March 1989

Recording Engineer/Producer

The Applications Magazine for Audio Professionals

AN INTERTEC PUBLICATION

$4.00

Engineer/Producer Index Debut page 65

Digital Technology

Also:
Signal Processing, Part I
Digital Filter Update
Sampling vs. Hard Disk Recording
Digital Signal Processing

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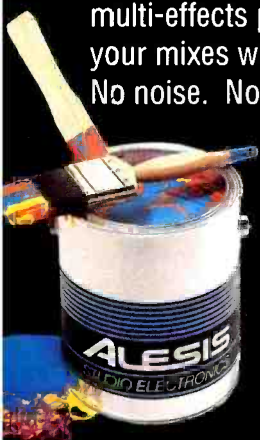

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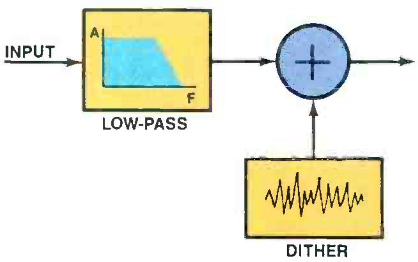
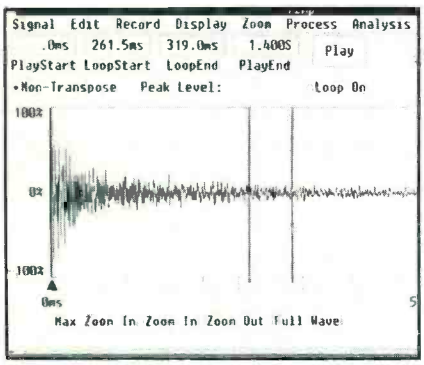




Other Features

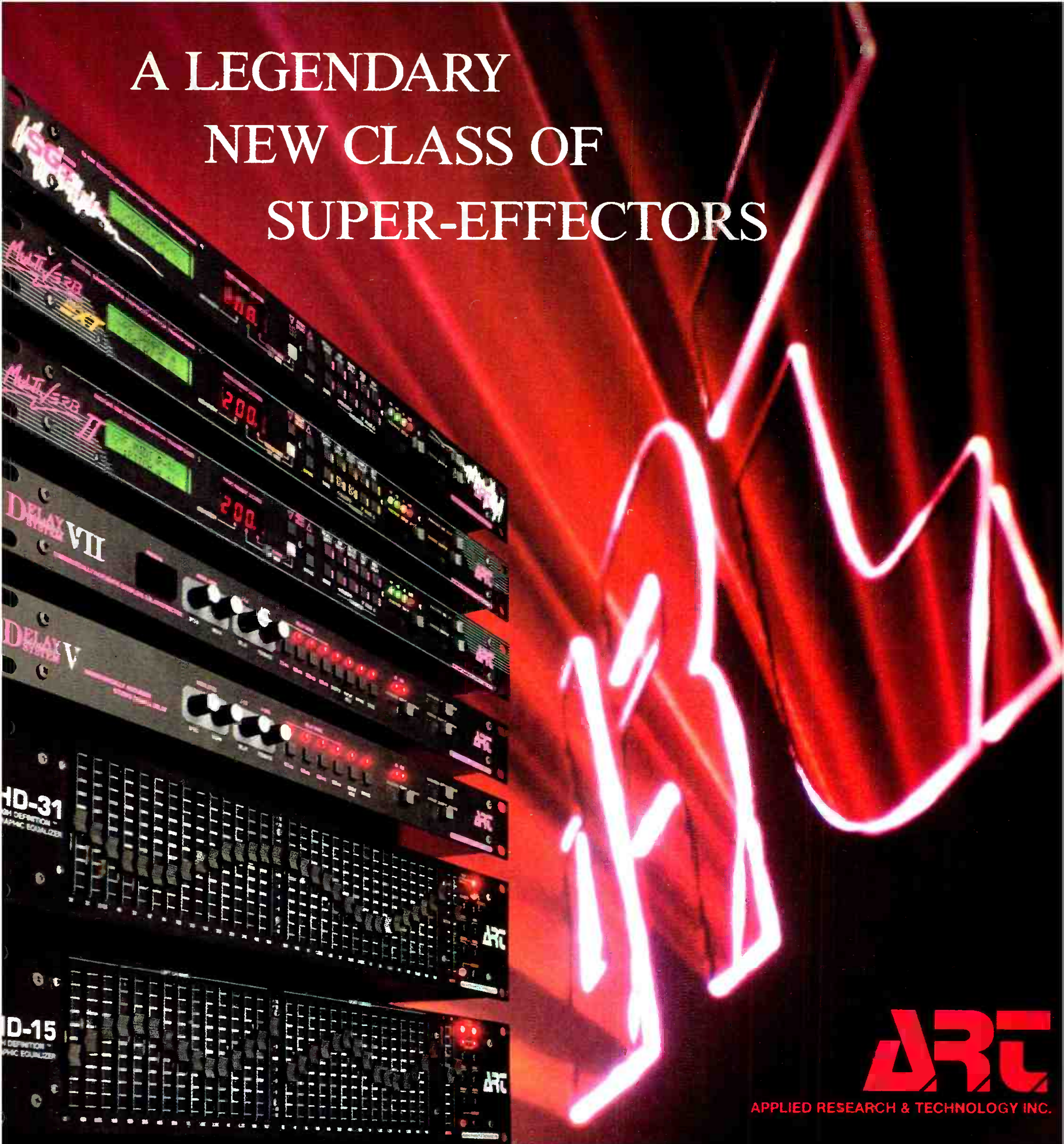
Departments

On the Cover

The Lexicon Opus demonstration facility, Waltham, MA.

Volume 20, No. 3

RECORDING ENGINEER/PRODUCER (ISSN 0034-1673) is published monthly for $26 to qualified readers, $30 to non-qualified readers per year by Intertec Publishing Corporation, 9221 Quivira Road, Overland Park, KS 66215. Second-class postage paid at Shawnee Mission, KS and additional mailing offices. POSTMASTER: Send address changes to Recording Engineer/Producer, P.O. Box 12960, Overland Park, KS 66212.



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EDITORIAL

Cooperation and Communication

Two developments in January bode well for the direction of our industry. Many people are finding ways to break down traditional barriers, at last realizing the value of shared information and resources. This can be seen on the manufacturing level, with more and more joint-manufacturer format agreements, and on the facility level, through groups such as SPARS.

Another example of this type of cooperation was demonstrated at a sound reinforcement seminar held during NAMM week under the auspices of Synergetic Audio Concepts. The three-day event brought together the combined experiences of five veteran touring professionals: Will Parry of Maryland Sound, Albert Leccerse of Audio Analysts, Mick Whelan of Electrotec, M.L. Procise of Showco and David Scheirman from Concert Sound Consultants.

