Macintosh front end is yet another option, of course.

I'm pleased with the way the DD1000 sounds, and I still feel that there is a tremendous amount of power packed into the comparatively small space that it occupies — even more so now that it sports these new facilities. Some of Akai's 'tricks' can't be matched by other machines at any price. It will not be long I'm sure, before some of the other Japanese majors will be hot on the trail with their own recorders using the removable optical disk concept, if they aren't already doing it by the time you read this! ■



Part Two: Avoiding RFI In Multi-Channel Wireless Microphone Systems.

SECRET KNOWLEDGE OF THE ETHER

By Bruce Jones

In last month's Secret Knowledge of the Ether, Part One, we addressed the various sources of radio frequency interference (RFI) in wireless microphone setups. Multiple order intermodulation and spurious emissions were discussed. In this month's article, we will cover RFI interaction in systems, and suggest numerous means and techniques for isolating, identifying and solving wireless microphone RF problems. We'll pick up where we left off — discussing the performance of systems.

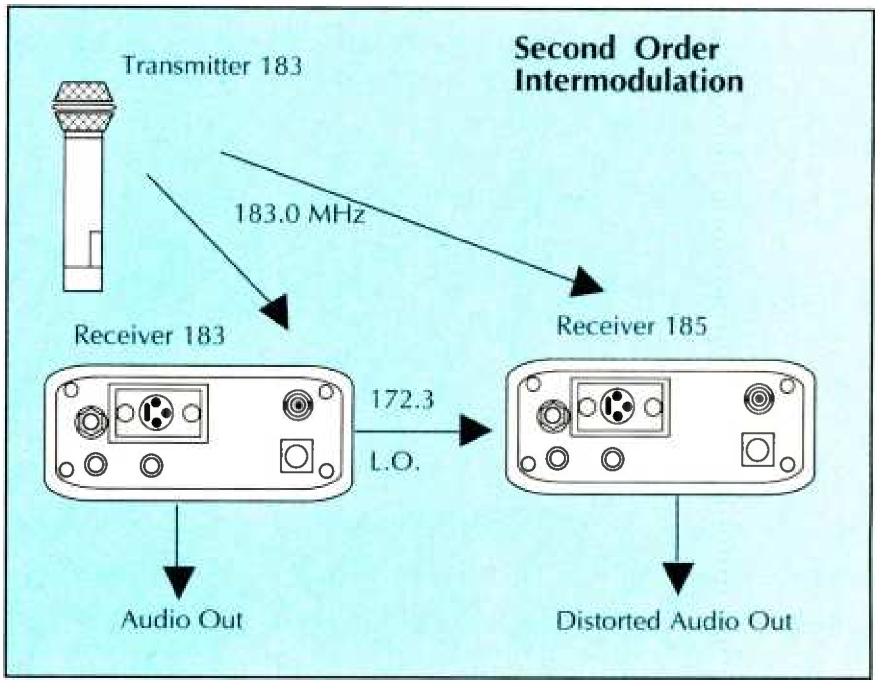
RFI IN SYSTEMS (RECEIVER/TRANSMITTER COMBINATIONS)

Second order IM is generally easy for a receiver to reject because the signals required to generate a problem must be far from the operating frequency of a receiver and can be easily rejected by the front-end filters. Remember that the frequency of a second order IM signal is produced by the simple sum or difference of the frequencies of two other signals. For instance, to generate an interfering signal at 185MHz would require two signals either 185MHz apart or two signals that would sum together to produce 185MHz. Mathematically, at least one of the signals must be at least 92.5MHz from the receiver's operating frequency (half the carrier frequency), which is easy for even standard front-end filtering to reject.

Second order IM can be more of a problem if you have two transmitters that are separated in frequency by the IF frequency of the receivers (commonly 10.7MHz). For instance, if you have transmitters at 185MHz and 195.7MHz, the difference is 10.7MHz. This difference in signal may interfere with any receiver with a 10.7MHz IF operating within 5MHz to 10MHz of these frequencies.

For instance, a receiver at 193MHz is only 2.7MHz away from 195.7MHz and only 8MHz away from 185Mz. A standard front-end may have only a few decibels of rejection for signals this close in, and the signals will pass through the front-end to the mixer stage, which will generate a 10.7MHz signal from these signals. Note again that it makes no difference what the receiver frequency is. As long as it is close to, or between 185MHz and 195.7MHz, it will have a problem with these two transmitters.

Obviously, you don't want to have transmitters spaced at the IF frequency



Bruce Jones is vice-president of marketing for Lectrosonics, a manufacturer and supplier of high-quality wireless systems of any receiver. Receivers with a highly selective front-end and high level mixers will help prevent this problem. Again, proper frequency coordination will alleviate this problem.

A subtle problem that can also occur in any multi-channel wireless system, no matter what operating frequencies are chosen, is similar to receiver crosstalk, as discussed earlier. Let's assume that two wireless systems are operating at 183MHz and 185MHz (you can choose any pair of frequencies as long as they are within 10MHz or so of each other). Now assume that both receivers have IF frequencies of 10.7MHz (the most commonly used). We'll designate the two parts of these systems as 183 and 185 to keep them straight. The basic problem is illustrated on page 40.

Receiver 183 has a local oscillator (LO) frequency of 172.3MHz (183 - 172.3 = 10.7MHz). If this local oscillator leaks into receiver 185, there would appear to be no problem since any competent receiver can reject a signal 12.7MHz off of the 185MHz frequency. But, a problem appears anyway when transmitter 183 is also on. Transmitter 183's carrier and receiver 183's local oscillator combine in receiver 185's mixer to produce a 10.7MHz signal (183.000 - 172.300 = 10.7 MHz). The problem is that both receivers will respond to the same transmitter, even though they are on different frequencies.

There are thousands of calculations that must be made in order to select a group of frequencies that are usable for any particular location.

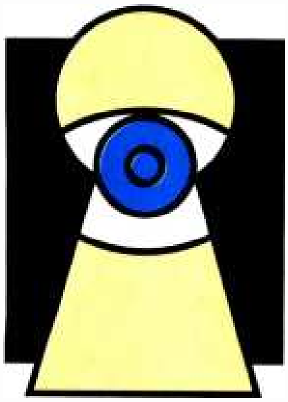
The reverse can also occur: receiver 185's local oscillator will be at 174.300MHz and can combine with transmitter 185's carrier to produce a 10.7MHz IF signal in receiver 183. In a well-designed receiver, the local oscillator leakage will be small and this problem will only show up when the corresponding transmitter signal is strong. If the receiver has a high selectivity front-end, this problem will be further reduced. Because the two interfering signals are spaced apart by the IF frequency (10.7MHz in this example), at least one of the signals will be attenuated by a high selectivity front-end.

To test for this problem, turn on all the receivers, hooked up exactly as they will be used, and turn on the transmitters one at a time. The transmitters should be about 10 feet away from the receiver antennas. The matching receiver will unsquelch (RF lamp comes on) of course, but watch for other receivers also unsquelching. If one or more other receivers unsquelch, turn off the receiver that matches the transmitter. If the other receivers then squelch properly when the matching receiver is turned off, you have this specific interference problem.

For a solution, you can try moving the transmitter farther away. At the actual operating distances this problem may disappear. If it remains a problem at 30 feet or more, you may need to make major shifts in system frequencies. Small shifts will not solve this problem, because the interfering signals are already 10.7MHz or more apart. If one receiver and matching transmitter cause all the problems, it is probably excessive local oscillator radiation from that receiver. You can simply try replacing it with a different receiver and transmitter.

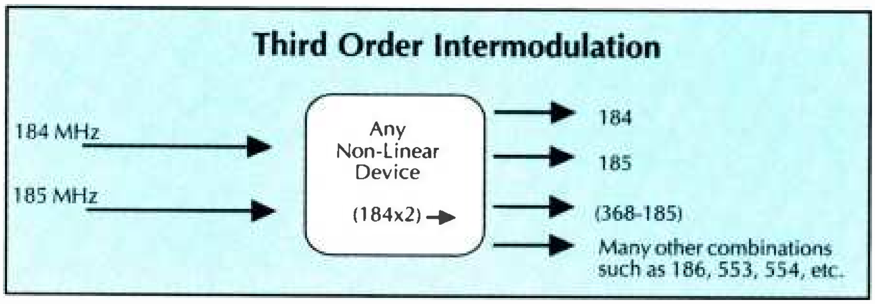
Some ways to reduce this problem are: Use antenna combiners that isolate the receiver antennas from each other; use receivers with low local

Circle (18) on Rapid Facts Card



oscillator radiation; use receivers with highly selective front-ends; or separate the receivers by several feet or more.

A basic problem that often occurs in multi-channel wireless systems is a matter of third order combining of the carriers. To illustrate this problem, assume that you have wireless systems on 183, 184, and 185MHz. The RF



front-ends of the receivers will provide only a small amount of attenuation because there is only a 2MHz spread between the frequencies (1.2%). High quality receivers can easily separate the frequencies in the IF filter section. Assume that the signal from transmitter 184 produces a 2nd harmonic (2× 184MHz) in the mixer stage of receiver 183. The signal from transmitter 185 (which also gets into receiver 183) is subtracted from it and the resulting signal is just as valid as the signal from the 183 transmitter. Obviously, you can prevent this problem by changing any one of the three frequencies. In a large multi-channel wireless installation, things aren't quite so easy to fix, because the possible combinations become mind boggling. In a 24-channel system there are 552 third order intermodulation products. Changing one frequency to get rid of one interference problem can create five new ones.

If you also include the case of three transmitters getting into a receiver, the calculations become even more tedious and a computer program becomes absolutely necessary. To make matters even more unpredictable, the wireless system carriers can also mix with signals from external sources, or the external signals alone can mix to generate the same sort of IM problems. It is virtually impossible to predict all the combinations that could occur in any one geographic area. The best advice is to use only receivers offering very high IM suppression and very high selectivity. A frequency coordination program is a must for any medium to large multi-channel system of six to 24 channels.

SOLUTIONS TO RFI PROBLEMS

Because most multi-channel wireless mic systems utilize inactive television channels, the geographic location must be determined at the outset. Television stations in the vicinity must be taken into account first, followed by at least several "runs" of a computer program to determine what sort of IM problems might occur within a particular group of possible frequencies. The better computer programs include automatic selection and testing of frequencies within available bands, followed by a report of the results. In most cases, choices have to be made between the frequency groupings with the fewest overall IM problems, and the practicality of delivering what is available on time, given the particular sound system application.

By conducting a "survey" of several manufacturers, you should be able to tell which ones really understand the common problems of multi-channel wireless systems and can offer solutions.

Even if you have some sort of computer program to do the calculations, there are still a number of choices and a few "judgment calls" that will need to be made. In addition, the particular filtering performance of each system must be taken into account.

Any wireless mic manufacturer who is truly involved with high-end wireless systems will have a computerized frequency coordination program available. In most cases, these programs are not available as "public domain" software because the parameters that make them valid change as the wireless systems are re-designed. There are thousands of calculations that must be made in order to select a group of frequencies that are usable for any particular location. This is why computers must be used. There is no way you could do all the calculations necessary by hand on a timely basis.

Always contact someone with experience with this process and work out a frequency coordination scheme before you agree to provide a large multi-channel wireless system for your client, especially in traveling situations. It is always best to implement a multi-channel wireless system using only identical systems from the same manufacturer on coordinated frequencies. Every wireless manufacturer has taken their own path in designing wireless systems. There are numerous choices made by the design engineer to determine the oscillator fundamental frequencies, the multipliers to be used and the IF frequencies in any particular design. Trying to mix different brands and models in a multi-channel system is just asking for problems. Putting three or four channels into a church in Buford's Point, Idaho is one thing. Trying to place 10 or more channels on the road, touring major cities in the United States, is an entirely different matter.

TESTING FOR COMPATIBILITY

Because we are living in the "real world," you rarely enjoy the luxury of purchasing or renting a new collection of wireless mic systems each time you must implement a multi-channel wireless system. It would be nice to go through all the steps and check out procedures previously mentioned for each job, but the real world just isn't that friendly. So, the following procedure is useful in determining the basic compatibility of a multi-channel wireless system.

■ Checking for receiver interaction: Turn on all receivers and place them in the same relative position they will be in the actual production. Leave the transmitters off. Check to see if any of the squelch indicators (usually labeled "RF") on any of the receivers opens up. If any receiver squelch opens, turn off the other receivers one at a time to locate the receiver generating the RFI signal. By repositioning the offending receiver, you may be able to alleviate the RFI problem which, in this case, could be the result of LO crosstalk.

If repositioning the receivers does not change the problem, you may have an external RF signal mixing with one of the receiver oscillators. In this case, turn on the transmitter for the receiver that the squelch opens on, and see if the audio sounds OK. If the external signal is fairly weak, the transmitter signal will bury it and the system will still operate OK. However, note that a strong external signal can create noise even when the receiver's own transmitter is on.

■ Checking for transmitter spurs and second order IM: Turn on all of the receivers. Then, turn on the transmitters one at a time. As each transmitter is turned on, its corresponding receiver will unsquelch (the RF lamp comes on). Look to see if any other receivers also unsquelch at the same time. If one or more others does come on, turn off the receiver that matches the transmitter and see if the other receivers remain unsquelched. If the squelch remains open, then you probably have a transmitter spur getting into the other receivers. If they squelch and shut down normally when the matching receiver is turned off, you may have a second order IM problem as was discussed in the earlier section entitled "RFI in Systems." Refer back to that section for more information on this type of RFI problem.

■ Checking for third order IM: Turn on all receivers and all transmitters. Place the transmitters as close in distance to the receiver antennas as they will be during the actual production. One at a time, turn the transmitters off, and then back on again. Do this at least five or six times for each transmitter as you move around. Moving around with the transmitters as you turn them off and on will ensure that you don't have one located in a "null" where the RF signal is very weak at a receiver where you would normally have a problem. Check to see if the corresponding receiver squelches (no RF lamp) as its transmitter is turned off. If it doesn't squelch, you will need to determine which combinations of transmitters creates the RFI signal.

Through what can sometimes be a rather lengthy "process of elimination," you can narrow down the problem and identify which particular combination of transmitters generates the offending IM signal. The solution to this type of IM problem often involves changing frequencies of one or more of the systems. Sometimes, moving the receiver antennas farther away from the transmitters can reduce the IM problem.

