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January 1992

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On The Cover: Liquid TV logo courtesy of Keith Hatschek at Music Annex. Photo by XAOS, Inc.

R*E*P: Recording*Engineering*Production (ISSN 1058-9678) is published monthly by Intertec Publishing Corporation, 9221 Quivira, Overland Park, KS 66215. Subscriptions rates are \$26 to qualified readers, \$30 to non-qualified readers per year in the United States, \$50 for qualified and \$60 for non-qualified per year outside the United States. Optional airmail for non-qualified readers outside the United States is also available for an additional \$55 per year. Foreign subscriptions are payable in U.S. funds only by bank check or money order. Adjustments necessitated by subscription termination at single copy rate. POSTMASTER: Send address changes to R*E*P: Recording*Engineering*Production P.O. Box 12960, Overland Park, KS 66282. Second-class postage paid at Shawnee Mission, KS 66202 and additional mailing offices.

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ISSN 1058-9678 \$4.00 + \$0.00

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THE DTR-90011, MEANS BUSINESS, INSIDE AND OUT.



When a digital machine has earned the reputation of being "the best sounding tape recorder in the world," and at the same time delivers Otari's legendary dependability, it can only mean more satisfied clients, and more business for you. However, DTR-900 users tell us there are other reasons to own one that are just as compelling.

First, you're not locked into a single 48-track machine. Otari's 64track system can be split for use in two simultaneous 32-track digital sessions. (One practical studio owner says he "uses one and rents one" when he doesn't need all 64tracks!)

Another owner appreciates what he calls, "the best autolocator-it makes production easy by doing most of the work for you." (He also likes the "900's" unique ability for simultaneous recording on digital and analog inputs.) Still another says, "The DTR-90011's adjustable time advance feature is a lifesaver when we're using an outboard digital console."

Obviously, the DTR-900 means business for these owners. Call Otari at 415/341-5900 for more information about what it can mean to you.



Circie (4) on Rapid Facts Card



Spending years on end cooped up in small, dark rooms with a bunch of engineers takes certain special qualities. Durability, for one. We've always been known for that. Of course, clear, uncolored sound quality doesn't hurt, either. Or hand-assembled components, with gap precision to plus or minus one-millionth of an inch.

These features got TAD speakers into studios like Record Plant, NOMIS and Masterfonics. And the same features are now getting us out of them.

See, we had this funny idea that if TAD could make music sound terrific in a small room, we could make music sound terrific in a huge arena. And every outing we've had with Maryland Sound has proved us right.

Not that we won't still work our woofers off in studios from L.A. to London all day. But, at night, we'd like to get out and jam more often.



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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

Who Do You Trust?

Here we are, living in a world where everything and every product is the best, has the most features and does the greatest job. Everything is great and performs perfectly, no matter the price. Just read the ads. Look at the fluffy articles and so-called product reviews that portend to expose the truth. The real truth is that it's common today for most magazines to be less interested in presenting reality to their reading public than they are in providing soothing words to the folks who buy ad space. My message today? Caveat Emptor. These are not honest times.

Let's talk about product reviews. Most other pro-audio magazines present themselves as if they were dealing in unbiased fact, but the truth is, many times they stop short of presenting the serious technical information that the rest of us pro's need. I don't know why. Real pro's are not scared of calculators. We can read graphs and understand measurements. Why do other mags chop out the tech and substitute fluff? Do they think we're all beginners or students? Or are they writing to the level of some least common denominator, some imagined unknowledgable reader? If I had a dime for every writer who came to me with a horror story about all the "good stuff" that another magazine chopped out of his piece, well ...

It's curious, but in the consumer audio, automobile, motorcycle and computer industries, to name a few, magazine reviewers really do take hard, detailed looks at products and compare them in mano-y-mano shootouts to competitive units. On specific features and performance merit, some units win and some lose. Even the losing products' manufacturers realize that the review's readers trust the magazine's writers, and that all the manufacturers are playing on a level playing field with their competition in regards to review ethics. If a product isn't as good, for whatever reasons, as a competitive piece, then it will fail in the marketplace. In effect, the reviewer is doing the manufacturer a service in the long run - it is showing him where and how to improve the product, and indicating (at no charge) what level of quality people will accept in a given price range.

In many industries, editors get notes of thanks from a given reviewed product's manufacturer, thanking them for pointing out the strengths and weaknesses saying, "We're going to improve that aspect you pointed out and we'd be honored to have you review it again." Mags often do. Usually it leads to a noticable improvement in sales. The readers respect the manufacturer for its honesty, and sales improve based on the large amount that they now know about the product, which contributes to familiarity and acceptance.

But in our industry? Usually not! If certain manufacturers get a truthful review, negatives and all, they stamp their feet and make nasty phone calls. Then they pull their ads. For most magazines, the fear of ruffling some feathers and losing ads is enough to kill any thought of conducting serious technical reviews. The readers lose in the end.

To their credit, quite a few manufacturers understand and support the concept of honesty. They are the first to say things to us like: "Great product review! As you pointed out, our \$60,000 board (for example) does have 'X' crosstalk between these buses. To get another 6dB separation would have added two grand to the cost and required a whole different, more expensive design. It was a compromise we made to keep to the price point. And as it stands, it is still a wonderful value, with superior performance for the dollar. Remember, perfection costs money. We manufacture that too, but it's \$200,000." We applaud the integrity of companies like that. They do the industry justice.

In the past several months, certain manufacturers who have had their products honestly and objectively reviewed in our pages have reacted loudly and rashly. Our response? We were sincerely surprised. In our reviews, we discuss things that anyone having access to an Audio Precision stack or a Techron TEF system can measure themselves, in real-world environments. We try to verify our measurements with manufacturer's staff engineers before the articles ever hit. These designers often tell *us* about things to watch out for.

We at R•E•P try hard to be fair and honest in our evaluations, pointing out the abovestandard and the sub-standard performance attributes, keeping in mind the technical capabilities appropriate for a device's price range. We think we are fair, and in fact, probably bend over too far to be factual and objective: One recent review was pushed-back for nine consecutive months as we verified and re-verified the results with multiple qualified sources, including over a dozen calls to the factory. Yet, to some companies, merely pointing out blemishes, if they truly exist, is unacceptable. They are uncomfortable with anyone discussing the real, unpublished performance attributes of their product. Go figure.

So let me ask you: Does the pro-audio world need a magazine like R•E•P that seriously reviews audio products? That tells it like it is, technically speaking? That believes a working professional has the right to know the truth about the performance of equipment they might consider buying? We obviously think so. It's been a hot topic of conversation around R•E•P lately. You see, we like to discuss ethics and our responsibility to the industry.

But what do you think? Do you, the pro reader, want more tech and more truth, or more hype? Are you happy with only getting a marketing department's version of performance reality, the kind of thing another magazine might sneakily reprint as if they wrote it themselves? Or do you want to see more techintensive product reviews? Inquisitive minds want to know.

Editor

Hands On Response

From: Meyer Sound, Berkeley, CA.

We have read your review of our HD-1 Monitor ("Hands On," November 1991). There are significant discrepancies between your measurements and ours. We cannot address the reasons for this, since R*E*P did not describe or identify the testing facility, list the equipment configuration employed, name the consulting engineers involved, or make public any information that would permit an independent judgement regarding the resolution and accuracy of the measurements.

We will endeavor here to address some of the major points in your discussion of our product.

l.) Given appropriate phase correction, it is, indeed, physically possible for a 2-way loudspeaker to behave as a true point source within a specific coverage area. By definition, a point source radiates spherical sound waves uniformly such that, for each doubling of distance from the source, the sound pressure level drops by 6dB across the entire frequency band of operation. This is relatively simple to verify through measurement, as can be seen in Figure 2 of your review.

2.) Our frequency response specification is an accurate and honest representation of the HD-l's performance. Respected engineers in several different countries have succeeded in confirming it through careful application of appropriate acoustic instrumentation.

3.) The use of an anechoic chamber does not, in and of itself, assure adequate accuracy to produce valid measurements of the HD-l. Even perfectly anechoic rooms (<10% pressure reflection from walls, ceiling and floor) exhibit $\pm 1dB$ residual amplitude error, and most anechoic facilities do not meet this standard. When attempting to characterize a speaker whose amplitude response falls within a $\pm 1dB$ window, this magnitude of error is unacceptable.

4.) It is not physically possible for a loudspeaker of the HD-I's design to exhibit 340° coverage at 2kHz (as depicted in the directivity plot of Figure 6). Moreover, the polar plots of Figure 5 and 7 do not show anything close to this figure, at any frequency in either axis. These discrepancies should have been sufficient to cause the data in Figure 6 to be rejected.

5.) It is impossible to determine the validity and meaning of polar plots if the measurement distance and the decibel scale of the polar grid are not specified; the same is true of ETCs with no horizontal scale marks (Figure 9).

6.) The action of the HD-1's dynamic limiting circuitry, which functions on a cycle-bycycle basis and only at the point of overload, cannot be accurately characterized with sine wave sweeps (Figure 10). Sine waves have the smallest crest factor of any test signal except square waves, and do not indicate, in any way, the dynamic response of a system to musical material. 7.) To reproduce the remarks of a competing manufacturer within the context of a product review is, we feel, inappropriate. The comments by Mr. D'Arcy of Miller & Kreisel regarding the effect of the HD-I's baffle on its tweeter are unscientific, and are not supported by measurements. Such effects have been known for many years, having first been described by Olsen. They are correctable in the transfer function of the system, and are dealt with electronically in the HD-1.

8.) We feel Mr. Levitin's comments that the HD-l is not "musical sounding" or "fun to listen to" are so vague and subjective as to be meaningless. And just as "I Love Lucy" reruns look grainy on HDTV, poorly recorded material will sound bad on the HD-l. This hardly seems an indictment of our product, nor does it indicate that the speaker is suited for only one particular style of music.

Professional recording engineers and producers require highly accurate audio monitors if they are to produce product that will translate on a wide variety of consumer loudspeakers. That a majority of top-echelon audio professionals worldwide have selected the HD-I as their primary (or sole) monitor is, we believe, evidence of this fact. The quality of their work speaks for itself.

Meyer Sound takes great pride in its products, and goes to significant lengths to provide complete and accurate data on the HD-I. All of our published data are fully documented, with test conditions clearly and completely specified. Those who wish to obtain this documentation are welcome to contact us directly.

Mike Joseph replies:

It is unfortunate that so much was misinterpreted by Meyer Sound concerning the HD-1 product review. Although we dedicated an unusually large amount of space to this high quality, most innovative and groundbreaking monitor speaker, due to space limitations we were unable to share *all* of the information we acquired (over 100 separate measurements, excluding the 38 graphs and charts contributed by the Meyer engineering department).

We felt we hit the major high points. In fact, many technical aspects were briefly touched upon which clearly could bear further investigation: a horizontal polar response which is wider at 5kHz than 1.5kHz; the large mid-range disparity in response linearity on-axis vs. 10° above axis; the strong down-firing lobe at the crossover frequency; the fact that Meyer claims linearity in their measurements only referenced to 0.5 meter, yet virtually all real-world nearfield listening is done at .75 to 1.5 meters distance (their own measurements show increasingly larger response aberrations beyond 0.5 meters); that at no point, even on their submitted measurements, is a true +1dB response across the stipulated bandwidth indicated (we have never seen, anywhere, documentation showing same); that any device with a crossover and two separate, displaced, band-specific radiators cannot be a true single wide-band point-source across its entire spectrum, at all measuring distances; that the horizontal pattern generally (with some exceptions) collapses with rising frequency, which is normal, opposing their published claim of 60°, etc.

The point here is not that the HD-1 speaker is or isn't perfect, however you measure perfect. The point is that, at almost \$5,000 a pair, they should be a known and definable quantity, with their limitations fully understood, expressed, and defined by the manufacturer via open discussion (short of publishing proprietary or patented design information, of course). Most companies provide this as a service to their customers.

More importantly, the speakers must be worth their value as a listening and mixing tool. Now, we know today's speakers have limitations they're not electrostatically driven ionic plasma - they're electro-mechanical devices! The HD-1 is constructed, after all, of fairly traditional components - wood, paper, steel, copper, fabric, plastic - and as such, no matter the proprietary processing electronically added, suffers, as I said in the article, from traditional physical limitations. Pistons narrow their propagated pattern - it's a wavelength-related phenomenon. Drivers reach excursion limits. Signal compression and/or limiting affects response linearity. Two drivers with centers separated by eight or so inches do not a true pointsource make across an entire frequency spectrum. Baffle edges contribute to diffraction effects. All dynamic loudspeakers in boxes suffer these phenomenon.

Specifically responding to the addressable points mentioned in the letter:

A) Information relating to the measurement devices and techniques applied are available in the November, 1990 issue of R-E-P, in the "Hands-On: Radian MS-8 Loudspeaker" review. As was pointed out in November, additional data was collected from other highly qualified sources using TEF, MLSSA, UREI, B&K and HP measuring systems in various and assorted environments, all compared by us for accuracy. Only material which correlated and was corroborable was used, although correlation between the various measurement techniques and locations was very high. We stand behind the measurement information presented. If anything, we can be accused of being conservative.

B) The definition of a point-source in the letter's item #1 is simplistic at best, and doesn't address wavelength, measurement distance, measurement angle, etc. The common acoustic definition of a point-source is a hypothetical point in space, *smaller than the wavelength of the sound being radiated*, which propagates omnidirectionally, in 3D. Few, if any, speakers are true point sources.

Any single component from a given multiway speaker might be referred to as a point source if viewed at only one specific frequency and at a very specific coverage angle. But "The Beta 58 delivers maximum SPL, to keep the vocals above screaming fans in a loud rock club — without feedback. Yet it has the sensitivity to reproduce the most subtle, breathy whisper for 80,000 people at an outdoor festival. And for guitar amps, the Beta 57 gives me the isolation I need without sacrificing the warmth and tone I want. For live sound, Shure Beta mics are the state of the art."

Steve Folsom, Sound Engineer for Melissa Etheridge and John Hiatt.



Shure Beta Microphones. Buy Them On Word Of Mouth Alone.

Beta 57

Beta 58

Before you select a microphone, listen to the leading pros who use the Shure Beta Series on stage. They'll tell you about the benefits of Shure Beta's advanced transducer design, extraordinary gain-before-feedback, and true supercardioid polar pattern, as well as its outstanding sensitivity and low handling noise. But most important, they'll tell you that nothing beats a Beta for live performance. And that's not just talk. Try Shure Beta today and get the final word for yourself. Or call us for more information at 1-800-25-SHURE. The Sound Of The Professionals[®]...Worldwide.



Continued from page 6

what does that mean? And does it fit any accepted definition? I'd like to hear from the readers on this!

C) As can be seen in the accompanying vertical polar plot, the measurements alluded to in point #4 were normalized on-axis at what happens to be the somewhat troublesome crossover area, in this case 2234Hz. Referenced to the cancellation-caused null, the -6dB down points are indeed 170° off-axis in both directions, for 340° coverage. Neither Figures 5 nor 7 show this worst case, but the 1.9kHz trace in Figure 7 of the review comes close. Also, the directivity plot mentioned (Figure 6) was made with 400 point measurement resolution for much greater detail, rather than the more typical 1/3-octave band plots. (Note: the pattern shown is a good example of the HD-1 not acting like a point source.)

D) As is customary, the polar grid decibel scales are 6dB/div. in Figures number 5 and 7 in the article. Measurements were made at 3 meters and normalized to 1 meter when appropriate. The Figure 9 scale is 1,000 Sec, as indicated. Our honest apologies for not captioning the original graphs in a more detailed manner. We promise that this will be corrected in future reviews.

E) In reference to point *6 in the letter, we should mention that we highly qualified the validity of sine sweeps on dynamic limiting in the original article. We suggest you please re-read the paragraphs on page 62 beginning, "It is important to emphasize, despite the radical appearance of the traces in Figure 10, that swept sine waves are not music or voice"

In conclusion, we feel, as do the many who have responded to us after the publication of the review, that the Meyer HD-1 is an excellent monitor speaker with specific unique charac-



teristics. The measurements speak for themselves. The HD-Is readily lend themselves to a range of high-quality audio applications. Obviously, they aren't for everyone, or for every purpose. Few things are. We highly recommend that every reader listen to the speakers personally and draw their own conclusions.

Dan Levitin Replies:

I feel that equipment reviews all too often ignore that audio equipment needs to be used by people with ears. R•E•P incorporated my subjective feelings about the HD-1s in a sidebar, separate from the technical portion of the review, and this seems like an appropriate treatment. You state that my comments are "subjective" as though this is an indictment. The whole point of my piece was to offer a subjective view of what it is like to actually use and listen to the speakers, a point which I'm sorry you missed.

I interviewed two people whose musical judgement I hold in the highest esteem. Jeffrey Norman is a top recording engineer, and as such, I felt his opinions about what it's like to actually *listen* to the speakers would be of interest to R•E•P readers. John D'Arcy is not a Meyer competitor; he is currently an electrical engineer in an unrelated field. Previously, he was a key player in the design of one of the most influential loudspeakers of the last 20 years. The opinions of these two professional listeners were indeed "unscientific" and that is precisely why I wanted them. Listening is what recording is all about, and these are two of the best listeners I know.

No one mixes in a vacuum. Engineers and producers routinely listen to other projects as a reference while they are mixing. Many people have played recordings that sound terrific on all other monitors, but sound strange on HD-1s. This can be disorienting and scary. Readers should know that other people have had this experience. Nowhere in the sidebar, however, did anyone say the Meyers weren't appropriate for mixing, or that they weren't accurate, only that they weren't ideal for listening in some people's opinion. I know of no professional engineer - or musician, for that matter - who lives by specifications alone: the feel and sound of equipment are the ultimate criterion

Vertical polar plot of Meyer HD-1 showing – 6dB down points at approximately 170° off-axis, up and down. Note 0° reference point on null. Measurement taken on cabinet axis, 3 meters, at 2234Hz, in a full anechoic environment using a B&K 4133 mic fed into a Techron TEF system, measured at 10° increments. Additional equipment included a B&K 2610 measuring amp, a UREI Model 200 chart recorder and a Hewlett-Packard 3325A function generator. Decibel plot scale is 6dB/division.

Send letters to R•E•P, Box 12901, Overland Park, KS 66282; or fax 913-541-6697. Letters must be signed and may be edited for length and clarity.

For The Artist In Every Engineer.

C

In recent years, recording methods have changed drastically. In today's studio, you're the artist, the engineer and producer. Designing today's professional recording or production studio presents unique challenges, like finding a flexible full featured console that fits your space, budget and audio requirements.

Soundcraft's Sapphyre lets you create both music and a recording environment the way you want it. Sapphyre gives you 32 track performance normally found in consoles many times the price. A noise gate on every channel. Great sounding four band EQ. A variety of modules to suit the way you record. And five frame sizes.

Sapphyre. Its unique, flexible routing and design approach puts everything you need where you need it. Direct to tape routing with grouping to any channel provides total creative flexibility. The in-line design takes up less space than conventional consoles and is as intuitive to use as any split console. Whether you're building a home studio, adding a video post suite or designing Studio Two, Sapphyre will bring out the artist in you. Now you can afford the best. Hear Sapphyre today at your authorized Soundcraft dealer. THE REAL PROPERTY OF

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Random Access

Perceived Value

Some of the below from Peter Watson, New York Times.

- **Item:** Eddie Murphy forks out \$26K+ for vest and black suede headband worn once on stage by Jimi Hendrix.
- **Item:** Collector pays \$14,780 for handmade Christmas card from John Lennon to his first wife, Cynthia.



Item: Madonna's soiled baseball uniform from unreleased film ''A League of Their Own'' nets \$2,100 at auction.

Item: "Reserved" table card with message "For guests of the Bruce Springsteen band" is purchased for \$990.

Message: Check your dumpster and keep that dirty hanky from the last session!

nt Record Label?

Roll Your Own Label?

Vanity recording was recently summarized by the *Wall Street Journal* this way: "The economics are steadily improving. Today, musicians say, it requires only a few thousand dollars of digital equipment to capture sound quality that could once be achieved only at major studios. In the last five years, the production costs of a typical quality CD has dropped as much as 50%. At the same time, cheap desktop computers are assisting musicians to write music, design album covers, publish fan newsletters, track cash flow, and maintain mailing lists.



As evidence of this trend, the *WSJ* cites bands that include former Electra records trio Shoes, which last year approached a \$250K gross on its own Black Vinyl Records. On the other, probably more common end, the *Journal* referenced industrial artist Claude Wiley, who had to delay upcoming projects while waiting on a \$1,500 outstanding debt from a record distributor. Clearly a cash flow susceptible proposition.

"Desktop recording labels" might seem in opposition to the mega-corporate recording industry. Certainly, the big studio project studio debate is tied to this trend. But positive long term effects are likely for all parties. A good way to think about desktop record production is that "homegrown labels" will assume a minor league posture within the music business. The quantity of non-commercial or fringe music gaining market access (however small) is absolutely beneficial. And the same engineers who mix and master successful desktop projects may soon be bringing honed chops to big rooms.

PEOPLE

In order to pursue their music industry careers unemcumbered by studio ownership, Britt Bacon and John Eden recently closed their Topanga Skyline Recording Company ... Carl Reavey has assumed the position of general manager for all Amek and TAC western hemisphere operations, while Lewis Frisch was been appointed regional Amek and TAC sales manager for U.S. operations ... Otari Corporation has promoted Emil Handke to national sales manager. Lee Pomerantz is Otari's new export sales manager. Roberta McKeehan has been named industrial product sales coordinator at Otari ... Karen Spriggs has joined Sony Broadcast and Communications as public relations officer ... Dr. Mike Smyth has been named to head the startup of Audio Processing Technology's Los Angeles sales office ... Neotek has announced that Julie Bacher has been promoted to business manager and Tom Lay has been named sales and marketing group leader ... Dave Collie has been assigned new manager of SSL's western operations ... Electro Voice has appointed Neil Andison general manager of Mark VI Audio Canada ... Hudson Fair of Ealing Mobile Sound was elected chairperson of EARS (Engineering and Recording Society) in Chicago for 1992.

Root

"The generalist knows less and less about more and more so that eventually he knows nothing about everything. The specialist knows more and more about less and less so that eventually he knows everything about nothing."

 Fred Harbert, uncle of unnamed East Coast record producer.

trend watch

JUST WHAT THEY NEED: The post-Marxist avalanche of decadence continues as a new "independent sound equipment and musical monthly" named *IN/OUT* is scheduled for publication next month in Russia. Editor Michael Subbotckin, said, "*IN/OUT* is dedicated to reporting and analyzing the sound equipment industry and market as they face a new era full of uncertainty and opportunity."

VAPORWARE: Tandy Corporation's long awaited compact disc player/recorder seems to have a blue smoke element. Since early 1988, Tandy has struggled to deliver its THOR (Tandy High-intensity Optical Recorder) which promised CD read/write capabilities for less than \$500.