These five people may be friends, but they also are competitors. Yet, for three days, they shared center stage on a program designed to disseminate their combined knowledge and experiences to a group of about 90 paying attendees. Those present represented a wide range of professional interests and experience. Design and management personnel from manufacturers, sound contractors, dealers, live sound mixing engineers, educators, regional reinforcement companies and even a few first-timers could be counted among the group who were there learning more about the touring sound business.

This seminar was a first-ever attempt to examine, on a hands-on basis, large-scale system installation and operation, from rigging to final performance. Some of the subjects covered included pre-production requirements, rigging, power distribution, monitor system requirements, microphone selection, micro monitoring, monitor and house mixing techniques, wireless systems, and plenty of lively Q&A discussions between the instructors and the class.

Generally, the response to the efforts and information presented was quite good. There were some occasional digressions from the topic-of-the-moment and some general criticism about the information being too basic or covered too superficially. But the important point is that this conference took place, not how refined the presentation was.

All in all, this seminar has to be considered a success. It marked the beginning of what I hope will become an era of mutual cooperation, education and communication involving shared information, rather than the proprietary or "everyone-for-themselves" approach that has historically prevailed in the touring sound community. There are bound to be more such seminars; given sufficient advance notice, *RE/P* will provide you with information about all such events.

There really are very few "trade secrets" that are significant enough to change the course or profitability of a service business. In the long run, when communication and resources are shared openly, all parties involved stand to benefit to an extent that is equal to or *greater* than they would have, had one of them tried to control the flow of information.

NAMM

Some are saying that this was the "best trade show in recent memory"—and I share those feelings. This was true because of one overriding factor, and it wasn't technology.

This winter's NAMM managed to finally capture the missing spirit that has plagued most of the audio-related conventions of the 1980s. The manufacturers were finally doing something more than talking about the development and marketing of their latest technology. They were making music.

All of the press conferences, not just one or two, and show performances that I attended featured very talented and, in some cases, big-name musicians. Those artists were demonstrating the real-world capabilities and applications of the products they represented in a purely musical form—*without* any hype about voices, megabytes, tracks, interfaces, updates, retrofits or compatibilities.

It seemed obvious that these exhibitors had finally matured to a point that allowed them to present their products, without catering to the more-is-better mentality of the past. In essence, their only message was, "We make these instruments and the artists on stage are going to show you how 'they' put the technology to work for themselves." Then the musicians just played.

For example, Yamaha sponsored three concerts daily that featured Chick Corea and his Acoustic Band. Admission was free to NAMM attendees, and every show was performed before a standing-room-only house. There were no new-product introductions, no marketing, no corporate profiles, just the band performing on Yamaha instruments, supported by the company's sound reinforcement equipment. The music was the message.

Another example was the Roland press conference. This was truly a benchmark effort. Roland didn't introduce any new products either, nor did they try to sell the press on their "leadership" role in the industry. Instead, they presented excerpts from their recently produced CD.

The CD contains original compositions, written and performed by some of the company's employees and consultants. There is no voice-over nor sales pitch anywhere on the disc, just skillfully orchestrated and carefully mixed music—performed using (nearly exclusively) Roland equipment. And it didn't stop there.

In the liner notes, the company documented the system setup, instrumentation, and any other noteworthy information that helped explain how the music was recorded and how they got the sounds that are heard on the disc.

Again, the message was, "Here is what you can do with our products." The company didn't present a laundry list of features. There really isn't anything proprietary about the processes of making this disc, so there's no reason why the process can't be shared with the listener. The combined efforts of a lot of talented and creative people made this music work, not the EQ and DDL settings.

It's still too early to tell whether this style of marketing is or will be successful, but it's got my vote. I speak for many who attended when I say, "It's about time—thanks."

Michael Fay
Editor

RE/P

Ramsa tests DAT format for professional use

To prove that the R-DAT format can be used in professional applications, Ramsa tested a single Panasonic DAT tape on an SV-3500 DAT recorder at the recent AES show.

The machine was interfaced to a computer running a program designed to continuously cycle the tape. The error correction rate was displayed at all times, and the program could be auditioned through the demo system at Ramsa's suite.

The tape ran a total of 2,800 cycles. The error correction rate at the start was 0016 to 0030, and 0250 at the end. A correction rate of 5000 would be the equivalent of one head blocked, but system redundancy would give normal performance.

Ramsa's Steve Wooley said the company was satisfied with the test results but also said that the quality of the tape would be critical to the results.

Syn-Aud-Con schedules TEF, computer workshops

Synergetic Audio Concepts has scheduled two additional workshops for the remainder of the year. A tutorial TEF conference is scheduled for March 24-25 in Dallas. A workshop on computers in audio is scheduled for June 15-17 in Norman, IN, where the company is located. For additional information, contact Syn-Aud-Con at RR1, Box 267, Norman, IN 47264; 812-995-8212; fax 812-995-2110.

Sound-Net debuts in Toronto

Sound-Net, a dedicated audio/acoustical BBS, is on-line in Toronto. Sponsored by the Toronto section of the AES, the system offers a variety of services, including news, conferences, software, electronic mail and classifieds. System membership is $45 for AES members and $55 for non-members. Additional information is available at 164 Sunnyside Ave., Suite 100, Toronto, Ontario M6R 2P6 Canada; 416-530-4423.

Live recording made without console

At a recent live performance at England's Marquee Club, blues guitarist Jeff Healey was recorded without a console, using Focusrite modules instead. According to engineer Barry Farmer, the modules were used because of their sound quality and the fact that there was not enough space to fit a console.

ISA 110 input signal amplifiers were used to feed drums, vocals and bass direct to the multitrack, while two ISA 130 dynamics processor modules were used on lead and backing vocal tracks to provide slight compression.

Engineers on the date were Farmer, Paul Riley and Mike Dignam from Eden Studios.

News notes

E-mu Systems was listed as one of the country's 500 fastest-growing privately held companies by *Inc.* magazine. During the past five years, the company's growth rate has exceeded 50% per year.

Neve's Nashville office has relocated to 1221 16th Ave. S, Nashville, TN 37212; 615-329-9584; fax 615-329-1473.

JBL Professional has added Seck consoles to its product line. The consoles, available in a variety of configurations for live and studio use, will be distributed through Soundcraft in the United States and Mexico. Harmon International, JBL's parent company, purchased Seck last year.

Ramsa has named Micworks, Huntington Beach, CA, as its 1987-88 dealer of the year.

Apogee Electronics supplied the sound system for cabaret singer Michael Fienstein at the Beverly Wilshire Theater in Los Angeles. Designed by Daryl Bornstein, the flying system incorporated 12 AE-5s grouped in threes, a cluster of three 3x3s and two AE-3s for side fills. Two AE-10s were used as subwoofers.

Neutrik AG has acquired Infomatic, a

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RECORDING ENGINEER/PRODUCER is edited to relate recording science to recording art to recording equipment, as these subjects, and their relationship to one another, may be of value and interest to those working in the field of commercially marketable recordings and live audio presentation. The editorial content includes: descriptions of sound recording techniques, uses of sound recording equipment, audio environment design, audio equipment maintenance, new products.