■ Final system check out: Turn everything on. Listen to the output of each system one at a time. The idea in this step is to check for bad connections, transmitter gain adjustments and receiver output levels.

WHERE TO FIND HELP

Manufacturers are generally your best source of assistance in choosing and operating multi-channel wireless systems. It is a good idea to be slightly skeptical of any wireless manufacturer making overly big claims about how many channels their systems allow, or how easily multi-channel systems can be put together. It will be informative if you contact a few different manufacturers and ask them questions such as these:

1) How many wireless systems can you reliably operate in a single location?

Good answer: Seven or eight per TV channel. They should ask what other systems you have, where the system will be, geographically, or what TV channels are in use in the area.

Poor answer #1: "We make 12 frequencies and you can use all of them together."

Poor answer #2: "Wireless systems are like light bulbs. You just turn 'em on."

Poor answer #3: "Our system is a great value for the money."

2) What do your systems offer to prevent interference?

Good answer: A discussion of front-end and IF selectivity, advantages of helical resonators, advantages of crystal filters or saw filters, advantages of multiple ceramic filters, etc. This should be placed on a level you understand, not just buzz words designed to snow you. It's a good sign if they offer to send you literature that explains the design philosophy of the wireless system in more detail. The literature should explain the problems and the design solutions, rather than just consist of pretty pictures.

Poor answer #1: "There's no interference in the TV channels, VHF band, UHF Band, etc."

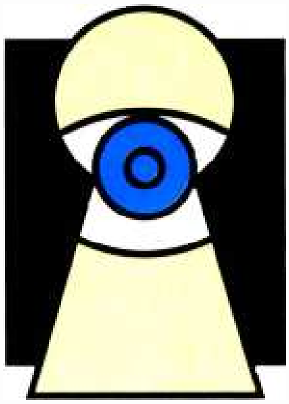
Poor answer #2: "I don't speak no technicalese!"

Poor answer #3: "Our engineer will be back Friday."

Poor answer #4: "We never have interference."

3) What sort of selectivity do your receivers offer?

Good answer: The front end should have a selectivity of at least 60dB of rejection at a bandwidth of 12MHz; 60dB at a bandwidth of 8MHz would be even better. A high selectivity front-end reduces adjacent channel TV interference and reduces other IM problems also. The IF selectivity could be as good as 60dB of rejection at a bandwidth of 90kHz if crystal IF filters are being used. If multiple ceramic filters are used, the selectivity won't be as good, but it is still respectable to provide 600kHz bandwidth for 60dB down points.



Poor answer #1: "Huh?"

Poor answer #2: "Our IF selectivity is 12dB at bandwidth of 600kHz."

Poor answer #3: "We use helical filters." (A helical filter is not a helical resonator.)

Poor answer #4: "Our engineer will be back Monday."

Poor answer #5: "Our front end bandwidth is 10MHz for 3dB down."

Poor answer #6: "You don't have to worry about that with our wireless."

Poor answer #7: "A diversity receiver doesn't have that problem."

4) How do you approach frequency coordination? Do you use a computer program?

Good answer: A computer program is used for each multiple channel wireless system sold. It lists problems in order of severity and checks for transmitter spurs, intermod, IF spacing, frequency spacing, etc. In addition, a printout of all tests and checks that were run should be available to the dealer and/or the purchaser.

Poor answer #1: "It's not necessary. Wireless systems are just like light bulbs. You just plug 'em in and turn 'em on."

Poor answer #2: "It's not necessary with diversity systems."

Poor answer #3: "Our Cray is down until Wednesday, but it's OK. We never have problems with that."

Poor answer #4: "Just space the frequencies far enough apart and it'll be cool."

Poor answer #5: "Our engineer got his thumb caught in the abacus and he won't be back 'til Saturday."

5) What happens if I have RFI problems?

Good answer: The manufacturer or their dealer will work with you to resolve the problem. If the problem is due to interaction between the vari-

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Reverberation, Reflections and Resonance

By David Scheirman

Prior to the development of sound reinforcement systems, the major concern in designing buildings for live musical programs was architectural. Before audio, a building designer was concerned with two things: the proper reverberation time that could be built into a facility to 'support' the natural sound of symphonic orchestras and operas, and the correct geometric relationship of walls, ceilings and seating areas to provide the most artistic listening effects for music, while preserving speech intelligibility—all without the use of a sound system.

Once sound reproduction systems began to become popular and effective for use in buildings of all types — movie houses, concert halls, public auditoriums, houses of worship and arts centers — a trend developed wherein many sound system designers assumed that most sound 'problems' would be solved with audio engineering alone. By the '50s and '60s, it was assumed by some that with enough of the right type of electronic gear and speaker systems, placed in the proper locations, the sound perceived by the audience would be better than ever.

A variety of new building projects and refurbishments have been completed in the U.S., circa 1970-1990, featuring nearly every conceivable combination of acoustics (passive or active), sound systems (monolithic central arrays and multi-source distributed 'surround' designs) and electronic assistance (artificial resonance systems and complex delay/distribution matrixes). The acoustical consulting and sound system designing communities have had a chance to see what works, and what doesn't, what the directions are for future technology development and what some of the limits are in different design flowpaths.

The 'Three R's' of building acoustics ... Reverberation, Reflections and Resonance, still hold court today for even the most rudimentary of listening rooms and it's good to know a bit about them. Let's take a look at what we know about the art and science of acoustics and see how it can apply to gaining an understanding of some of today's new buildings that are designed for the presentation of live music.

David Scheirman is R•E•P's live sound consulting editor and president of Concert Sound Consultants, Julian, CA.

HISTORY

The intentional design of acoustics for concert halls is not a new craft. Some of the world's best-sounding concert halls are 'old-world' rooms in Europe, built more than two centuries ago. Venues such as the Berlin Philharmonic, the Concertgebouw in Amsterdam and the Stadt-Casino in Basel, Switzerland, are among the best, built after a classical tradition first pioneered in western Europe nearly 250 years ago. The Grosser Musivereinsaal, Vienna, is considered by many to be the best concert hall in the world. If there is a common problem, however, with any such facilities, it might be said that they are 'too small.' In other words, to achieve the 'right' natural acoustics, it takes a limiting of the room size. This obviously limits the amount of seating space, typically to about 1,000-1,200 people. Building designers have always had to wrestle with this dilemma: how to best present the live show on stage visually and acoustically, without making the enclosed room space so large that the sound suffers?

A pioneer in the relatively new scientific approach to acoustics was Wallace Sabine. In 1889, the Harvard University physics professor undertook a survey of lecture halls on campus. He found that 'good' acoustics were entirely dependent on reverberation time. Working at night for three years, Sabine and his students visited different rooms, using an organ pipe and a stop watch. He eventually formulated certain basic concepts that are now taken for granted, for example, the larger a room, the longer its reverberation time. The more people or furnishings in the room, the shorter its reverberation time. Shorter reverberation cycles enhanced sound clarity; longer ones contributed to 'liveness.' Reverberation time was arbitrarily defined by Sabine as the time required for the sound to decrease to one millionth of its original intensity after stopping the original sound source. (For full results see "Collected Papers in Acoustics" by W.C. Sabine, published by Harvard University Press, Cambridge, MA.)

Employed as a consultant for the new Boston Symphony Hall (opened in 1900), Sabine was able to influence the construction of the facility to such an extent that the hall is still world-famous for its acoustics. To this day, we use the term 'sabin' for measuring sound absorption coefficients of different building materials.

Another great name in acoustics is Leo Beranek, coincidentally, also based in Cambridge. In the mid-'90s, Beranek became known as one of the outstanding experts on the design of music halls around the world. (See Beranek's classic book, "Music, Acoustics and Architecture," published by Macmillan, NY.)

Beranek was the acoustical consultant for Philharmonic Hall (Lincoln Center, NY). Unfortunate design changes in the building (more seats than Beranek's design called for, and the use of budget-saving concrete 'clouds' in the ceiling) caused a fiasco: Opened in 1962, the Philharmonic's sound was panned by critics and musicians alike. It was eventually gutted, renovated and re-opened as Avery Fisher Hall.

Modern, recently built concert halls such as the Eugene McDermott Hall at Myerson Symphony Center (Dallas); Evangeline Atwood Hall (Anchorage, AK); and Segistron Hall (Orange County, CA), each hardly five years old, make use of classical building design considerations and the most advanced computer-modeling techniques and electronic architecture technologies to help ensure that a repeat of New York's Philharmonic fiasco won't be repeated. And, these newer halls seat far more than 1,000 people. With the help of 'electronic architecture,' they are able to break the 2,000-seat barrier, enabling shows to serve larger audiences.

Such technologies were anticipated by Harry F. Olson in his landmark book "Elements of Acoustical Engineering," first published in 1940 by D. Van Nostrand Company, and now available from Professional Audio Journals (Philadelphia) as "Acoustical Engineering." Olson says, "It is quite evident that reproduced sound offers greater possibilities for obtaining the proper artistic effects by the use of the following expedients: incidental sound, a wide volume range, the control of the reverberation of the room characteristics and various sound effects."

From its early beginnings, less than a century ago, the science of acoustical consulting has grown such that the Acoustical Society of America boasts more than 7,000 members. Today, well-known acoustical consulting companies such as Jaffe Acoustics (Norwalk, CT) and Artec Consultants (New York) use technologies such as retractable sound-absorbing panels, electronically assisted building resonances, adjustable-height sound reflectors and controllable reverberation chambers to coax rich, classical sounds from larger and larger rooms, while providing a means to control or vary the acoustics to suit the type of performance.

NORMAL IS ...

Typically, a reverberation time of 1.4 seconds is considered best for the good hearing of speech in lecture halls. For opera, an average time of approximately 1.6 seconds enables audiences to hear a singer clearly, yet ensemble music still sounds 'live.' For symphonic music in concert halls, two seconds is considered to be an ideal design target for reverberation time.

As an example, the previously mentioned Evangeline Atwood Concert Hall lets the technical staff switch from a 'normal' reverberation time of 1.6 seconds (good for light opera and amplified Broadway-style musicals) to a full two seconds for symphonic music. A large number of distributed speakers mounted in walls and ceiling allow the presentation of a reverberant 'surround' program that can increase the apparent (perceived) room size.

With the science of room acoustics design improving each year, and with audio engineering technology offering better transducers, cleaner electronics and quieter, more efficient power amplifiers, the listening public now has an opportunity to hear live music presented in a way to large audiences that was once only possible in an intimate, limited-seat classical European concert hall. It takes a combination of proper building architecture, correct acoustical modeling predictions and discretely applied sound reinforcement systems. If you haven't yet experienced a concert at one of today's modern concert halls, do so at your earliest convenience. Chances are that you'll find it well worth the listen. ■

Continued from page 43

ous elements of the wireless systems and if the manufacturer has handled the frequency coordination, they will change frequencies at no cost to you.

Poor answer #1: "True diversity systems don't have RFI problem."

Poor answer #2: "We never have those problems with our UHF system."

Poor answer #3: "We'll forward your problem to our engineer in Tasmania."

Poor answer #4: "Try another frequency, any one at all."

Poor answer #5: "Oh, you can trade up to our Super Blasto system."

Poor answer #6: "Take it back to your dealer."

6) If I need to change frequencies how long does it take? And how much does it cost?

Good answer: Five days or less after receipt at the factory, and for $100 (or less).

Poor answer #1: "We've got systems stacked to the ceiling. We can't get to your system 'til next month."

Poor answer #2: "Two or three weeks and $150."

By conducting a "survey" of several manufacturers, you should be able to tell which ones really understand the common problems of multi-channel wireless systems and can offer solutions. If nothing else, you can at least make some acquaintences and locate sources of information, if and when you do get involved with a multi-channel wireless system. There are now more than 40 different brands of wireless mic systems being distributed and sold by various manufacturers. You should be aware that only a handful of them actually manufacture the products themselves. Many are OEM units, with designs purchased and/or created overseas. This is only important in that it could make a big difference concerning the level of expertise and services available from *the factory*, if and when RFI problems arise. After all, when you have some really strange RFI problems, who you gonna' call? ■

Note: The information in this article is available as part of a larger publication covering the design concepts and implementation of wireless microphone systems for professional and commercial users. Please call 1-800-821-1121 for a copy of this publication. Special credit belongs to Larry Fisher, vice-president of engineering at Lectrosonics, for his technical expertise and generous assistance.

The 1992 OLYMPIC WINTER GAMES

By David Scheirman

For a 16-day period in February, 1992, people around the world looked to the XVI Olympic Winter Games in the Savoie provincial region of France for a chance to witness the world's best skiers, skaters and other winter athletic competitors. This is the third time that the French Alps have hosted the Winter Olympics, the first were at Chamonix in 1924, and the second at Grenoble in 1968.

While the 1992 events were centered in the city of Albertville, in reality, the various hockey, bobsled, ice skating and skiing facilities were scattered around the French Alps, from Tignes to Moutiers to Les Saises, with many venues being several hours apart even in good weather. The Olympic area covered world-famous ski areas, including the Three Valleys (Vourchevel, Meribel and Les Menuires) that boasts 500km of ski slopes, over 200 ski lifts and 600 snow-making machines. Thus, the equipment, crew and logistical requirements were monumental.

Bose S.A.R.L. (Bose France) was chosen as the official professional sound system supplier to provide, install, operate and maintain all sound system needs for these Winter Games. Drawing on experience first gained by Bose Canada at the XV Winter Olympics in Calgary, and with the assistance of Bruce Myers, Bose U.S.A.'s

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Reliable, intelligible audio for athletes and audiences in the face of snow and freezing temperatures.

senior special projects engineer and the company's general manager, Andy Smaga, Bose France undertook to make the sound for the XVI Games the best yet.

"I first met with the French Olympics Organizing Committee in 1988, four years prior to the event," says Smaga. "I showed them that we weren't trying to sell them speaker products, or just treat this like any other profit-making job. What we wanted to do was to provide a complete service and deliver what they needed — good-quality sound. It would be a showcase for us. I could not afford to do a poor quality job in my own country!"

In mid-1989 Bose France S.A.R.L. received a letter of intent to contract for services from the Olympic Committee. "28 contract drafts later, we had a signed deal," says Smaga. "We elected to do this for a flat, all-inclusive fee. There were so many variables about sites, audience sizes and so forth that the constant changes and contract revisions would have made us run out of valuable time."