It didn't happen for at least two reasons: First, Tandy insists the ongoing copy protection battle intervened. Secondly, and realistically a bigger obstacle, THOR could write-erase-read only 10,000 times before disc crash. That limitation effectively removed THOR from any computer storage applications.

Meanwhile, Sony has announced development of a \$500 magneto-optical technology which will tolerate millions of write-erase-read sequences.

INVITATION TO A SIDESHOW: As Dire Straits began its fall (1991) tour, concert site demonstrations were planned for Philips' new digital compact cassettes (DCC). By sponsoring the 300-city tour, Philips apparently hopes to jumpstart the new format for the inevitable marketplace confrontation with Sony's DAT technology. Let the battle begin.

ANOTHER ONE BITES THE DUST: It is with sadness that we note the passing of Freddie Mercury. Born in Zanzibar as Frederick Bulsara, Mercury died Nov. 24, 1991 in London from AIDS complications. Flamboyant and controversial, the longtime Queen lead singer maximized his 45 years with unrivaled bravado.

DID YOU KNOW: The United States Copyright Law (Title USC 101) identifies a "public performance" as one that occurs "at a place open to the public or at any place where a substantial number of persons outside of a normal circle of family and its social acquaintances is gathered." As such, the act of broadcasting, cablecasting, communicating a performance to the public , or even playing music-on-hold on a phone system is considered "performance" under copyright law.

Random Access

STUDI	O UPDATE	NEWS NOTES					
Facility/Location	Details	President George Bush toured Peavey E					
NORTHEAST		tronics and addressed nearly 2,000 Peavey					
Hit Factory/New York	Studio A2 received retrofit of SSL's Ultimation console automation system to its 64-input SL 4000 G series desk.	ployees on a December 3, 1991 visit to Me an, MS. While acknowledging the prolon downturn in global economy, President E					
Acme Recording Studios/ Mamaroneck, NY	Has upgraded its facilities with a second Otari MTR-90, MKII 24-track machine and Lexicon 480L digital effects processor.	pointed to Peavey as a prime example of An can industry's potential for international gro through an aggressive export posture.					
G <mark>olden Studios</mark> / Hancock, NH	New studio, designed by chief engineer David Torrey, features the Spectral Synthesis Digital Studio.	As a memento of the trip, Hartley and M Peavey gave the president one of their new of electro-acoustic guitars. The customized white, and blue axe was appropriately nar					
OUTHEAST		"The Chief." [See Feb. 1992 Random Access					
U <mark>ltrasonic Studios/New Orleans</mark>	Expanded to 48 tracks with its recent addi- tion of a new Studer A827 24-track recorder with 24 channels of Dolby SR.	further coverage of the Bush visit.] The Society of Professional Audio Reco					
The Music Mill/Nashville	Installed the Focusrite recording console with GML automation into Studio A.	ing Services (SPARS) board of directors de ed to retire the SPARS code at a board m					
New River Studios/Fort Lauderdale	Recently retrofitted flying fader automation to their Neve 8108 console and added a Mit- subishi X850 32-track digital tape machine with Apogee filters.	ing held during the 1991 AES convention New York. The SPARS code, introduced in mid-80s, was used to identify which porti of the recording process were digital and wh were analog.					
Omega Studios/ Rockville, MD	A Mitsubishi X-850 32-track digital recorder has been added to the Studio A SSL system.	Ampex Corporation and Sprague Mag					
Windmark Recording Studios/ Virginia Beach	New digital mastering facilities anchored by a Sony SDP 1000 and a Studer Dyaxis.	ics have reached an agreement that will all Sprague to provide ongoing worldwide s					
MIDWEST		ice and support for Ampex audio record					
Triad Productions Des Moines, IA	Completed main room installation of the new Euphonix CSII digitally controlled analog stu- dio system.	Robert G. Shaw CEO of International Jen Incorporated , recently announced the for tion of the corporate technology division					
Zeta Recording Studios/Toledo, OH	Installed a 32-input Hill Concept 8400 console.	group dedicated to advanced research in dio and sound reproduction.					
OUTHERN CALIFORNIA		TimeLine and Digidesign have formed ar					
University of California at San Diego/Burbank	Recently purchased an API 48×24 Discrete series "Touch Reset" console.	liance based on TimeLine's new video cl card, known as the Pro Tools Interface.					
Kingsound Studios/North Hollywood	Installed a Neve V3-48 console with 60 chan- nels of flying fader automation, 12 custom in- puts with choice of API 550A, 550B and Mas- senburg 8200 EQ. A Studer A827 24-track	card allows Digidesign's Mac-based pro system full multimachine synchroniza					
	and an Ampex ATR 102 2-track were also added.	Focusrite Audio Engineering, Bourne, gland, has signed a distribution pact of George Massenburg Labs of Los Angeles,					
MUSIC Annex Post Production/	Has added a second New England Digital	GML to serve as sole North American dist utor of Focusrite studio consoles.					
San Francisco	Post Pro editing system.	API Audio Products and SONTEC have					
JNITED KINGDOM	Her ordered on AMS VCS 40 (s low evil and	tered into a licensing agreement to begin n keting the new API 554B parametric equali					
BBC Radio/London	Has ordered an AMS VCS, 48-fader assignable console for the Maida Vale studios.	Rivera Research and Development offi					
CTS Studios/London	Has invested in a 60-channel Neve VRP con- sole with recall and flying fader.	ly joins the JBL Professional family during winter 1992 NAMM conference.					
DESIGNERS							
Walters-Storyk Design Group/ New York	Currently working on two new confer- ence/listening rooms and a suite of offices for Mercury Records Worldwide Plaza loca- tion in Manhattan.	ADDRESS CHANGES					
Harris Grant Associates/United Kingdom	Has been appointed by Sony Classical Productions to construct two new mix- ing/editing suites in its Manhattan engineer- ing facility.	Beyerdynamic has moved to 56 Central A Farmingdale, NY 11735; 516-293-3200; fax 293-3288.					
Russ Berger/Dallas	Completed design of a personal recording studio addition for the residence of produc- er/engineer Mike Galesi in Bernardsville, NJ.	The new corporate headquarters for W burn International is 255 Corporate W Parkway, Vernon Hills, IL 60061; 708-913-					

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The Benny Green Trio: "Greens"

Label: Blue Note Produced by: Matt Pierson Engineered by: Jim Anderson Recorded at: Power Station (New York) Mastered by: Ron McMaster at Capitol SPARS Code: AD



Comments: Green is one of the most exciting new piano players we've heard in years. This album of straight-ahead jazz is a model of jazz trio recording at its best.

Of special interest: The instruments are beautfully recorded — very clear and bright. The ambiences are very natural. All of the instruments are given their own position in the soundscape. The piano is remarkably even throughout its tonal range, and the drums present and wellblended. Recorded direct to 2-track, 15ips, ¹/4-inch analog using Dolby SR. To these ears, the fidelity is indistinguishable from the best digital. Definitely worth a listen.■

Nancy Argenta with the Chandos Baroque Players: "Alessandro Scarlatti Cantatas"

Label: EMI Classics Produced by: Nicholas Parker Engineered by: Tim Handley Editor: Adrian Hunter Recorded at: Concert Hall (University of Cardiff) SPARS Code: DDD



Comments: Argenta possesses a remarkable voice, not just for her technical facility, but for her ability to blend so well with the chamber orchestra. The timbre of her voice weaves in and out of the instruments in just the way the composer must have intended. Sonically, the most amazing thing about this recording are the ambiences from the conert hall. Warm and full, it sounds as if only ambient mics were used, without even a spot mic on the vocals (or one which was barely used). The recording realistically conveys the sense of being in the audience during the performance.

REISSUES AND COMPILATIONS

Stan Getz: "The Best of the Verve Years, Vol. 1." We've always felt Getz's best recordings were done during this period and they are assembled here, including his sessions with Bill Evans, Jobim, Dizzy, Chick Corea and Charlie Byrd. Getz has never sounded better than on the tracks compiled and collected herein. The tapes sound as if they were recorded just last week. Engineer Phil Schaap works restoration magic on tracks dating back to 1952.

Les Paul: "The Legend and The Legacy." (Capitol) If there is one single individual all of us owe our careers to, it is of course, Les Paul, inventor of multi-track recording, flanging, variable recording speed effects and a host of other studio mainstay techniques. This beautiful 4-CD box is a must-have for everyone in the recording business, not just for the historical value and inspiration it provides, but for the four hours of great and innovative music. Produced by Ron Furmanek, remastered by Bob Norberg and Rus Paul.

. . .

Howlin' Wolf: "The Chess Box." Another superb boxed set from MCA reissue producer/wizard Andy McKaie, this 3-CD set captures the best of one of the great blues artists of all time. The packaging and sonics are first rate, and the liner notes informative.

Dan Levitin is R+E+P's music production editor and a producer based in Stanford, CA. We asked Phil Ramone to field test our new AT4033 studio condenser microphone.

He wouldn't give it back!

> Phil Ramone knows exactly what he wants from a studio microphone. And when he tested a sample of our new AT4033 cardioid condenser microphone, he knew it was right for him and ideal for the artists he records.

He liked being able to concentrate on getting the right music from the musicians, rather than first spending time experimenting with EQ to get the right sound.

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> The high head-room and wide dynamic range, plus low noise floor, make the AT4033 ideal for the most demanding digital recordings. And the maximum input SPL is an awesome 140 dB, so important when recording high-output instruments and very close-up vocals. In addition, transformerless design contributes to overall sonic transparency. The AT4033 also includes a switchable 10 dB pad and lo-cut filter, plus a built-in pop filter and internal shock mounting.

> We're not certain we'll ever get the sample AT4033 back from Phil Ramone, but no matter. We're busy making your AT4033 right now. For more details on this impressive new microphone, ask your A-T sound specialist to schedule a test of the AT4033 today.



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Phil Ramone photos by Michael Bloom

AT4033 Studio Condenser Microphone



The Modern Mandolin Quartet: Tchaikovsky, "The Nutcracker Suite"

Label: Windham Hill Produced by: Mike Marshall Engineered by: Oliver DiCicco Recorded at: Mobius Music (San Francisco) Mastered by: Bernie Grundman SPARS Code: AD



Comments: The ambitiousness of transcribing "The Nutcracker Suite" for four mandolins is mind-boggling and the result is truly brilliant. The transcription (by MMQ members Paul Binkley and John Imholz) not only works, it excels, adding unanticipated newness to the music.

Of special interest: Oliver DiCicco's recording is flawless. His use of stereo imaging and his sense of tone at once blend the instruments just the right amount, while allowing them to maintain the essence of their distinctiveness. And DiCicco is flexible — his past projects include engineering for The Dead Kennedys as well as Michael Hedges' "Aerial Boundaries" (for which he received a Grammy nomination for best engineered recording). ■

FOCUS:

OLIVER DICICCO, Engineer, "The Nutcracker Suite"

R*E*P: How did you mic the musicians?

OD: Each instrument was recorded with a stereo mic pair, so there were eight mics close-miking the band, and a pair of B&K 4006s that were used for ambient mics. Mike Marshall's mandolin was miked with a pair of Neumann KM54 tube mics; the second mandolin and mandola with KM84s modified by Klaus Heine. Mike has a couple of Klaus Heine KM84s too, and the mandocello was recorded with one of his KM84s on the neck and a tube U47 on the body. We recorded direct to stereo at 15ips using Dolby SR on a ¹/4- inch Studer 820, with Agfa 468.

The record was done without any EQ at all, and we went for tone by placement of the mics. We went through the mic pre's on my Neve 8068, and we went for the minimal signal path that was available with this particular console and configuration. I put the mics in approximate positions, and then Mike went out into the studio with headphones and he'd move the mics to the point where he wanted them to be. He likes a certain brightness and a certain presence to the instruments, but he wants them to be blended as well.

R-E-P: How did you keep such good separation between the instruments? **OD**: The musicians stood in a semi-circle, roughly 15 feet in diameter. They were in fairly close, to allow them to interact with each other the way they normally do on stage.

R-E-P: What about moving around? There's never a time on the recording where it sounds like their tonality changed because of moving away from or toward the mics.

OD: They're pretty disciplined as a group and they all have recording experience — they're aware of mic placement.

R•**E**•**P**: What did you use for reverb?

OD: We used an early Lexicon 224 with some of the original programs, and an Ultra-Harmonizer H3000SE on a modification of the concert hall program. We changed the parameters to match the physical dimensions of the room in terms of pre-delay, to get the reverb to sound fairly natural. The other thing is, we used the room mics as the primary reverb sends; the idea was to try to create as natural a sound as possible but still maintain the presence that Mike wanted, which is kind of mutually exclusive.

R•**E**•**P**: With all those mics in such close proximity, weren't you worried about phase problems?

OD: You do have phase information going on all the time. The way I look at phase is that if it sounds good and it still sounds good in mono, then the phase information is part of the sound. If you have more than one mic on an instrument you're gong to have phase information present, but that's what gives the instruments a sense of space, as opposed to having a single point of sound emanating from a speaker. You create a dimension for that particular instrument — you give it some size.

Basically, we're dealing with an illusion, but it's trying to create a good illusion of space while still maintaining a sense of definition and presence. The [phase] leakage works to your advantage — it adds more dimension to the sound.

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Keith Whitley: "Kentucky Bluebird"

Label: RČA Production director: Garth Fundis Produced by: Garth Fundis, Blake Mevis, Keith Whitley, Kix Brooks, Fred Koller, Don Cook

Engineered by: Bill Harris, Gary Laney, Garth Fundis

Mixed by: Garth Fundis

Recorded at: Music City Music Hall, Woodland Sound Studio, Sound Emporium (Nashville) Mastered by: Denny Purcell at Georgetown Masters

SPARS Code: N/A



Comments: When Whitley died a couple of years ago, he left behind tracks "in the can" in various stages of completion. Hardly a "dregs" album of previously forgotten tracks, "Kentucky Bluebird" may contain Whitley's finest material ever. Fundis tastefully rerecorded some of the backing tracks to accompany Whitley's vocals. The result is a cohesive and poignant retrospective of this highly talented country vocalist.

With the recent death of such greats as Whitley, the Vaughan Brothers and Miles Davis, it gives one cause to stop and reflect about the importance of the recording engineer's job, capturing a moment in musical time to allow it to live on for posterity, and in some cases, documenting the great musical artists of our day.

FOCUS:

GARTH FUNDIS, Production Director, "Kentucky Bluebird"

R•**E**•**P**: How much of the original tracks for these songs did you keep and how much did you scrap?

GF: "Going Home" for example was a complete restoration, where I only kept the vocals. I took the 24-track analog masters and transferred them to 32-track Mitsubishi 850, and then basically we put down the new click track, with Eddie [Bayers, drummer] playing hi-hat to the old track. That was the first pass. Then for the second pass, everyone played at once. I didn't do this pieceby-piece — I tried to do this the way we would have done it normally, when Keith was there. That is, to have everyone there playing on the session live, all the musicians in the studio at the same time. When solos come up, 'go for it', you know, don't lay out and overdub it later. In some cases we did go back and work on solos, but we were always trying to go for it.

The other tracks that were complete restorations were "Somebody's Doin' Me Right," "Brotherly Love," "Kentucky Bluebird" and "I Want My Rib Back." The latter two were originally demos from Tree Publishing that Keith had sung early in his career. The other songs had originally been produced by Blake Mevis, who had produced two of Keith's earlier albums, but those songs had gotten shelved because Keith and the label weren't happy with them the way they were.

R-E-P: You take production and co-engineering on most of your projects. How do you decide when to engineer and when not to?

GF: During live sessions there are so many things to take care of as far as communication with the artist and the players, that I really don't feel like it's fair for me to try to do all of it and make everyone wait while I'm working on a drum sound or trying to change a mic or something. I work closely with my engineer Gary Laney, but I don't actually sit behind the console during tracking. But I do all my own mixing.

R-E-P: What reverbs did you use on "Kentucky Bluebird"?

GF: For vocals the EMT 250, I guess set at 2.2 sec. Sometimes I put a tight pre-delay on the voice, of about 60ms I use the AMS reverb on the ambience program for some of the drums, a couple of PCM70s, Rev 7. I used an old tube Fairchild compressor on the vocals — I really love the sound of those things, they just give it a 'sound.'

R-E-P: You mixed on a Neve?

GF: Yeah, I do all my mixing on the Neve 8128 in Studio A at the Sound Emporium. For years they had a Harrison console which I refused to use — it was awful. I hate VCAs, which is why I haven't used SSLs much and when I do, I try to bypass the VCAs. People have tried to persuade me over the years that the new VCAs are improved, (that) you can't tell they're VCAs anymore. That's fine for them, but I prefer the Neve sound. I've used Sontec mic-pre's from time to time, but I have no complaints at all about the Neve pre-amps.

[Fundis has also produced recent albums by Trisha Yearwood and Don Williams.]

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it's a pleasure to use. Because everyone likes working with it. "But, here's the news.

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mere \$5,500 MSRP. So obviously, they're for those situations where you want the best console available. But you don't have the space or the budget to get the 3000. "The PM1800A was just updated. So it has an improved signal-to-noise ratio (6 dB better). And 0dB insert points for easy

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has the same mute grouping feature you find on the 3000. But that's not the end of it.

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"Obviously, they're both ripoffs of the Yamaha PM3000." YAMAHA

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Be There Now

By Tom Scott and Tom Kobayashi

here's a technological revolution taking place that has been creeping up so slowly, you might not have noticed it. But, in the remaining years of this millenium you won't be able to avoid it if you're in the entertainment business. What we're talking about is the inexorable movement to connect all of our facilities together into a worldwide shared workspace, that will eventually transform the business of recording and post-production.

External conditions are conspiring against the status quo. As you're surely aware, schedules and budgets are getting tighter and tighter. Any way to cut expenses is fair game. Travel is increasingly expensive, across town or across country. Cities have become gridlocked to the point where it may take two hours to deliver a tape across town. Between cities there are fewer flights and they're more crowded. The price paid for a cross country flight also includes an unhealthy dose of stress and airline food. As city life becomes more difficult, the high-priced performing talent prefers to live farther away, sometimes way out in the country. Producers have to pay more to bring them in for a session. All of these conditions argue for a better workstyle, one that progresses beyond the traditional studio work scheme - the one where everyone important to the process must be scheduled together in one room to do the work, whether it's for a meeting, a mix, an overdub, a voiceover, a mastering session or a playback.

While we've been grumbling over the price of gas and the traffic on the freeway, technology has been getting ready for easier collaboration over long distances. Remember when, not too long ago, if you wanted to send a schematic, a score, or a budget to a colleague, you photocopied it and put it in an envelope with a 13-cent stamp? In a couple of days it was probably delivered. Last year you faxed it for telephone comments. This year you are just as likely to E-mail a file and have the corrected score modemed back. Next year it's probable that your PC will call your colleague's Mac and you'll both work on the document simultaneously. Remote collaboration is the key concept here, and there are signs everywhere that it is a revolution whose time has arrived.

Here's an interesting experiment you might not have heard about: In Northern California, our local SMPTE chapter has members spread out over a 500-mile area from Eureka in the north and Fresno in the south to Reno in the east. That's too far to expect more than local folks to attend monthly meetings. So, throughout 1991, the enterprising program committee used Hi-8 video cameras and a news-gathering truck to broadcast the 2-hour meetings via satellite. Members in Sacramento or Ukiah get together at the homes of local members who own an ordinary satellite dish, participate in the meeting over beer and pizza, and call in questions on an 800 number to the meeting chair. Of course, you could tape the meetings and send out VHS copies, but the simultaneous remote collaboration is the key to a real meeting. Otherwise it's just a taped lecture.

Another related tip-of-the-iceberg development: Announcers and other voice talent have begun using satellite links and telephone program lines to record their lines for advertising agencies and television stations. For example, rather than paying to fly a famous announcer from Los Angeles to New York for a few hours work on a car comercial that is due in 24 hours. a Manhattan ad agency books time at a nearby studio, which in turn schedules satellite time and books a second studio in L.A. The talent arrives at the L.A. studio while the producer or director goes to the N.Y. studio. The latest script is faxed to L.A. minutes before the session begins. The studios patch from L.A. to N.Y. with a high-quality, one-way satellite audio link for the recording, and the director makes comments via an ordinary long distance call over a speaker-phone. An hour later, the ad is complete and the talent is off to another session. The producer pays for two studios and some satellite time, but is overjoyed to have beaten the client's deadline and incurred no travel expenses or per diem.

It's a simple step from that voiceover scenario to music overdubs. Imagine that you're a record producer (or maybe you are one). For about the cost of a one-way flight to Muscle Shoals, you can stay in your own town and overdub those horns remotely. You save on hotel, rental car, per diem and stress. You won't miss that emergency meeting for your next job, or worse, have to tell your kids for the second year in a row that you can't make the Cub Scout banquet or their birthday party. The musicians will have similar savings in time, stress, and family angst. The studios are happy with the collaboration because it means more work, and in some cases, work that would never have occurred without remote cooperation.

Long distance collaboration is already a proven option. At Skywalker Sound, we have been doing remote mixing, playback and ADR sessions via digital telephone tielines connecting our Northern and Southern California facilities. For more details, refer to Rick Schwartz's Digital Domain article, "The Audio WAN" (R•E•P March 1991). In that article, Schwartz may have coined a new buzz-word for the '90s. In computer parlance, a WAN is a Wide Area Network. What seems to be building here is a movement toward the Worldwide Entertainment Business Wide Area Network. Say it all together now: "The WE-B-WAN." That's it! We Be One! Wow, cosmic! Seriously, though, there are powerful foces at work here. The telecommunications and computer giants are working toward providing the interconnection tools for us, but we must be smart enough to use them to the best advantage. The price of high bandwidth communications is coming down. Low bit-rate coding methodologies are evolving that will deliver the high quality demanded by the recording industry over inexpensive telephone lines. The proliferation of optical fiber networks will provide terrestrial digital paths that can carry not just audio, but video teleconferencing that the average studio owner can finally afford.

The economy and the march of urbanization have us squeezed between the hard place of budgetary cutbacks and the rock of traffic gridlock. We want to be able to have more family time and less travel time. Can this be the advent of the entertainment business version of telecommuting? Perhaps, instead of building another 24-track mix room, your next studio expansion should be a transcontinental overdub room or a teleconferencing setup.

Beyond the personal life and company business benefits, there are world economic implications here as well. Communicating in place of commuting will save resources and cause less pollution. There might even be tax savings in some situations. If you take a tape overseas and overdub on it, you may have to pay a Value Added Tax. If you record remotely by satellite or telephone, the data contained in the call is not taxed (yet).

The most successful companies today are those that are thinking years ahead and forming alliances. They see the futility of short term defensive isolation and the benefits of a long term, global strategy. The current world economy rewards companies that have stretched out from their local bases to participate internationally. The American Entertainment Business is one of our most successful and far-reaching international exports. The power of these interconnections may help us maintain that preeminence while allowing us to participate in global entertainment markets that are far beyond our present reach.