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Tascam's proprietary head technology allows you even more production speed, with head quality so uniform that EQing decisions can be made right in sync without rewinding to repro. Special circuitry provides transparent punch-ins for completely gapless and seamless edits.

But there's only so much of the ATR-80 that can be described in features. For the rest you must sit down in front of it and lay your hands on the controls. That's when you'll sense the craftsmanship and quality of it's design. The power, the speed, the smooth response of the transport.

See your Tascam ATR-80 dealer. After you use it, you won't miss those preconceptions one bit.

TASCAM

Circle (7) on Rapid Facts Card

NEWS

manufacturer of modular digital sound recorder and reproduction systems.

Communications Engineering Inc. has been retained to design and build a 32,000-square-foot production facility for Black Entertainment Television. When completed, the facility will be the largest production and post-production plant in the Washington, DC, metropolitan area.

HM Electronics has appointed four rep firms to handle its product lines. Northshore Marketing, Seattle, will cover Washington, Alaska, Oregon, western Montana and northwestern Idaho. Brian Trankle and Associates, Belmont, CA, will cover Northern California and northern Nevada. Powerlines Marketing, Evanston, IL, will cover northern Illinois and eastern Wisconsin. R.J. Throckmorton Sales Co., Ballwin, MO, will cover eastern Nebraska and southern Illinois.

Apogee Sound has named RPM Sales as its 1988 representative of the year.

WaveFrame has appointed a manufacturer's rep and three new European distributors. Audio Images will cover Northern California, Oregon and Washington. European distributors are Amptown Electroacoustic, West Germany, Wave Trade, Sweden, and New Musik, Denmark.

ATM Audio has moved to 17104 S. Figueroa St., Gardena, CA 90248; 213-538-2004.

Frazier has appointed several rep firms. Burcaw Company will serve Michigan. Henry W. Phillips Company will cover North and South Carolina, Tennessee, Georgia, Alabama and Mississippi. SRT Marketing will cover Florida. Bi-State Marketers will serve northeast New Jersey, metropolitan New York and lower New York state.

People

Dr. Per Bruel, co-founder and director of Bruel & Kjaer, was awarded the Silver Medal Award at the AES Convention for his contributions to and continued refinement of microphones used for acoustical measurements.

Colin Pringle has been named head of marketing for Solid State Logic.

Forrest "Bud" Rees has been named president and CEO for Sound Technology.

Bernard Howard Fryman has been named eastern regional sales manager for Numark Electronics.

Bob Jenkins has been named vice president/managing director of FirstCom/Music House.

WaveFrame has made two appointments as part of its expansion of its service facilities. **Craig Damon** has been named customer service manager. **Cathy Curley** has been named field service coordinator.

Circle (9) on Rapid Facts Card

NEWS

JBL Professional has named **Ted Telesky** as loudspeaker product manager.

Phil Wagner, Neve's eastern regional sales manager, has been named the company's salesman of the year.

Ted Pine has been promoted to marketing communications manager at New England Digital.

John Carey, marketing manager at Otari, has assumed the sales manager duties of David Roudebush, who resigned in December.

Electro-Voice has announced two appointments. **Bill Mullin** has been named market development manager. **Bob Doebel** has been named director of employee relations.

Circuit Research Labs has announced several three appointments: **Gerard Vargas,** international sales manager; **Clayton Creekmore,** western U.S., radio and TV sales associate; and **Debra Ernzen,** marketing coordinator.

Larry J. Repasky has been named district sales manager, professional audio and video tape products, in the southern U.S. for BASF.

Carver has appointed **Mark Friedman** senior vice president and director of sales.

Kent W. Sheldon has been appointed national sales manager of Klipsch & Associates.

1989 Black Book addition

The following entry should be added to the Dealer/Distributor directory under Pennsylvania:

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MANAGING MIDI

By Paul D. Lehrman

Sequentially Speaking, Part 2

Last month, we talked about some of the major features I like to see in a sequencing program. This month we delve into some more esoteric functions.

Free barring. Bar lines are essential divisions for a sequencer to work with, like pages in a word processor. But just as you can move page breaks around in a word processor to make your text look good, you should be able to move bar lines around to accommodate your music. This can come in handy when you need to add a partial pickup bar to an already-recorded sequence, or when you want to play a track in free time, not listening to a metronome, and be able to insert beat and bar divisions after the fact. Some programs let you do this musically, by recording the notes on one pass, and then tapping a key or a pedal to denote where the beat is on a subsequent pass. Other programs let you do it graphically, by physically moving bar lines back and forth on the screen.

Two-dimensional regional editing. Early sequencers only allowed you to do transformations (i.e., quantize, transpose, copy and paste) on an entire track at a time. Nowadays, most sequencers let you specify beginning and ending times for an operation. The next step, which some sequencers have already taken, is to be able to specify upper and lower *values* for an operation—e.g., only notes between C3 and G#5 get quantized, only controller values below 27 get doubled, only program changes between 63 and 126 get incremented by one.

Logical editing. And the step beyond that is to be able to specify Boolean operations for editing: all E flats that sound on the same beat as a G get changed to an E natural; all kick drum hits with a velocity greater than 45 get transposed to a low tom; all pitchbend values from -20 to +20 get set to zero while all others are doubled. Quantize and increase the velocity by 20% of only the top note of each chord; or take every fourth note in a melody line and transpose it down three octaves while kicking it over to the bass track, but only if it falls within 5% of a beat, and only if it's in the C#-major scale.

You get the picture. A properly designed program can allow very complex operations that can save enormous amounts of time.

A properly designed program can allow very complex operations that can save enormous amounts of time.

Mapping. By which I mean being able to take a specific note or value and transform it to another specific note or value, and to be able to do that for a large set of values in one operation. If you've ever had a sequence recorded with one drum machine in mind and tried to play it back on a different drum machine, you know how useful this can be. It can also be used to do modal transpositions in which, for example, you want to raise the pitch of a melody a whole step, but keep it in the same key you started in.

Randomizing functions. I used to pooh-pooh the idea that computer-generated randomness could be of any musical use (except as a white-noise source), but I've changed my tune in the last couple of years. Especially if you're lazy and depend on step-time entry for a lot of your work, a little randomization can do an awful lot to make a track sound human. Particularly with something like an orchestral string flourish, randomizing slightly the timing, velocity and duration can make the difference between something that sounds like it was played by a robot and the London Symphony Orchestra. It's also a nice way to break up piano chords and downbeats on drum tracks where everything hits at the same time.

And speaking of downbeats, a very small amount of randomization in note attack times on highly cluttered beats can help relieve the symptoms of mild MIDI choke. It won't actually cure the problem, but because one track or note won't *consistently* lag behind the others—instead the various tracks will chase each other around—the problem will become much less obvious.

The best randomizing functions can be combined with scaling-over-time functions to produce what some folks call "tendency masks"—for example, a passage might start out using random velocity values between 20 and 30, and end up with random velocities between 85 and 90, while the velocities for the notes in between are interpolated. The crescendo effect is definitely there, but without the artificialness that can result from simple velocity scaling.