Smaga's project director, Dominique Marphay, reached the early conclusion that sound for this event would be successful only if each specific system was well-planned in advance, including original design, equipment list and special hardware needs and appropriate personnel.

Initial sound system predictive work, undertaken with Bose's Modeler computer sound system design program, were used to format preliminary layouts for the different venues. Basic system requirements included the need, in most instances, to operate out-of-doors in temperatures that could get as low as -30° F (on downhill ski runs, for instance). Myers did his first site surveys in November 1989. Preliminary venue drawings were mapped out, and targets were set for each system's coverage and sound-pressure level needs. Every competition site was different and required its own design approach to answer questions such as whether to install point-source arrays or distributed sound with delay zones, or full-bandwidth systems for music support or vocal reinforcement only. Each building and site had its own requirements.

"We started to choose what equipment would be used at each site, and this was based on crowd size, grounds or building layout and program input," says Bose's Bruce Myers. "The 27 different temporary sound systems fell into several different categories. Some were simple, portable units used wherever a press conference might be held on short notice. Others had to be prerigged and installed with hanging gear just as if they were permanent installations, even though they would be pulled out a day or two after the competition events at that site. Still others were intended to remain behind and become part of the building facilities that would be turned over to the local civic authorities."

Generally, a full variety of Bose's products was used to format the different systems, ranging from the tiny RoomMate speakers (used as cue monitors in amplifier rack rooms) to the venerable model 102, 402 and 802 systems, all of which rely on Bose's small 4.5-inch full-range cone loudspeakers and active equalization circuitry. The Acoustimass Pro system, containing its own power amplifiers, was used in many systems. Where increased bass response was required for music playback, the unique tube-type Acoustic Wave Cannon and the Model 302 Tandem-Tuned low-frequency enclosures were available.

Most systems would experience full-time use for only about two weeks and then be removed, but some would have to be pre-installed months in advance. There were many variable such as where to locate hidden cable runs, how to place mounting poles and brackets, etc. In all, there were 13 different Olympic sites that required advance sound system designs and installation, in addition to the portable systems.

DOWNHILL

Outdoor ski sites typically relied on multiples of Bose's model 802 enclosures erected on poles and set up in distributed, signal-delayed zones. Indoor venues, such as the Hall De Glace, used for ice skating events, were given a central array for music playback along with multi-zone distributed vocal announcement reinforcement systems.

Of primary importance was the fact that all events would be televised. "The stress at world-televised events can happen when the broadcast and sound reinforcement people don't speak the same language," says Smaga. "For instance, the Olympic Committee wants to please the spectators, who travel great distances and pay to buy tickets to see the events. The TV people have a different set of considerations. They want to have an audio signal going out to millions of people that doesn't contain public-address interference."

The stress at world-televised events can happen when the broadcast and sound reinforcement people don't speak the same language.

Care was taken to lay out the different sound reinforcement systems so that a minimal amount of live sound "bleed" would be present at locations

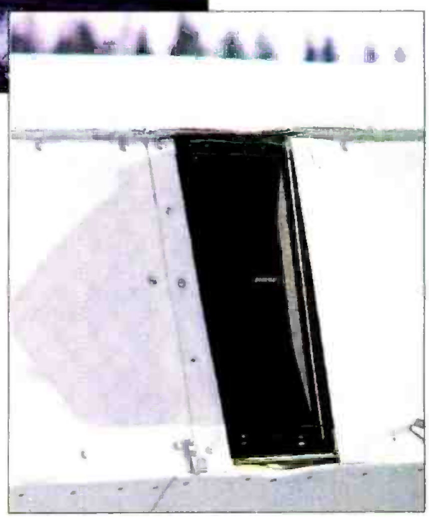
An example of the unusual sound coverage problems presented by many Olympic event venues, the bobsled run shown here required a multi-zone distributed audio system.

where broadcast-audio microphones were located. While Bose France was no stranger to providing sound reinforcement for major sporting events (having worked on the LeMans auto race and the Tour de France bicycle race), test systems were set up during World Cup events in 1990 that enabled the company to gain more experience with winter sporting events that included TV broadcast involvement.

"The tests revealed valuable information," says Smaga. "For the broadcast people, it is important that their mics pick up the natural sound of the skier, the bobsled or the skater going by. For competitive skiing, a large part of the excitement of the broadcast audio is the sound of skis and poles on snow and the rush of the air. Some of the events include downhill skiing to music. The skier needs to hear it, the spectators need to hear it, yet the TV people don't want to hear the P.A. It is very tricky."

A bobsled whizzes past a loudspeaker location at 90mph. The sound reinforcement system handled both vocal and music playback input.

The sound team also learned that system placement and levels that worked well at the Val D'Isere site for men's skiing were not applicable to the Meribel site where women's events were held. "As it turned out, the men are more aggressive in their skiing, so they make more noise," says Smaga. "The TV mics picked up the men just fine with no P.A. interference, but the women's events required some basic changes because it was harder for the broadcast audio to pick up their natural sound."



A Bose model 402 loudspeaker enclosure mounted in the custom-built bobsled run.

INSTALLATIONS AND CREW DEPLOYMENT

The installation of parts of some systems for the Winter Games began in the summer of 1989. This included cable runs and pole-mount positions. Dominique Marphay worked with a hand-picked crew numbering up to 60 people at various times, allowing the same technicians who would be operating the different systems to work on installing them as they became familiar with the different sites. Many new buildings and competitive sites were also constructed as new projects, built from the ground up, so sound system work needed to keep on schedule with the construction track.

"Each different system in a venue had its own dedicated crew under the direction of a site manager," says Smaga. "We didn't want to get trapped by trying to rely on people, or systems, to do double-duty. For instance, let's say there were a few days off at one site for one system, and another venue could maybe make use of those resources, but, what if the weather changes, or an event gets canceled and then re-scheduled? We wanted to leave nothing to chance."

With event sites spread out over an entire province of France, and some of them being several hours drive away from each other (often more with traffic delays), the plan made sense. About three dozen sound technicians were required to staff the systems during the events, and an additional two dozen laborers and specialists (including riggers, electricians and such) were used during the installation phase. Some sound technicians were chosen for their ability on skis, because many times the compact Bose loudspeakers had to be taken up steep alpine slopes for mounting, and this task was best accomplished by one person on skis with a back-pack.

The 34,000-seat outdoor stadium at Albertville, used for the opening and closing ceremonies, featured a central production tower to hold lighting and sound equipment.

THE LA PLAGNE BOBSLED SITE SYSTEM

It is beyond the scope of this feature to examine all 13 site installations in detail. We can, however, take a look at one site and see how things fit together, from original design to facility construction to the use for competitive events.

The La Plagne site, used for bobsled and luge events, was based around a custom-built, ammonia-cooled bobsled run. It was predicted that temperatures at the outdoor facility would range from -20° F to +45° F. A system was needed to provide an average of 95dBA sound-pressure level reinforcement for both voice and incidental music. Background noise would include crowd sounds and wind projected to measure up to 80dBA in the audience area.

Elain Francais, sound designer for the opening and closing ceremonies, shown here at the Yamaha DMC-1000 digital console.

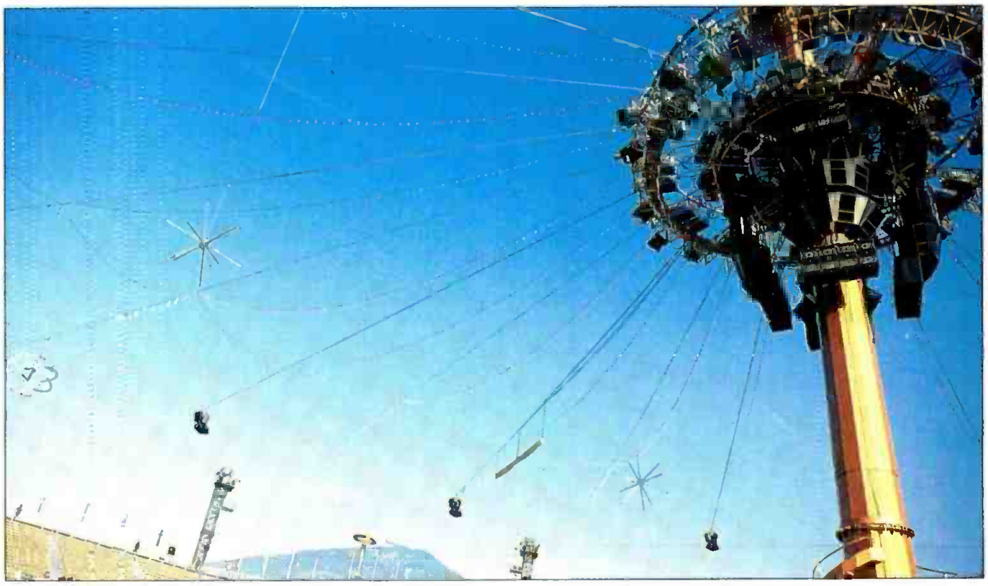
This site required the early installation of cable troughs and speaker mounting locations, because the concrete track was first poured more than a year before the competitive events would take place. The system design relied on multiple units of Bose's model 402 column-type speaker enclosure, that were installed in recessed cavities in the walls of the bobsled track.

Including 36 of the 402s and another 30 of the smaller model 102 enclosures, the system would not only serve to present events results announcements in both French and English, but to cover the entire audience area with sound when musical programs, playback messages and broadcast narrative was presented from the International Broadcast Center. Most of the loudspeaker system was laid out as a 100V constant-voltage line.

As installed, the system featured an 8×2 Yamaha mixer that received input from six hard-wired cardioid mics and one wireless vocal mic system, along with a cassette player, CD player and FM tuner. Two graphic equalizers and a program limiter were used on the mixer's outputs. A 1×10 distribution amplifier fed signal to nine Bose model 1800-IV power amplifiers (distributed in Europe). A tiny Room-Mate speaker was used as a cue monitor in the equipment room.

"This was one of the more interesting advance system designs," says Bose's Bruce Myers. "The bobsled track wound downhill like a serpent, and the audience areas were scattered. There was no single point at which you could say, 'sound here' or 'no sound here.' It was a case of needing to use good judgment in having enough speaker location density to reach the people where they were, yet not having a system that spilled out too much from the spectator areas, because all of the various broadcast commentators needed to be able to use their remote mics without picking up P.A. interference."

The system described offered good, clear and intelligible audio signal distribution to the intended spectator coverage areas, while minimizing the spill outside of the venue. Announcements were crisp and musical interludes were dynamic, yet the system did not call attention to itself. The compact, recessed enclosures rendered it visually unobtrusive, if not nearly invisible.



The central production mast showing U.S. Sound's loudspeaker arrays for 360° coverage. Additional signal-delayed speaker units were suspended from support cables anchored to the stadium perimeter.

OTHER EVENTS

In addition to the bobsled site detailed above, there were of course many other sites, each with a sound system tailored to the needs of the sport involved. Ski jumping, ice dancing, slalom and freestyle skiing all required a different approach to find the delicate balance of sound reinforcement that worked for athletes, spectators and broadcast executives.

The installation of parts of some systems for the 1992 Winter Games began in the summer of 1989.

"One of the more challenging tasks had to do with the skaters," says Andy Smaga "We supplied a complete rehearsal system in a different building, and a compact audio production facility where our audio techs could receive the different musical program tapes from each country's skaters and then standardize them for playback during the events. For example, one from Lithuania might bring us an L.P. record. Another from Sweden might have a studio-produced reel-to-reel tape. An American might have their music program on a cassette tape. They were all different formats, different qualities, different levels and so forth. We had not only the skater's own musical programs that they do their routines to, but also different national anthems to play when the winners were announced at an event. Our techs worked to get everything standardized on DAT so that we could have the best, consistent quality and quickly go back over the same things if they were needed again."

Indoor skating and hockey buildings received custom-configured central arrays constructed of 10 to 12 model 802s and Acoustic Wave Cannon low-frequency devices. Custom-constructed hanging frames were used for rigging the systems. Additional speakers (up to 32 per venue) were hung on catwalks and configured in delay zones using Yamaha's DDL-3 digital delays to maximize vocal clarity for announcements.

Outdoor ski events at seven different sites often featured 40 or more model 802s or AM-Pro enclosures located on 12-foot to 16-foot poles, usually spaced at 24-foot intervals to cover spectator groups and to provide music playback reinforcement for some of these outdoor events.

In all, for 16 straight days, up to 25 different events were serviced with sound reinforcement on a daily basis, on time, without technical difficulty, and in spite of traffic snarls, winter weather conditions and crowds that often exceeded the event planners' expectations.

OPENING AND CLOSING CEREMONIES

Every Olympic Games has its opening and closing ceremony to help define the spirit of the event and to honor the locale in which it is being held. In Albertville, France, these programs took place in a 34,000-seat outdoor stadium constructed of temporary high-rise bleachers with a tower-like production mast all erected on a soccer field. Telema, a European production organization, was the primary contractor in charge of all technical aspects of the opening and closing ceremonies. Bose France was subcontracted to supply sound reinforcement services to Telema.

The primary sound designer for the events was Elain Francais, a Paris-based sound professional who chose to use digital sampling and storage equipment to present an unusual, highly-entertaining audio program to complement Telema's production. This included diverse aspects such as 100 costumed clowns riding bicycles, gymnasts and jugglers, and 300 regional folk-dancers in the Savoie provincial costume.

Francais relied on three of Akai's DD1000 optical disk recorders, linked via a Macintosh computer, to sequence and cue-up a wide variety of sounds for the opening and closing ceremony programs, including special effects such as wind, birds and insects, jet planes, accordions, ticking clocks and bagpipes. The DD1000 gave Mr. Francais the tool he needed to assemble complicated, time-cued music and sound effects segments with the assistance of Akai's DD-QMAC software. Each DD1000 features a removable magneto-optical disk and can record and store up to 90 minutes of material with instant random access.

"We have had many changes within the musical program since I first started putting it together for this event," says Francais. "The digital system allows me to quickly re-configure things if we need to accommodate changes in the show's choreography."

The equipment, crew and logistical requirements were monumental.