Reflecting the SPARS goal of industry networking, we've tried in this article to point out a few of the early harbingers that point the way to our future as surely as the first few swallows into Capistrano mean "they're ba-ack." The coming years will see these small steps dwarfed by a giant networking growth allowing collaborations that would have been unthinkable a decade ago. We will enjoy improvements in the quality of our lives, while saving resources, time and money. The wise studio owner, engineer, and performer will investigate these new techniques and start thinking of ways to put the technology to work.

Tom Scott is director of engineering and Tom Kobayashi is vice president/general manager at Skywalker Sound, CA.

The Society of Professional Audio Recording Services is the industry's best source of business information. For details on membership or activities, contact SPARS at 4300 10th Ave. N., Suite 2, Lake Worth, FL 33461; 407-641-6648; fax 407-642-8263.

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Digital Domain

Digital Speak: Part II

Picture yourself as an explorer in an ancient civilization. Before you could find work, you would first have to learn the language of the local tribe. After travelling to many villages, you would eventually find that everyone used similar tools - although they looked slightly different and were called different names. Little has changed in the past 200 or so years. Last fall, this column dared to ask the question - what makes our high-tech audio tools so hard to use? We're talking about digital audio work stations here. Wouldn't it be great if someone was foolish enough to go around to each tribe - I mean company - and translate their terminology into plain English. (Although someone was rumored to at tempt such a noble task some time ago, I heard they lost all of the information when their hard disk crashed.) We attempt to do just that, backing up our disk daily.

Rick Schwartz is a contributing editor to R+E-P and director of post-production at Music Animals. Los Angeles.

THE DAW DIALECT

We've all heard someone ask "what is 'such and such' called our your system." This article is intended to bridge-the-gap between different products. It is not intended to be a shootout or feature comparison. Think of it as a digital audio workstation glossary or language translator. Imagine trying to learn a foreign language such as Spanish without first knowing English. It's much easier to understand something if you have a familiar point of reference. With the premise: once you've learned one system, you've learned them all, let us continue on with our journey.

THE CATTLE CALL

Workstation manufacturers were faxed a brief questionnaire and encouraged to fill in as many blanks as possible. More than 20 manufacturers were contacted, including Akai, AMS, Digidesign, Doremi Labs, Hybrid Arts, Lexicon, Microtech, New England Digital, Otari, Roland. Sonic Solutions, SSL. Spectral Synthesis, Steinberg Jones, Studer EdiTech. Symetrix, Turtle Beach Systems and WaveFrame. (Unfortunately not everyone responded by press time, which is why some didn't make the chart.) Although we tried to include as many major, shipping systems as possible, we are aware that there are over 50 workstations rumored to be on the market. Needless to say, it didn't take long before our fax machine was ringing off the hook with responses.

Data was grouped into four main areas of interest: system conventions, general definitions, play commands and editing commands. The top section on the accompanying chart describes system conventions. Most programmers use a tape-based model with on-screen animation to simulate tape movement. Some systems move the tape, others move a simulated tape head. Not everyone rolls tape from left to right, depending on your perspective to the heads and the intended application.

PLAY TIME

Things are not as straightforward as they were in the days of the mechanical tape transport. In the good old days, when you pressed play, the machine started playing from the point the tape had last been stopped. These days it's not uncommon for a workstation to have up to 10 play commands. Because of their randomaccess nature, a hard disk can stop-on-a-dime. A workstation can play right up to an edit point and stop with single-sample accuracy. This feature is great for checking edits. Even if you catch a few milliseconds of the next beat. you

Description	Akal DD-1000	AMS Audiofile	Digidesign Pro Tools	Digidesign Sound Tools	Doremi Labs Dawn
Indicates movement on the screen		circuit and	equater	0	time Ine
	play cursor	play head	counter	Cursor	
Direction the screen moves The recording media is called	left to right MO drive	left to right hard disk	left to right disk	left to right disk	top to bottom rec. media
eneral Definitions	NIC GIVE	noro orsk	CISK	UISK	IC. MOUN
Reference marker	GPM	mark	marker	marker	marker
A recording is called a	take	cue	sound file	sound file	sound file
Bundled audio data with edits	take	flies	session file	sound file	edit list
A selected part of a recording	cut	segment	region	region	event
Play a specified region of a sound	play cut	play	play	play	play region
An item in a playlist	cue	event	region	region	cue or event
An EDL is called a	Qlist	eventilist	playlist	playlist	event list
A graphic editing window	Q list edit	trim window	ProEDIT	editing window	mix window
Multiple versions of the same recording	renamed takes	takes	takes	copies	takes
Internal Storage Buffer	dipboard	cue library	region list	regions	clipboard
ay Commands	appead	coe acrosy	10 grout has	Togloria	ciipocora
Play sound before selected area	move GPM before	play up to	option left arrow	left arrow	play selection
Play sound between selected area	play cut	play	play	speaker Icon	play selection
Play sound after selected area	move gpm after	play from	option right arrow	right arrow	play selection
Play sound across an edit	n/a	play across	option play	opt. cmd. speaker	play selection
liting Commands					
Remove sound & shift all other sound to the left	define & assemble In Q List	Insert segment	shuffle	cut region	bite & splice
Remove sound w/o changing sync (destructive)	cut & retake	overlay segment	delete (sip mode)	silence	clear
Remove sound & shift all to the left (destructive)	copy & select new cut	replace segment	n/a	n/a	delete
Remove part of a sound file & leave the rest	copy + T	spot segment	use sip mode	remove region	bite
Insert a sound at a start marker	copy + T	spot segment	paste	paste	Insert
Insert a sound at an end marker	select new sound	spot segment	paste	paste	Insert
Insert a sound immediately after another	Insert in song mode	spot segment	drop region	paste	splice
Insert a sound, substituting one sound for another	drag & edit TC number	replace segment	paste	replace	place
Insert a sound and move all others to the right	block slip	n/a	shuffle	Insert region	Insert
Insert a sound at a specific time code location	glip	sync segment	use spot window	insert & set time	capture
Move a group of sounds in a playlist	change number while offline	block offset	shift select & move	shift select & move	offset selection
Move an element by one frame	zoomin	nudge	use spot window	adjust time code	bump
Resync an element while locked to timecode	n/a	n/a	n/a	n/a	capture
Sample level editingto match zero crossings	yes	n/a	use zoom tool	use zoom tool	auto-zero
To chase timecode you need to	set SMPTE as timesource	select external	go online	go online	ext. sync
To jump between markers is to	data +/-	nudge	hit autolocate #	type marker #'s	tab
To ungroup elements is to	exit block, slip page	offset	click outside	click outside	n/a

will hear a suction-type sound, instead of a clean stop. The ability to play across an edit gives you the ability to preview an edit without actually performing it. In addition, randomaccess capability allows the user to play any sound at any time code location instantly. Some work stations also give you the ability to play a sound file backwards, half-speed or twice play speed.

THE ELECTRONIC GREASE PEN

Before you can edit, you need to tell the system what to remove. Sound is marked using an electronic grease pencil. If your client changes direction during an editing session, just delete all of your marks and start over. The ability to label marks helps to keep things straight. It's possible to make very accurate marks onthe-fly. Once you've made your marks it's time to start cutting.

Actually, cutting is a misnomer, because most systems employ non-destructive editing. The system just plays around the cuts. Nothing is removed from the sound file. No matter how much you chop up a sound file, you can always go back to the original recording. Nondestructive editing is great, but there are times when you may want destructive editing as well. Let's say you're cutting dialogue for a western movie and a pager goes off on the set - you may want to remove that sound never to see the light of day again.

In addition, the disk space left after you cut out a file is available to be reused again (it seems like you can never have enough disk space available). There are several caveats to destructive editing, aside from the fact that it permanently changes your master recording. Unless the disk operating system rewrites the file, deleting many small chunks of sound will leave your disk looking like a piece of Swiss cheese. Disk fragmentation limits your maximum recording time and makes the disk work harder than it needs to. To avoid excess fragmentation, some systems will rewrite a new contiguous file on disk, which takes time.

SOUND PROCESSING

Digital audio workstations operate much like a word processor in the way they edit sound. Most editing commands are based on cut, copy and paste. Copy is self explanatory and was the only command everyone agreed upon, so I removed it from the chart to make more room. In the analog world, you copy or dub something like a chorus, so it could be used elsewhere in a song. Cut is similar to copy, except for the fact that it removes a selection from your original file. Unlike analog, you can cut out parts of a sound file without changing sync. Normally, sound after the edit moves to the left to fill in the hole.

Another difference between analog and diskbased digital is that once you cut or copy something, you can paste or insert it more than once. Sound can be inserted at a specific time code location or immediately after another. Digital editing allows the user to decide whether the next sound is ignored, replaced or shifted to the right. Cut and paste can be destructive or non-destructive in nature. Sound can also be moved simply without making a copy of it.

Some manufacturers included key commands or F-keys for their editing commands. It's important to strike a balance between command keys and menu commands. As companies add even more features, it will become increasingly more difficult to include everything in pull down menus, icon palettes or soft keys. Nested menus or command keys must be used. Using command keys is faster than pointing and clicking at the screen, but it takes time to memorize them. Ideally a system should work on both levels — it should be easy-to-use for the beginner, without being slow or clumsy for the experienced user.

Continued on page 64

Lexicon Opus	Micro Technology MicroSound	NED Post Pro	Otarl Prodisk	Roland DM-80	Studer Editech Dyaxis 2+2	Turtle Beach 56k	Waveframe Audioframe
position Indicator	play cursor	scrolling events	shuttle	scrolling	play head	play cursor	play head
left to right	R to L, L to R	right to left	L to R, T to Bottom	right to left	left to right	left to right	side to side
job/reels	disk	clisk	disk	hard disk	disk	Clisk	disk
cue point	flag	marker	cue polínt	marker	log tic	marker	sync point
segment	sound file	cue	CUB	take	sound file	sound file	recording
safe segments	mix files	events	n/a	project	view files	n/a	reel or track
section	segment	region	event	phrase	view	zone	event
section	play mark	audition	play	play-phrase	view file	zone audition	play event
segment	segment	event	cue	phrase	mix element	zone	cue or event
track list/file screen	mb: file	EDL	cue list	playlist	track list	playlist	edit decision list
play/record screen	waveform window	Editvlew	cue editor	tape window	mix window	main editing screen	signal window
copies	user titles anything	takes	track versions	renamed phrases	takes	takes	re-takes
track	positional window	dipboards	clipboard	dipboard	clipboard	clipboard	clip board
play to edit	preroll	play to handle	cursor left	preview to	play before	play from	n/a
play	play mark	audition	shift cilck	play phrase	play	play sound	płay
play from edit	post rol	play from handle	cursor right	preview after	play after	play from	end select, play
play edit	skip zone	n/a	n/a	preview thru	cut play	play across	n/a
track aut	mark zone, skip	delete	cut	cut	cut	playlist	close
segment cut	n/a	aut, replace aue	delete	n/a	cut, save selection as	n/a	replace
cut delete	n/a	n/a	delete & ripple	n/a	cut, save selection as	cut	close
cut Insert	mark zone, slience	cut	erase	erose	cut	mute	replace
copy insert	snap to grid	paste or insert	place in	move	paste to left cursor	paste	lay-in
replace	snap to grid	paste or insert	place out	move	n/a	butt to zone	replace, split or lay
cut Insert	Alt. S+A	paste	chain	move to marker	abut	paste	replace, split or lay
replace	n/a	exchange	delete/place	take charge	paste, cut	paste fill	replace or lay-in
Insert	move all	150	ripple	Insert	n/a	paste	open
insert	snap to grid	place	place record time	move	paste to time code	n/a	replace, split or lay
track align	control click & drag	move	ripple all	insert	nudge or cut & paste	n/a	slip
align	drog to grid	dip	nudge	offset a move	bump	n/a	slip
n/a	n/a	align	n/a	yes	slew	n/a	split, slip
n/a	yes	trim	n/a	offset edit	fine locate	n/a	n/a
enable time code	enable chase lock	set ext. sync	chase on	set timebase to SMPTE	chase time code	n/a	select chase
go to cue point	click go to icon	locate	click on	next/previous	tab	marker tabs	click on time in
n/a	unlock	de-select	n/a	solit	deselect	n/a	n/a



THE R-E-P Interview:

STEVE LILLYWHITE

By Dan Levitin

Ritish Producer/Engineer Steve Lillywhite is best known for the drum sound from hell — the huge reverb and compressed kick-snare sound that rode with Big Country and U2 to the top of the international charts. His sounds have been often imitated but never copied. On the release of his newest production — wife Kirsty MacColl's second Charisma release, "Electric Landlady"

- Lillywhite took time out to speak with R*E*P about the album specifically, his approach to production in general and his feelings about rap music.

R-E-P: I like Kirsty's new record a lot. It struck in me the same reaction as when I first heard The Smiths — alternative music with these light, bouncy guitars...

SL: Great! That's exactly what we tried to do.

R-E-P: The album seems a bit of a departure from your other work. Compared to Big Country, U2, The Furs and so on, this is lighter. **SL**: Yes, well, personally I've gotten off the big drum sound, which is something I used to do in the mid-'80s.

R-E-P: A lot of people say that you are responsible for having introduced that big drum sound to the airwaves in the '80s. The Steve Lillywhite drum sound that began with these alternative bands became mainstream with artists such as Bruce Springsteen. It's been said that you changed the sound of radio in the '80s. SL: My goodness, that's very nice to be credited with that. Well ... I'd been experimenting with ambience around the time of Siouxsie and the Banshees' first album, which is a long time ago, and then I suppose when Peter Gabriel's album came out, and that had "The Intruder" song and stuff like that, we were using a lot of compression on ambience. Really, that's all it was

R-E-P: And the snare was huge. It was drums from hell!

SL: That's right — we like that. At least we did in those days. The reason I don't like doing it so much these days is that everyone has that drum sound in a microchip. You know how it

Dan Levitin is R•E•P's music production editor and a producer based in Stanford, CA. is when you're always looking for something a little bit different. This [Kirsty MacColl] is what I've been doing lately. I prefer drummers not to play so hard now, and I don't mic them so widely.

R•**E**•**P**: So what is the new Steve Lillywhite creative inspiration these days?

SL: Oh, I'm not sure I've got any brain cells left! I don't know...I'm into music more. I'm learning more about music rather than sound. I suppose when I worked with David Byrne on "Rei Momo" that influenced me a lot.

R•**E**•**P**: The musicians on that album were great, and you used them for some of the tracks on "Electric Landlady."

SL: Right. I really enjoyed the musicians who played on the "Rei Momo" album a lot, and so did Kirsty; it was her idea. She said, "Let's get those guys in to do some tracks on my album."

STEVE LILLYWHITE PARTIAL DISCOGRAPHY

Big Country	"The Crossing,"
ů ·	"Steeltown"
David Byrne	"Rei Momo"
The La's	"The La's"
Peter Gabriel	"Peter Gabriel 3"
Kirsty MacColl	"Kite," "Electric
	Landlady"
Psychedelic Furs	"Psychedelic Furs,"
	"Talk Talk Talk"
Rolling Stones	"Dirty Work"
Simple Minds	"Sparkle in the
	Rain"
Siouxsie and The	"The Scream"
Banshees	
The Smiths	"Ask" (track)
Trash Can Sinatras	(In production)
U2	"Boy," "October,"
	"War," "Joshua
	Tree" (remixes)
Ultravox	"Ultravox," "Ha Ha
	Ha''
XTC	"Drums and Wires,"
	"Black Sea"
	-

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R•E•P: And Johnny Marr and Elliot Randall play brilliantly.

SL: Well, Elliot was a real find, because he now lives in London. He's married to this English girl, and no one knows he's around.

R•**E**•**P**: What former work of his was it that stands out in your mind?

SL: I suppose the Steely Dan stuff.

R•E•P: Yes, he plays that solo on "Reelin' in the Years." Jimmy Page once called it his all-time favorite guitar solo.

SL: On one hand, I kind of want to keep him to myself, but on the other he's such a good player, I think he should play with everybody. He's great.

R•E•P: He *has* played with everybody — Paul McCartney's Ram, The Blues Brothers, Sha Na Na, Chuck Berry ...

SL: Yes, he's great. Johnny, too, of course, although Johnny Marr and Elliot weren't ever together at the same time on this album.

R•E•P: The song "My Affair" is a wonderful combination of a sort of "Copacabana" type of groove with a Smiths or XTC approach to songwriting.

SL: She co-wrote that with Mark Nevin, who is well-known for the Fairground Attraction. He also co-wrote some of Morrisey's new album.

R•**E**•**P**: There's tons of music going on in "Electric Landlady." How much of that was the producer, and how much was the songwriters and musicians?

SL: The game plan for the album was to record with as many musicians as it would take to make the backing tracks sound like a record, which was actually the game plan for "Kite" (MacColl's previous album) as well. But on "Electric Landlady" we went further, culminating in a session at Electric Lady studios in New York. For "My Affair" sessions, we had eight players at the same time, which is why you get this feeling of players playing off each other. It really was eight players all interacting.

We have a studio at our house, with a Sony/ MCI 24-track, and so we just took the tapes and did all the vocals at home. Then we sat back and thought about what else we needed on it. In most cases, we found we didn't really need much more.

R•**E**•**P**: And what percentage of your input is musical?

SL: Well, it's hard to say. I'm not the kind of person to tell a bunch of Latin players what notes to play, though. I get in the right people, the people I know will play what I want to hear.

R•**E**•**P**: I guess everybody asks you this in interviews, but...

SL: No, I don't do many interviews at all. I really don't like doing interviews. Most of 'em make me nervous.

R•E•P: What unique problems are there producing your wife?

SL: Well, one of the big advantages is you get to sleep with her afterwards.

R•**E**•**P**: I suppose if you had a fight in the studio and then had to sleep with her afterwards, that might not be an advantage.

SL: Funny enough, we don't fight too much in the studio; she is a talented girl. She has some great ideas, and I just let the ideas flow.

R•**E**•**P**: Tell me about the miking for the album. **SL**: Everything was pretty much close miked. I used an AKG 414 on her vocals up until the very end when I borrowed a TLM 170.

R•**E**•**P**: How did you like that?

SL: The TLM was brilliant! I wished I'd had it all the way through. Normally I have to EQ Kirsty's voice on a 414, to add top end because she's very quiet, but on the TLM I didn't need to do that. It had a crisper top end.

R•**E**•**P**: Because it's more sensitive to a soft voice you think, or because of its EQ curve? **SL**: I think because of the sensitivity, yes. I was really very impressed. I have been using it a lot recently.

We were all young, you know, and felt like "Yeah, let's go for it!"

R•**E**•**P**: You're an engineer yourself, and yet lately you've been hiring engineers. **SL**: I used to be an engineer, but I've taught myself not to look at the meters. I like sitting at the desk, and I like to balance [mix], but I don't really like getting sounds too much. I do like to get the echoes and balance the monitors. I'm not really that confident as an engineer anymore. The engineer who did Kirsty's record, Jonn Fausty, is very, very good. He's engineered something like 3,000 Latin albums. He's a free-lancer in New York.

R•E•P: What else can I ask you about Kirsty's album?

SL: I don't know, let me think...it is a bit of a departure for me, I suppose. I'm quite pleased with the sound of it. It's not my sort of sound, really ... it's a bit wider sound. You can hear things more clearly than in my other records.

R-E-P: You make an interesting point. Your U2 albums ...

SL: They all sounded a bit dense, I think.

R•**E**•**P**: Yes, it's a funny thing. They were very influential, and they're very powerful, but from a technical standpoint, they sound terrible. **SL**: Yes, they're dreadful, aren't they?

R•**E**•**P**: And yet there's some genius in them. The drums are too loud, the bass is too loud — everything sounds too loud, and too wild, too dense. It was a very bold way to record. It established, I think, a whole new sound and way of using the studio.

SL: Well, we were all young, you know, and felt like "Yeah, let's go for it!" It sounds much like a band playing live, but it wasn't done like that. There were a lot of overdubs and layering and stacking up of sounds.

R•**E**•**P**: I think the genius in the recording, and why everyone is so enamored of the sound, is because it sounds like some maniac just stuck up a couple of mics at random and recorded some wild band live, while the sound guy was on acid and making a lot of wrong moves. **SL**: (Laughs) I don't know. I've never done a mix on acid, so I don't know. I did three albums with U2, and they never did a song live before recording it after the first album. "October" and "War" were written entirely in the studio. After the recording was done and I went to hear them live, I thought they sounded better.

R•**E**•**P**: Where did you come up with the concept for recording them the way you did? **SL**: The drum sound came about because the sound I wanted had to be done out in the hallway at night. It had to be nighttime because during the day there was a girl at the end of the hall answering the telephone.

R•**E**•**P**: Your career has followed an interesting path in that, with the exception of the Rolling Stones record you produced, everything you've worked on is alternative music. They are groups and albums who have strong and devoted followings, but for the most part, they are not Top 40 smashes, they don't sell into the mutli-platinum category. They are influential albums, though, the kind that other musicians and producers hear and are influenced by. Why haven't you become involved with more mainstream artists?

SL: Because there aren't many mainstream artists that I like. There is one, R.E.M., that I would die to work with. They're great artistic people. But, I mean, who is there? Who would I work with? Bruce Springsteen!? I would have liked to work with him when he was younger, earlier in his career.

R•E•P: Vanilla Ice!

SL: No! Why? What would be the point?

R•E•P: I don't know — you could make a ton of money.

SL: I don't want to make a ton of money. I want to make music I like, and to work with people I like. Money! How much money can you spend? God ...

R•E•P: Will you ever make a rap album? **SL**: Well, there's a rap on "Walking Down Madison," but I wouldn't want to make a whole rap album because it's such a limited musical concept. I'm aware that it's good, and I like some rap records, and I'd like to maybe use it in something, but not a whole album. It would get boring, wouldn't it?

I've always just done artists and people that I like, with the exception of the Rolling Stones, because you can't say "no" to the Rolling Stones. It's just a pisser that it was at a time when Keith and Mick weren't friendly with each other.

R•**E**•**P**: Well, that's a crap shoot.

SL: So there you go. It would have been nice to work with them when they were younger, but even so, it's nice to know you've done a Rolling Stones record.

R•E•P: Yeah, it kind of fits nicely into one's discography.

SL: I don't have a discography. Or did you say earlier you have something that has all the things I've worked on?

R•E•P: Yes, your management gave it to me. Would you like me to give it to you? **SL**: No, that's OK, I'll try to get one from them.

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Circle (14) on Rapid Facts Card





By Anthony McLean

Some day, archaeologists digging down to our century will declare the 1990's to be the end of transitional analog technologies. Slightly deeper, the spoon diggers will unearth the fact that Bell Labs formatted audio signal transmission down copper cable by defining decibels and 600Ω lines in the 1920's. They will undoubtedly observe and chronicle analog's slow drift toward extinction. And they will surely notice that sometime in the 21st century planet earth went totally digital.