System exclusive capability. I'm not talking about librarian functions, where you can dump and load whole banks of patches through your sequencer—although that can be a nice feature, especially if you don't have a dedicated librarian program. I'm referring to the ability to specify brief strings of system-exclusive data and send them out at particular times as part of a running sequence. Although MIDI controllers are being used for more esoteric things all the time, there are still some functions that will probably continue to be beyond them, like LFO speed, pitchbend range and microtonal pitches, but which can be addressed perfectly well using relatively short, system-exclusive commands. Obviously, you don't want them to get too long, or you could jam up the data stream pretty badly.

Being able to record short bursts of system-exclusive isn't a bad idea, either—I haven't seen this anywhere, but I could imagine a feature that would let you place incoming sys-ex commands into a text buffer where they could be edited, and then placed back into the sequence wherever you like. No sequencer that I know of has all of these features, and perhaps none ever will. But with MIDI File compatibility becoming more common, folks are beginning to be able to combine programs to get all the features they need.

As I said last month, clipboard-style MIDI File compatibility, in which you can move small pieces of a sequence in and out of several programs at will, should be the next step. If we're lucky, what will then crop up will be little "editing module" programs that you can customize and use to perform those incredibly esoteric editing functions no sequencer will ever support. RE/P

Paul D. Lehrman is RE/P's electronic music consulting editor and is a Boston-based producer, electronic musician and free-lance writer.

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SAMPLING. 16 bit stereo or mono multi-sampling (up to 32 seconds) with pitch shift, adjustable attack and decay, flexible output mixing, up to 12 simultaneous voices, support for MIDI sample dumps, triggering by audio input, MIDI or foot pedals.

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SPARS ON-LINE

By Bruce Merley

The Next Decade

As the newly elected president of SPARS, I am deeply honored to be serving an organization that has given so much to me through the years. Compared to past presidents, I am a relative newcomer to the recording industry, and I owe much of my success to the insights I've gained from the organization. Now it's my turn to repay the debt.

I'd like to start out by mentioning a fine book by Allan Bloom entitled, "The Closing of the American Mind." This study is an indictment of the present state of education in America, particularly in the colleges and universities. One of the great hallmarks of our educational system has been the search for truth. It is reflected in the open mind of our society. This search for truth is in dire straits these days, and during the past few generations we have seen a change in our schools and in our society. We no longer seem to have a firm cultural foundation for evaluating facts, figures, and the massive amounts of information we are presented with. We are living in a cultural void where people have lost a sense of place in society.

Our minds are open, but with no foundation or standards by which we can evaluate and judge. We have become disinterested observers who are tolerant of everything. As an example, we view the social problems of the homeless and say it's a shame. We don't see it for what it is—a deplorable situation that must be changed. We have become alienated, isolated, and distanced participants in our own culture. Each of us must struggle with this problem if we want to save ourselves, our society, our planet.

What does this have to do with SPARS? Well, this situation has its parallel in the recording industry. To serve as president, I feel I must address these types of problems. As an industry, we are always open to new ideas, new technology, new ways of making our lives more productive and profitable. We are bombarded with data at an ever-increasing rate, and it seems to me that we are losing our ability to examine and evaluate.

Without a foundation within our audio society, we give equal weight to all new technologies, new market demographics and changes in business climate. As a result, we are at risk of destroying ourselves as an industry, as a profession and as an aesthetic endeavor. We must examine our roots, reconfirm our foundations and conscientiously measure information from a common viewpoint.

I see this as a big challenge, both for myself and for SPARS. We are fortunate to have within our organization the leaders who have shaped the industry we work in today. We have the members who created SPARS and made it a voice in our industry, and we have many new members who are further shaping the future of the organization and the industry as a whole. My objective is to use this significant human resource as the foundation, to create a clear standard by which we can analyze and evaluate the deluge of information that is thrust on us.

In my work with the members of SPARS, I hope to develop a stronger profile and a point of view that represents what we feel is the best philosophy—the spirit of the recording industry. We are a group that represents every facet of audio, from major studio to manufacturer to free-lance engineer to home studio operator. We must pull together to strengthen our common goals.

Because we serve the established elements of the industry, as well as the newcomers, it is vitally important that the information we dispense, and the conferences and dialogues we promote are of the highest standard. The information we share can make SPARS a stronger force, and the information we convey to the industry as a whole will pave the way to a better future. I apologize if this sounds pompous, but I believe the time is critical, and as we celebrate our first decade, I hope that it points the way to an even more productive second decade.

Specifically, what can we do? We begin our second decade by going back to the beginning and doing what our founders had the good sense to do—communicate. Our goal is to share information, identify our common problems and the issues that burden us all, and use SPARS to solve those problems. To that end, you will be hearing a lot from me, from other members of the board, and from our executive director, in a unified effort to generate the kind of dialogue that can make SPARS even more useful to its members.

From this dialogue we develop our long-range agenda. We clearly identify the issues that must be faced for the good of the industry. It's all too easy for us to complain about the individual problems of our respective businesses, but if we pull together and identify shared problems, SPARS can do something about it. Taxes, foreign competition, rate-cutting, non-professional practices? SPARS has played a vital role in everything from product design to modifying tax laws. We will determine the issues together and use SPARS to act.

But in order to act, we need strength. We need the unified force of our entire industry, not just part of it. In order to speak with authority, we must increase our membership. From my point of view, this entire year will be a membership drive to double the number of members we represent in all categories. In this period of renewal, it's a good time to join us. We are planning many conferences this year to deal with the problems of survival that face us all. Our educational and internship program is firmly in place, but we'd like to have even more participation from across the country and the world. We've got some surprises in store for the 10th year gathering at AES in the fall, and we'd like new members to share in the celebration. When you join SPARS, you join people like yourself. You learn from those who have prospered, and you make friends with those who are launching their careers in audio. SPARS is an information network, and a safety net to help you in times of need.

In a recent conversation I had with a SPARS member, we discussed the problems and issues we are all facing. Although he has always been clear in his reasons for being a member, the simplicity of his thought was potent. "How," he said, "can you be in this business and not belong to SPARS?" Indeed.

Want to talk about SPARS? Call me at 212-246-2444, or contact Shirley Kaye, executive director, SPARS, 4300 10th Ave. North, Suite 2, Lake Worth, FL 33461; 407-641-6648. RE/P

Bruce Merley is president of SPARS and president of Clinton Recording, New York.

UNDERSTANDING COMPUTERS

By Jeff Burger

Modems

Computers do not inherently get along with telephones. They require peripheral devices know as modems to get onto Ma Bell's wavelength. The word *modem* is short for *modulator/demodulator.* To communicate via the telephone, a computer's digital signal must be converted to the analog domain for transmission, and then back to digital for reception. The modem is really a type of D/A/D conversion device.

Modems typically connect to a serial port on your computer, though for the PC-clone family, some are available as plug-in cards. Modems that plug into a modular jack are called direct-connect modems, while those that have receptacles to mate with a telephone handset are called acoustic coupler modems. The one advantage of the latter is the ability to plug into virtually any phone in the world.