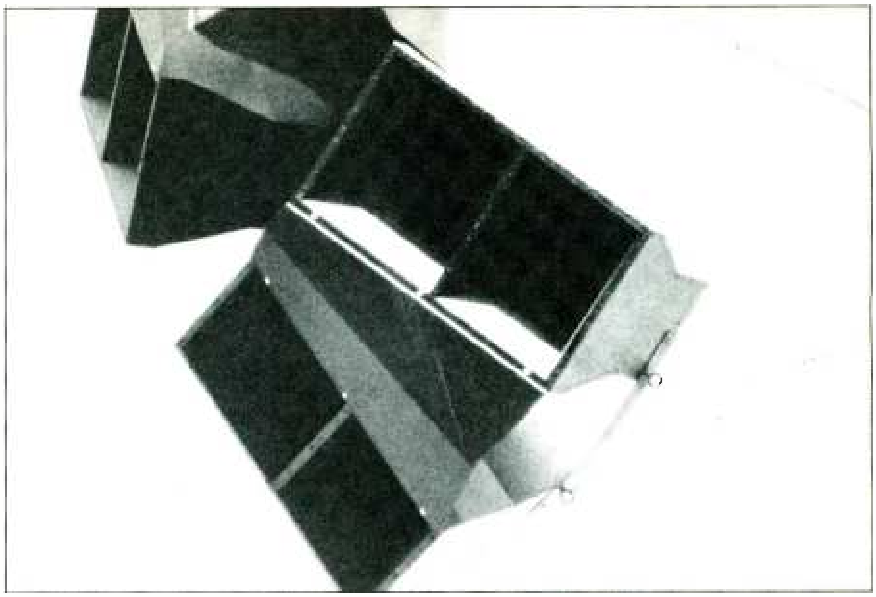
The digital outputs of the DD1000s were fed into a Yamaha D-1000MC digital mixing console, that was routed to a Yamaha PM-3000-40, used to distribute sound to the many different speaker zones. Also feeding the PM-3000 were 22 of AKG's new battery-powered RF mic systems. The new AKG systems, operating in the 800MHz-900MHz band, were picked up by a mast-mount antenna located high on the stadium rim; a single 70MHz line carried 12 diversity channels to the receivers, located in the sound control booth. Hans Radda, in charge of export sales and marketing for AKG Acoustics in Vienna and present at the Winter Games Ceremonies, stated that this system represented a wireless-technology breakthrough. Currently, other similar systems can carry no more than six diversity channels in such a narrow bandwidth, whereas the new AKG wireless system doubles that capacity.

The wireless mic systems found many uses. Program announcers, some of whom rode above the audience suspended in ski-lift chairs, relied on them. Tiny C410 mic elements were attached to the blades of ice skaters, with transmitters hidden in their costumes; the crisp sound of blades meeting ice was then panned around the large outdoor stadium, following the skaters around the ring via placement in one of nearly two dozen signal-delayed overhead speaker arrays.

SPEAKERS

Utility speaker systems installed in the ceremony stadium included Bose model 102s, placed beneath the bleachers for use as a paging system to alert program participants of upcoming event cues, and an array of model 802s used for music playback needs in the performer staging area.

The primary sound reinforcement system for the round, open-air facility was obtained from U.S. Sound (New Jersey). The company has been quietly at work for the past five or so years developing new-technology, horn-loaded integrated loudspeaker systems for use in large-scale sound reinforcement. Designer Cliff Henricksen (formerly with Electro-Voice and Community Light & Sound) has used large-format compression drivers to a good advantage. Combining new driver technologies with advanced enclosure-building materials, U.S. Sound has created a lightweight, powerful system that is exceptionally articulate. Two patents have been granted to the company, one for a method of constructing lightweight enclosures, and the other related to narrow-pattern array technology for covering large audiences. The highly directive arrayable enclosures represent an interesting and effective new design trend for large-scale sound reinforcement.



A cluster of four of U.S. Sound's mid/high enclosures, suspended from Kevlar cables at a point 50 meters from the central tower.

"A primary criteria for the system design here was a low weight factor," says Henricksen. "The center mast that the 360° system hangs from supports a very complex lighting system, along with being the central anchor point for the Kevlar suspension cables used for hanging production hardware and even our own delay arrays. Traditional sound system packaging would have just been too heavy. Our entire 64-box speaker enclosure complement weighs just over three tons." The weight per low frequency box is merely 110 pounds, components included, and the weight per mid/high box at 98 pounds!

Four identical arrays made up of the 3-way system's modular boxes were suspended from the center mast to cover the round stadium area. This included a total of 32 low boxes and 32 mid/high boxes for the central array. Additional subwoofers were located below the seating sections.

A ring of signal-delayed mid/high enclosures, arranged in pairs and suspended from Kevlar cables, was set out at 50 meters from the center mast and oriented down toward the audience seating areas. These were primarily used to reinforce vocal announcements and for special-effects sound imaging placement. Other similar enclosures were positioned atop the stadium's perimeter and mounted to lighting-support towers, thus enfolding the audience in a huge "surround-sound" field that gave sound designer Francais a full speaker system toolkit to work with.

SUMMARY COMMENTS

Major, globally-televised events such as the Winter Olympics require the highest level of professionalism in terms of sound reinforcement, from design to installation to operations. Audiences for these "mega-events" come with a sense of expectation to see and hear things that have never been experienced before, to be wowed by sight and sound, and to be not only entertained, but to be drawn into the group experience of participating in a spectacle. Millions more watch at home on their television sets (and listen to broadcast audio). It is at this type of special event that a fine line is walked between the proven reliability of known technology and the risk-taking that often accompanies cutting-edge technologies.

Bose France's project manager Dominique Marphay did a world-class job of assembling the equipment and operators needed to handle an extremely challenging sound project. Bruce Myers of Bose U.S.A. played an integral part, specifying and designing what would work best for the different venue sites. New technologies from other companies as represented by U.S. Sound's lightweight, high-definition loudspeaker system and AKG's new wireless microphone systems played their part, as well. ■

Wow and Flutter

By M. Raymond Jason

Digital recording can claim one absolute advantage over analog, which no future analog innovations will challenge: immunity from time-dependent problems. While in some circumstances (read: poor design) one can measure, or even hear digital phase-jitter, analog recording is and will forever be stuck with wow, flutter and (unless time code is used) timing problems. This and next month's focus will be troubleshooting these time-dependent ATR transport problems.

Wow and flutter are the audible effects of disturbances in tape motion, manifesting as periodic pitch variation. Wow refers specifically to low-frequency variation, from 0.5Hz to approximately 2Hz, while flutter encompasses all higher-frequency variation. At 15ips, rotating components down to approximately two inches in outside diameter can contribute to wow or flutter, depending on how many geometrical or rotational imperfections are present over the span of a circumference. Smaller components including small rollers and the capstan can contribute only to flutter.

Flutter frequencies up to several thousand hertz can result from transport servo-system oscillations or intermittencies, and from mechanical disturbance or oscillation of the tape sliding past the heads. These more than 200Hz problems are collectively measured using wide-bandwidth flutter meters as "scrape flutter."

Four major standards organizations publish specifications for wow and flutter measurement, including the NAB in the U.S., the IEC and DIN in Europe, and the JIS in Japan. Because the IEC and DIN specifications are identical, and little equipment available in America uses JIS specs, the following discussion will only consider NAB and IEC/DIN measurements.

NAB VS. IEC/DIN MEASUREMENTS

There are three main differences between NAB and IEC/DIN wow and flutter, resulting in measurement differences depending on the standard used (See below). Use the standard by which your ATRs are specified.

Parameter:	**NAB**	**IEC/DIN**
Frequency:	3kHz	3.15kHz
Bandwidth:	0.5-200Hz	0.2-200Hz
Detection:	Average response (rms cal.)	Quasi-peak

Both standards share a requirement for measurement during playback, as distinct from measurement in repro mode during recording. The NAB requires a special wow and flutter alignment tape, while the IEC/DIN specifies that a recording be made on the machine under test, the tape rewound, then played back for measurement.

DERIVING A SINGLE NUMBER

Because indicated wow and flutter varies from moment to moment, there is ambiguity regarding how to associate a single-number performance specification to a variable analyzer indication. In general, peak wow and flutter is more interesting than average or minimum, for the obvious reason that the goal in optimizing a tape machine is elimination of all audible problems.

On the other hand, brief bursts of flutter that occur very rarely (and are often due to tape sticking within the supply pack) are unlikely to be noticed even during critical listening, and are often uncurable. A compromise technique, which allows for occasional bursts, selects the point at which the indicated value is exceeded only approximately 5% of the time.

Automated analyzers make this calculation ... well, automatically. To approximate this scheme using a conventional meter, note the peak indications for each of ten 10-second readings. Then, simply take the second-highest reading to obtain a single-number wow and flutter measurement. Averaged repetitions of this process increase accuracy.

SCRAPE FLUTTER

Scrape flutter can dramatically degrade sound quality, yet is excluded from the standardized measurement techniques. Few, if any, ATR manufacturers provide a scrape flutter specification, probably precisely because there is no standard. Furthermore, while wide-bandwidth flutter meters are available, they are generally expensive, making them rare in the field. Overall, this useful and revealing measure of ATR performance has acquired the status of "outcast."

If you don't have a wide-bandwidth flutter meter, you can still obtain relative measurements of scrape flutter by recording a 12.5kHz tone, rewinding, then playing back through a harmonic distortion analyzer set to null the tone. Simply listening to the output of the analyzer permits fine adjustment of scrape-flutter rollers. Position the roller for minimum noise. Take care not to give the roller so much tape penetration that it lifts the tape from the heads. At the optimum penetration, signal strength should not be diminished by more than 0.1dB.

SEARCH AND DESTROY

To eliminate a source of wow or flutter you must first locate it. Check the capstan and the pinch-roller. If either has any grease-pencil or 3-D oxide build-up, cleaning them off usually solves the problem.

Pinch-rollers age and incur damage over time. Swapping pinch-rollers is a simple and fast way to tell if you need a new one. Incorrect pinch-roller pressure can also promote flutter, so check the pressure with a spring scale according to your service manual.

The supply motor and its rotation-counting or velocity-sensing mechanism are common wow and flutter-producing failure sites (wow and flutter are rarely transferred from the take-up reel through the capstan/pinch-roller system). Problems of this type appear as periodic tension variation on the supply side of the capstan. Large-magnitude tension variations may be visible as dancer arm oscillations. Use an in-line tape-tension gauge to catch smaller amplitude oscillations.

If you don't have an in-line tension gauge, you may have success using a 3-inch to 4-inch length of 1/8-inch heat-shrinkable tubing (or even a plastic drinking straw: use its whole length because it's quite stiff) held vertically between thumb and forefinger and pressed against the tape. Even very small tension perturbations will be revealed in the jitterings of the tubing. If your ATR's transport logic allows it, holding the dancer arm fixed may help, because the dancer arm, in doing its job, may reduce the amplitude of a

Continued on page 64

M. Raymond Jason is an electronic engineer at National Public Radio in Washington, DC.

CUT ALONG DOTTED LINE

Digital Domain

Dealing with DAT Machines, Part Two

By Rick Schwartz

Back in January we carefully looked at many of the DAT recorders used in professional recording studios. In the column, we included both rack-mount and portable units, and focused on problem areas such as program indexing and digital I/Os. Since then, DAT manufacturers have been hard at work on new products, as well as enhancements of existing products. This update will look at new developments from companies such as Denon, Fostex, Marantz, Sony and Teac.

DENON

Although the DTR-2000G is a grey-market machine, there is a U.S. version with a black face called the DTR-2000. The 2000G is a special edition machine with a gold front panel, gold-plated connecters and oxygen-free wire inside the unit. The company didn't skimp on quality, as the 2000G features 20-bit LAMBA type D/A converters and the same heavy-duty transport as the Panasonic SV-3900. This model has an interesting feature I have not seen before: While in pause mode, you can advance at half-speed with a fine cue control. It also has a digital fade in/out feature, and records digitally with SCMS at 48kHz, 44.1kHz and 32kHz sample rates.

FOSTEX

Although the PD-2 was first shown last June, it was just starting to ship at the time this article was written. The PD-2 is clearly the Swiss Army knife of professional portables. It includes almost every feature imaginable: a built-in TC generator, 4-head design for off-tape monitoring, MS decoding for headphones, filters, phantom power, error search capability and much more. Compatible with IEC standard time code as well as time code from the original Fostex D-20, it will lock to external references including word-sync, composite video or frame pulses. If there is a drawback it would have to be its price and weight, even though it is comparable to a Nagra in both of those categories.

The evolution of the D-20 and PD-2 continues with Fostex's introduction of the new D-20B. An updated version of the D-20 offers added time code generator and chase lock synchronizer, and is capable of transcoding time code from tape. The time code generator has standard time code types and jam sync capability (24, 25, 29.97, D/ND and 30). If a DAT tape has a different format code, the D-20B can be set to playback the format you are using and chase the master, eliminating conversion problems.

JVC

Although the DS-DT900 was included in the earlier feature, we decided to run it again to include additional information recently provided from the manufacturer. I learned that the DT-900 has an internal DIP switch that allows the digital input and output to be configured alternately (i.e. AES in, S/PDIF out; or S/PDIF in, AES out, etc.).

DAT manufacturers have been hard at work on new products, as well as enhancements of existing products.

I also learned the machine has the ability to write start IDs via the AES port as audio passes below a -60dB threshold. The machine also has software available that will expand its time code features. See the chart for more information.

MARANTZ

New in the queue is this portable stereo DAT recorder of extremely small proportions (4" x 2.5" x 8.5"). At less than 3.5 pounds including batteries, it features 1-bit A/D conversion at 48kHz and 1-bit D/A conversion at 48kHz, 44.1kHz and 32kHz rates. Capabilities are quite extensive, with AES/EBU and S/PDIF digital I/Os supported, balanced mic and line inputs, unbalanced analog line I/Os, ABS time code, no SCMS, a built-in auto limiter, start ID editing and integral re-numbering.