Fiber optic systems have traditionally been Fortune 500 domain; AT&T territory. Phil Lambert, a Siemens fiber optic systems engineer at their Princeton, IN, Potter and Brumfeild plant, emphasizes that fiber optics' big work is to link data networks and video and facility controls such as climate and security. Michael Pascazi of Fiber Optek Interconnect in Wap-

Anthony McLean is features editor of R-E-P.



pingers Falls, NY, believes the relatively tiny professional audio market hasn't held the same allure for vendors as seven-figure fiber optic installations. Audio, it seems, is the least common fiber optic application and the smallest piece of the fiber optic pie.

DIG THIS

Causing no capacitance, no inductance, and no hum, multiplexed digital signal transfer is clearly superior to analog. For simple audio signals, even twisted pair copper wire can be used to transfer digital audio. But at the high end, fiber optics easily manage audio and video signal, plus all other production-required data exchanges such as SMPTE, MIDI and RS-232.

Eventually, audio workstations (a.k.a. digital production workplaces) will force universal transition to bi-directional fiber optics within the studio production environment. In 1992, audio engineering standards and practices are surfing the second wave of digital technology. Near term, digital transmission interface seems



Figure 1. Centralized Fiber Optic Processing Topology.

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finally positioned to penetrate professional audio. It's the current wave. Fiber optics, arguably the most elegant of transmission formats, are up ahead on the beach.

Since early fiber optic cabling was relatively brittle, live sound applications were out of the question. Expensive A/D and D/A conversion inhibited fiber optic penetration into production suite environments.

But as chip-based digital technology has improved and optic cable production capacity has skyrocketed, fiber optic hardware now approaches justifiable performance and expense territory for professional audio applications. In the '90s, the question changes from "if" to "when" pro audio will go fiber.

The fiber optic lexicon includes familiar terms such as transmitter/receivers (a.k.a. data links), transceivers, relays, interface modules and connecter/jumper hardware.

Giants such as Siemens International, in an effort to cross over from phone to commercial audio, manufacture components meeting FDDI (Fiber Distributed Data Interface) standards. While unsuited to professional audio, FDDI successful applications include CCTV systems in Munich's new airport and mega scale computer networking. Other standards, such as the TI and T3 telephone standard also fail to reach professional audio's requirements.

GLASS/COPPER

A single optical fiber, or twisted pair, can simultaneously transfer various data types (its all data to the cable) via digital multiplexing. This encode-decode process integrates multiple audio and other inputs into a hyper-speed serial data stream output which is essentially loss free and signal phase/flat out to gigahertz.

The serial data string is primarily dependent only on its relative timing location to be accurately captured and processed.

As such, the key to any fiber optic data exchange is the system's fail-proof capacity to encode and decode, via timing control, all of the signal data (digital information) within the data stream.

Two basic approaches to fiber optic systems have evolved. Central processing, which is common to telco systems, is the original and most common application topology. Central processing strategy consolidates sub-systems, concentrates hardware and is perfectly matched to the telco universe. (See Figure 1.)

Less common is distributed processing topology. This approach, which is analogous to multiple outboard racks of routing, distribution and digital signal processing, seems well suited to professional audio. In general, distributed processing emphasizes minimum lengths of analog signal paths by making A/D conversions in relatively close proximity to the source and maintaining digital signal and processing control for as long as possible.

In the event of component or cable failure,

The marriage of fiber optics and audio production seems long overdue.

the distributed technique can immediately construct an alternative signal path through application of a fail-safe switching program. While differing according to vendor, distributed processing can automatically execute glitchfree, hyper-fast bypass and reconnect commands. (See Figure 2.)

Since signal strength and risk management

(i.e. keeping the signal flowing) constitute pro audio's universal "Job #1," the marriage of fiber optics and audio production seems long overdue.

THE LONE STAR CONNECTION

Lester Audio Labs of Dallas expects to deliver their first audio-fiber commercial system in January 1992. Named the DAS 2000, the rackmount package fits eight channels per card to execute the A/D/A signal conversion, and is expandable to 64 send and 16 return channels.

Lester's software-based "soft patch" control uses an expandable 32×32 matrix for multiple routing, 1/O, gain staging, and phantom power configurations.

Non-volatile memory stores 190 patch configurations for point-to-point or point-to-multipoint switching.

Designed for mobile applications such as broadcast remote trucks and touring sound reinforcement, Lester beta-tested their system with Turner broadcasting at the 1990 Seattle Goodwill games.

Paul Trimble is a veteran sound engineer and Lester's director of new product development. As the portable systems begin to ship, Trimble is developing fixed installation systems for new construction and retrofit.

Acknowledging the global production market, Trimble noted that, "The broadcast situation, the sound reinforcement situation, and studio installation, as well as theme-park/hotel complexes are all areas where multiple stages to some master control and multiple output points," define a potentially large fiber optic user base.

Acknowledging that "smart buildings, completely interactive, hold the greatest international market potential," Barry Thorton of OptoDigital in Austin, TX, anticipates an immediate \$1.15 trillion annual market in fiber optic broadcast sales for improvements and capital expenditures.



Figure 2. Distributed Fiber Optic Processing Topology.





Aphex Modular System

Once hooked on Aphex signal processing, people have an insatiable appetite for more and more. Space then may become a problem. That is why we shrank four of our best ... the Aural Exciter[®], Compellor[®], Expressor[™] and Expander/Gate.

These modules feature all the processing power and performance of their standalone counterparts including our servo-balanced inputs and outputs. You can fit 11 modules in our 3RU Model 9000 rack* (or nine in the compatible dbx 900 Series rack).

Here's a brief rundown on these powerful tools:

- Aural Exciter- the signal enhancer that increases intelligibility, presence, clarity, and detail.
- Compellor the "intelligent" compressor/leveler that controls levels as if a mixer were riding faders.
- Expressor a full featured compressor/limiter that lets you tailor the sound your way.
- Expander/Gate simply the world's finest gate, no one ever met our \$10,000 challenge to find a better one!
- And, more to come. See your nearest professional audio dealer to rack up more processing power per inch than ever before.

Circle (16) on Rapid Facts Card



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Aphex is proudly American ... 100% owned, engineered and manufactured in the U.S.A.

* Unpowered, separate power supply available

Particularly strong is the state-supported European broadcast market. Continental governments use TV as a sociological tool, and don't hesitate to spend on transmission technology. Specified for all new Spanish TV installations, OptoDigital has also equipped many European video remote trucks.

TO NODE OR NOT TO NODE

For professional audio, the less time and distance that audio signal remains as analog the better. The fewest possible D/A and A/D conversions are also best, so OptoDigital recommends A/D conversion near the source. Once digital, all signals are merged and multiplexed at a device OptoDigital calls their "Node.

Nodes act as the ultimate electronic traffic cop. Essentially, a node is just smart enough to do its specific job, but lacks the knowledge to run the entire system. Some nodes simply switch and route. More intelligent nodes, after receiving electrical digital signal output from the A/D converters, output in optical, and simultaneously multiplex, switch and route within proprietary Bendix connecters.

optic universe.

president of engineering, produced his first prototype 40-channel multiplexed, wired-based system in 1988.

Initially, Bec systems were large format (64to 80-channel) mainframe configurations. Internal market research concluded this was not what the market desired. Bec then redesigned its twisted pair and fiber systems so that modular components could be strategically assembled for various sized applications.

Bec's current hardware is fifth generation and uses an AT&T drive chip set. The 5-device system can be assembled in multiple configurations, features dc backup and is anchored by a programmable logic device Bec calls its PLD. Easy pin-to-pin chip set changeout is inherent to Bec's upgradable design.

Using three levels of redundant communication, Bec seeks to eliminate risk through its Fault Tolerant Redundant Communications. According to their marketing statement, "Fault tolerance uses redundant systems to monitor performance, switch any faulty piece out of the system, bring one of the backups on-line and let the engineer see that a failure has occurred

so that he can service the unit at a more convenient time." Bec's system synchronizes on every sample to remain fault tolerant. Creamer uses the analogy of "a light pipe with taps" to

communicate fiber optic signal circuitry. With the equivalent of a file server at each tap location, Creamer insists fiber optics is simply a better mouse trap. When asked about fiber optic's obvious utility within a digital environment Creamer responded, "I don't think it's so much the digital, it is just a better way to transmit."

OLD DOGS, NEW TRICKS

"On the horizon," Creamer says, "is a generation of (performance) consoles with mic preamps and A/D conversion taking place near the mics." But for various reasons, audio vendors seem reluctant to adopt digital transmission technologies.

At the big picture level, Creamer feels digital signal transmission eventually "... will change the way people do things." By removing time consuming and gut wrenching signal distribution hassle, Creamer envisions a time when "... (engineers) can spend their time on more important things, like the creative process of mixing."

Sudden impact of a change to fiber optics in drive line technology would be profound. If, and when, touring technologies assimilate digital signal transmission, distro gurus will likely change status from drop-dead important to nonessential commodities.

Factor the reality of a 1- to 3-D/A signal split without signal loss, add the fact that a fiber optic cable splice can be executed in five minutes, and the future seems clear. Fiber optics are definitely the audio snake and splitter for all of us.

PUSHING THE ENVELOPE IN THE REAL WORLD

Craig Dory spent six years in strategic planning and exploration development at Bell Labs. Eventually, Dory left Bell Labs to merge his fiber optic experience and self-educated recording chops.

The result, Dorian Recordings, has redefined

classical music recording. Dorian customizes nearly all of their gear. Using Steven Paulmodified microphones, mic specific Jensen preamps and Wadia DigiLink A/D converters and fiber optics, Dorian has established an entire environment appropriate to digital recording.

These electronics comprise what Dorian calls its "pod". A Dorian pod integrates non-phantom (direct voltage to mic element) powered preamps and highly evolved A/D converters with special dithering circuitry. Close (within 15 feet) coupling between pods and transducers minimizes inherent analog signal problems such as capacitance, inductance and temporal deviations. Optical output maintains this pristine signal until its destiny with a DAT recorder.

Dorian records globally, but uses the acoustically renowned, 1,200-seat, Troy Savings Bank Music Hall as home. Labor intensive preproduction earmarks all Dorian recordings. Foregoing equalization, the Dorian technique places a premium on input signal, with mic placement as the ultimate detail. Nothing gets fixed in the mix, because nothing is left to fix.

Dorian assures success, via redundancy, by simultaneously running tape on as many as four recorders, each with a particular mic array. The results are nothing short of spectacular.

Dory candidly points out that, "I was surprised. (When You) synergize all your electronics, and eliminate mic cable, even at the 50foot level, it's definitely worth it." Dorians' modifications aren't for sale, but Dory did speculate that his approach might transfer to a bigger production universe, depending on budget.

Obviously, this is audiophile stuff, but one wonders what 48 digital tracks of this would sound like?

LETS GET META-PHYSICAL

Marshall McLuhan distilled the future into his conceptual world village and the production environment is moving, via hyper-media upgrades, to online status with whole earth scope. McLuhan could not, however, have imagined how his concept would become say ... virtual reality.

When the inevitable happens, and universal fiber optic interconnect between our production facilities occurs, the effect will transcend mere production.

It will accelerate the evolution to a world village. Creative communities, such as your favorite studio, will assume the role of international artist colonies. By eliminating distinctions between video, audio and allied production technologies, digital interconnect will sledgehammer the way we do our work.

FOR FURTHER INFORMATION

Lester Audio Labs 214-637-9311; fax 214-637-9314.

OptoDigital 512-338-4707; fax 512-794-9997.

Bec Technologies 407-855-8181; fax 407-856-7516.

Transition to fiber optic is, according to Thorton, "A migration path from the old to the new, not a stampede." Thorton added that,"Everyone agrees that by pulling the fiber you wont get hurt. Pick some small, almost irrelevant,

Redundancy is the order of the fiber

part of your business ... put the fiber in there. When (users) forget it's there, it is assimilated." OptoDigital has petitioned the EIA to have RS 9000 standardized for protocol and cable pin-outs. Reasoning that a 450Mbyte standard is needed for professional performance and to facilitate global interface, OptoDigital's goal is to achieve some common ground for the

"migration" to fiber optic practices. While some structures, such as Hewlett Packard's HPPI exceed the 450Mbyte rate, (up to 622Mbytes) none have achieved a true "default" status. One seductive byproduct of OptoDigital's sys-

tem is the DSP resident in its A/D converters For users needing interactive, temporal control of multiple signal paths, the possibilities loom large.

Signal distribution reliabilities are, in a high risk business such as live broadcast or performance, serious as a heart attack. OptoDigital's routing and processing control, as applied to risk management, is designed for "can't-everfail" venues. The alternative path controls inherent to their software driven fiber optics are intended to eliminate potential interconnect failure. Redundancy is the order of the fiber optic universe, and OptoDigital understands that clearly.

ANOTHER PLAYER

Michael Creamer, vice president of sales/ marketing for Bec Technologies agrees that, "There are no second chances!" Interface failure of a remote truck at a Stones pay-for-view, or a Holyfield-Tyson fight, could ruin the company at fault.

Bec, with offices in Orlando and Seattle, came to the audio industry from the military/industrial complex. Bob Proctor, vice



TC 2290: Frequency Response: 20-20KHz, +0 / -0.5 dB THD: < 0.05% 1KHz, 0 dBm; Dynamic Range: >100 dB Digital Conversion: Dynamic Differential Sampling Rate: 1 MHz; Max. Input Level: +22dBm M5000: Frequency Response: 10-22KHz, -0.5 dB, Fs=48KHz THD: < 0.01% 1kHz, 0 dBm; Dynamic Range: >100 dB Digital Conversion: Linear 18bit 64X in, 20bit out Sampling Rate: 48KHz, 44.1KHz, 32KHz; Max. Input Level: +22 dBm

The legendary TC2290 has a new partner! Meet the M5000 Digital Audio Mainframe, the world's first user expandable DSP system. You've relied upon the TC2290 for the ultimate in digital delay effects for years, now the M5000 is the latest tool for Reverb, Ambience and Pitch Shift effects. Like the TC2290, the performance of the M5000 is simply astonishing. In fact, the M5000 will change forever the way you think about moderately priced digital signal processing. Why? The M5000 uses T.C.'s proprietary DARC (Digital Audio Reverb Co-processor) technology. DARC boosts the M5000's digital processor well beyond normal levels of performance and provides the power to support the complex algorithms desired by even the most demanding audio professional.

The M5000's High Speed 24bit buss supports up to four modules. The standard M5000 configuration of 1 AD-DA module and 1 DSP module leaves an additional pair of empty slots. Want two completely independent true Stereo processors with Analog and Digital I/O in two rack spaces? Simply add a second set of modules for a fraction of the cost of an additional unit! All Digital Studio? Configure the M5000 accordingly with up to Eight channels of Digital Processing with AES/EBU, SPDIF and Optical I/O. Standard Interfaces include MIDI, Ram Card and SMPTE (in). Options include SCSI, LAN, & Floppy Disc Interfaces. Future M5000 modules will allow you to upgrade your unit as technology evolves. Say good-bye to planned obsolescence!



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Circle (17) on Rapid Facts Card

A 1992 analysis of the important professional features today's DAT recorders provide.

DAT ANALYSIS

By Rick Schwartz

ven though most of today's professional DAT machines seem to offer similar features and capabilities, one almost has to look past the spec sheets and feature sets to find out what they really provide, functionally. We've done the footwork for you.

Here's an up-to-date analysis of the current crop of DAT recorders. We hope this helps you in your shopping and spec-ing activities.

AIWA HH-B1

A popular new portable from AIWA (with help from the U.K.'s Hilton Hire) uses 1-bit A/D converters. The HH-BI supports both AES/EBU (via an XLR splitter lead) and IEC digital formats. SCMS can be avoided by using the AES outputs. According to the manufacturer's literature, the unit will not record at 44.1kHz using the analog inputs; weighs only 0.9kg (without batteries); will record for up to 220 minutes using AA dry cell and lead-acid rechargeable batteries; has the distinction of being the only portable that can write, erase and renumber program IDs; and also will record absolute time code.

FOSTEX D-20

This unit achieved popularity by being the first DAT on the market with SMPTE time code capabilities. The D-20 reads and writes SMPTE time code, which conforms to the AES/EBU standard, as well as the Fostex time code standard. It has word-clock inputs and outputs. The D-20 is a 4-head machine with monitoring offtape during recording. Because the unit has been around for awhile it escaped SCMS. Indexes are not transferred during digital dubs, because the unit only has an AES-EBU digital port. It has no auto indexing capability. Poststriping SMPTE time code is useful, but program IDs are erased in the process. Although the unit will perform audio punch-ins, they are not always glitch free. The D-20 is still the only machine on the market with pitch-control, although the process creates problems on digital transfers.

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

JVC DS-DT900

The largest DAT player on the market, it has $64 \times$ oversampling A/D converters with 3-stage FIR digital filter and fourth order noise shaper; reads and writes SMPTE time code and will slave to external sync; has an hour meter that indicates tape head wear. Also features a built-in head cleaner; has AES/EBU digital interface only; will perform digital fades; and may be remote controlled via a 45-pin parallel interface, or a 9-pin serial interface.

PANASONIC SV-255

This is Panasonic's popular portable with lower noise specs and better-sounding filters than its predecessor - the SV-250. The unit weighs only 3.2 pounds and has a rechargeable battery that will last up to 2.2 hours; has a unique dual-channel mono mode that records one channel -15dB lower to protect from signal overload; uses a non-standard unbalanced connecter for S/PDIF digital output. Because it was designed to be used in the field, the unit has no digital input, and it will not record at a sampling frequency of 44.1kHz; does not allow the user to write or erase program IDs or renumber start IDs; program IDs are not transferred during a digital dub; and the unit does not read or record absolute time.

PANASONIC SV-3500

Although this is an older machine, we include it for reference purposes because of its use in so many professional facilities. It will copy protected CDs, by changing an internal switch; does not acknowledge SCMS, so a user can make unlimited digital copies; and program IDs are not transferred during a digital dub. Head cleaning is extremely difficult. To properly clean the heads the entire unit has to be disassembled, which can take several hours. It also includes a selectable error rate readout.

PANASONIC SV-3700/3900

Except for their remote controls and the fact that the 3900 has a RS-422 port, both of these machines are similar, using same transport and one-bit converter technology. Both use the IEC-958 format for consumer digital I/O. Minor digital transfer problems caused by the new format can be solved by cutting a capacitor inside the unit, which voids warranty. Contrary to popular belief, copy-prohibit is completely avoidable on both consumer and professional digital inputs. Copy prohibit bits can be ignored, passed on, added or removed although control over SCMS would be easier if the back panel DIP switches were labeled. As long as there is ample time in-between indexes, the consumer ports will transfer program IDs during a digital transfer. Panasonic is one of the only manufacturers that post-records absolute time code. Both machines write AES/EBU standard time code that can be read on any SMPTE time code machine. The SV-3900 will locate to absolute time using optional Macintosh software. However, without purchasing the optional remote for the SV-3900, a user cannot access several important features, including end search. Ironically, several essential features are missing from the remote, including auto record mute and door open. It has a built-in error rate readout display for both heads.

SHARP FXD200/YAMAHA DTR-2

Some people feel this machine is one of the best-sounding units on the market, with its dual 1-bit A/D and D/A converters with oversampling digital filters. Although the unit has fiber optic and coaxial digital I/Os (using a locking industrial-type BNC connector) it doesn't support AES/EBU. Sharp does not acknowledge copy prohibit flags, so users can make unlimited digital copies. The Yamaha DTR-2 has balanced and unbalanced connectors. Both units will locate to absolute time location and both offer 32kHz extended record and play capabilities.

SONY PCM-2000

Although Sony's professional time code portable has been out for almost four years, the time code section hasn't worked effectively until recently. There is no way to pre-stripe or post-stripe a tape with SMPTE time code, which limits the device's usefulness in a postproduction environment. It is one of the only units on the market that will record at 44.056kHz. It has only AES/EBU digital I/O; cannot write, erase or renumber program IDs, and will not record absolute time. Weighing in at 9 pounds 4 ounces (without the BVG-200 time code generator) the PCM-2000 is the heaviest portable on-the-market.