Modems are measured by their *baud rate*, a baud being one bit per second. When modems first became popular, the standard rate was 300 baud. But today, in addition to the 300-baud rate, most models offer a communication speed of at least 1,200 baud, and 2,400- and 9,600-baud models are becoming more affordable. The catch is that the systems on each end of the connection have to be operating at the same speed.

Parameters

Baud rate is a parameter that is set within a telecommunications software packages—the second ingredient of modem communication. In addition to baud rate, several other parameters must be properly set on both ends before two computers can communicate. The first of these is the *duplex*. Half-duplex is similar to a CB connection where the parties on either end must take turns speaking, while full-duplex is similar to a 2-way telephone connection.

Next, the operators must determine the size of the *data words* to be used, typically seven or eight bits. The first 128 characters (0 to 127) of the ASCII code (containing the code for such characters as the alphabet, common punctuation marks and numbers), are standardized for all computers and require seven data bits for transmission. The eighth (or high) bit can be used to transmit the remaining 128 characters (128 to 255) of the ASCII (containing the code for such characters as the Greek alphabet, mathematical signs and other special symbols). But there is no standardization of these special characters beyond a given computer system and they can create compatibility problems. (Refer to an ASCII table for a listing of these characters.)

Most problems are the result of not having the parameters properly set on both ends.

Another common use of the eighth bit is error detection. In a process called *parity checking*, odd or even parity is specified. At the end of a transmission the ones and zeroes of the 7-bit words are totaled using simple arithmetic, after which the the eighth bit is set either to zero or one, making the total an odd or even number. The receiving computer adds up the eighth bits and verifies that the sum is still odd or even as specified. While this method isn't infallible, when the parity check is OK, it's a good bet that the data has not lost its integrity during transmission.

Finally, the number of *stop bits* or *framing bits* that delineate data words has to be agreed upon—typically it is one or two.

Let's say that you want to call your best friend's modem-equipped computer. You call each other by voice and set the parameter the same on both ends, with your system set to originate and your friend's set to answer. Your friend will let the computer answer the phone when you call back. Next, disable call-waiting if you can, because incoming calls will surely destroy your connection. Now have your computer dial your friend's number. When the remote computer answers, it sends an audio tone called a *carrier* that your modem latches onto. Assuming the gods of data are smiling upon you, you are now "on-line" and anything either one of you types will show up on the other's screen!

In addition to dialing, answering and the setting of parameters, the typical telecommunications package provides for *uploading* and *downloading*. *Uploading* refers to sending a file from your computer to another just as if you were typing. *Downloading* refers to capturing another computer's transmission on disk or in a buffer (holding area in memory).

Macros

Most software packages also include *macro* programming capabilities that allow you to recall all the necessary parameters and keystrokes required to log onto a remote computer. Procedures can even be set for auto-execution of a given task, such as downloading the status of your Dow Jones stocks in the middle of the night.

Telecommunication is not hard, it's just fraught with variables. Most problems are the result of not having the parameters properly set on both ends of the connection. If you can only see one side of the communication, the duplex setting is probably wrong. If you see garbage on your screen, it probably means that the baud rate, data-word size or stop-bit size is wrong. Occasionally bad telephone lines are the guilty party, and random dropouts occur.

Another common problem involves line feed and carriage return confusion between software packages. When you send a "return" or "enter," you essentially send a carriage return. The remote computer also needs a line feed command associated with it to prevent lines from overprinting one another. Either you have to set your software to send line feeds with carriage returns or your partner must tell his software to add them. If both systems are adding line feeds, you'll have unwanted spaces between the lines of text. Some systems default to a line length, such as 80 characters, and automatically insert line feeds and carriage returns at those points. Since this may result in unwanted line feeds breaking up the file on the recipient's end, it's usually a good idea to disable any arbitrary line length settings.

RE/P

Jeff Burger is *RE/P*'s computer consulting editor and is president of Creative Technologies in Los Angeles.

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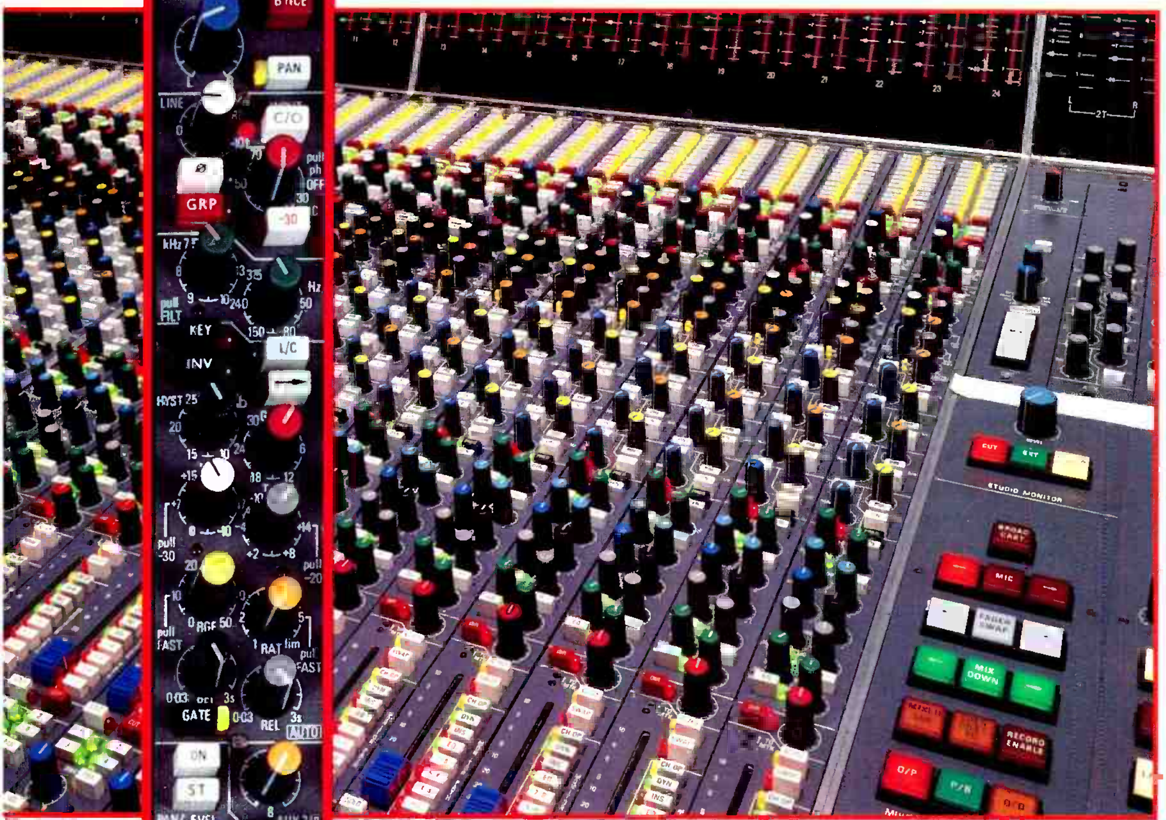