SONY

Since last fall, when our original DAT column was written, Sony has released the PCM-7000 Series of DAT recorders. The 7000 Series was designed from the ground up to be professional workhorse machines that offer an array of standard and optional features, including glitch-free punch in/out, confidence monitoring, internal time code synchronization, instant start and serial or parallel interfaces. All Sony PCM-7000 Series recorders join Fostex by offering varispeed capability. As of last September, the Sony TCD-D10 Pro was replaced by the TCD-D10 Pro II. The new model adds absolute time code and also addresses the AES/EBU interface and clock jitter issues mentioned in my previous article. If you own an older TCD-D10, you can upgrade it by calling Sony customer assistance at 201-833-5300. The Sony PCM-2000 was included in the earlier story, the only thing that has changed since then is its new lower list price of $5,000.

TEAC

The DA-P20 is one of the lowest priced portables on the market, with balanced XLR mic/line inputs. It includes IEC-958 digital I/O and records digitally at 48kHz, 44.1kHz and 32kHz. The unit weighs 3.1 pounds and has a unique table of contents feature that displays the total performance time on a DAT tape, as well as its total number of indexes.

UPDATES ON-LINE

If you own a DAT machine that we have not yet featured or would like to discuss these issues in greater depth — please contact me on our new bulletin board on Compuserve. Go to the section 14, REPMAG in the MIDI VENDER area. My CIS address is 70672,1377. See you on-line. ■

Correction: The Vocal Splicer mentioned in the March 1992 column is available from Studio City Sound, 818-710-9306.

Rick Schwartz is a contributing editor to R•E•P and director of post-production at Post Complex, Los Angeles.

Special thanks to Tim Murray of the DAT Store 310-828-6487 for his assistance with some information included in this column.

The Issues	Denon DTR-2000G	Fostex PD-2	JVC DS-DT900	Marantz PMD700	Sony PCM-2300	Sony PCM-2700	Sony TCD-D-10 II	Sony PCM-7010	Sony PCM-7030	Sony PCM-7050	TEAC DA-P20
Digital I/O:											
AES/EBU (XLR-3 type)	No	Yes	Yes	Yes	No (9)	No (9)	Yes	Yes (15)	Yes (15)	Yes	No
IEC-958/Type-2 (RCA type)	Yes	No	No	No	No (10)	No (10)	No (10)	Yes (15)	No	No	Yes
S/PDIF (RCA type)	No	Yes (1)	Yes (5)	Yes	No (11)	No (11)	No (11)	Yes (15)	No	No	No
OPTICAL (EIAJ CP-340)	Yes	No	No	No	No	No	No	No	No	No	No
Metering & Display											
Internal Metering Reference Level (+4 dBu)	-12	-12 (2)	-18	-12	-20	-20	-20	-20	-20	-20	-12
Meter Range	-60 to 0	-60 to 0	-60 to 0	-40 to 0	-60 to 0	-60 to 0	-60 to 0	-60 to 0	-60 to 0	-60 to 0	-50 to 0
Error Rate Readout	No	Yes	Yes (6)	No	No	No	No	Yes	Yes	Yes	No
Program IDs											
Will Renumber Program IDs	Yes	Yes	Yes	Yes	Yes	Yes	No	Yes	Yes	Yes	Yes
Manual Writing of Start/Skip IDs	Yes	Yes	Yes	Yes	Yes	Yes	Yes (14)	Yes (16)	Yes (16)	Yes (16)	Yes
Will Copy Program IDs During a Digital Dub	Yes	No	No	No (8)	Yes	Yes	No	Yes (17)	Yes (17)	Yes (17)	Yes
Copy Protection											
Will Record from a Copy Prohibited CD	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Adds SCMS Copycode to Digital Copies	Yes	No	No	No	No (12)	No (12)	No (12)	No	No	No	Yes
Will Ignore SCMS Bit on S/PDIF Ports	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	n/a	n/a	No
Will Remove SCMS Bit During a Digital Dub	No	Yes	Yes		Yes (13)	Yes (13)	No	Yes	Yes	Yes	No
Time Code:											
Stripes Absolute Code While Recording	Yes	No (3)	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Reads Absolute Time Code	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Post-stripes Absolute Time Code	Yes	No	No	No	No	No	No	Yes	Yes	Yes	No
Will Locate to Absolute Time	No	Yes (4)	Yes (7)	No	No	Yes	No	Yes	Yes	Yes	No
Records SMPTE Time Code	No	Yes	Yes	No	No	No	No	Yes (18)	Yes (18)	Yes	No
Post-stripes SMPTE Time Code	No	No	No	No	No	No	No	Yes (18)	Yes (18)	Yes	No
General											
Portable	No	Yes	No	Yes	No	No	Yes	No	No	No	Yes
Confidence Head Playback	No	Yes	No	No	No	Yes	No	Yes	Yes	Yes	No
List Price	$1,200	$10,950	$4,850	$2,500	$1,590	$2,900	$3,300	$3,900	$8,000	$11,500	$1,100

Notes:

(1) Using an optional RCA to XLR adapter connected the the AES/EBU port
(2) Has a Variable reference level of -12, -14, -16 or -18
(3) Stripes IEC-standard SMPTE instead of A-Time
(4) Front panel controls will locate to A-Time or SMPTE time code
(4) Via internal DIP switches (see article)
(6) Error rate can be read with a small mod. Contact JVC for more info.
(7) Software is available that will allow the DT900 to use A-Time if SMPTE is not present.
(8) Does not copy program IDs, but will copy start IDs
(9) The IEC-958 Type I interface accepts input from an AES/EBU source
(10) The IEC-958 Type I interface accepts input from an IEC-958 Type II source
(11) The IEC-958 Type I interface used on this model accepts input from an S/PDIF source
(12) For IEC-958 Type II & S/PDIF it writes ID6=10; For AES/EBU & IEC-958 Type I copy prohibit is manually selectable (Not on TCD-D10 Pro2)
(13) IEC-958 Type I only
(14) Start IDs Only
(15) Requires optional DABK-7011 digital I/O
(16) Write/erase of Start IDs available on front panel. Write/erase of start/skip/end IDs available on parallel port and RS-232C option
(17) Requires an ID dub cable in addition to the AES/EBU connection
(18) Requires optional DABK-7010 SMPTE time code board

R•E•P: On-Line

Service is ...

By Tim Sadler

The other day I was having trouble accessing CIS (CompuServe Information Service) via my usual modem connection in Cambridge, MA. Seems that all 20+ nodes were occupied by others who had logged on before me. It made me stop and think. If the ultimate goal is to get everybody networked together, are there enough phone circuits or modems to do it? The fact is that it is expensive to provide an access point to an international network such as CIS, and few of us want our home phone line tied up by the computer. As the number of people seeking information sources other than TV, radio and newspaper grows, the problem just gets worse.

Well, I wouldn't have dragged you this far if I hadn't found a very interesting answer to this info-bottleneck, and it's probably as close as the nearest TV set. That fat wire we call *the cable* has got a lot more coming out of it than just 104 channels of video programming!

I've been experimenting with two new technologies that take advantage of the data channels already present on the cable. The first is the X-Press Information Service. If you've ever gotten excited by watching a newswire teletype machine as it spewed out reams of paper covered with breaking news from around the world, then you'll like X-Press. The day the interface arrived, I couldn't wait to connect it to my cable to see if there really was news gushing out of the end of the wire. I plugged in the power supply and two out of three lights lit up! But when I screwed on the cable connection, there it was. The third little LED just started to wink at me. I knew by the way it pulsed that I was looking at the ebb and flow of digital information. I know, it's going to sound reverential, but I just sat there and stared at that pulsing light for a whole minute.

A simple connection to my Mac and there it was on the screen. News, weather sports, stocks — all scrolling by faster than I could read them. OK, I can hear it now: "Who needs an expensive teleprompter feeding you stories faster than you can digest them?" I won't deny that, as a news junkie, I get a thrill just knowing all that information is pouring across my threshold, but the real power of this service is in the application software that arrived with it and on it. Software updates are also delivered via this cable service.

As has been my custom after many years with the Mac interface, I just plunged right in without reading the instructions. Kids, don't try this at home. After poking around for several minutes while the news stories flew by, I thought it advisable to give the docs a once over. When I went to close the program, it announced that I had actively been archiving 227 stories to disk! Now, I subscribe to three dailies, six weeklies and at least as many monthlies, but 227 articles in under five minutes! I was stunned. I had the media monster by the tail.

Fortunately, it took only minutes to tame the beast. The software allows you to select which services you wish to save information from at any given time. I started by turning off all the weather, sports, entertainment and financial news, and severely limiting the business, national and world news. What I was left with was news about the X-Press service, and information about its use in education.

As an experiment, I initialized a *clipping folder* to snag any stories relating to 'computers.' Because the flow of information from X-Press is omnipresent, the software is designed to run unobtrusively in the background as you proceed with your other desktop assignments. Later in the day, I checked back with X-Press and in my personal directory were nine articles about 'computers,' a collection of general stories from the services I had left active, and a special directory file informing me of the software files that would be available for downloading over the next week. The software list included a patch or bug fix that would correct a windowing problem I had discovered earlier in the day.

So I marked that file to be delivered to me that evening and went to read the articles in my 'computer' clipping folder. With a little refining this thing could be the Personalized Virtual Newspaper we've heard so much about. The X-Press interface includes a whole set of commands that are aimed at stock and commodity market tracking.

In vaguely-related news: A Massachusetts company has developed digital communications hardware that will permit schools, businesses, hospitals, or yes, even studios to use their local CATV systems as high speed computer networks. Applitek Corporation of Andover, MA, has created a system called LANcity. Without getting into the nuts and bolts, it supports the integration of smaller Ethernet Local Area Networks into one big Metropolitan Area Network (MAN). Literally, any classroom, office, studio or living room with a cable TV outlet and a PC can be hooked into this computer network.

Potential benefits might include real time database research and maintenance, resource sharing (sample libraries anyone?), administrative paperwork and operating cost reductions, low-cost/hig -speed two-way data communications for users such as a production facility spread over several floors of a big building, or even among separate buildings. Interesting, yes?

If you ever wonder where I dig up some of these esoteric digital services, I feel compelled to take just a moment to plug the annual Seybold Digital World Conference coming to Beverly Hills, June 1992. The event is three days of total immersion in digital communications, storage and retrieval, with sound and lights, moving or not. It's about computers and telephony multimedia and music — in fact any thing that can be converted into bits and moved around. If you need to know where the digital world is headed, this conference will give you lots to chew on. Examples of subjects addressed include Broadcast Quality Video in a Box; a Multimedia Platform Shoot-Out; The New Consumer Devices, discussing 'personal digital assistants' and *producing* for them; Delivery of Entertainment to the Home will focus on the radical changes occurring to music, film and video distribution channels and; The Other Information Networks looks at cable, cellular and direct-broadcast satellite technology as additional distribution channels.

The bottom line is, if you want your studio to have an edge in the next millennium, you need to know this stuff. The Digital World Conference is a great place to get your feet wet. ■

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Who's Laughing?

By Dwight Cook

Are you fascinated with the concept of the "tapeless studio?" Are you wondering which workstation to buy? Are you worried about whether you bought the right one? Are you as tired of this recession as I am? Let's start there — it's time we all did something about it!

When I think about this economy I am reminded of my trip to L.A. for the SPARS Business Conference last January. I went to glean some tidbits of wisdom from my many SPARS colleagues. I sure came back with some bits, and with a little effort I might even apply a few ideas.

Being serious is important and has its place, but it's also important to loosen up and look on the lighter side — something my wife is always telling me. I firmly believe that we should never go through the day without a good laugh. It's therapeutic. Also, we should find time to get our minds off work. Use that time to do something with the family or for recreation. Find some time to be alone. I set aside a little quiet time Monday through Friday by getting up 30 minutes before the rest of the family.

The laughter, getting away from work and being alone can help you to focus your energies more productively. If you plan on being a leader in this business, you have to focus and have a clear idea of where you are going. And if you don't know where you're going, don't expect anyone else to follow.

After a full day at the SPARS business conference, Barry Landrum, (a great motivational speaker and one of the funniest human beings I have ever met) and I were ready for a laugh and decided to visit a comedy club on Sunset Boulevard. We were given front row seats and believe me, we heard some bonafide bad jokes that night. A lot of the humor was in poor taste and the comedians were simply going for the shock value. I think a truly funny person makes you laugh without going for the gross-out.

One comedian didn't have any jokes at all, just props. Barry and I were not laughing — it was simply a bad act. Midway through his performance, the comic threw a sheet over Barry and me, stuck his head under, and said "What's wrong with you guys? You haven't laughed once during my act." Then the comic climbed out from under the sheet and continued his promenade across the stage. I quickly removed my head from underneath and began to laugh. I was laughing because Barry was still under the sheet. Barry had it pulled up close to his face and was trying to watch the performance through the sheet. After about 10 minutes he came out from under the sheet and gave me a wild look when he realized that I was already out. To this day, Barry still says, "Dwight, I can't believe you left me alone under that sheet." This recession is like that bad act — it's no laughing matter. But you have to find something to laugh about.

My wife got one of those "word-a-day" calendars for Christmas. And nobody can accuse you of misspelling your own word. One day last week the word was "ostrichize," meaning you didn't believe or accept things the way they really were.

Just how long did it take our government to accept the fact that we were in a recession? Too long. And how long is it going to take us to get out of the recession? Too long — unless we take the offensive position. Let's think positive and do something positive. You can make a difference within your business, and within your sphere of influence. Get out from under that sheet and face the music!

Workstations: there are a bunch to choose from. Are you thinking of buying one? Do you have to have one? Which one? Let's get back to the basics. Don't cave in to peer pressure. Buy one if you need one and buy the one you need.

I do some computer consulting as well as this studio stuff. I am reminded of a conversation I had with a client who asked me, "What computer should I buy?" The answer was obvious to me. "First, find the software you need for your task," I said. "If you can find commercial off-the-shelf software, get it. Otherwise, have custom software written. Then find the kind of computer you need to run it. Of course, you may need a faster, more expensive computer if you have big storage needs, want graphics or require networking." Purchasing a workstation demands the same logic. Find the workstation that has the software that does the job. Expect to pay more for faster boxes with more storage.

If you are not computerized, you better get with the program — now. But let's remember that computers by themselves don't make one successful. They are just a tool, as is a workstation. Our tools are changing, and not always for the better.

Many people currently can't edit a voice track with a mouse as fast as a good engineer can with a blade. If we're not careful we could replace razor cuts with carpel tunnel syndrome. I think the ideal workstation would use standard computer hardware that users can upgrade in the field. After all, we've heard it said that the audio industry is at the mercy of the computer industry.