SONY PCM-2700

A much-improved version of the PCM-2500, this unit has confidence playback heads for read-after-write capability and 1-bit D/A converters, and will record and playback at 32kHz for extended play capabilities. Although the
(1) AES/EBU is accessable via a mini-connector and a splitter cable which is included

(2) Start IDs will be transfered during a digital dub if the S/PDIF port is used and AUTO is on

(3) Full Control over SCMS is provided via rear panel DIP switches

(4) The Panasonic SV-3900 will located to absolute time using optional Mac software

(5) Digital output appears on a 2.5mm miniture pin plug which requires included adapter cable. No digital input

(6) Uses a professional BNC connector

(7) Uses a 1/4-inch Jack with optional adapter cable

- (8) New version of the TCD-D10 will read and write absolute code
- (9) With an optional 30 pin cable

(10) Although there is no time code input, the units record SMPTE-compatable absolute timecode

N/A = Not Applicable

List Price	Confidence Head Playback	Portable	General	Post-stripes SMPTE Time Code	Records SMPTE Time Code	Will Locate to Absolute Time	Post-stripes Absolute Time Code	Reads Absolute Time Code	Stripes Absolute Code While Recording	Time Code:	Will Remove SCMS Bit During Digital Dub	Will Ignore SCMS Bit on S/PDIF Ports	Adds SCMS Copycode to Digital Copies	Will Record from a Copy Prohibited CD	Copy Protection	Will Copy Program IDs During Digital Dub	Manual Writing of Start/Skip IDs	Will Renumber Program IDs	Program IDs	Error Rate Readout	Meter Range	Internal Metering Reference Level (+4 dBu)	Metering & Display	OPTICAL (EIA) CP-340)	S/P DIF (RCA type)	IEC-958/Type-2 (RCA type)	AES/EBU (XLR-3 type)	Digital I/O:	PROFESSIONAL DAT RECORDERS	
\$2,000	No	Yes		No	No	No	No	Yes	Yes		No	No	Yes	Yes		No	Yes	Yes		None	-40 to 0	-18dB		No	Yes	No	Yes (1)		Aiwa HHB1PRO	
\$8,500	Yes	No		Yes	Yes	No	No	Yes	Yes		N/A	N/A	No	N/A		Yes (9)	Yes	Yes		Yes	-50 to 0	-18dB		No	No	No	Yes		Fostex D-20	
\$4,500	No	No		No	Yes	No	No	Yes	Yes		N/A	N/A	No	N/A		No	Yes	Yes		1 LED	-60 to 0	-18dB		No	No	No	Yes		JVC DS-DT900	
\$2,700	No	Yes		No	No	No	No	No	No		N/A	N/A	No	N/A		No	No	Yes		Yes	-50 to 0	-18dB		No	Yes (5)	No	No		Panasonic SV-255	
\$2,500	No	No		No	No	No	Yes	Yes	Yes		No	Yes	No	Yes		No	Yes	Yes		Yes	-60 to 0	-18dB		No	Yes	No	No		Panasonic SV-3500	
\$1600/\$2100	No	No		No	No	Yes (4)	Yes	Yes	Yes		Yes	Yes (3)	Yes	Yes		Yes (2)	Yes	Yes		Yes	-60 to 0	-18dB		No	No	Yes	Yes		Panasonic 3700/3900	

- (1) AES/EBU is accessable via a mini-connector and a splitter cable which is included
- (2) Start IDs will be transfered during a digital dub if the S/PDIF port is used and AUTO is on
- (3) Full Control over SCMS is provided via rear panel DIP switches
- (4) The Panasonic SV-3900 will located to absolute time using optional Mac software
- (5) Digital output appears on a 2.5mm miniture pin plug which requires included adapter cable. No digital input
- (6) Uses a professional BNC connector
- (7) Uses a 1/4-inch Jack with optional adapter cable
- (8) New version of the TCD-D10 will read and write absolute code
- (9) With an optional 30 pin cable
- (10) Although there is no time code input, the units record SMPTE-compatable absolute timecode
- N/A = Not Applicable

PROFESSIONAL DAT RECORDERS	Sharp SX-D200	Sony PCM-2000	Sony PCM-2700	Sony TCD-D10	Tascam DA-30	Technics DA-10	Yamaha DTR2
Digital I/O:							
AES/EBU (XLR-3 type)	No	Yes	Yes	Yes (7)	Yes	No	No
IEC-958/Type-2 (RCA type)	Yes (6)	No	Yes	No	No	Yes	Yes
S/P DIF (RCA type)	٥N	No	No	Yes (8)	Yes	No	No
OPTICAL (EIA) CP-340)	Yes	No	No	No	0 N	No	Yes
Metering & Display							
Internal Metering Reference Level (+4 dBu)	-18dB	-20dB	-20dB	-20dB	-16dB	-18dB	-18dB
Meter Range	-50 to 0	-60 to 0	-60 to 0	-50 to 0	-50 to 0	-60 to 0	-50 to 0
Error Rate Readout	No	No	oN	No	g	Yes	٥
Program [])s							
Will Renumber Program 1Ds	Yes	No	Yes	No	Yes	Yes	Yes
Manual Writing of Start/Skip IDs	Yes	Yes	Yes	No	Yes	Yes	Yes
Will Copy Program IDs During Digital Dub	No	Yes	Yes	No	No	Yes	Ŋ
Coox Protection							
Will Record from a Copy Prohibited CD	Yes	N/A	Yes	No	Yes	Yes	Yes
Adds SCMS Copycode to Digital Copies	٥N	No	Yes	No	Yes	Yes	Yes
Will Ignore SCMS Bit on S/PDIF Ports	٥N	Yes	No	Yes	No	No	No
Will Remove SCMS Bit During Digital Dub	No	No	٥N	No	No	No	No
Time Code.							
Stripes Absolute Code While Recording	Yes	No	Yes	No (8)	Yes	Yes	Yes
	Yes	No	Yes	No (8)	Yes	Yes	Yes
Post-stripes Absolute Time Code	°N N	No	No	No	No	Yes	No
Will Locate to Absolute Time	Yes	No	Yes	No	0N N	No	Yes
Records SMPTE Time Code	No	Yes	No	Ŷ	9Z	No	No
Post-stripes SMPTE Time Code	No	°N No	٥N	٥N	٥N	No	°N No
General							
Portable	No	Yes	No	Yes	9N	No	οŊ
Confidence Head Playback	No	οN	Yes	٥N	9Z	No	No
List Price	\$2,200	\$8,520	\$2,900	\$2,900	\$1,500	\$1,000	\$1,495

Continued from page 34

unit will accept signals from an AES/EBU source, signal level adjustment may be required. By entering an absolute time address via numeric keys, location to any point on tape can be achieved. It has a rehearsal function that allows the user to trim start, skip and end IDs. The digital fader works on analog and digital inputs and outputs. Tapes can be time- and datestamped using an internal clock.

SONY TCD-D10 PRO

This is the best-selling portable from Sony, and one of the only units on the market with a built-in speaker. Tapes can be time- and datestamped using an internal clock. Although current machines do not have absolute time code, the unit is being upgraded to include it. It will not record at 44.1kHz using the analog inputs. Unfortunately, the device cloes not fully support AES/EBU specifications (factory update may be available) and has some clock jitter problems. As a result, the device will not work with SoundTools without a mod that voids it's warranty. It doesn't read program IDs from other machines, and indexes are not transferred during a digital dub.

TASCAM DA-30

This unit has $64 \times$ oversampling delta-sigma A/Ds and 8× oversampling 18-bit D/As. It has some digital clocking problems. Tapes recorded on almost any other machine won't have start IDs read correctly. If a tape has any type of copy flag, you will not be able to make copies using the consumer ports. With the AES/EBU ports this is not a problem. Margin display shows available headroom in decibels; it reads and

writes absolute time code; indexes are not transferred during a digital dub; 32kHz recording is possible using digital inputs; has 15-pin parallel I/O port; and has +4/-10 signal on XLR-3 and RCA connectors.

TECHNICS DA-10

A best buy. Similar in many ways to the popular SV-3700, but uses lower grade converters. It has MASH noise shaping and 1-bit converters; analog I/O is -10 on RCA connectors; can stripe absolute time after the fact; and has builtin error-rate readout. Like the SV-3700, program IDs can be post recorded or erased. The DA-10 has coax and fiber digital I/Os; SCMS; but no AES/EBU. Digital fades only work on analog inputs.

If digital I/O problems have got you down, cheer up.

DIGITAL INTERFACING

If digital I/O problems have got you down, cheer up. There are a number of problemsolving interface boxes on the market that perform useful functions including copy prohbit bit stripping, SCMS elimination and format conversion. My favorite is a device manufactured

by Digital Domain. (We like the name, too -Ed.) The FCN-1 format converter lists for \$500 and will do conversions from consumer to professional formats including IEC-958, which is new type of consumer digital standard. Although IEC-958 is designed to be interchangeable with S/PDIF, it has a slightly different word clock, which can sometimes cause problems. according to Jesse Jacobson at The DAT Store in Santa Monica, CA. The FCN-1 will add or remove C-bits which allow the SCMS to keep track of how many copies of a tape have been made. It also adds and removes emphasis (early machines such as the Sony 2500 and the Sony DTC-1000 could only add emphasis). It also has a digital distribution amplifier, so you can go in from one machine and out to four machines simultaneously. It will also make sure that tapes have a sampling frequency flag and it claims to be compatible with future formats such as DCC and optical recording. Options include polarity inversion, digital "over" indication, and channel reversal in the digital domain. If your budget is a little tighter and you just need to strip out the copy prohibit flag, to record protected CDs, for example (professional, not-forprofit applications only, please!), another device called the Digital Inserter will do the job.

Rick Schwartz and R•E•P would like to thank Jesse Jacobson of The DAT Store for his assistance in the preparation of this article.



Out on the wildest edge: audio design and post production for MTV's Liquid TV.

THE SOUNDS

iquid TV, airing in prime time this past summer on MTV, presented a unique, irreverent 30-minute blend of diverse animation segments by a variety of artists and writers. San Francisco-based Colossal Pictures originated the concept for the weekly se-

ries and Music Annex Audio Post Production, also of San Francisco, handled the audio for the shows. Described as 'dangerous, delicious and disposable,' Liquid TV parodies existing TV and comic book forms with no segment lasting more than three minutes.

Fifty-seven different segments comprised the first shows, so hundreds of audio elements on many different formats were delivered. In addition to the nine different Colossal Pictures directors creating segments, 13 other directors provided animated segments as well. All of these pieces had to be woven into a continuous tapestry of animation and sound by the Colossal/Music Annex team. According to Music Annex sound designer Jon Grier, "Having sat with pen in hand for months in silence, animators tend to have unique and well-developed ideas regarding sound. Each show was made up of 9 to 12 segments and various intros and bumpers that we were completing in a very compressed time schedule. Audio elements were flying in the door from directors in almost as many formats as you can imagine, but everyone retained their sense of humor ... after all, this is Liquid TV?

Keith Hatschek is vice president of marketing for Music Annex, Fremont, CA

Initially, dialogue segments were recorded on 1/4-inch tape and transferred to mag stock for the directors. They then edited the original dialogue elements on film, and in a process called "Track Reading" assigned frame numbers (cel numbers) as a road map for animators to create pictures. Once animation was assembled,

earnest.

to what they heard for the individual segments. Then we rolled

up our sleeves and got to the fun

stuff," says Grier.

One of the principle areas requiring sound design was the innovative host segments that set the tone for the shows. "The host segments feature body parts floating in water with the host voice talking to the audience," says Grier. "In one host segment, a pair of lips is suspended in water talking ... these lips got to be known

By Keith Hatschek



Liquid TV sound team Jon Grier, Will Harvey and Mary Ellen Perry in Music Annex, Studio III.



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"The layout of Studio III and Studio IV made these changes easy to perform and the producers could peek into both rooms alternately to monitor any changes, as well as keep tabs on

Imagine a steamy soap opera performed by stop-motion animated bars of soap and a soap pump named Judd.

the next show mix" says Harvey. "D-2 offers four channels of audio as well and we were able to lay the stereo show mix on tracks 1-2, a mono mix on track 3, and an M&E mono mix on track 4. With this layout the show was ready to go out for foreign language versions, with all elements on one piece of video tape."

As the mixing process headed into the home stretch, word came down that a U.K. version would be required. The air dates for the U.K. shows were very soon, so supervising engineer Randy Bobo dumped the completed American show mixes from the D-2 digital master into

SERIES CREDITS

MTV

Executive in charge of production -Abby Terkuhle Creative supervisor - John Payson Special thanks to - Judy McGrath

COLOSSAL PICTURES

Executive producers - Japhet Asher and Kit Laybourne Producer - Prudence Fenton Co-Producer - Nicole Grindle Production coordinator - Amy Capen

VIDEO POST - WESTERN IMAGES -SAN FRANCISCO

Senior editor - Pat Caballero Editors - John Henkel and Alan Chimenti

MUSIC ANNEX AUDIO POST PRODUCTION

Supervising engineer -Randy Bobo Sound design/segment mixes -Jon Grier Show mixer - Will Harvey SFX editor - Mary Ellen Perry Transfer engineer - Linda Lew

and just played out a new ending and matched levels and EO for a fix. These types of changes would be almost impossible using tape. We would have had to call in the composer, and there just wasn't time."

High technology like the Synclavier wasn't the only thing to save the day in some instances. When creating a track for a segment titled "Footworks - Dog Flirting," where the picture showed the footprints of two dogs in a courtship ritual, unnamed Music Annex engineers rose to the occassion and provided Foley for the dogs breathing, panting, sniffing and even creating an effect fondly named "puppy piddle" using two pans of water to match the on-camera "action." Another segment entitled "Soap Opera," which details an unrequited love affair between a bar of soap and a liquid soap dispenser using stop-motion photography, required some quick Foley work as well.

"In one scene, the soap pump is getting aroused and we had to create a good sound for his pump, so we sampled the sound of a balloon being stretched and then played with the pitch in the Synclavier until it matched the oncamera action," says SFX editor Mary Ellen Perry. "Bringing the bars of soap to life was tremendous fun."

TRACKING THE CHANGES

Because of the varying amount of off-line editing that occurred to so-called "finished edits," often substantial changes occurred to what had been thought to be final dialogue





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Designing a flexible all-digital audio system for video post production.

VIDEO POST & TRANSFER'S DREAM

The digital audio revolution isn't just confined to CD production — the video world is going digital, too. "People are going to have to look a lot harder at digital audio, now that we have D-1 and D-2," said Neil Feldman, president of Video Post & Transfer, a leading postproduction house in Dallas. The professional video formats he's referring to offer unprecedented audio quality, thanks to their four 48kHz digital audio tracks.

"In the world of 1-inch, the video machine acted as an audio filter," explained Feldman. "The quality could not exceed what that machine could do. When you replace 1-inch with a digital machine, that filter is gone. Now you can hear the limitations of the audio mixer and the electronics."

To maintain the audio quality the new formats demand, Feldman and his staff have been building a "Dream" — a Digital Real-Time Editing for Audio Mastering system — "to keep digital, digital." While pursuing that goal, they found they could solve an other problem: keeping client costs down and minimizing production time.

According to Ron Evans, one of VP&T's audio engineers, "Clients wanted to mix in the post room. They might have only a couple of channels of audio, and they didn't want to go sweeten somewhere else. They wanted to be finished when they were finished."

Said Feldman, "We set out to keep things digital inexpensively, and in the course of putting things together, we realized we could do a lot more in the on-line bay."

The Dream system combines three technological advances of the past few years: harddisk digital recording, the AES/EBU digital audio interface, and MIDI. It is built primarily around off-the-shelf components, which means its cost is considerably lower than dedicated systems, whose price tags inevitably reflect their manufacturers' R&D costs. It also means that the system is modular, and as new technologies and products become available, they can be incorporated at low cost without obsoleting the rest of the system.

At the heart of Dream are a Macintosh Ilfx computer equipped with a 650Mbyte hard-disk drive with Digidesign's "SoundTools" system, and two Yamaha DMP7D digital audio mixers. The mixers are ganged together into a 16input/4-output configuration. They use Yama-

Paul D. 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 is a member of the Executive Board of the MIDI Manufacturers Association.



Far field at the first Dream system desk.

ha's proprietary digital I/O format, so signals from the D-2 machines, SoundTools and a Studer 730 CD player, which are all in AES/EBU format, go through Yamaha digital format converters. The mixer outputs on their way back to videotape are also converted.

Analog audio decks — for instance Nagras and multitracks — as well as analog audio from ³/4-inch, Betacam/SP and 1-inch video decks go through the Yamaha A/D converters. "Whatever comes in," said Feldman, "we digitize and work with in the digital domain. Clients are getting sophisticated enough to know that if they are using digital mastering, they need digital all the way through — that's what they're paying for."

HARD-DISK AUDIO

Much of the audio work is done directly from D-2 to D-2, but when more tracks are needed, or when sound has to be moved around in time, SoundTools is used. SoundTools, Digidesign's 2channel recording and editing system, consists of a hardware card that fits into one of the Mac's NuBus slots, an AES/EBU interface, and Sound Designer II software. It provides visual display of waveforms and a wide array of tools, such as mixing, crossfading, equalization, dynamic compression, time-slipping, time compression and expansion without pitch alteration. It records and plays directly from the Mac hard disk, with 650Mbytes providing more than one hour of stereo sound. It locks to SMPTE time code by first converting it to MIDI Time Code (MTC), a task handled at VP&T by an Opcode Studio 3 interface.

Evans, principal architect of the system, explained how it is used: "We can fly in sound effects from a CD in digital, stretch or shrink them, and move them around in time, with frame accuracy, then drop them over to D-2. Normally, the four D-2 channels are sync sound, voice-over, music and effects, but with the Mac we can also do stereo effects or music and keep it on line.

"We also use the Mac for music editing, and it does a great job with crossfades. We do dialogue replacement, and we can do both hard sync and 'squeeze' stuff to fit. Sometimes we do corrective editing on it, too. We had one track come in that was very distorted, and we were able to redraw the waveforms where the distortion occurred, and really clean it up."

Feldman added, "Repairing distorted audio is not a booming business, but it's something clients can use."

Off-line storage is handled by Digidesign's "DATa" program, that can archive files on any storage format which uses the AES/ EBU protocol, such as the D-2 machines themselves. "We'll be using DAT in the future," says Evans.

Don Clark, one of the system operators said, "We're looking at magneto-optical disks, but they are still a little too expensive and too slow. But the more a client is into audio, the more they're willing to pay for the quality and flexibility of doing it all digital, and that includes the cost of storage."

MIXING AND MIDI CONTROL

The Yamaha mixers, which work entirely in the digital domain, do more than just combine the signals. Each channel has 3-band fully parametric EQ, and three internal effects buses, one of which can be patched externally. Moreover, they are entirely MIDI-controllable using ordinary note, program change, and continuous controller commands, with instant total reset and moving faders.

"Yamaha solved the automation problem, and at the same time prevented generation loss," Evans said. Even the EQ doesn't bring up any extra noise. We can now bounce sound back and forth without any signal degradation. Dubs are clones."

On the Macintosh computer is another Digidesign program, "Q-Sheet A/V," which is designed to handle MIDI-based studio automation, for controlling the mixers. The software, which also locks to MIDI time code, can work in real time, recording fader and other control moves, trimming and editing them. It also works in an edit-list mode in which individual events are entered and tied to frame numbers. Q-Sheet can also play back hard-disk files recorded with Sound Tools, so they can be shifted around in time as well from within the program. The software will even import a CMX edit list to make it easy to match audio events with visual ones.

The automation information and playback list is actually in the form of a MIDI sequence and is stored on the hard or floppy disk as a Macintosh file.

"It's really important to be able to store all the events and get to them later," said Evans. "People are always coming back to us with revision s for their projects, and we just call up the automation data from the disk."

The system also makes it easy to create and store multiple mixes for a job. Audio editor Pat Couch: "It's not an ordeal to listen to a mix two or three different ways, using different tracks on Q-Sheet. The clients have options as to what to use."

Although it's possible to control every parameter on the DMP7Ds with realtime MIDI messages, it's often easier to save a particular setup as a "scene" in the mixer's internal RAM or on RAM cartridges. These scenes can also be off-loaded into the computer using System Exclusive messages, and can then be stored as a file using Opcode's "Vision" sequencer and recalled quickly at any time.

ROOMS AND MONITORS

The creation of Dream has meant that VP&T has had to re-think its approach to room design. Edit suites now have to be acoustically constructed, because the final audio mix is being done in the same room as the video. "Now the room is the audio filter," said Feldman.

To assist in this, the Dream system includes a custom 4-channel monitor mixer designed by vice president of engineering Brad Walker. "It handles two different mixdowns at the same tīme," he said, "one for the main speakers, which the producer and video mixer hear, and one for near fields, for the audio mixer. It has 16 inputs to choose from, digital or analog, with independent front and rear mutes on each. Six

The creation of Dream has meant that VP&T has had to re-think its approach to room design.

inputs are active at one time. With the four outputs, we may be able to have it do surround sound in the future."

"We have to be able to use the four channels on D-2 however the client wants," said Feldman, "whether it's surround sound, multiple languages, or separation of the music and dialogue. The mixer has to be able to handle it."

There are two mono auxiliary buses for listening to any external processing that might be necessary (a Drawmer M500 compressor/limiter and Yamaha SPX-1000 effects processor are currently part of the system), and two onboard memories for signal routing. There's a digital tone generator, with automatic speaker muting, and in a tribute to the staff's favorite rockn-roll movie, "This is Spinal Tap," monitor level controls go to "11."

The mixer can also be configured to follow a video switcher so that it changes inputs when the source deck changes. A special mode is available for preview editing in which the mixer does not switch when the pre-read starts, thus maintaining proper audio continuity. "The technology is nothing new," said Walker, "but the user interface is."

Metering on the Dream system is handled by Dorrough digital program meters, which use a large array of colored LEDs to display progam and peak levels simultaneously, referenced both to analog and digital "zeros." In addition, there is a Klark-Teknik spectrum analyzer. Clark explained its use: "In the digital world, the frequency spectrum is very broad, and we need to be able to see all of it, even if we can't hear it. In a noisy environment like a post suite, when people are talking, it's often easier to see HF noise than to hear it."

SYNCHRONIZATION AND CONVERSION

Another important task for the mixer is to distribute word clock among the various digital components in the studio. Synchronizing the signal sources in a digital audio studio is not nearly as straightforward a task as syncing tape machines, as VP&T has learned the hard way. The AES/EBU standard has no requirement for a separate word clock signal. This means that signals coming from two different sources may be unsynchronized, and if

they are to be mixed in the digital domain, some

Full re Dreamspace: The artist's rendition.



way must be found to force them into sync.

"We need to be able to take CD input at 44.1kHz," said Brad Walker, "and other sources at 48kHz, mix them together, and varispeed them. What we need is an AES/EBU receiver, a converter that will take any sample rate up to about 53kHz which is 48kHz with a 10% error and decode it, track it, and then output it to 44.1kHz or 48 kHz, locked to something that is independent of its own input: word clock, video sync, house sync or some other audio signal.

"Sony has a unit, the DFX-2400, that does it, but it takes up two rack spaces and costs over \$10,000 per stereo input. The Yamaha converters we're using can take different signals that aren't perfectly in sync, but they must be at the exact same sample rate, and can't walk past each other."

VP&T is working with an independent company called GNP Research to develop a small, low-cost converter. Greg Basile, president of the company, explained further: "When you're using asynchronous AES/EBU generators, even if they're all using the same sample rate, you still get clicks and pops."

Walker added, "We would like to be able to lock everything to a master video sync signal, which is independent of the video tape recorder. Having the VTR as master is dangerous: If it goes into the wrong mode or powers down, it takes the whole room down."

The need for independent sync has caused some trouble for the facility when it comes to dealing with MIDI samplers. VP&T would like to use samplers as additional on-line audio sources. Q-Sheet can play back notes from a sampler, and even impose pitch change for added flexibility, and the post suites have Yamaha KX76 keyboards for "live" triggering of effects.

"We'd like to be able to fly them in from the keyboard," said Evans, "and then go back and trim them if we have to."

However, VP&T hasn't found a sampler that meets its requirements. The problem is that no sampler exists with digital outputs that accepts external word clock to drive those outputs. The principals are talking with several manufacturers to overcome this problem, and a solution may soon be at hand.

The digital audio revolution isn't just confined to CD production — the video world is going to digital, too.

THE FRONT END

One remarkable aspect of the Dream system that has emerged during its development is that the distinction between designer and user is breaking down. Thanks to the Macintosh and Hypercard, those on the front lines of audio editing at VP&T are increasingly involved in deciding how the system is going to work. The front end of the system is actually a Hypercard stack, written by assistant editor Doug Wilson, which gives immediate access to the system's various functions, including the commercial Macintosh software (Q-Sheet, Vision, Sound Designer and Digidesign's "Softsynth" for synthesizer-like manipulation of recorded sounds), and a custom program for controlling the Studer CD player, using RS-422 serial protocols.

The CD driver program includes a database for on-line keyword searching of specific sound effects. The studio uses the Sound Ideas effects library, all of which is catalogued in the database. "You type in the keyword, and one or two sub-keywords, and the program finds it in the library and tells you what disc it's on," said Dan Clark. "You load the disc, and the computer cues it up."

Wilson added, "Our search function adds skip capability to the player. It can actually do it by itself, but Studer didn't put in a panel control for it."

The Hypercard stack is constantly being upgraded, to improve both its speed and flexibility. "Soon it will be controlling Nagras, D-2 and Betacam," said Wilson. "It will be able to handle tape shuttling, and you'll even be able to edit video directly from the Mac." It also has a time code calculator, a telephone directory, and a detailed take-out menu from a local eatery.

Other software is being developed to improve on Q-Sheet's capabilities. "Q-Sheet does 95% of what we want to do," said editor John Muller, who is writing a new program in Pascal, "but we sometimes need a better mousetrap. Q-Sheet can't trigger the CD player, for example,

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or any of the video machines. We also want a timeline display of mix parameters, with line segments for fades, and more automation. I want to be able to type 'Fade 30,' and have it automatically insert a one-second fade."