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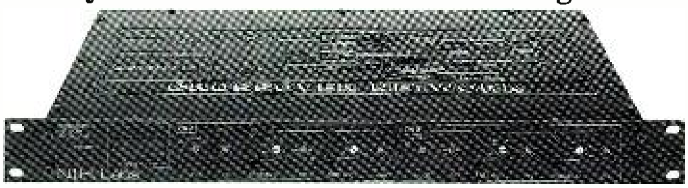

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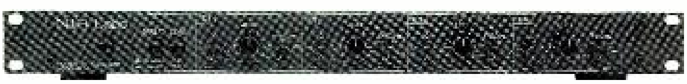

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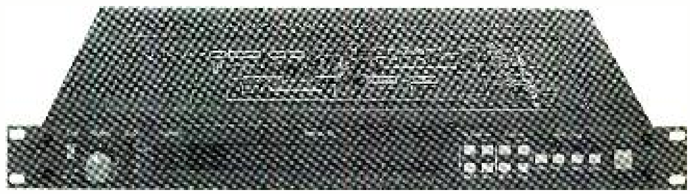

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Performance Aspects of Digital Oversampling

By Dr. Richard C. Cabot

Nothing is free, particularly with regard to digital oversampling. D/A and A/D conversion can and will affect other performance aspects of digital audio devices.

As was the case with early applications of transistor technology to audio, many people do not understand the detailed trade-offs inherent in D/A and A/D conversion technology. The drive for impressive numbers on the current buzz-word specifications has resulted in serious degradations of other important specs. As a result, commercially available digital devices differ significantly in audible and measurable performance.

Richard Cabot is principal engineer at Audio Precision, Beaverton, OR.

Signal conversion

A classic, textbook digital audio device is diagrammed in Figure 1. Circuitry at the input converts the audio signal to digital data. At the output, the data is converted back into an audio signal. The sample and hold section captures samples of the audio signal for conversion by the A/D. These samples are taken *at least* twice per cycle of the highest audio frequency in the signal. For example, to handle a 20kHz bandwidth, most professional audio equipment samples the audio 48,000 times per second (the sampling rate). Theoretically, a 48kHz sampling rate allows signals up to 24kHz to pass, but because of practical circuit complexity limitations, the highest frequency that passes is actually about 22kHz.

The Nyquist frequency represents a low-pass cutoff point equal to one-half the sampling rate, and was named after the engineer who originally derived the

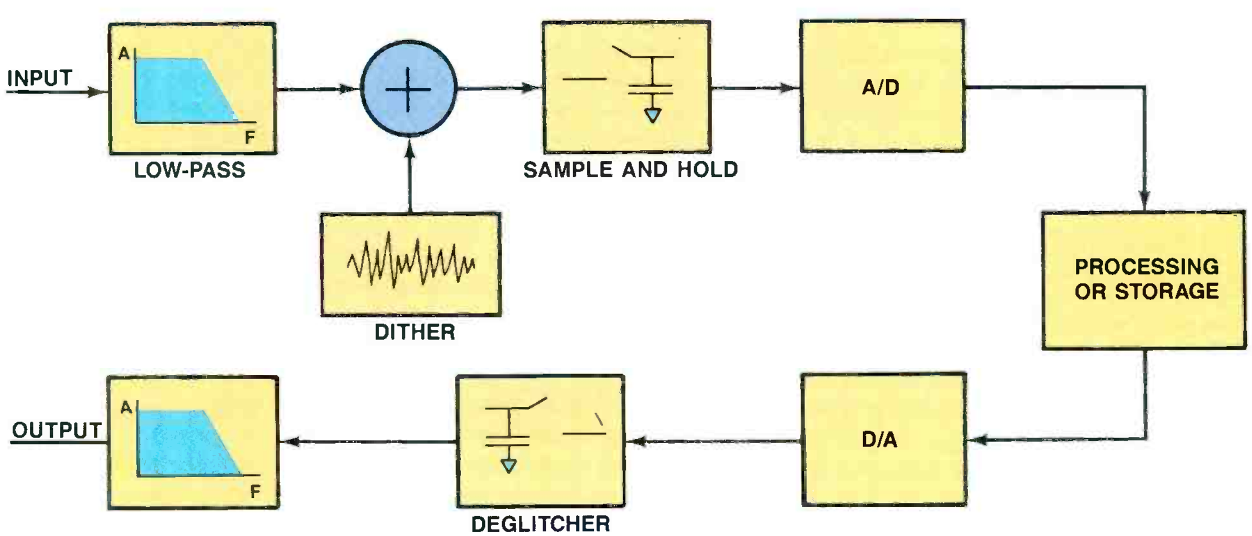


Figure 1. A basic digital audio system.