I would also like to see more manufacturers coming to our facilities to see the kind of work we do and how we do it. This would result in more efficient, faster workstations with better interfaces. Thankfully, this is already beginning to happen. ■

Dwight Cook was the first vice president of SPARS, and is currently president of Cook Sound and Picture Works, Houston, TX.

VARIABLE ARCHITECTURE SYNTHESIS TECHNOLOGY

HANDS ON:

KURZWEIL K2000

By Trammel Stark

The meteoric evolution of samplers and synthesizers has made them an unavoidable commodity in today's production environment. It is a fact of studio life that facilities in today's hyper-competitive marketplace are required to be fluent in the real world applications of sampling and synthesis, whether for music production or for the manipulation of sound effects and dialogue.

The latest trend in affordable music workstation design has seen the development of hybrid instruments that seek to effectively combine the worlds of sampling, digital synthesis and digital effects processing into one practical and easy-to-use unit. This concept had its beginnings in the development of the Synclavier and Fairlight instruments of over a decade ago (notice the term "affordable" in the previous sentence). These high-end instruments (or more correctly, DAWs) continue to push the technological envelope of sound design, and may even be given credit for changing the complexion of recording studios as we now know them today. All this progress, however, does not come without a price tag.

Trammel Stark is an Atlanta producer, composer and arranger.

For those of us whose equipment budgets do not reach into the lofty heights of six figures, MI manufacturers are creating instruments that are capable of performing ever more complex tasks while retaining a price structure that will allow the majority of professionals to sustain more mundane callings. Such as, possibly, food and shelter.

Kurzweil (now a division of Young Chang) has quite possibly set a new standard for these instruments with the introduction of the K2000. The compact, unassuming appearance of this unit belies the impressive power that resides inside.

OVERVIEW

Kurzweil has appropriately given the name VAST (Variable Architecture Synthesis Technology) to the K2000. This synthesizer generates sounds from a variety of sources. The sounds emanate from instrument samples and sampled synth waveforms stored in ROM, as well as from samples loaded from floppy disks into sample RAM. These sounds may then be modified by an impressive list of DSP functions. The 24- voice K2000 is shipped with 8Mbytes of ROM, expandable to 24Mbytes through the addition of two optional 8Mbytes ROM kits. The standard 2Mbytes of sample RAM is expandable to 64Mbytes through the addition of Macintosh simms. (That's right, 64Mbytes!)

So there you have a workstation with the potential of 88Mbytes of samples and waveforms on line at once! For storage purposes, the unit comes standard with a built-in floppy drive that utilizes 720Kbytes or 1.44Mbytes MS-DOS formatted disks. In addition to the floppy drive, there is a provision for an optional internal hard disk, and the unit comes with a 25-pin SCSI port to facilitate use of an external hard disk and/or CD-ROM. The internal effects processor is essentially a Digitech 256 that is capable of providing four simultaneous effects and is controllable in real time or through MIDI. Finally, an optional stereo sampling board is currently under development that will incorporate 1/4-inch stereo analog inputs as well as digital AES/EBU/S/PDIF and optical digitals inputs and outputs.

The K2000 has a 5-octave, 61-note keyboard that is spring- loaded and sends and receives attack velocity, release velocity and mono pressure. The unit receives, but does not transmit poly-pressure. The front panel has eight mode select buttons that each have secondary uses in edit mode, a volume control, an assignable control slider, six soft keys underneath a 240 × 64 LCD readout, cursor buttons, a large alpha wheel and an alpha-numeric keypad. There are also 'mark' and 'jump' buttons that allow access to previously used pages while in edit mode.

On the back panel one finds the power switch with a voltage selector that will allow the unit to use local power in any part of the world, a headphone output, six audio outputs configured as a stereo mix with two additional stereo outs (that also function as four independent mono outs), two footswitch jacks, a control pedal jack, MIDI ports and the SCSI port. The group A and group B outs may be used as insert loop-throughs for outboard effects devices through the use of a stereo insert cable, with the tip functioning as a send and the ring functioning as the return. The mod wheel, control slider, footswitch jacks and control pedal are all independently programmable.

OPERATING MODES

The K2000 is divided into eight operating modes. The first is Program Mode, where one chooses, edits and plays the various programs. The program, keymap and sample editors are all found in this mode. Programs begin life as individual instrument samples or as sampled waveforms that are assigned to a keymap. The keymap is then routed to an algorithm (more on these later) where the samples are modulated, shaped, filtered, sliced, diced and otherwise manipulated to your heart's content. This resulting sound is referred to as a layer.

Finally, up to three layers may be stacked, each with different keyboard ranges that may split or overlap and with individual controller response characteristics. This layered combination comprises the completed program. The layer is the basic unit of polyphony, therefore if a program is made up of three layers, depressing one note will take up three of the 24 available voices.

As defined by Kurzweil, an algorithm is basically a signal path that serves to route the sound source through a variety of user-selectable DSP functions. The DSP functions chosen within each segment of the algorithm determine the type of synthesis that is employed. There are 31 predefined algorithms in the K2000, with up to five DSP modules depending on which algorithm is chosen. Each segment of the algorithm may also be modulated by a variety of control sources.

Some of the many DSP functions included are: high frequency stimulator (a type of exciter), parametric and graphic equalization, four-pole low-pass and high-pass filters, a band-pass filter, various low frequency oscillation waveforms, distortion and numerous others.

A multitude of other functions may define such parameters as the program's response to pitch bend, velocity, assignable controllers, and keyboard tracking, reverse samples, adjust fine tuning control, adjust the wet/dry mix of the effects, create envelopes, save and copy layers, or implement other basic utilities and programming options. It is safe to say that there are ample structural and control options to keep even the most jaded sound designers occupied for the duration of life as we now know it!

The next step in the layering scheme of the K2000 is the setup mode. In this mode, up to three programs may be combined as a single performance combination. Each of the zones may have its own program, MIDI channel, controller assignments, transposition, key range and effects mix.

Quick access mode is used to arrange random programs or setups into groups of

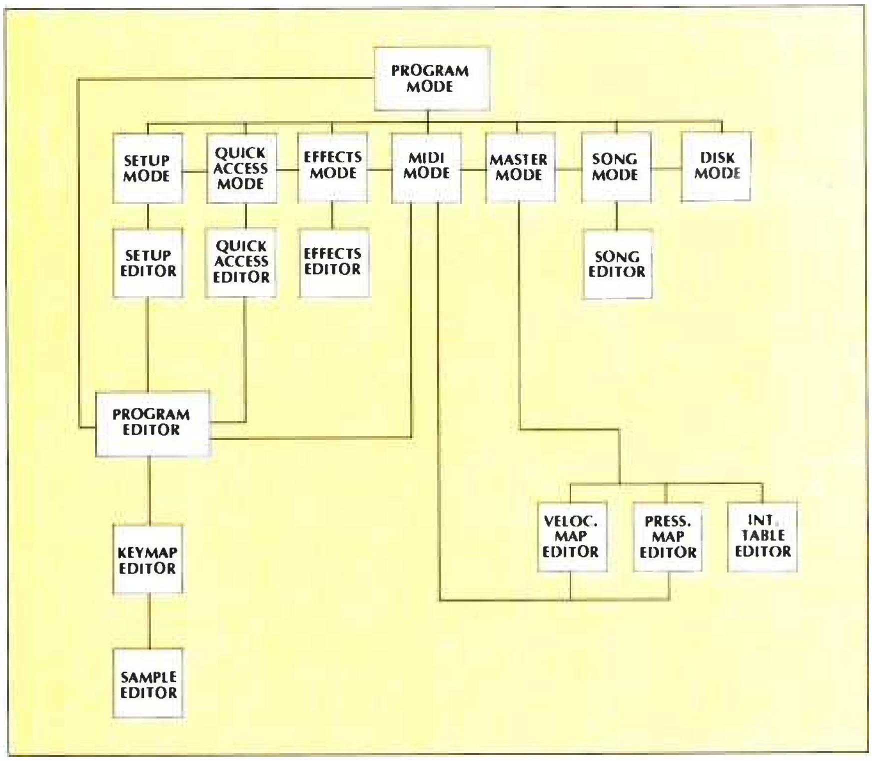

Figure 1. Flow chart of Program Editor showing "nested editors," each related to the parameters that make up different components of a program.

VAST CONSIDERATIONS

By Paul Lehrman

The first time I ever got incredibly excited about a synthesizer was my sophomore year of college, when I first laid eyes on the Buchla modular system at the Columbia-Princeton Electronic Music Center. I can't say the Kurzweil K-2000 gives me exactly the same rush, but after spending the last month with it, I can say it comes awfully close. This is a box with just about everything that synthesizer, sound design, and MIDI heads have been lusting after for the past decade, at a price you won't have to re-mortgage the house to afford.

It slices, it dices, it samples, it synths, it filters, it effects, it even processes external audio. And it does everything very well. To understand the concept behind it, imagine sticking an Oberheim Matrix-12, with all of its filtering, modulation routing, and real, twiddleable knobs in the same chassis with an Akai S1000. The filters, processors, equalizers, "stimulators" and other programs are incredibly flexible, and sound beautiful to boot.

Okay, let me calm down a bit. Is this the ultimate workstation, the last piece of MIDI gear that you will ever need? No, for a number of reasons that I'll get into a little later. But it does more than any other single less-than-$50,000 piece of equipment I've ever seen, with very few compromises. As the center of a MIDI-based studio, it is ideal, and as a live-performance instrument or portable music and effects studio, it is unsurpassed.

Philosophically, the K2000 is closer to Kurzweil's original K250 than to the 1000 or 1200 Series. The K250 was a real sampler with substantial processing power, but it was extremely difficult to use, so most users stuck with the factory sounds. The company's engineers took this into account when they put together the 1000 Series, which were playback-only modules, featuring twice as many voices, a complex but relatively understandable operating system and price tags running about 80% less than the K250.

The K2000 handles both sampling and synthesis with ROM and RAM sounds. It does straight sample playback like the 1000, but also provides an enormous amount of DSP power on top of the basic waveforms. The ROM samples are no slouches and sound fine just the way they are, but the fun begins when you start using the filters and processors to make subtle or gross

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10 entries each. This is very useful for arranging programs or setups into a specific order for live performance, and to arrange categories of sounds into related groups for studio situations.

As previously mentioned, the effects processor in the K2000 is effectively a Digitech 256 chip set. There are 47 factory presets in ROM and 80 user-programmable preset locations. The digital effects may be set to respond to program FX assignments, setup FX assignments, or to ignore these pre-programmed assignments. It is also possible to assign a MIDI channel to the effects (FX channel) that will change effect programs independently and that will allow real time control of the wet/dry mix and two parameters, variable from preset to preset.

There are 27 configurations of effects that range from stereo chorus, delay, flange, various reverbs, and parametric and graphic EQ, to combinations of up to four simultaneous effects. This is probably the finest onboard effects device that I have heard on any synthesizer.

MIDI mode configures the K2000's response to incoming MIDI messages and configures MIDI information that the keyboard sends to other devices. The MIDI controls are flexible and serve to make the K2000 a capable master controller. While in 'program' mode, the K2000 transmits on one MIDI channel at a time, but in 'setup' mode, it is capable of transmitting on three MIDI channels at once, with each channel having it's own key range. One exciting function called 'buttons,' enables a user to record a sequence of button presses into a sequencer and then play back the stream of events as a macro. One possible use of this feature would be to record a sequence of events that on playback, will enter disk mode, select a specific SCSI device, and load one or more banks of samples or programs. The K2000 also responds to the MIDI controller 0 program change command.

This latter allows direct MIDI access to program numbers that exceed the original limit of 127 without the limitations inherent in using patch maps. The K2000 numbers programs and setups in 10 banks of 100 locations each (000 - 999).

Master mode is a global utility mode that controls tuning, transposition, velocity and pressure sensitivity, panning of outputs A and B and 'drum channel.' This is a specific MIDI channel that is used for drum programs, allowing 32 split points in the program rather than the usual three layers. It also allows other manipulation of the drum sounds such as individual drum EQ, filtering, envelopes and note assignments.

The master mode also contains a list of 17 possible intonation tables for those of you who are into non-western tonalities (a little Tibetan hip-hop anyone?). These scales are not editable, however, it is possible to create custom microtonal scales within the keymap editor. Another interesting feature is called MIDIScope. This displays MIDI messages that are also sent and received by the K2000.

Song mode calls up the 'scratchpad' sequencer. This is a very rudimentary 15,000-note, 1-track sequencer that will allow multi-channel overdubbing, but no editing at all. Its only metric setting is 4/4. The tempo is adjustable from 1 to 255 BPM, and the sequencer may be locked to external MIDI sync. It will, however, record multi-timbral sequences from an external sequencer and will play back type 0 MIDI sequence files (1-track, multi-timbral sequences).

It is satisfying to see an instrument manufacturer that does not force one to purchase an onboard sequencer that will rarely, if ever, be used. Granted there are many users for whom these sequencers are just the ticket for casual gigs or songwriting, but I would assume that most users who purchase instruments in the $2,000 to $3,000 range are probably already using software-based or hardware sequencers and have no desire or need for an elaborate onboard sequencer. I would like to think that these resources are put to use in other areas of the synth that are of more practical benefit.

Disk mode handles all disk utility functions. In the K2000, disk files contain what are referred to as 'objects.' An object may be defined as a program, setup, quick access bank, keymap, sample, effect, etc. The K2000 will load Akai S-1000 sample files directly. It does not recognize program or drum input information due to a different operating system and architecture than the S-1000, however,the K2000 will ask if you want to create a keymap for the S-1000 sample before loading and will assign the sample root to C4 if you answer yes. Otherwise, a keymap must be created in the program edit mode before the sample can be played.

IMPRESSIONS

Now that we've waded through a brief synopsis of this keyboard, what, you may ask, does this thing sound like? For starters, the architecture of the K2000 allows for the simultaneous creation of various types of synthesis.

In addition to sample playback, the instrument is capable of emulating the L.A. synthesis of the Roland 'D' Series synthesizers, emulations of vectoring synthesis in which the sound may evolve over a span of 20 or 30 seconds. Additionally, the K2000 creates some of the most authentic analog sounds that I have ever heard coming from a digital synth. For all you proponents of the "more is more" school of music production, it must be noted that this synthesizer has the equivalent of four oscillators per voice, thereby making it possible to stack up to 96 oscillators in mono mode! Sufficiently huge sounds are possible, however, without resorting to such excess.