With all of this input, Dream is easy for a new user to get running on. When we visited the facility, Clark had been on staff only six weeks, and by his own admission "had never run a Mac before."

"The system is really easy to understand," he said. "It quickly stopped being a gadget and started being a tool. You can get your mind on mixing and assembling, rather than on which hutton to push."

Couch, who has been on staff about 18 months, sumed up: "The more anyone learns about it the more we challenge it, and figure out what it can and can't do. We can now do things we couldn't before."

THE FUTURE

Video Post & Transfer is going in two directions simultaneously with its Dream: The company has built a facility that includes three video suites containing Dream systems and one Dream-only room; and it is exploring the possibility of marketing turnkey systems to other facilities.

"It's about half the price of anything on the market now with its capabilities," said Feldman. "We're considering marketing a secondgeneration system for about \$90,000, total, which includes the recording system and the mixer. Compared to what's out there, it's ridiculously low.

"Before, if you wanted to bring audio post

into the video suite, you'd probably have to put half a million dollars of audio equipment in when you total up the synchronizers and other kludges — but now it can be done for less than \$100,000. And that means you don't have to raise the price of the room. That is what we're doing at our new facility, and what we hope others will want to do."

With all of this input, Dream is easy for a new user to get running on.

The systems at the new facility will eventually have 24 or possibly 32 inputs. "Ganging more Yamahas is always possible," said Feldman.

The facility, about a mile from Dallas' Love Field Airport (where the company was formerly located), opened in June. The architectural renderings for the new plant were done in-house on the Macintosh, using Claris CAD software and a Houston Instruments color plotter. (Not surprisingly, the system has seen additional use in designing the front panel of the monitor mixer.)

Hand in hand with the low initial cost is the

system's modularity, which means it will be in expensive to keep up to date. "It is easy to change with the times," Evans said. "The beauty of it is that instead of being locked into one big box from a company, if Yamaha or another manufacturer comes up with a new mixer or converter, we can replace just those components."

Feldman said, "With big dedicated systems, you have to wait for the manufacturer to make the feature, and you're locked into them. The advantage of the Dream is that this is our first pass. When new equipment comes out, we will change."

Evans concluded, "If there's a new Mac program, a better CD player, a cheaper storage device, or better processors, we can add them just by plugging them in. It means I can continuously improve the quality of the system without any significant increase in the overall cost."

Video Post & Transfer may have hit on a unique solution that solves many problems at one time, and will go far toward realizing the potential for superb audio quality to match the new high-quality video formats. If the company is successful, its Dream will become reality for many facilities and clients.

Tour Business Slowdown Prompts Diversification

By David Scheirman

The previous concert touring season has been one of the "softest" in the past decade, and many live sound professionals and concert touring companies have looked to other related business endeavors to ensure a steady cashflow. With the reluctance of many concert acts and promoters to tour at the beginning of last year due to the Persian Gulf war, the industry never quite got back on its feet before the summer season was upon us ... and the year stayed soft.

1991 saw very few "blockbuster" shows such as those that drew record crowds during 1989-90 (including major tours by Paul McCartney, Phil Collins, Steve Winwood, The Who, The Rolling Stones, Madonna and such). In '91, many of the outdoor amphitheaters had unusually quiet seasons, and there were literally no nationwide stadium tours booked with the exception of some late-summer dates by badboy rockers Guns 'n' Roses, a collection of major outdoor shows by perennial favorite Jimmy Buffett, and a successful tour by the Grateful Dead.

"We decided to go ahead with our plans for stadium dates, despite the economic slowdown," said Cameron Sears, tour manager for the Grateful Dead. "It turned out to be a killer year for us. No one else was out doing the major shows, and we ended up with recordbreaking crowds in many markets." Indeed, the Dead were consistently rated as the #1 concert act draw throughout 1991 by *Amusement Business*... and many of those shows were stadium dates.

Many other major concert acts touring in arenas and other large venues were not so fortunate, and many saw less-than-capacity crowds. Even well-established acts such as Paul Simon, Diana Ross, The Judds, The Moody Blues, Tom Petty & the Heartbreakers, Aretha Franklin and Manhattan Transfer reported audience draws of 55-70% capacity for shows in some regions of the country.

The touring business slowdown saw the rise of an innovative new trend: "package tours." Creative promoters put together "packages" of different bands that would all travel together and perform on the same stage ... sometimes as few as three or four acts, sometimes as many as eight or nine. As was the case of the Lol lapalooza Tour which featured artists like Souxsie & The Banshees, Living Colour, Jane's Addiction, and IceT. One of the unexpected tour

David Scheirman is R+E+P's live sound consulting editor and president of Concert Sound Consultants, Julian, CA. successes of the season, the Lollapalooza Tour drew large crowds to major venues, giving exposure to a variety of rock acts that would not normally have been playing to such large houses. Other hastily-assembled tour packages, some more successful than others, featured oldies acts, heavy-metal rock groups and jazz artists.

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When the touring business softens up as it did in '91, what happens to companies that are used to an active concert business? Many such companies are finding that it is very important to branch out into other profit-generating directions in order to keep gear and personnel working. High on the list of alternative endeavors: permanent installations, long-term leases or sales of gear to civic venues and specialized product manufacturing for retail distribution.

A close look at some major touring companies such as Clair Bros. Audio and Maryland Sound Industries shows the year-round operation of dedicated permanent systems sales and installation divisions. The design, sales and installation of turnkey sound systems for venues such as outdoor amphitheaters, city auditoriums, theme parks, major churches and government facilities helps to keep road crews active during slower seasons, and leads to new opportunities for employees looking to broaden their skills in the area of system design, crew supervision, fabrication and such.

"Our installation department is only in its third year, but business is booming," said Gene Clair of Clair Bros. Audio. "There's always something cooking. We have staff that works on nothing but that, but our road guys can jump right in when they are not out on tour, and provide project management and expert training for our installation clients."

Some concert sound companies find that the high-tech field of doing special events for industrial clients (new product introductions, national sales meetings and conventions, and corporate retreats) offers a new market for both their specialized audio skills and their rental systems inventories. While this field (a.k.a. "corporate theatre") is often controlled by longestablished companies in New York, Chicago or Los Angeles, there are significant opportunities to break into this field for regional companies and their personnel. Oftentimes, an audio-visual consulting company or event producer seeks sound system resources in different parts of the country. For instance with a Broadway show, a production "design" is generated for the corporation's events, and then locally available crews and sound systems are sought for use in convention-area locales such as Phoenix, Las Vegas, Orlando, Honolulu, Dallas and other resort cities.

"There are very high quality expectations in this field," said Bruce Cameron, an independent sound system designer/operator based in New York. "The equipment requirements are getting more complex. I just completed a series of shows for IBM and we had top-caliber rock entertainment talent such as Huey Lewis & The News; week-long shows for the company's sales force were staged in Florida and California. These events require a blending of resources from the audio-visual and the concert audio industries."

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Some companies find that creating a speciality "niche" for their services can keep them in demand ... and in the black. For some, such as Compact Monitor Systems in Los Angeles, this may mean the design, construction and rental of stage monitor systems that take up as little truck space as possible. For others, such as Hi-Tech Audio in San Francisco and Mercury Sound Leasing in New York, mixing console rentals may be the key to notoriety. Still others, including ATM Audio in Carson, CA, are actively pursuing spin-off manufacturing arms. (ATM is the creator and manufacturer of ATM Fly-Ware, a line of rigging products for the modular speaker enclosures now supplied by a variety of speaker manfuacturers.)

Perhaps nothing offers as great an opportunity for regional sound companies as the growth of community-sponsored festivals, fairs and concert series. With many city governments and regional arts councils seeking ways to boost local culture and pride, and to create new draws for tourist trade, a wide variety of special events are becoming known around the country.

Events such as the Telluride Bluegrass Festival, New Orleans Jazz & Heritage Festival, Chicago Blues Festival and many others are drawing larger and larger crowds each year. This means that there is a requirement for more sophisticated sound reinforcement systems and the qualified crews to operate them. When these regularly scheduled events are combined with the rise in popularity of benefit-type rock concert events to aid charity and non-profit causes, a significant regional market can be seen in many cities that has little or nothing to do with concert "touring"... and yet, the sound system requirements are virtually the same.

The 1992 season should see a change for the positive, with many artists returning to the road after a few seasons of down-time (John Mellencamp, U-2 and a re-formed Journey to name but a few). And, there are new "megatours" reportedly in the works. (Genesis will tour to support their new album, maybe Madonna again, perhaps Bruce Springsteen, and there are even whispers [and denials] of Michael Jackson's show plans for '92.) However, the mega-tours belong to the megacompanies, and often such rumors don't turn into reality.

With as many as a dozen or more arena-sized concert sound systems to keep busy, the major companies are keeping a watchful eye on the entertainment industry's plans for the near future. It's significant to note that many veteran concert promoters are picking and choosing their booking commitments very, very carefully. For smaller sound companies, regional touring companies, hard-working tour individuals and everyone else in the business, wisely chosen diversification activities and a hope for the return of a booming economy are on the schedule for 1992. Sound reinforcement and event production for an industrial product release — with an English spin.

INDUSTRIAL



The \$1.8 million GM/Vauxhall product launch used an extensive audio-visual show to introduce the updated Astra to thousands of delegates from 600 dealers.



ngland's GM/Vauxhall automobile manufacturers are updating their small-car Astra series to take them deeper into the European fleet car market. To take the news to their dealers, they recently organized a series of product launches at the new International Conference Centre in Birmingham, England, inviting a total of 7,000 delegates over a period of six days.

Organizing the sound on behalf of production company Spectrum Communications Ltd, Delta Sound Inc. designed a sound system around the new Midas XL3 desk, taking full advantage of its VCA control and multiple output facilities.

The launch show used a combination of A/V and 35mm film interspersed with presenters from Vauxhall's management team — all leading up to the final product reveal.

The sound was a combination of multitrack for the A/V and product rollout sequences, stereo for the film sequence, walk-on/walk-off background music and radio mics for the presenters. In total, the system demanded 13 independent loudspeaker zone outputs. It was this element of the specification that tempted Delta's Managing Director, Paul Keating, to move away from his traditional consoles and try the Midas.

"On productions of this size we normally require a higher level of outputs compared to the number of inputs. With the 13 feeds to speaker zones we needed a console with around 16 output buses, and the Midas having 22 separate controllable outputs gave us even more control. The alternative of busing two smaller boards together would have been impractical from an operational viewpoint and without the VCA control, we would have been manually trying pull down 16 outputs in a split second.

TAPE TRACKS

The main recorded sound source for the show system came from a Fostex G16 16-track deck, which contained all of the music, effects, film sound and voice-over elements, each recorded on separate channels premixed for the separate loudspeaker chains. With time code coming off tape, the G16 became the master machine for the event, with the code controlling the A/V slide system, film, lighting and some of the mechanics.

For immediate safety backup, a Tascam 34B with a stereo mix of the soundtrack and additional time code track was run in parallel. If the Gl6 failed in the middle of a section, then they could immediately fade over to the 34B until the next presenter support section. At that point they would connect in a complete duplicate standby Gl6/34B system and continue feeding the show from a multitrack source.

This double level of backup extended to other areas. Each of Vauxhall's management presenters wore two complete Micron radio mic systems, and there were also cabled mics to hand off should both of the radios fail for some reason. With \$1.8 million invested in the launch, and 600 dealers waiting to be impressed, no-one wanted to risk any foul-up midpresentation.

The electronic music used throughout was commissioned specially for the show, and the composer had supplied various stereo submixes and sound effects sections on DAT tapes. These came without time code and were laid off onto an 8-track DAR Soundstation and then synced together to create the full soundtrack.

Tim Frost is an international free-lance journalist specializing in the audio, broadcasting and computer fields.







All of the signal sources were routed first to the main racks and then distributed from there to the others via multicores.





The speakers were flown at 22 motor hoist positions around the hall's 43 feet high ceiling, which comprises a network of steel grids with access for the riggers.



The control room housed a 40-channel console with 22 separate controllable outputs, along with a ½-inch 16-track, which contained all of the music, effects, film sound and voice-over elements, each recorded on separate channels and pre-mixed for each of the loudspeaker chains.

During the four days of remixing at the London-based Soundtracks studio, Delta used a speaker layout that mimicked the final system as much as it was possible in the studio environment. Dedicating one tape track per speaker channel, they created a final mix where the bulk of the fades and effects were already placed on tape. This meant that very little mixing and additional effects work needed to be done during the show itself, simplifying the operator's job considerably.

The live effects were primarily small amounts of stereo echo and reverb added to the side and rear effects channels to give more depth to one or two of the sequences, while firmly maintaining the focus on the picture rather than distracting delegates with noises coming from above or behind them. Keating had considered using MIDI control on the Yamaha outboard gear to automate this function, but felt that MIDI presented some potential software problems when set-up time was at a premium. So the occasional effects were added manually using the effects return channels routed to the relevant speaker positions.

The 13 main and effects tracks were recorded from the 8-track Soundstation, back onto tape in two passes, using the time code track to ensure that they stayed precisely in sync.

The G16 tape contained all of the recorded sound for the show. including the soundtrack for the 35mm film, with each channel being fed to an input on the Midas.

In addition to the 16 tape inputs, the XL-3 handled 6 channels of microphones, 9 channels dedicated to back up systems, and assorted returns and stereo inputs, using up all but a couple of the XL3's 40 channels.

CONTROLLING FACTORS

On the console, mics, FX, music, radios and backups were each allocated to an individual VCA. The master VCA pair was set up to control the overall output of the desk and used for master fading functions.

The ICC's Hall Three, the venue for the launch, is some 60m deep and 40m wide, and to increase the impact of the product rollout, the main seating area was surrounded by false walls, which with the stage surrounds, lifted away revealing the full range of cars placed around and behind the stage area. This meant that a lot of the hall's depth was lost for the main presentation, with the seating area becoming 23m wide and only 20m deep, and resulting in presenters being only 3m away from the front row of seats.

With virtually no depth to play with, the biggest problem facing Delta was to design a sound system that offered adequate coverage and level without running into feedback problems. This was further complicated by the fact that the live presentation elements were given by Vauxhall executives not speaking from a fixed position but moving around the stage area.

Normally for a presentation, Delta would install a split system — a screen position for music, and a separate vocal system for the radio mics. But without the depth of stage and seating area, they had to use the main A/V system for the radio mics as well. With loudspeakers placed 5 meters upstage of the main presenting area, level, especially when using the omnidirectional radio mic capsules, became Keating's most critical concern.

"When you are dealing with non-professional presenters you need to have in hand as much headroom in the system as possible, and the headroom situation was certainly not helped by the upstage presenter positioning. To resolve this problem we controlled the coverage by making sure that no one loudspeaker system was used to cover more than a 10m depth of seating area."

SPEAKER SYSTEMS

The twin main speaker systems used Nexo units, each with a pair of SI2000s for stage left/right and a pair of the smaller MSICs for stage center. The systems covering the rear half of the seating were mounted high at the front of the stage, but the system covering the front half was the one set 5m further back, deep within the set and well behind the presenters.

To assist the clarity of the speech elements, Delta put four additional Martin Audio CX units half way down the auditorium, and these were fed from the mic channels and the voice-over tracks of the A/V and film sections.

These, like the main speakers, were fed via delay lines, a mix of Klark-Teknik, Yamaha and BSS units. Delays of between 16mS and 42mS were applied to focus the speech elements to a main reference point downstage center. The only advantage of having the loudspeaker systems behind the presenters was that Delta didn't have to bother with any form of foldback, as the presenters could hear perfectly well from the main system hanging behind them.

The rest of the loudspeaker layout was primarily concerned with atmospherics and effects, with additional SI 2000s on the left and right exhibition hall walls. Nexo MSIVs were hung at the rear and along the sides of the false auditorium walls, and a total of eight Martin Audio BSX sub-bass units placed under the stage and strapped under the seating.

Every speaker channel had its own ¹/₃octave equalizer tuned using a mix of ears and a dbx RTA-1 analyzer. On the main speakers, the EQs were set relatively flat, with no boost, and cuts of up to 3dB at the most. Firing through the heavy cloth of the false walls, the side speakers needed a certain amount of HF

TECHNICAL DETAILS

Tape Track Allocations for Fostex G16

- 1 Front left
- 2 Front right
- 3 Front center
- 4 Left side
- 5 Right side
- 6 Rear left
- 7 Rear center
- 8 Rear right
- 9 Exhibition left
- 10 Exhibition right
- 11 Sub FX/Drones
- 12 Voice-over cine
- 13 Mixed cine
- 14 Blank
- 15 Backup SMPTE time code
- 16 Master SMPTE time code

EQUIPMENT LIST

Mixers

- 1 Midas 40 channel XL3
- 1 Soundcraft 8000 16/8/8

Tape Machines

- 2 Fostex G16 ¹/2-inch 16-track
- 2 Tascam 34B ¹/4-inch 4-track
- Tascam 112R Cassette 1
- Tascam 401 CD 1
- 1 Sony FS55 DAT

Processing

- 3 C Audio 312R stereo graphic
- 3 Klark-Teknik DN300 graphic eq
- 4 Yamaha 2031 EO
- 2 Klark-Teknik DN410 parametric EO
- BQ3 parametric EQ 1
- BSS TCS 804 Delay 1 1
- Yamaha YDD2600 Delay
- 2 Klark-Teknik DN716 1 SCV Pro Source Director
- Yamaha SPX50D 1
- 1 Yamaha SPX 90 II
- 1
- Yamaha SPX900
- 4 Dolby SR noise reduction cards 2 dbx 150X noise reduction cards
- 1 AKG TDU 7000 Delay
- 1 dbx RTA analyser

Microphones

- 6 Micron diversity radio mics
- 6 Sennheiser MKE2 capsules

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sensitive, the positior-

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achieved by bonding a special

thermoplastic to a quality poly-

fluted surface is complemented

propylene base. The velvety black,

ing highly positive.

Toucl™ knob

- 2 Sennheiser MKH40
- 2 Beyer M400

Speakers

- 8 Nexo SI2000
- 4 Nexo MSIC
- 7 Nexo MSIV
- 4 Nexo PC212

- 4 Nexo PC sub
- 6 Martin CX2
- 8 Martin BSX
- 4 Fostex 6301 self powered

Amps/Speaker processors

- 12 Yamaha PC2602M
- 6 C Audio SR808
- 3 C Audio SR606
- 2 C Audio SR404 4 C Audio TR850
- 11 Nexo processors
- 2 Martin EX2 processors

Multicores

THE VELVET TOUCH.

- 1 100m 9 pair Belden
- 4 100m 19 pair Belden
- 6 50m 6 core speaker
- 4 Nexo SI2000 racks



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lift, but even this was limited as these channels carried effects only, and intelligibility and accurate tonal balance was not a major issue.

The RTA-1 was one of Paul Keating's mostused tools, being employed for a lot more than simple EQ measurements.

"We use the RTA to analyze the response of each of the speaker clusters from a central point. The RRC room response facility allows us to compare the input to the speaker against the output in the room, which is helpful. But we also use the dbx system to check mic capsules for consistency and for tape machine line-up."

Hard disk system reliability is now good enough for live work and the cost is coming down.

Keating was also very happy with the design, access and facilities of the new hall. Set-up was over a period of three days, with the first day allocated to system installation, the second day for control and the third for system testing and alignment. In all, 22 motor hoist positions were used to fly the speakers. Although 13m high, the hall's ceiling has been designed as a network of steel grids with access for the riggers to get almost anywhere in the roof area. Unlike some venues with a restricted number of load-bearing points, this grid allowed Delta to fly speakers wherever they needed.

The only omission in the hall's facilities is a service lift to take hardware to the balcony operations position. The Midas got there by 'Mandraulics,' Delta's term for the crew members taking a corner each and manhandling the console up into position.

Cable trunking is also well catered for at the ICC. The main racks driving the Nexos were at the rear of the hall, 20m to the side of the mixing position, with additional Yamaha-based amp racks back-stage and by the projection gantry.

The main racks contained the Nexo electronic speaker system controllers running the S12000s bi-amped from C-Audio SR series amplifiers, and all of the signal sources were routed first to these racks and then distributed from there to the others via satellite multicores.

Instead of using a separate mixer output channel for the sub-bass, the main outputs were routed first to an SCV Source Director which passed the signal unaltered onto the main system and central clusters, whilst generating a separate low pass filtered mono output for the Martin BSXs.

SHOWTIME

On performance day, everything went smoothly. Back-up was required only once when the synchronizer controlling the 35mm projector failed, unlocking it from soundtrack coming from the G16. For this eventuality Delta had an audio follower in the projection gantry with a duplicate of the soundtrack recorded using Dolby SR, so when they switched over to it, there was no quality loss.

Having used the DAR Soundstation for preparing the soundtracks Paul Keating is now developing the idea of using a 16-track hard disk system for playback on-site at a similarsized event next year. Using a Soundstation for preparation and on-site would mean that different work practices can be brought into play. The show could be constructed from the outset so that sections would be easily updated at the last minute, and balances, that are going to be venue dependent, can be quickly tuned on-site.

Hard disk system reliability is now good enough for live work and the cost is coming down. But the flexibility to cut and paste, and adjust fade times and track allocation on-site, Paul believes is going to have to be explained to the producers who will initially just see it costing ten times more than a tape machine.

"Within the conference industry you are up against time. You don't have theater schedules with a week of previews; you have a couple of technicals, a couple of rehearsals and then, in this case, straight into a £1 million (\$1.8 million) show which makes or breaks in 45 minutes.

As well as the sound quality advantage, a hard disk system will allow the producer flexibility around a carefully pre-programmed production. They will have much more power to fine tune their productions within the limited time they have on-site."



Circle (29) on Rapid Facts Card

H<mark>ands on</mark>:

By Mike Oxlong

ebate continues over the viability of music "workstation" platforms for professional and private use. Key to this debate is the price vs. performance feature continuum. Second-generation music workstations are in use, and third-generation units are close.

At issue is the comparative hassle of connecting discrete components to assemble a personal studio. Multiple keyboard controllers, FM or alternative sound generating modules, sampling keyboards and/or modules, a drum machine, outboard effects, plus a MIDI mixing console, and an external sequencer or PC computer are required. Add interface software and hardware, and this larger-than-a-roadcase proposition gets pricy. Technical expertise is mandatory.

As alternatives to integrated MIDI studios, workstations bundle components and functions into a stand-alone professional platform. Less cash (often under \$5K) is needed. Learning curves (one box, one manual, one nation under God) are shorter. And rather than honing your retail purchasing/deal-making skills, more time is spent doing real work.

THE DPM 3 KEYBOARD FAMILY

While getting past the Peavey M.I. image might be hard for some, most professionals are now aware of the user-driven innovation springing from Peavey. Cornerstone to Peavey's first all-in-one workstation is the DPM 3 Composition Center. It looks like any other live performance MIDI synthesizer. The DPM's software-based architecture engages three powerful Motorola 56000 series chips and is aimed at recording applications and artistoriented MIDI production studios.