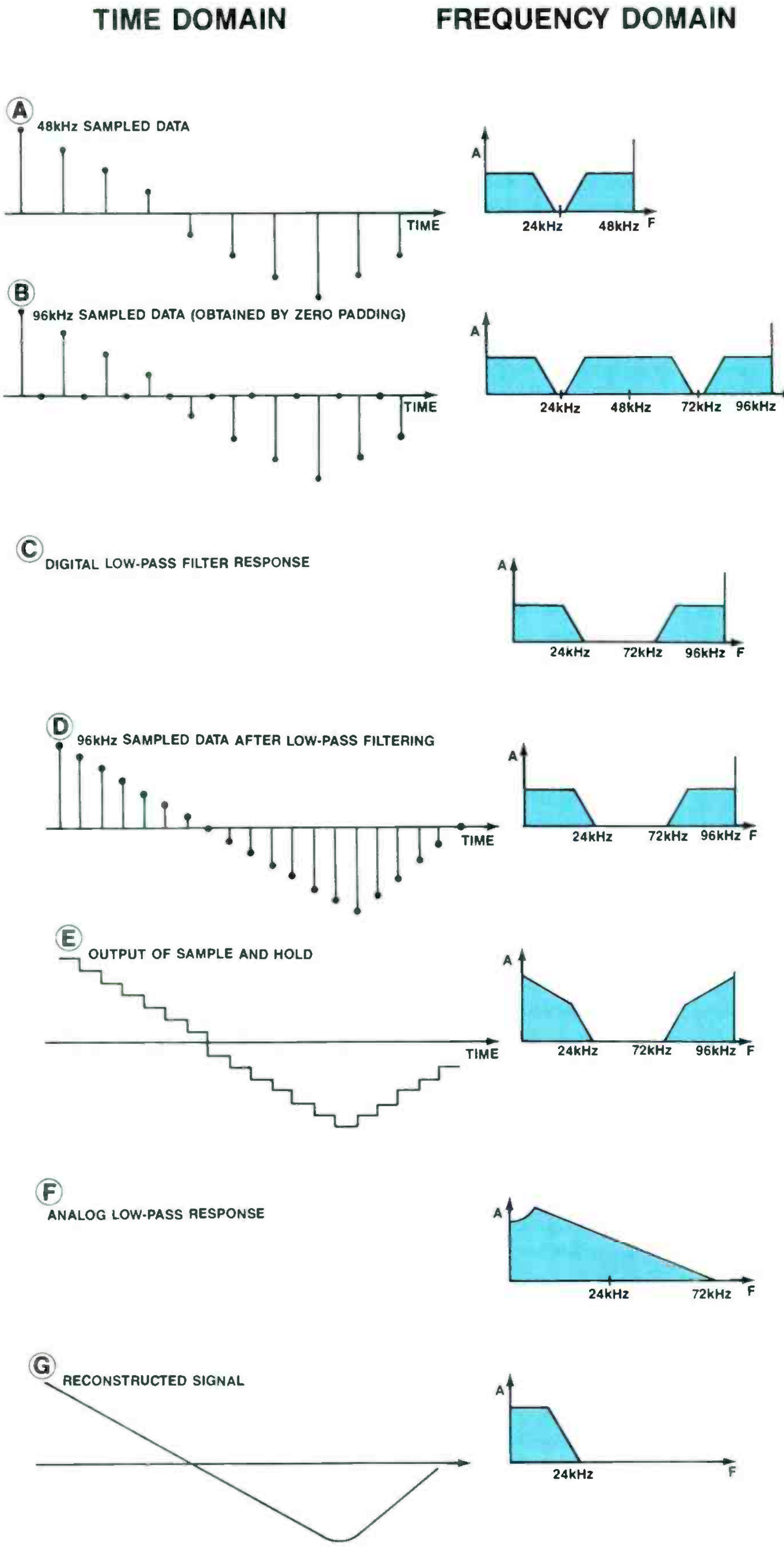


Figure 2. Illustration of 2X oversampling D/A operation.

theory and identified its need. Input signals above the Nyquist frequency cannot be properly converted and will create new signals in the audio band that were not present in the original signal. For example (assuming a 48kHz sample rate, therefore, a 24kHz Nyquist point), a 27kHz analog signal will appear in the digital output as 21kHz, and a 28kHz signal results in a 20kHz output. These are called *alias components* and are prevented by low-pass filtering the input signal a little below the Nyquist frequency. This low-pass filter must have flat response over the audio frequency range of interest and must attenuate the frequencies above the Nyquist point enough to put them into the noise floor. Some designers choose to simplify these filters by not fully attenuating alias frequencies that appear in the digital signal above the range of hearing (usually assumed to be 20kHz).

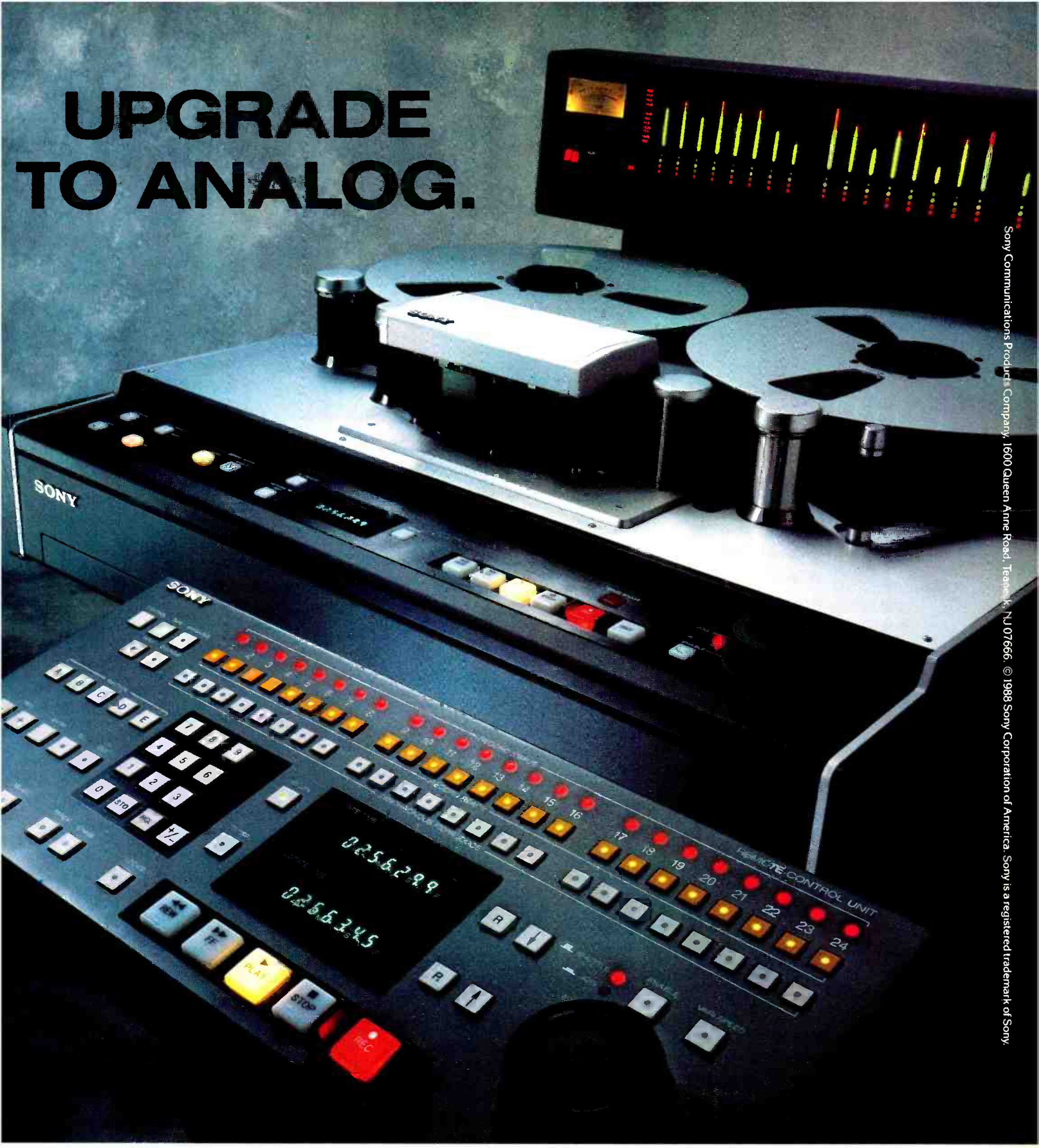
The samples are converted to binary digital codes by the A/D converter. Although the original signal is continually changing, the output of the sample and hold is constant between samples. This allows the converter enough time to look at the analog voltage and come up with the digital code that represents it.

The process of converting an analog signal with an infinite number of possible voltages into one of a finite range of numbers (quantization) introduces an error signal that is dependent on the signal being sampled. This quantization error will be on the order of one least-significant bit (LSB) in amplitude, quite small compared to full-amplitude signals.

Full-amplitude distortion measurements, therefore, look quite respectable, possibly even impressive. However, when the signal gets smaller, the error becomes a larger percentage of the total signal. What's worse, as the signal makes small changes in amplitude or frequency, this error will change radically. When the signal is removed, the error will also vanish, leaving a totally silent background.