The factory samples, as one would ex-

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changes in the sound. Want to make a hyper-realistic trumpet patch in which aftertouch controls vibrato rate and foot pedal pulls a Harmon mute in and out? It's actually easy to do. Want to create an entire symphony orchestra spread across the keyboard, with different balances, pan positions and room simulations available at the touch of a button? That's not difficult to do either.

The optional sound blocks that will be coming out over the next few months will contain, among other sounds, the K250's and 1000's "greatest hits," and in all will let you have up to 24Mbytes of ROM. Add to that 64Mbytes of RAM (using 16Mbyte SIMMs) and you arguably have the largest sampler in the world. You can't do direct sampling into the K2000 yet (that option will be out this summer), but the designers have included SCSI-based sample transfer, using the Peavey-designed, and hopefully soon-to-be-standard, SMDI protocol. For the first time in all my experiences with samplers, this actually worked the first time that I tried it.

This means that you can record samples with a Sound Tools or Pro Tools system, an Audiomedia card, a Peavey module, or any other SCSI-based sampler, and load them into the K2000 quite quickly. (No, sorry, you can't play the thing while you're loading samples over SCSI.) Right now, the only computer software that supports the K2000 correctly is Passport's Alchemy (in a version that as of this writing is still in Beta-test), but that program handles all types of Digidesign files, and we can hope that Digidesign's software will be upgraded soon as well.

In fact, the SCSI implementation lets you do something brand new: You can connect a Syquest removable drive to a Macintosh and the K2000 at the same time, and use it with either device, without switching cables. When you pop a K2000-formatted cartridge in the drive, you can read it and write to it from the K2000, and when you put a Mac-formatted cartridge in, the Mac takes over without even blinking. Furthermore, regardless of what's in the drive, the Mac and the K2000 can continue to communicate with each other.

You can also put a hard disk drive (any Macintosh-compatible 1/3-height Connor or Quantum mechanism, up to 240 Mbytes) inside the case, making the K2000 truly self-contained.

SPECIAL F/X

The digital processor's effects are programmable in real time. Each sound program has an effects program asso-

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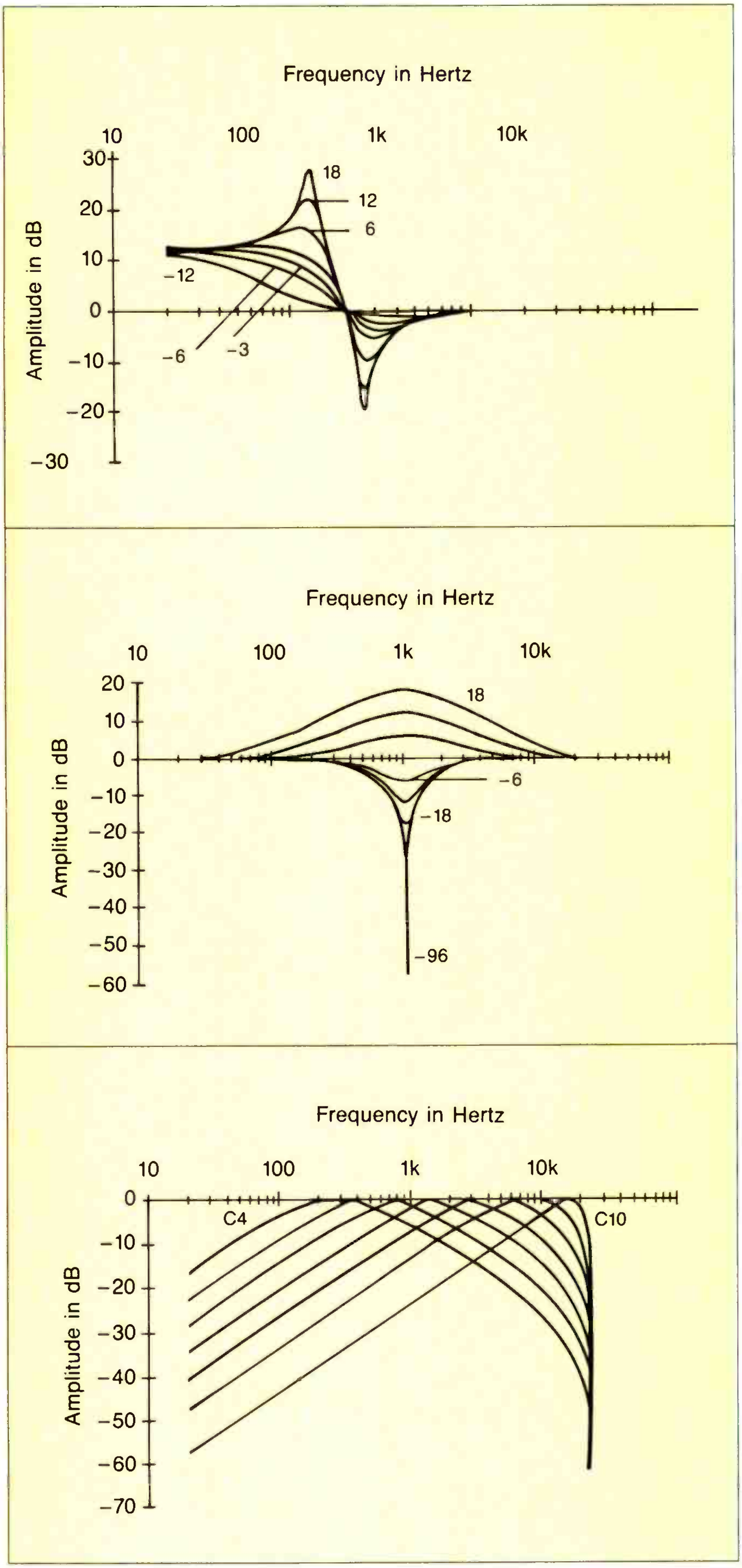

Figure 2. Examples of K2000 DSP filters: Top graph—steep resonant bass; middle—parametric equalizer; bottom—two-pole bandpass filter with fixed width.

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ciated with it, and the effects parameters that you can control with the sound program are determined in the program editor, not in the effects editor — an unusual arrangement, but one with a certain logic behind it. Beside wet/dry mix, each program allows two effects parameter assignments.

In master mode, the effects are "decoupled" from the programs, so changing a sound program doesn't change the effect. This is the way the unit will be used most often in the studio, and in this mode an unlimited number of effects parameters can be controlled over MIDI. However, which MIDI controllers adjust which parameters is not changeable by the user, and unfortunately is also not documented, so the feature is a bit difficult to use.

So, are you ready to buy yet? Hold on — there are a few things you need to consider. First of all, as of this writing, the K2000 isn't finished — all of the drool-inducing options that Kurzweil/Young Chang has promised are not implemented or available yet. Which is not to say the K2000 doesn't work: on the contrary, unlike some computer-based pro audio products that are released before their time, the K2000 won't wipe your hard disk, fry your power supply, or wreck your production schedule. In fact, the thing has never so much as hiccupped since I first turned it on. It just doesn't yet do everything that you might like it to.

Right under the sampling option on your priority list will probably be expanded program RAM. This is different from sample RAM, because it's non-volatile, and therefore more expensive. The stock K2000 has 128kbytes of program RAM. That may sound like a lot, (I'm told the 1000s had about 1/10th of that) but programming objects in the K2000 are *big*. The stock voices are more intriguing than they are strictly utilitarian, which is a trend that I welcome, because they will inspire you to dig into programming the thing, but it means you are going to want to customize most of your voices, and you will need space to put them in. Right now there's room for a couple of hundred RAM programs, but (especially if you've got 64Mbytes of samples to play with) you may quickly find that's not enough.

The $3,000 price tag certainly offers a lot for the money, but it's a bit misleading if you plan to use the K2000 to its full potential. A full-blown system, including the sampling option, sample and program RAM and a big hard disk will more than double that price.

Some users will consider the 24-voice limit a problem. With three voice lay-

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pect from a company with Kurzweil's sampling pedigree, are all quite good. Acoustic and electric pianos are very well represented. There is an especially nice Rhodes program called 'tine piano' that is reminiscent of a "dyno my piano" modified Rhodes.

Electric bass samples cover the usual range of fretless, picked, and funk bass, and as one might expect from this instrument's analog emulation capabilities, synth basses are very good. There are a variety of guitar programs ranging from aggressive leads to acoustic six and 12-string. One of my favorites is the '12-string with Leslie' program.

All the other usual sounds found on high-quality synthesizers these days exist in abundance, such as 'tone wheel' organs, lush strings, sax, trumpet, and brass samples, synth brass, ('Toto brass' is very nice) and cherubic vocal choirs. There are also great drum samples. Kurzweil has recorded all new drum and percussion samples for the K2000, and you get a wide range of sounds, from huge rock drums to tight dance drums, as well as all the usual percussion. Some of the drum samples were recorded dry, and others recorded with ambience. As stated earlier, all drum sounds may have independent filtering, EQ and envelopes applied in the drum channel mode.

AMAZING SYNTHS

The K2000 really shines with its indescribable synth programs. These run the gamut of delicate and sweet to downright frightening. There is one particular speech-like program that may best be described as a recitation by Cousin It's big brother. In setup mode, the sounds really get interesting, with all the layering capacity of the K2000 taken to extreme. These sounds really must be heard to be fully appreciated.

As is true for all synthesizers, these factory programs represent a starting point for the creation of sounds. Several third-party programmers are already at work creating new sounds for the K2000, such as Sound Sources and Stratus Sounds. Kurzweil also has a team of programmers at work converting all of the existing Kurzweil 250 and K-1000 samples for use in the K2000, as well as creating new samples and programs. Additionally, Opcode has developed a K2000 module for its Galaxy Ed/Lib.

All in all, Kurzweil has come up with a formidable contestant in the never ending war of the synthesizers, and the K2000 may well prove to be the Mother of all Keyboard Workstations. The sheer potential for onboard memory alone is truly remarkable. With the addition of the upcoming digital and/or analog stereo sampling option, the K-2000 will prove itself a very capable sound design and production centerpoint.

In spite of all that was done right with this instrument, I see one potential shortcoming. This has to do with the 24-voice polyphony. Normally, 24 voices is more than adequate for most synths and samplers, and in many cases may be enough for the K2000. But given the abundant layering capacity of this instrument, it is possible to run short of voices very quickly. Granted, not all music requires massive simultaneous layering on multiple parts, but consider that in program mode, with the maximum of three layers playing at once, a 5-note chord will require 15 of the 24 available voices. In setup mode, with up to nine possible layers, well, you add it up. Also, stereo samples require two voices per note. It would be interesting to see some sort of expansion module (maybe be along the lines of the Akai S-1100 expander) that would offer additional voices under the control of a single K2000.

Another area of concern has to do with the fact that there are only six outputs. In many cases, once again, this is sufficient. However, while recording, especially in a virtual tracking situation, six outputs may be used up very quickly if one desires to route sounds through individual channels on the console.

However, all things considered, Kurzweil has released a tremendous synthesizer/sampler in the K2000, one that will no doubt take its place among the most prominent and important of the instruments currently available. ■

Circle (100) on Rapid Facts Card

SPECS AND DESCRIPTION

Manufacturer:	Young Chang America
Contact:	David Fox 13336 Alondra Blvd. Cerritos, CA 90701 310-926-3200 FAX: 310-404-0748
Model:	Kurzweil K2000
Price:	$2,995
Description:	VAST (Variable Architecture Synthesis Technology) Synthesizer/Sampler Stereo, 24-voice 8Mbytes ROM, expandable to 24Mbytes 2Mbytes RAM, expandable to 64Mbytes 720/1.44Mbyte floppy disk drive Optional internal hard disk SCSI port for external hard disk and/or CD-ROM Digitech 256 internal DSP Optional digital I/O, stereo sampling board 5-octave, 61-note keyboard Six audio outputs (stereo mix plus four individual mono outs) Loads Akai S-1000 sample files directly

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ers available per program, it is true that you could use up voices quickly. However, these are not sine waves we're talking about here: Each of those voices is a rich waveform in itself, and with all of the DSP in the various algorithms available, you can create incredibly interesting sounds with a single layer. In addition, when it comes to intelligently allocating and stealing voices, Kurzweil has always been one of the best in the business (most people who played a K250 could never *really* believe it only had 12 voices), and the K2000 is no exception.

If you want to use it in post, you're going to have to hook it up to a good MIDI sequencer.

Will you run out of voices somewhere along the way while you're doing major production? Undoubtedly, but you can always take your more conventional sounds (pianos, strings, brass) and "off-load" them onto a 1200 module, an E-Mu Proteus, or for that matter a Roland Sound Canvas, leaving the K2000 to do what it does best. (And yes, there will be a rack version of the K2000 before long.)

The number of audio outputs may be a more serious issue. Although there are six holes in the back, there are really only four output channels (of course, this is still better than the two found in the 1000s and the original K250). The "mix" outputs merely combine various configurations of the A/B/C/D signals, dry or through the onboard digital processor, and do not accept voice assignments of their own. This is a very nice scheme if you have a limited number of mixer inputs available, but since mixer inputs are a dime a dozen these days, I don't see it as much of an advantage. If you want to do discrete outboard processing of many different sounds emanating from the K2000, this will be a limitation.

On the other hand, the designers have done all they can to make the output structure flexible. One positive thing to be said for it is that, as on the Proteus, the individual outs are actually channel inserts, so you can route their signals through an external processor and then back into the K2000, to emerge again from the mix outputs,

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with or without further processing from the K2000's own chip. Having a digital equalizer available for each voice helps as well. Furthermore, not unlike the Korg Wavestation A/D, you can take an audio signal from an altogether *different source*, bring it into the K2000, process it with the on-board processor, and mix it with the K2000's different sounds.

For the amount of MIDI control this beast provides, it's surprising that there are so few physical controllers available: two foot switch inputs, one control pedal input, and a data slider, plus modulation and pitch wheels. If you want to add more real time control, you'll have to add something such as a Lexicon MRC, and such devices don't come cheap.