Peavey has merged the sound production capabilities of several analog, digital and sampling sound modules to an onboard 9-track, 20,000note sequencer. Also included is 24-bit processing from four internally controllable "effects racks." As with many MIDI mixers on the market, track levels can be controlled via the sequencer's sends and automated amplitude controls.

Sampling rates of the DPM 3 deluxe version (known as the DPM 3 SE) top out at 48kHz/16 bits. DPM 3 SE sample input is accessed via its 1-rack-high companion: the DPM SX. A professional front-end A/D converter, the DPM SX features both SCSI and MIDI Sample Dump Standard (SDS) interfaces. These protocols allow SX samples to be shared with third-party sampling technology.

Once a sample has been recorded by the SX module and routed by MIDI cables to the 3 SE, sounds can be edited in much the same manner as "resident" DPM 3 SE ROM sounds. Digital effects processing and automated mixing of sound files (once they're recorded to sequence) complete the package.

PEAVEY

The DPM 3's comprehensive manual, written by MIDI guru Craig Anderton, details the potential of these machines. The front-panel control surface is separated into five primary sections: system controls, voice editing, sequencing, data control and performance control.

SYSTEM CONTROLS

In the upper left-hand chassis corner, six push-buttons access 100 on-board user programmable (RAM) sound files. Sounds are matrixed into 10 banks of 10 files each. Individual sound files can be constructed of onboard factory ROM waveforms, sound files from sample libraries, including Prosonus (stored in RAM), or user samples loaded from the SX (which are also stored in 3 SE RAM).

Whereas most commonly used factory and third-party sound files offer multiple sources, the DPM's production value is enhanced by access to user samples.

The DPM is defined by the concept of sound file. Single sound files may consist of a variety of basic sounds. 'Basics' include two soundgenerating oscillators that draw from ROM and/or RAM sound files (i.e. factory vs. usersupplied samples/libraries).

Each sound file can then be "combined" with up to three other sound files (each with another two oscillators), taking advantage of up to eight separate oscillators simultaneously. DPM 3 SE Composition Center,

Finally, multiple user samples can be mapped across an entire keyboard (i.e. an entire drum kit can be mapped with an entire orchestra of samples under one keyboard without patch changes.) Memory capacity is the only limitation.

DPM 3 SE

SPECS AND DESCRIPTION

Manu-	
facturer:	Peavey Electronics
	Corporation
Contact:	Doc Adkins
	711 A Street
	Meridian, MS 39302-2898
	601-483-5365
	Fax: 601-484-4278
Model(s):	DPM 3 SE
12/27	Synthesizer keyboard
	\$2,999
	DPM SX
	6-bit sampling expander
	\$349.99
	Mega Sample RAM board
	Increases sample RAM to
	IMbyte
	\$159.99
	DPM SP
	16-bit sample playback
	module
	\$999.99

Mike Oxlong is an independent MIDI system consultant and free-lance writer.



The Peavey DPMSX sampler/expander.

SPEAKING OF MEMORY

DPM 3 SE factory units are equipped with 64K of user RAM. Installation of the optional Mega Sample RAM Board upgrades the original 64K of static "user-definable" samples to professional standards compatible (read: longer sampling time). With 100 ROM- or RAM-based sounds, a Mega Sample RAM Board provides the user 13.6 seconds of 16-bit samples at 38.4kHz. Internal playback speed of the 3 SE is 38.4kHz.

Two varieties of memory upgrades are offered. Users can install eight, 28-pin, 32K chip sets or eight, 32-pin 128K chip sets, yielding a total of either 256kbytes or 1Mbyte of static RAM. Internal programming structure (synthesis architecture) and RAM allotment is further enhanced by upgradeable factory revisions of software versions. The current software revision is Rev. 2.3.

The 3 se offers other storage media via an on-board 3.5-inch double-sided, 720kbyte disk (which can be formatted by the DPM or by any MS-DOS or Atari computer). More storage happens when a 32K ROM card is inserted into the rear panel memory cartridge slot which doubles 3 SE onboard programs to 200. (Storage is also available via porting MIDI data to an external hard disk drive.) The 3 SE's SysEx dump feature permits upload and download of file libraries in seconds.

VOICE EDITING

Many users desire to access and edit raw sample data or the DPM SE's resident ROM "waves" according to needs. The 3 SE does this through its voice editing section.

The 3 SE can store up to 48 raw samples and 32 multisamples in sample RAM. A total of 105 internal ROM waves are the building blocks of the synthesis process.

Voice editing is achieved through a synthesizer-like control matrix. Waves and samples can be modified through two oscillators (one wave each), two digitally controlled amplifiers (one per oscillator), a low-pass filter, four envelope generators, two low-frequency oscillators, an output amplifier stage and an output effects router (similar to a console's effects send). Waves and samples can also be combined using a "multi" option to layer sounds via keyboard ranges (splits), velocity and pressure.

SEQUENCING

Finished sound files next enter the DPM's sequencer. This digital MIDI recorder functions similarly to an external, computer-based sequencer. Depending on the application, it makes it possible to stay inside the machine until the final 2-mix. The sequencer can use five discrete drum kits, a 9-track recorder with automated mixing and level controls, and a digital onboard signal processing module whereby four effects can simultaneously process sounds.

Sequencing track commands emulate multitrack recorders. Such operations as recordready mode, track selection, rewind, fastforward, pause, stop and play transport controls are straightforward.

DATA CONTROL

Sound generation by the DPM 3 SE is managed by a simple soft-button user interface, front-panel accessed. Sound file menus are manipulated by the 12-button "voice edit" section located front and center above the keyboard. Menus can be viewed and edited by using a combination of "+" and "-" touch-buttons to select page menus increments or to fine-tune parameter values. Coarse parameter value adjustments are obtained by using either a linear slider or data wheel.

Peavey's 3-edition Composition Series library provides more than 100 rhythm patterns.

Interface layout is comfortably designed for sampling, editing and recording functions. The five sections are color-coded for easy identification. A 40×2 back-lit LCD serves as the main character display and features angle-viewing controls.

PERFORMANCE CONTROL

The 5-octave, velocity and pressure sensitive synth-action keyboard also edits sound files. Two mod wheels (one dedicated for pitch bending and the second assignable to any modulation source) enhance performance control. Footswitches and other MIDI controllers can be assigned via a back panel ¹/4-inch jack and MIDI receptacles. A headphone jack has an independent level control. Two ¹/4-inch jacks provide mono and/or stereo left/right unbalanced outputs.

MIDI IMPLEMENTATION

The DPM's MIDI capabilities define control of the unit's three MIDI ports by assigning MIDI transmit and receive channels. MIDI omni-, poly- and multi-control modes, along with complete or partial parameter filtering, are present. Four discrete multimode presets enable an external sequencer to record and drive as many as 16 separate sound files at once.

EDITING FX, DRUM MACHINES AND SAMPLING

DPM's master mode is used when editing and mapping the onboard drum kits, FX processors, user-tuning, external MIDI controllers (such as footpedal and breath controllers) and the sample RAM section. Drums, effects, tuning and controller selection are also edited to onboard sound files. The 3 SE differs from most music workstations in the sample RAM section. Here, users download samples from the SX sampling unit for more elaborate editing.

With the DPM 3 SE connected to a DPM SX (or connected to another SDS sampler) by MIDI cables at both in and out ports, the 3 SE can receive, load and save user samples to diskette and transmit compatible 16-bit samples to other DPMs or SDS units. SX specifications include full 16-bit delta-sigma A/D conversion, selectable sampling rates (16kHz, 24kHz, 32kHz, 38.4kHz, 44.1kHz or 48kHz, and line/level or microphone [with phantom] inputs).

The 3 SE is optimized for the SX. Beyond traditional trim and looping control, editing options include sampling frequency adjustments, threshold level, digital word length, sample record arm/start and sample dump which are all available at the front panel of the 3 SE.

The sample transfer rate is slow. Of course, the normal sample data transmission speed over any MIDI bus is hopelessly sluggish. Anyone working in time-is-money situations should save all samples to disc for immediate access.

OPTIONS, OPTIONS EVERYWHERE ...

In addition to the 3 SE's Mega RAM Sampling Board, the SX sampling front end expands up to 16Mbytes by using standard Macintosh 80ns or 100ns (SIMM) chips. Because the DPM 3 SE, as many other workstations, can only use of a maximum 1Mbyte, the SX offers an inexpensive, professional alternative to costly singlechannel A/D converters.

For engineers or producers new to sound design, and/or pressed for time, an 8-edition library of factory supplied waves and samples enables user access to 100 unique sound files per edition. The library is read by the DPM's onboard 3.5-inch disk drive or cache card port.

Peavey's 3-edition Composition Series library provides more than 100 rhythm patterns, which will be helpful to novice sequencer users and those needing generic rhythm tracks. Experienced users, seeking to construct a strictly sample-based library, will fancy Prosonus' 8edition Peavey set (SPARS Code: DDD). These fully digital samples were culled from Prosonus' stellar compact disc library and range from traditional orchestral samples to sound effects.

Although basic samples from Prosonus hover around or below 64kbytes, heavy users will opt for the full 1Mbyte option. Immediate access to Prosonus' sound files seems mandatory to anyone who's in it for the money.

For sample editing, the recommendation from Peavey designers is Turtle Beach System's



The Peavey DPMV3 Synthesizer Voice Module.

SampleVision 2.0. SampleVision overlaps the DPM 3 SE's onboard user sample editing screens. Based on intuitive graphical user interface (GUI), SampleVision offers superior visual editing, digital signal processing, and has drivers for various sampling keyboard devices. Both the DPM 3 SE and DPM SX models are supported by Turtle Beach's adherence to the MIDI SDS format.

For those not needing a 3 SE, Peavey's SX offers a logical alternative to expensive A/D converters. SampleVision and the SX together create a cost-effective digital sampling editor. If you've already dedicated a sampler for playback purposes, the SX becomes your recorder while SampleVision links the front end and existing equipment.

SPOT PRODUCTION AND MUSIC COMPOSITION

To really find out about the performance of the Peavey, we took the unit out for a real-life spin. In a typical session in a notable Chicago studio, The DPM worked well, as advertised. Background vocals and spoken trailer tracks were recorded to 2-track analog before sampling and assembly sessions. Once committed to tape, vocal tracks were then re-recorded by the SX sampler. Each vocal sound bite was transferred individually to the DPM 3 SE for digital editing (trimming, looping, balance and deletion of extraneous noises). This sent the talent home early and brought talent fees in under budget.

Once inside the 3 SE, samples were assigned to particular sound files. Because each sound file can be modified as if it were a non-sampled waveform (i.e. a musical instrument) the engineer could then modify the amplitude time and levels of certain background vocals to better suit the producer's requirements. Filtering of the spoken trailer by the engineer eliminated brittleness of the narration. Panning each vocal sound file into a discrete position broadened the streeo sound field.

For creation and sequencing the musical portion to the vocal samples, the producer chose to track the electronic keyboard and drum directly from the DPM 3 SE's internal sound files to the internal sequencer. Keyboard talent was ready to assist in the creation of a simple drum groove (using two of the five 3 SE onboard drum/percussion machines) and the synchronization of slap bass and some lush keyboard pads.

A decision to fly previously loaded vocal samples over the musical sequence was made because of time constraints placed on keyboard talent. Rather than record separate vocal track(s) in the internal sequencer along side of the rhythm section sequence, the keyboardist "performed" the vocal sound files live directly to the 2-track mix. Because the DPM 3 SE allows sound file selection for performance, along with a sequenced pattern or song, this task was simple.

With multiple parts (combining musical sequence and vocals) needing to be mixed directly to DAT, a combination sound file represented all of the sampled vocal parts under one master soundfile (combination mode). Thus, all four vocal sound files appeared under the entire keyboard and eliminated any need to stop tape and switch sound-file memory locations.

TO BUY OR NOT TO BUY?

Eventually, music/production worskstations will be as indispensable as the notebook-sized

PCs now flooding the market. At press time, however, can't-live-without-it users are individuals who must remain mobile. Fixed studios most need workstation technology for second rooms and project development.

Portable keyboard sales are off nearly 20% for 1991, and several established competitors offer workstations. It's a buyer's market. With this in mind, it is fair to say that the DPM compares favorably with second-generation competitive units. Depending on the application and budget, the DPM 3 SE may match your needs, and studio.

Peavey's recent track record of intensive R&D and commitment to customers bodes well for potential users. The company is equipped to go the long haul.

Circle (100) on Rapid Facts Card



First Look

Things with Time Code

By Laurel Cash-Jones and Fred Jones

his is our favorite kind of product: one that solves a problem that arises from the omission of a *necessary* feature by one manufacturer and is built by another manufacturer to help out all of the people who needed it in the first place. It is toward this happy solution that Cipher Digital introduces the CDI-825. Before we delve into this product, let us consider the other product that it works in conjunction with. (Bear with us, we'll straighten it out in a moment).

In case you are unaware, Sony has been trying to introduce Hi-Band 8mm to the professional, industrial and consumer market for a few years. This format has several advantages over VHS, not the least of which is the fact that they have included a time code track plus stereo digital audio tracks on the professional versions.

However, don't get to excited about the time code track. Unfortunately for us, there is no way to get to the time code other than using the serial interface that is meant to work with a video editing system.

By the by, the digital audio tracks are currently 8-bit with a 32kHz sampling rate, but a new version (at least for consumers) is expected early next year with 16-bit and either 44.1kHz or 48kHz sampling rates.

If you're reading this at Sony Pro, a Hi-Band 8mm deck with time code in and out, and high quality audio would do *very* well as a professional alternative for home/MIDI studios to use instead of a ³/4-inch or VHS deck. If you're reading this in the consumer division, we would also like to have one without the time code track for our system at home. Just a hint.

Back to reality. The Sony model EVO-9800 has time code (sorry, out only) available on it's 9-pin RS-422 control port. Those clever folks at Cipher Digital decided to make a serial-tolongitudinal time code device so that those of us who do not own a video editing system can take advantage of this feature.

The CDI-825 operates in either of two modes: stand-alone or monitor mode. In the standalone mode, the 825 requests the time code information from the deck each frame and outputs it in LTC form. In the monitor mode, the 825 is inserted between the deck and a controlling device, such as a video editing system. In this mode the 825 does not request any data, it spies on the communication between the controlling device and the deck (not unlike the CIA). When the editor requests the time code data (typically each frame) the 825 also receives it and outputs the data as LTC, without affecting any other function of the editor, thus becoming transparent to the editor. The CDI-825 also outputs LTC at any speed, and compensates for "on-time" displays in time code readers by subtracting a frame from the code, thus enabling the reader to display the correct time.

Circle (101) on Rapid Facts Card



MORE TIME CODE THINGS

From Fostex comes the portable DAT deck we have all been waiting for. If you are into field recording for television or film and need to use time code, this is the 4-head DAT machine you *must* have. It's called the Fostex PD-2, and from its built-in monitor speaker and slate mic, to its extremely ergonomic design, we were unable to find any field-usable feature that was not included. Indeed, it has so many features, we are sure we are going to leave something out, but rest assured that Fostex did not.



Since we don't know where to start, let's take a look at the display. The large LCD shows all the usual things such as DAT absolute time and SMPTE/EBU time code, but it also shows you the pre-selected operational settings, such as frame and sampling rate, and sync source.

As far as getting in and out of the PD-2, in addition to all of the normal analog and digital audio connections, it has BNC-style input and output connectors for both video and external sync in with video composite, 25, 29.97, 30fps frame, 24, 25, 29.97, 30fps field, 48, 50, 59.94, 60 field/second rates standard. Digital word sync in accepts three sampling frequencies (44.1, 48, and 44.056kHz). It will also record in each of those frequencies. Of course, an internal time code generator that can generate code or jam sync to any external source or regenerate actual camera time code is included.

On the input stages mic/line switching with a 30dB pad and phantom powering of 48V and T12-type voltage is standard, and a 3-position steep bass cut filter is integral, with settings at 40Hz, 80Hz, and 400Hz, plus Channel 2 has a phase reverse switch for stereo coherence.

The indexing and slating features are also unique. Take numbers can be entered manually or automatically; You can slate via the builtin mic or use just the tone generator. You can also mark the PCM errors (they are recorded as PNOs from 700), input overload or time code dropout errors. Errors can be marked so that you can search by them, or you can search by time, or index points) The PD-2 can also be used in an edit suite using the RS-422 9-pin port to talk directly to most existing editing systems. We could go on, but space does not permit.

Circle (102) on Rapid Facts Card

Laurel Cash-Jones is R+E+P's editorial consultant and a Los Angeles-based free-lance writer. Fred Jones is an audio industry observer and a Los Angeles-based free-lance writer.

Cutting Edge

RANE LINE TRANSFORMER

Rane is now offering the FLT 22 Line Transformer module, which provides two lowdistortion, wide-bandwidth nickel core audio transformers to convert unbalanced line-level signals to isolated balanced outputs. The FLT 22 is housed in a Flex Series HR format chassis and can be expanded to four channels via the Option 44 expander board. FLT 22 input/output connectors are terminal strip and the unit may be rack-mounted horizontally, vertically or left as a stand-alone shelf-top unit. Where space and/or budget is tight, the Option 44 card may be used as-is, without the HR chassis — the terminal strip is mounted right on the PC card.

Circle (106) on Rapid Facts Card

ARIEL DIGITAL AUDIO INTERFACE

Ariel Corporation has debuted the ProPort Model 656, a self-contained digital audio interface that brings recording studio quality analog audio to ISA/EISA, Sun, VMEbus, Macintosh, Hewlett-Packard, and NeXT computers via the DSP Port serial interface.

ProPort provides two channels of 20-bit, 8times-oversampled digital-to-analog conversion with a selectable sample rate from 5kHz to 96kHz, for any signal I/O including speech processing, laboratory data acquisition, signal generation and speech research.



resulting in superb imaging and greatly reduced phase distortion.

The curved surface of the injection-molded ABS baffle serves to direct possible reflection of the shorter wavelengths away from the listening position and reduce baffle diffraction distortion. The ducted port was moved to the rear of the enclosure to further reduce distortion. Vertical alignment of the transducers across the baffle center produces natural mirror-imaging.

Circle (105) on Rapid Facts Card





ProPort employs 16-bit oversampling technology for the input converters, electronically balanced microphone and line-level inputs, switchable phantom microphone power, peak reading level indicators, continuously adjustable gain controls with 60dB range and overvoltage/surge protection. Analog outputs are driven by active-balanced, low-impedance line level amplifiers.

Circle (107) on Rapid Facts Card

LEXICON OPUS UPGRADE

Lexicon is now shipping OPUS Version 3.0 featuring extensive enhancements to the OPUS digital production system. System enhancements affect both hardware and software, and make OPUS the industry's only fully automated digital mixing console, as well as the only system to integrate every major audio postproduction function.

Circle (122) on Rapid Facts Card

STEWART ELECTRONICS POWER AMP

a radical new outward appearance in the quest

for superior console-top monitoring. The 4206

features a 6.5-inch woofer, and the 4208 an 8-

inch woofer and were designed specifically for

use in the near field. The Multi-Radial sculp-

tured baffle directs axial output of the individu-

al components for optimum summing at the

most common console-top listening distance,

approx. 1 to 11/2 meter (3 to 5 feet). The Multi-

Radial baffle also positions the transducers to

achieve alignment of their acoustic centers

Stewart Electronics has added the model PA-800 power amplifier to their amplifier arsenal. A single rack-space unit, the PA-800 employs the same technology as the larger PA-1200. The PA-800's high efficiency "Switch Mode Power Supply" delivers significant increases in the amp's efficiency, and simultaneously allows most of the amplifier's major components to be downsized, saving space, weight and cost. Typically, six PA-800's can co-exist on a standard 20-amp circuit. Each PA-800 delivers 400W per channel into a 2 Ω load.

Circle (115) on Rapid Facts Card

SOUNDCRAFT IN-LINE CONSOLE

The Sapphyre is Soundcraft Electronics' newest recording console. Designed for the mid-level recording market, Sapphyre's performance specifications and design make it ideal for recording and post production applications. The console is available in 20-to 44-input versions with or without integral patchbay.

Each I/O module incorporates individual noise gates with an advanced 4-band EQ design, splittable between the two signal paths. The dual line input option enables increased input capability for effects returns or virtual tracks. A combined I/O module gives access to dual signal paths, one for monitoring and the other for recording, with sub-grouping and routing architecture enabling different modes to be easily configured and controlled.

Sapphyre boasts eight sub-group buses that can be used as virtual patchcords, allowing signals to be re-routed to any tape channel directly, or to an input, and then sub-grouped to tape. Frame sizes are 20, 28, 36, or 44 I/O modules, each with six stereo effect returns.



Circle (109) on Rapid Facts Card

Cutting Edge

TECHRON INTERFACE

Equipped with the same powers as the TEF 20, Techron's TEF 20HI includes a high-speed interface that allows the device to be used as a digital audio recorder. This same HI interface also permits third-party software programs to utilize the TEF 20 DSP chip to control the entire system. The TEF 20HI's high-speed interface provides the user with a wide variety of tools and functions not usually associated with TEF, such as the ability to become a top-notch digital oscilloscope, real-time analyzer, filter generating unit, and data capture and display/analysis device.

Circle (110) on Rapid Facts Card



API CONSOLE

Audio Products introduced the fourth in a new series of discrete consoles during the 1991 New York AES Show. Configured with up to 128 inputs and 49 buses, this console's new customized features include an automated send module capable of assigning individual sends (available in configurations of 8, 10, or 12 per module) to pre or post on either the channel's large or small fader.

Sends can be individually muted through the GML Automation Environment. The console's "Touch Reset" control creates resettable switch settings on the entire console by the main computer.

Circle (108) on Rapid Facts Card

TIMELINE CONTROL

The TimeLine Console Control Unit (CCU)is a miniature keypad that mounts directly into standard Neve, SSL, Euphonix, and other consoles. The CCU operates the TimeLine System Supervisor multiple machine controller which interfaces to standard console automation software with no changes or updates required. Utilizing Lynx Time Code Modules, the CCU controls up to six analog or digital audio tape recorders, VTRs or sprocketed film transports. All data communication is processed by the TimeLine system; all machines are operated directly by the console automation.

Through the CCU any transport may be designed as the master without switching cables. The CCU system offers variable speed control of the master for pitch changes of an entire synchronized machine group. An optional jog/shuttle wheel is available.

Circle (112) on Rapid Facts Card

CREATIONS TECHNOLOGIES MIDI DRIVER

Anatek's MIDIMatch line driver is designed for people needing to control MIDI lighting systems, sequencers or computers up to 4,000 feet away. MIDIMatach data can be sent through any two conductor-shielded audio cable, and the signals can be treated like an audio signal and sent through a patchbay with no signal loss or MIDI delay.

Each unit has two completely independent transmit and receive circuits that can be used to send and receive two different data streams. MIDIMatch signals are immune to noise and will not interfere with any other signal carried on adjacent lines. The system consists of two identical bi-directional units; one for each end of the cable, and two 9V power supplies.

Circle (125) on Rapid Facts Card

MARK OF THE UNICORN TIME PIECE

Mark of the Unicorn's new IBM PC/compatible version of the MIDI Time Piece is a multicable MIDI/SMPTE interface for computerbased music production systems. MIDI Time Piece features eight independent MIDI Input/Output cables. Each cable has 16 MIDI channels for 128 MIDI channels per unit. MIDI Time Piece also has complete MIDI merging, routing, channelizing and event muting capabilities. The device functions as a standalone merger/mapper when the computer is turned off. An advanced MIDI/SMPTE reader/generator/converter for tape synchronization completes this three-in-one device.

This IBM version features an 8-bit interface card that allows connection of up to two MIDI Time Pieces to the computer for a MIDI network with 256 channel and 16 MIDI inputs and outputs. Two separate computers can share the same MIDI Time Piece network.

Circle (116) on Rapid Facts Card

SONIC SOLUTIONS-PREMASTER CD

Sonic Solutions has unleashed advanced technology for complete tapeles, mastering of audio CD and CD-ROM. Sonic Solutions' PreMaster CD offers economy, reliability and the ability to quickly check the master before it goes to the plant.

With the Sonic System and a CD Maker or CD Printer, a record company or mastering studio can record a finished program onto a PreMaster CD (PMCD). In addition to the audio program and a table of contents for standard CD player, the PMCD contains information required by the code cutter such as precise timing information for all track starts and indexes, plus data relating to copy prohibit, emphasis, ISRC code, etc. At the CD plant, the MasterMaker, a Sonic-System/CD Maker combination outfitted with special software, reads back and relays data to the code cutter which in turn writes the glass master.

Circle (113) on Rapid Facts Card

ELECTRO-VOICE ACTIVE CROSSOVER

Electro-Voice has introduced the EX-24, a stereo 2-way, mono 3-way crossover designed to maximize biamped and triamped system performance.

The 1-rack unit EX-24 offers 12 selectable crossover frequencies per channel. Frequency settings are 80Hz, 125Hz, 160Hz, 250Hz, 500Hz, 630Hz, 800Hz, 1250Hz, 1600Hz, 2500Hz, 500Hz, and 6300Hz.

1/Os are¹/4-inch balanced or unbalanced and XLR balanced. Discrete channel controls over

low and high output levels, polarity switches and on/off output switches serve to ease system setup.

A switchable horn equalization circuit is present to flatten system response when using constant-directivity horns. Infrasonic filtering (-3dB at 30Hz) and stereo or a mono subwoofer option enhance the low pass circuitry. The internal power supply may be configured for global applications. Included is an IEC connecter with detachable ac line cord.

Circle (114) on Rapid Facts Card



AUDIO TECHNICA STEREO MICROPHONE

Audio Technica has introduced the AT822 One Point X/Y stereo condenser microphone designed specifically for DAT and high performance cassette recording. Mono compatible, the AT 822 is also intended for television, FM and field recordings. Inside of the AT 822 are a pair of wide-ranged and close-matched cardioid condenser elements delivering natural response across an arc of 170°



The high output stereo AT833 terminates its standard cord with two mini plugs threaded inside a pair of ¹/4-inch phone plug adapters. The AT822 operates on a standard 1.5V AA battery and includes a switchable low-cut filter, windscreen and camera shoe mount adapter. **Circle (135) on Rapid Facts Card**

APHEX 9901 PARAMETRIC EQ MODULE

The single channel 9901 Parametric EQ from Aphex is intended for live sound applications as well as recording, film sound and broadcast environments. As the latest addition to Aphex's 9000 Series modular processing rack system, the unit offers three overlapping bands of EQ with a peak/shelf filter on each band. Each adjustable band has ± 15 dB of boost or cut with transformerless servo-balanced input and output circuitry. Like the other Series 9000 modules, the 9901 fits into both the 9000 rack and the dbx series 900 modular frame.

Circle (126) on Rapid Facts Card

CROWN AMPLIFIER

From the output end, the first in the latest generation of Micro-Tech amplifiers, Crown's Macro-Tech 3600 VZ produces a maximum of 3,600W of power in a 2-rack-space frame. Operable in stereo, bridged mono, and parallel mono modes, the 3600 VZ claims a 105dB (A-weighted) S/N ratio at full output, with 26dB of gain. Frequency response rates at $\pm 0.1dB$ from 20Hz to 20kHz at 1W, while THD was measured at ± 0.05 from 20Hz to 1kHz with lineal increase to 0.1% at 20kHz at full output. (Evaluations were made in stereo mode with both channels driven into an 8Ωload.) **Circle (127) on Rapid Facts Card**

STUDIO O You CAN Direct-To-Disk Audio Editing have a on PC/ATs with Digital MicroSound[™] Audio Studio NOW! · 2 or 4-track analog 16-bit A/D and 18-bit D/A 64x/8x oversampling converters То · Direct-digital transfers to/from DAT or CD find · Windows 3.0 based graphic waveform editor out Record from 8KHz to 48KHz rates how -· Rearrange segments in any order using crossfading or butt-splicing for CALL linear arrangements playing in real-time WRITE Disk-Layering[™] will mix overlapped segments with unique fade/gain settings, from up to 20 files to play 4, 6, 8...32-track simultaneous mixes or FAX Zoom from sample level to track-hours to make edits (undoable anytime) today - for Hear edits using Play mark, Scrub, Audition and more a free SMPTE/MIDI Chase-Lock to incoming timecode, locking and playing brochure within 1 second that Overdubbing optional explains Ready-to-use MicroSound[™] WorkStations also available how to 156 Wind Chime Court meet your Raleigh, NC 27619 audio Phone: (919) 870-0344 needs. Micro Technology Unlimited Fax: (919) 870-7163

Circle (24) on Rapid Facts Card



Cutting Edge

QSC POWER AMPLIFIERS



QSC Audio has released their new MXa power amplifier series. The "a" designation behind the MX1500a and MX2000a models signifies increased power and lighter weight versions of the previously released MX1500 and MX2000. Additional improvements include automatic fan speed control, an input slot for additional connecters, indicators and both active and passive input accessories. Prices have not increased. The model MX1000a is an all new product. Stereo. 80 output ratings for these units are 450W per channel for the M2000a, 250W per channel for the 1500a and 250W per channel for the 1000a.

Circle (137) on Rapid Facts Card

HYBRIOS ARTS RECORDER

Hybrid Arts, Inc. has begun shipping the Digital Master, an affordable direct-to-disk recorder/editor. The complete systems includes a CPU, monitor and Mouse, 105Mbyte hard disk, A/D and D/A converters, MIDI, SMPTE interface, and software. Software features include comprehensive graphic editing functions and various playback functions such as non-destructive editing, a sound effects cue page, over an hour of continuous recording time, and up to 14 hours of sound on line. The hardware includes an S/PDIF (AES/EBU compatible) digital audio interface and a SCSI port for connection of common hard disk drives.

The Digital Master is a stereo, direct-to-hard disk audio recording system for the Atari 1040STE (with 4Mbytes of RAM), Mega4ST, and Mega4STE with 16-bit, 64× oversampling A/D converters and dual 18-bit, 8× oversampling D/A converter. It also offers selectable 48kHz, 44.1kHz, 32kHz, 31kHz, 25kHz, 22.05kHz, and 15.25kHz sampling frequencies. Digital I/O is via S/PDIF and AES/EBU (RCA Connecters). Support is provided for 24, 25, 29.97 and 30fps SMPTE time code. Options include time compression, real-time and offline digital filtering. The Digital Master has a frequency response of 10Hz to 20kHz (0.1dB), greater than 96dB dynamic range, a S/N ration of greater than 90dB (full-scale, at 1kHz, A-weighted), and less than 0.02% THD plus noise. Digital Master works with an SCSI hard disk with a minimum transfer rate of 350kbyteB/second and a maximum seek time of 50ms.

Circle (117) on Rapid Facts Card

KLIPSCH LOW FREQUENCY LOUOSPEAKERS

Klipsch and Associates has also begun shipping the K-1200, K-1500, and K-1800 Series, professional-level 12-inch, 15-inch and 18-inch woofers, marking the company's entry into the "raw frame" component marketplace. High frequency drivers will soon join the K series. K series speakers can handle 300W of continuous pink noise from 40Hz to 2kHz for eight hours, with peaks to 3kW. The 12-inch speakers have 77 ounce magnets, while the 15-inch and 18-inch speakers have 96 ounce magnets and employ 3-inch Kapton voice coil forms. K-1200 speakers are 8Ω designs, and the K-1500 and K-1800 speakers are produced in both 4Ω and 8Ω . The 15-inch and 18-inch speakers are also available in models for small bass reflex

Circle (131) on Rapid Facts Card

enclosures as well as horn enclosures.

LEMO CATALOG

LEMO USA has released its newest catalog for connector specifiers of audio and video equipment. Designed to include the most up-to-date technology of the audio/video industry, the catalog features coaxial and triaxial connectors for audio, video, and TV camera applications; multicoaxial connectors for the state-ofthe-art HDTV industry; and 50Ω or 75Ω connectors combining coaxial, triaxial and signal (low voltage) contacts for these applications. Connectors come in a variety of sizes and shell styles including a sealed version. Also detailed in this catalog are LEMO's newest patch panels for audio and video applications.

All LEMO connectors feature a selflatching, push/pull-type design to ensure reliable connections and avoid signal interruption by accidental pulling of the cable.

Circle (124) on Rapid Facts Card

RAMSA MIXING CONSOLES

The Ramsa WR-S4400 console series offers 12-, 16- and 24-channel professional 4-bus mixer consoles featuring professional length faders (100mm), two selectable inputs per channel, individually switchable 48V phantom power, flexible 3-band EQ with sweepable midrange and layout similar to Ramsa's popular WR-S840 series of concert desks.

WR-S4400 outputs feature four main groups, plus left and right stereo masters from channels or groups and four aux sends. To increase the available Aux groups, Ramsa added a D- out switch and output to each input channel. The switch routes the channel's signal through its Aux bus control, and off Aux 1 bus to its direct output, creating up to 15 aux sends on the 12-channel board, 19 on the 16-channel and 27 on the 24-channel without affecting any other channel operation. The result is an aux group multiplier that operates only per input channel and allows the use of individual effects like voice limiters or reverb devices without taking up one of the conventional four aux groups.

Circle (129) on Rapid Facts Card



INTERFACE MIXING CONSOLES

Interface, a new series of modular mixing consoles, is now available from several Mark IV audio companies. The Interface desks are being manufactured under Mark IV's "multi-brand concept" and will be marketed by Altec Lansing, DDA, Dynacord and Electro-Voice. Available in 8-, 16-, 24-, 32- and 40-channel mainframes, Interface features include five LED level indicators on each channel, padded mic inputs. Aux sends which are switchable to direct channel outputs and a pre/post switch on Aux sends one and two. The group module is equipped with extensive switching for PA or recording applications.

Four group mixing buses allow the use of up to four group output modules. Six Aux buses are also provided, giving six additional mixes with master level controls. Optional input and output transformers are available to isolate the electronically balanced XLR connecters. The 8-channel model is also available as a rack-mount.

Circle (142) on Rapid Facts Card

FOSTEX MIXING CONSOLE

The new Fostex 2412 mixing console is a $24 \times 12 \times 2$ recording console that can manifest two simultaneous stereo mixes via two complete mixing paths. Both mixes have EQ, Solo, Aux sends, and Foldback/Cue sends. This configuration results in the equivalent of 60 avail-

able inputs during a stereo mix. Automated MIDI muting, stereo in-place Solo, six Aux returns and split equalization (Hi, Lo and two bands of sweep) complete the a 22inch deep package.

Circle (138) on Rapid Facts Card



MARION SYSTEMS SCSI

Marion Systems recently confirmed availability of the MPC-SCSI hard disk SCSI interface for the Akai MPC60 and MPC60-II. With the MPC-SCSI and a Macintosh-compatible hard disk, up to 780Mbytes of on-line storage can be accessed. Load and save sequences are four times faster than the floppy equivalent. The MPC-SCSI has been tested with multiple Macintoshcompatible drives, including Syquest, Conner, Quantum and Seagate.

The MPC-SCSI consists of a small circuit card and associated cables. Installation requires opening the MPC60, installing the new software ROMs, the circuit card, and attaching two cables.

Software for the MPC-SCSI was designed and written by Roger Linn and his team of engineers. Other than the hard disk portion, the software is identical to Akai's latest software revision for the MPC60.

Circle (121) on Rapid Facts Card

SOMICH ENGINEERING HEADPHONE AMPLIFIER

The new HPX high performance headphone amplifier from Somich engineering was designed to drive dynamic headphones without the coloration inherent to most high current

MCCAULEY SPEAKERS

McCauley Sound of Puyallup, WA, has announced a new line of extended low range loudspeakers available in 12-, 15- and 18-inch sizes. Designed to handle extremely highpower applications (400-450W RMS) with minimal distortion and breakup, all three speakers share the same field-serviceable magnet assembly. Removal of three allen-head screws enables users to separate the magnets from the baskets to inspect and service the speaker, thus reducing down time and even the need for reconing. The interchangeable magnet will fit any size basket, allowing speakers to be changed to accommodate user needs.

The 18-inch 6254 provides 450W RMS with a frequency response of 20Hz to 800Hz. The 15-inch 6242 is rated at 450W RMS, with a frequency response extending from 1.2Hz down to a clean 25Hz. The 12-inch 6232 model takes low frequencies from 40Hz to 2kHz.

Circle (128) on Rapid Facts Card

headphone amplifiers. The HPX circuit topology uses a minimalist approach, relying on highest quality monolith components to recreate the loudspeaker sound stage, to provide 20dB of class-A voltage amplification. Circle (133) on Rapid Facts Card





Circle (26) on Rapid Facts Card

Cutting Edge

AUDIO PRECISION FASTEST SYSTEM



Audio Precision has introduced the FASTest system which can acquire conventional analog measurement results in seconds. The FASTest program for the System One+DSP and System One Dual Domain can characterize an audio device, system or channel for frequency response, distortion and noise, in a few seconds

ALPHA AUDIO PORTABLE BOOTHS

Alpha Audio Acoustics is shipping its new Audio Seal Portable Sound Booths. The booths are made of flexible panels constructed from the Audio Seal barrier and quilted fiberglass absorber combination blankets. Used as an onlocation sound and video booth, the Audio Seal product is durable, yet easy to construct and dismantle. A secondary application for the booth is as a residential practice studio. The units have a steel frame and are assembled using component parts and velcro fasteners. These booths have a Standard Transmission Coefficient (STC 29) Rating and are Class I fire rated for flame spread and smoke density.

Circle (120) on Rapid Facts Card

HYBRIDS ARTS RECORDER ADAP IV

Hybrid Arts has introduced the third generation ADAP IV into the multitrack, direct-to-disk, digital recording/editing market. ADAP IV's standard hardware consists of four channels of high quality analog input and output, 16-bit, 64× oversampled delta-sigma A/D and 8× oversampled dual 18-bit DAC's, 4-channel digital I/O, a custom DSP module, a CPU with mouse, keyboard, and monitor, MIDI and SCSI ports and a built in SMPTE interface for true "chase-lock" operation. Up to seven hard disks can be connected for a total recording time of over 12 hours.

Circle (118) on Rapid Facts Card

without compromising accuracy or resolution. Using FFT analysis of a composite multi-tone test signal, the relative amplitudes and frequencies of each discrete signal are analyzed and compared to yield these multiple measurements.

Circle (134) on Rapid Facts Card

API COMPRESSOR/LIMITER

API has also unveiled the new 525B, a higher evolution of the original 525 compressor/limiter. Continuing the longstanding API philosophy of building around the original circuit, the 525B will still only have two discrete op-amps in its signal path. The new compressor/limiter is enhanced with a gating function, a frequency sweepable de-esser and will be available for use in the 500-B4 Lunchbox, the 500-VPR, and 500-HPR power racks and the API Discrete Series Console.

Circle (123) on Rapid Facts Card

DRAWMER QUAD GATE

Drawmner is now shipping the DS404 Quad Noise Gate featuring "program adaptive" circuitry capable of handling an extremely wide range of program input signals. A hard/soft gating switch is a key part of the adaptability of the DS404. In "hard" position the DS404 provides ultra fast response, while in "soft" mode the unit assumes expander functions. Gentle release characteristics complement the "soft" gate mode. The DS 404 also exhibits frequency sensitive gating and a slave function when linked channels are required.

Circle (136) on Rapid Facts Card

CYCLONE SYSTEMS SIGNAL INJECTOR

Faults in audio equipment can easily be located in almost any location using the YIBBOX. New to the American market, the ingenious YIB-BOX provides 400Hz outputs at 0dBm, -20dBm, -30dBm and -50dBm to check broadcast, microphone and keyboard feeds. It can also test loudspeakers and headsets. Portable, the YIBBOX weighs seven ounces and is powered from an internal 9V battery with a life of 24 hours continuous operation.



Circle (139) on Rapid Facts Card

CROWN MULTIPLEXER

Sensing technology is the major addition to Crown's new SMX-6 multiplexer. Based on the design of its predecessor, the MPX-6, this unit houses the power and intelligence to switch and route each of its six inputs, two summed outputs, and two relay-switched output buses. The SMX-6 is also capable of integrating with Crown's IQ System 2000 for digital control of

time delay, equalizers, etc.

The inclusion of eight level detectors on both the units' input and output stages provides the next level of real time system status displays, such as monitoring of voltage and current across loudspeaker lines and summed representations, in bar graph form, of all input levels. Circle (132) on Rapid Facts Card



CREATIONS TECHNOLOGIES HARDRIVES

The Anatek DS series of rack-mount hard drives are designed to overcome data storage space limitations by combining a large, fixed hard drive with a removable media drive in the same chassis.

This combination enables storage of CD-quality digital audio on the fixed disk which can be transferred to a removable media for transport or archival. The data remains in unfinished form as it is shuttled on-and off-line from the faster, fixed hard drive. Work may resume from precisely the same point once reloaded.

Anatek DS features for the professional recording environment include: worldwide power supply, dual fans that may be disabled for absolute quiet, high quality ac spike and surge protection for both the drives and auxiliary outlets, and proprietary Macintosh software for partitioning, password protection, spanning and automatic backups.

Preconfigured DS systems range from a 101Mbyte fixed hard drive and 44Mbyte removable hard drive for approximately 13 minutes of stereo to a 1Gbyte fixed drive and 1.3Gbyte DAT for a total of four hours of stereo sound storage. CD ROM drives are also available for access to pre-recorded sound libraries.

Circle (119) on Rapid Facts Card

AREIL DATPORT

Ariel's new DatPort is a self contained digital audio interface that can be used with any computer or DSP processor board that has serial DSP capabilities compatible with the NeXT DSP port. DatPort accommodates standard 48kHz, 44.1kHz and 32kHz sample rates and also is



capable of generating other sample rates through its on-board crystal generator. When connected to the DSP port of the Motorola DSP56001 chip, non-audio data is interpreted and formatted by software running on the DSP. For other chips, data is controlled by front-panel switches.

Circle (141) on Rapid Facts Card

STELLAVOX D/A CONVERTER

Stellavox has collaborated with Swiss sister company Goldmund, to produce the new Stellamode D/A converter. The unit can convert from an AES/EBU or S/PDIF input. It also includes an AES/EBU output, so the Stellamode can be inserted into an existing AES/EBU loop without breaking or changing any connection. Analog outputs can be adjusted between 0,+4dB and +6dB and in polarity. To keep pace with technological evolution, the Stellamode's digital interface and converter-filter are molded in pluggable modules which can be upgraded as required.

Circle (143) on Rapid Facts Card



SPIKE and MIC



CUTTING EDGE

Continued from page 63

WOHLER TECHNOLOGIES LEVEL METERS

The new MSM series audio level meters from Wohler Technologies provides accurate monitoring of levels for up to 20 mono or 10 stereo audio sources in a single rackspace unit. The array of 10-segment LED bar graph meters, each with green/amber/red signal level indication, assures at a glance recognition of potential problems with missing signals, mismatched levels, or input overloads. "Zero" calibration level and VU or PPM ballistics are individually selectable on each meter.

Circle (130) on Rapid Facts Card

WHARFEDALE LOUDSPEAKERS

The new British-made Wharfedale Force 9 loudspeakers are now available in the U.S. from Optimum Audio of Stamford, CT. The Force 9 features a newly designed SMS co-axial driver with a 12-inch silicon impregnated paper cone and a 1-inch titanium compression driver sharing a common magnet in a trapezoidal cabinet. Extended hass response is achieved by two front firing tuned ports.



Force 9 drivers are protected from overload by two temperature coefficient devices with indicating LEDs flashing on front cabinet. Sensit-vity is rated at 98dB at 1 meter on axis, 1W pink noise, a maximum SPL of 122dB at 1 meter continuous with 60° vertical and horizontal coverage. Power handling is 250W and frequency response covers 70Hz-20kHz at –6dB.

Circle (140) on Rapid Facts Card

DIGITAL DOMAIN

Continued from page 23

STANDARDS AND PRIDE

Some of you are probably thinking, why don't all of the companies standardize on the same terminology to eliminate all the confusion? While it would certainly be nice if workstation manufacturers shared a common command set, I don't believe it will ever happen. One reason is pride. Some of these companies are intense rivals, who would rather die than adopt the same terminology their competition uses. Another reason is that standards often create mediocrity.

Hopefully, when manufacturers see how they stack up against each other, they will rethink their choice of language to make sure it's concise and easy-to-understand. Unfortunately, it's hard to change horses in the middle of the stream. I've talked with programmers who feel it is important to be consistent with their terminology, even when it contradicts programning guidelines or is just plain confusing. I understand their fear of alienating existing users, but digital technology is still in its infancy. The time we have behind us is small compared to the time we have ahead of us.

THE SUE SYNDROME

Most graphic user interfaces (GUIs) use icons. pull-down menus and a mouse. GUIs have slowly evolved over the last 20 or so years. No one company or person can take the credit for the concept. It's unfortunate that the 'look and feel' lawsuits, waged by the software companies, have gained such widespread attention, because they discourage commonalty between software. Imagine if 30 years ago the courts had ruled that Ampex was the only tape machine manufacturer that could use the terms fast forward, rewind and play on their tape transports. That situation would be no more foolish than some of the debates going on in the software/hardware world today. Apple Computer's real competitor is not IBM or Microsoft, it's a pad of paper and a calculator. There are still millions of people who have never used a computer. I say, if someone has a good idea, share it. Otherwise each company is forced to spend precious time reinventing the wheel or changing things slightly to avoid lawsuits. In the end, everyone loses.

While writing this piece, it occurred to me that maybe the ultimate user interface should be completely language independent. While this is an interesting concept, current products have evolved to the point that an icon-based design or control surface would have to be quite large to encompass all of the features companies incorporate today. This chart just scratches the surface of all of the commands available. I plan to discuss more advanced editing commands in a future column.



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