WRAP IT UP

But enough of its shortcomings. There *is* room for improvement, but there is also no doubt that the K2000 is a truly magnificent musical instrument, a giant leap forward in the integration of sampling and synthesis, with great sound and a surprisingly comprehensible operating system. I foresee it having a long and fruitful life in many different facets of the industry, from composition, to post-production, to live performance.

Is it a true audio workstation? No. It doesn't do direct-to-disk recording, it has no direct SMPTE inputs, and it has no word clock input, so its samples cannot play back locked to any external source. If you want to use it in post, you're going to have to hook it up to a good MIDI sequencer. But then it will do plenty of amazing things: play full scores; create, edit, and place sound effects; fly in good-sized chunks of dialogue or sung vocals; and accomplish some heavy-duty real time processing. And, you can carry the sucker underneath one arm to your next nightclub gig, as well. ■

Paul Lehrman, a long-time contributor to R•E•P, is a composer, author and consultant in the Boston area. He is on the faculty of the University of Massachusetts-Lowell, and serves on the executive board of the MIDI Manufacturers Association.

First Look

By Laurel Cash-Jones and Fred Jones

Normally in this column we go in depth about a particular product or two and give you all the skinny on it, hopefully before anyone else does. Because we have been flooded with a bunch of new things, we thought we would deviate from this tried and true formula and get to all of them in one month. Hey, we're flexible.

TAKE IT TO THE LIMIT

With that introduction, you probably expect us to talk about some new limiters. You're right. UREI is introducing three new compressor/limiters; the LA-22, LA-12 and LA-10. They are all single rack space units, and are designed to handle signals in excess of 24dBu, but that is where the similarity ends.

The LA-10 is a single-channel unit, while the LA-12 is a dual-channel unit. The LA-22 is also a dual-channel unit, but it features a parametric filter that can be adjusted to change the function from being a broadband unit, to a frequency specific one.

Circle (125) on Rapid Facts Card

WE'RE KEEPING IT ALL UNDER CONTROL

Since the MIDI manufacturers of America and Japan MIDI standards committee released the specifications of their new MIDI Machine Control protocols (called MMC), tape recorder builders have been staying up late trying to add this new feature to their current line of equipment. This newly defined protocol allows you to control and automate most of the functions of a standard tape recorder via MIDI commands from a synthesizer or computer.

Fostex has been a serious advocate of this idea. In fact, they have already had a product in the field for some time that allows machine control via MIDI — the R8/MTC-1. They are now introducing a new version of the product that conforms to the newly released MIDI MMC standards, which makes them among the first to have a product with MMC.

The Fostex MMC-1/B is now available, and as an added bonus to current owners of the R8/MTC-1, they will update your system at no charge. (You *should* take them up on this.)

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Laurel Cash-Jones is an editorial consultant to R•E•P and a free-lance writer. Fred Jones is a free-lance engineer, producer and writer.

DO YOU BELIEVE IN MAGI?

If so, you will really appreciate this. J.L. Cooper Electronics is announcing the addition of the P-1 Mounting Pan to their MAGI II automation system. As you are probably aware, the MAGI II is a cost-effective console automation system, controlled by a Macintosh computer, that can provide up to 64 channels of automation to your existing console in either an external or internal version.

The new P-1 Mounting Pan standardizes and simplifies the installation of the internal version, the MAGI IIi, and is particularly useful for small consoles that don't have space available inside for mounting the internal version and have to use external faders. A single P-1 can hold up to 32 channels, and because all automation controls can be factory installed, it reduces installation time at your studio.

Circle (127) on Rapid Facts Card

ADAT IS ADAT BUT ...

DataSYNC is here to solve your problems before you even have one. What we mean is, the folks at J.L. Cooper Electronics (remember them?) are building a much needed accessory to the Alesis ADAT 8-track digital recorder before it is even widely available on the market.

J.L. Cooper's new dataSYNC generates MIDI time code from the Alesis ADAT's sync out. This bi-taccurate unit plugs into a 9-pin port on the back of the ADAT and converts the sample clock to MIDI time code. This means you can drive a sequencer or digital audio workstation without having to lay down MIDI or SMPTE time code on one of the eight tracks on the ADAT, thus saving you some very valuable space on the tape.

A MIDI input allows new MIDI data to be merged while locked to the ADAT. Lock and run LEDs tell you what is happening, and the dataSYNC converts the ADAT's transport functions into MIDI Machine Control (MMC) messages. The dataSYNC is a 1/2 rack-space workhorse, just ready to get to work for you as soon as you as you have an ADAT. How do they do all of this for $349.95? Will wonders ever cease?

Circle (128) on Rapid Facts Cards

FROM MASTERMIND TO MACROMIND

MacroMind-Paracomp is announcing the release of SoundEdit Pro and the

First Look
Continued from page 63

MacRecorder Sound System Pro, which are (as you may have guessed) completely new versions of their hard disk recording software for the Macintosh operating system.

SoundEdit Pro's new features include the ability to record, edit, and play sounds directly from the disk, thus eliminating RAM constraints. You may also now input sound from any hardware device whose driver is compatible with the Macintosh Sound Input Manager, open and save sounds at 8 or 16 bits, and sample sounds at any sample rate up to and including 48kHz.

You may also view the sound spectrum waveform in 2D and 3D simultaneously, and work with additional file formats, such as Sound Designer II, System 7.0 SND, AIFF-C and System 7.0 disk-based sound resources and virtual memory. Current registered users of earlier MacRecorder versions can upgrade for $75. New versions:

MacRecorder Sound System 2.0.5, $249; MacRecorder Sound System Pro, $349; SoundEdit 2.0.5 (software), $195; SoundEdit Pro (software), $295.

This program is particularly well-suited for multimedia applications. A suggested add-on is the new CS-1 control station from J.L. Cooper. It is a universal controller, with five user programmable function keys and two modifier keys that allow for a total of 20 possible key combinations.

Circle (129) on Rapid Facts Card ■

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problem but not eliminate it, thereby making diagnosis elusive.

Most ATR dancer arms are designed to smooth out residual tension variation caused by imperfect supply-motor drive. The position of the dancer arm may affect wow and flutter performance, as may fine adjustment of the dancer arm's damping mechanism.

Curing wow and flutter problems is usually time consuming and mentally taxing. Ease the process by taking careful wow and flutter measurements when you first start your repair, and at each step along the way as you make adjustments or replace parts. This will provide a written record and will be useful next time the ATR comes down with a case of wow and flutter. ■

Too New/In Progress

Cutting Edge breaks tradition slightly for this issue to continue Workstation coverage of products that we haven't been able to evaluate or are not yet available on the open market.

Product vendors in 1992 Update:

AMS AUDIOFILE OPTICA

A simpler, 4-channel version of AudioFile PLUS, the AMS AudioFile OPTICA is intended for simple track laying operations. Using a MOD, the OPTICA system allows fast transfer between AudioFile systems. Capable of recording manually or automatically with reference to a video-style EDL or as an ADR device, the OPTICA can also operate as a background recorder in a video edit suite.

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DYAXIS LITE

Scheduled for mid-summer release, the Dyaxis Lite is a low budget, basic task, 2-track digital editor. A specially authored version of Studer Editech's MacMix software provides digital editing features with industry standard commands and terminology. The engineer-friendly Dyaxis Lite Remote Controller is configured with familiar audio controls. Dedicated keys execute most editing functions while 10 softkeys allow operators more commands to meet specific needs. Dyaxis Lite data files are fully compatible with the Dyaxis I and the Lite can completely upgrade into a Dyaxis I. And, little-or-no computer experience is required.

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SSL SCENARIA

SSL's new Scenaria includes a 38-channel automated digital mixing console, a 24-track random access recorder/player, multitrack audio editor and automated audio/control routing. Compatibility with ScreenSound and SoundNet and multiple machine control is provided. The anchor to this package is the VisionTrack function that integrates digital random access video storage. Combined with the 24-track random access audio recorder, VisionTrack permits instantaneous location and rollback of audio and video to any point in the soundtrack.

Circle (132) on Rapid Facts Card

WAVEFRAME 401

WaveFrame recently introduced a moderately priced disk recorder — the WaveFrame 401 — which uses the same disk recording and editorial software used on the newly reconfigured WaveFrame 1000 and 400. This new system offers 8-track recording, editing, and mixing, and is intended for music production, CD mastering, digital post production and other applications. Digital I/Os, VITC sync, and multitrack punch-in recording are also part of the package.

Circle (133) on Rapid Facts Card

Other Approaches:

DIGITAL AUDIO LABS

The CardD, from Digital Audio Labs, is (more or less) a PC workstation "kit," used to deliver two simultaneous recording/replay channels and intended for audio post for video, music editing, broadcast applications and CD pre-mastering. Version 2 of The CardD editing software, called The EdDitor, delivers fade/crossfade functions, markers, edit history, a save/load edit list, and recording VU meters. "Cataloged" cues can be arranged and triggered by assigned keys on the alphanumeric keyboard. The system can be installed in some portable PCs, but requires 386/16MHz and 2Mbytes of RAM or higher. For successful operation, the RAM should be configured as expanded memory with a memory manager program.

The available developer's software toolkit seems to indicate an open invitation to third-party vendors. To that end, third-party party software is already available for time code slaving and events list editing. Third Party MIDI interfacing software has been announced.

Circle (134) on Rapid Facts Card

SUNRIZE INDUSTRIES-STUDIO 16 AND AD1012

The Video Toaster, used with the Commodore Amiga computer, is revolutionizing video production. Now comes the AD1012 digital audio card with Studio 16 (Version 1.0) for the Amiga. Bundled with SunRize's Studio 16 editing software, the AD1012 allows Amiga owners hours of hard-disk audio recording. The AD1012 digital audio card mounts to an Amiga 2000 or 3000 and will record/playback with 12-bit resolution.

Included on the AD1012 is an Analog

Devices 2105 "sound accelerator" digital signal processor that allows real time digital effects such as flanging, chorusing and echoes. Studio 16 offers fade in/out, 4-track playback with a 4-channel mixer, LEDs and graphic meters. AD1012 owners can upgrade to SunRize's upcoming AD516 16-bit stereo card.

SunRize is working with Blue Ribbon Sound Works and other developers to provide direct software support for the AD1012. The "Bars & Pipes" program allows users to trigger the AD1012 as a MIDI channel.

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STEINBERG TOPAZ

For some time now, Steinberg has been developing the 4-channel Topaz based on Cubase audio sequencing software and an MOD. This system reportedly generates eight channels and currently provides digital mixing. Future plans include automation and EQ. Another software program, Time Bandit, can be imported for realtime compression/expansion and can construct a chord through recognition of the pitch of a cue.

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YAMAHA CBX-D5

Speaking of Steinberg, a Yamaha partnership with Steinberg and fellow software developer Mark of the Unicorn ("Digital Performer" for the Macintosh) has resulted in the release of the CBX-D5 4-track recording system. This unit offers 2-track simultaneous recording and 4-track CD-quality playback. Audio data such as vocals, instruments, etc., and MIDI data can be simultaneously computer-controlled. Computer interface occurs via SCSI connections. Hard disks can connect directly to the CBX-D5, reducing the processing power required from the host computer. The result allows the CBX-D5 to be paired with computers such as the Mac SE 30 and Atari ST.

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Other Options:

PUBLISON-INFERNAL WORKSTATION

The Publison-Infernal Workstation can reportedly be expanded from four to 16 channels via 4-channel modules. A proprietary hardware and user interface features graphic tablet and pen. Recording and editing take place on the same screen. With nearly a six-figure price, intrigue continues about the Publison, which may be attributed to its French mystique. While unable to discuss the machine with North American users (are there any?) our European agents indicate the device should be considered a strong work "In Progress."

Circle (138) on Rapid Facts Card

SINGULAR SOLUTIONS

A 2-channel recording editing system, using the NEXT computer platform, is being offered by California-based Singular Solutions with I/O help from Metaresearch. On-line endorsement of the NEXT Platform/Singular Solutions combo came from veteran (St.Louis-based) classical recording and broadcast producer Barry Hufker of Hufker Recording, who strongly indicated he "would not change systems" even if offered free alternatives.

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SIDETRACK SYSTEMS (ALPHA AUDIO DR-2)

The Alpha Audio DR-2 digital 2-track hard-disk recorder was designed to emulate a standard 2-track open-reel deck and more. While Alpha Audio no longer manufactures the DR-2, exisiting users (in some cases seven year veterans) continue to be serviced and receive software upgrades from Sidetrack Systems of Richmond, VA.

Although the DR-2 name has disappeared, Sidetrack now manufactures and distributes an upgraded system based on the original DR-2 design.

One major Sidetrack system improvement removes pop at edit (a problem with the original DR-2 system) via new edit-to-subframe capabilties. While it doesn't crossfade, provide EQ, sample-rate conversion or time/compression, the Sidetrack does resolve to house sync (24, 25, 30, or 30DF), samples at 44.056, 44.1 and 48kHz and allows users to slave up to eight SCSI memory devices.

The company is also working on a 3rd generation of The Boss editing system while supporting and offering upgrades for existing 8400 and The Boss 2 systems. Sidetrack systems cost approximately 50% less than original Alpha DR-2 systems due to new economies in hardware.

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Processing; Sony PCM601 conv, $850; Eventide H3000SE, $2.4K; **Mics**; Neumann U47 Nuvistor, $2.4K; U47FET, $1.4K: AKG C12A (tube), $1.4K; Pultec Mavec (tube EQ/pre), $1.4K.

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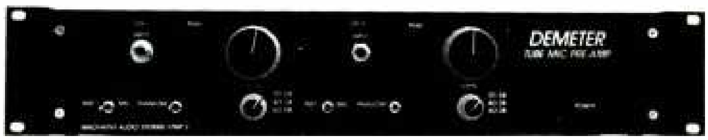

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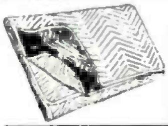

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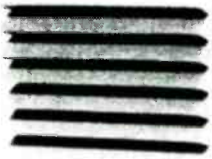

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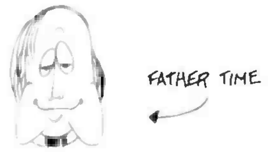

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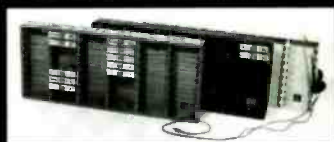

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