This constantly changing error signal sounds like a harsh, gritty noise that "breathes" during low-level passages in the program material. It is largely responsible for the early criticism of digital audio and for the some of the continuing problems with specific models today. [For a more detailed discussion of modulation noise and quantization distortion in digital audio, see "Noise Modulation in Digital Audio Devices or Who Wrapped the Mics in Sandpaper?" July 1988, p. 20.]

To make the quantization error unrelated to the input signal, a noise-like, low-amplitude random signal is mixed with the input signal before conversion or sampling. This random signal is called *dither*, and the process of mixing it with



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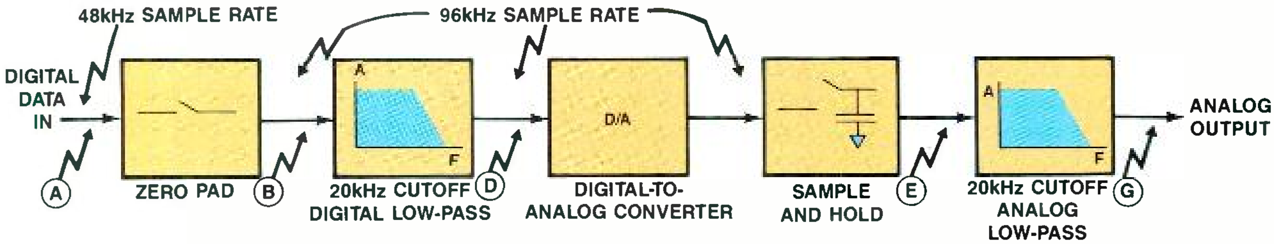

Figure 3. *Block diagram of 2X oversampling D/A.*

the signal to be converted is called *dithering*. It is quite effective at changing what would otherwise be obnoxious distortion into benign background noise.

The op-amps and the anti-alias filter in many early digital systems created enough noise to provide an adequate dithering effect without its specific addition. Unfortunately, because of this, some designers have not understood the need for correct signal dithering and have produced undithered conversion hardware with low-noise front ends. The result is a "non-musical" piece of equipment, best suited for recording political speeches, or anything else destined never to be heard again.

To re-create the analog signal from the digital data, the samples are fed to a D/A converter. This creates a staircase-looking waveform that changes value at each new sample. Since the converter will often create incorrect values (glitches) as it is changing from one sample to the next, the signal is processed by a deglitcher. This disconnects the D/A from the output while it settles to the new value. Deglitchers work in much the same way as the sample and hold circuit located at the input.

To eliminate the staircase appearance, and the harmonics that go with it, a low-pass filter is used. It rounds off the sharp edges from the waveform and smoothly creates values between the samples. This is called a *reconstruction filter* because it reconstructs the analog waveform from the sample values.

Analog filter trade-offs

The anti-aliasing and reconstruction filters are very similar in their design requirements and performance. These performance limitations appear in all areas: frequency response, time domain response, distortion, noise and dynamic range. As mentioned earlier, completely filtering all alias products requires a very complex filter that has more filtering stages. Each stage adds noise, distortion, phase shift and cost. Using more stages necessitates tighter control of component tolerances to keep the frequency response flat in the audio band.

Conventional analog filters introduce a delay into the signal that is not constant with frequency. This shifts the phase of harmonics by a greater amount than the fundamental. Consequently, square waves applied to the filter come out looking decidedly non-square.

If the filter had a constant time delay for all frequencies (linear phase response), the square wave would be rounded at the edges a little, ringing on the top and bottom. The non-constant delay causes the waveform to ring substantially more on one edge than the other. It is possible to compensate partially for the frequency-dependent phase shift by adding more phase shift at lower frequencies as necessary. Phase-corrected filters require additional analog stages to provide this phase shift, which increases cost.

> ***The op-amps and the anti-alias filter in many early digital systems created enough noise to provide an adequate dithering effect without its specific addition.***

Many companies manufacture analog low-pass filter modules for use in digital audio equipment. The most common design for the internal circuitry of these modules is called a *ladder filter* because the connection of inductors and capacitors looks like a ladder when drawn on paper. Most modern implementations replace the inductors with op-amp circuits called *frequency-dependent negative resistor* (FDNR) elements. These avoid the traditional inductor problems of distortion at low frequencies when the core saturates, and allow much tighter control of tolerances since capacitors are easier to manufacture accurately than inductors. The phase correction stages, if included, are also most easily implemented with active filters.

But extremely sharp-cutoff active filters, such as those required for digital audio, have their share of problems, too. Sharper rolloff results in higher noise gain from the filter op-amps to the filter output. This necessitates the use of very low-noise op-amps to achieve adequate performance. This also forces the use of high signal levels to maintain the signal-to-noise performance of the digital system. The filter topology exposes the operational amplifiers to large common-mode signals at high frequencies, a problem exacerbated by the large signal levels. This can generate great amounts of distortion with many op-amps common in audio, especially FET input types.

To keep power consumption and cost low, many manufacturers have nonetheless used FET input op-amps in their filter modules. The resulting distortion may not show up in THD measurements (because of the low-pass action of the filter), although twin tone IM measurements catch it nicely. This has created a large aftermarket for anti-alias filter and reconstruction filter upgrades in professional digital tape machines.

Oversampling digital-to-analog conversion

Recall that the complexity of the reconstruction filter is dictated by how close to the Nyquist frequency we want to reproduce signals. If the highest frequency to be output in a 16-bit, 48kHz sampled system is 20kHz, the low-pass filter must be flat to 20kHz and yet have more than 96dB attenuation at 28kHz. This is an extremely sharp filter. If the sampling rate is raised to 96kHz, the filter will only need to reach 96dB of attenuation at frequencies above 76kHz, (the frequency above the Nyquist point that when folded over, represents 20kHz).

With 96kHz sampling, filter design becomes much simpler, cheaper, and has lower noise, distortion and phase shift.

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Oversampling Pros and Cons

- Oversampling can give more resolution.
- Oversampling A/Ds can provide freedom from low-level non-linearites.
- Digital filters can be linear-phase.
- Digital filters can be lower-distortion.
- Digital chips are usually cheaper to make.
- Digital filters give better control over response.
- Oversampling requires faster circuits, but faster circuits are less accurate.
- Oversampling D/As raise the noise floor unless more bits are used to suppress filter round-off.
- Oversampling D/As limit headroom unless an extra bit is used to handle overshoot.

However, to do so, the signal must be converted from a 48kHz sampling rate to a 96kHz sampling rate. Since the sampling rate is doubled, this technique is commonly called *two times oversampling*. If the rate is increased by four, raising the sample rate to 192kHz, it is called *four times oversampling*.

A term more accurate than oversampling would be interpolating, since the process actually interpolates the additional samples from the existing sample values. Oversampling implies that the signal is sampled at a higher rate, but by this point in the process, the signal is not available to be resampled. Despite these incongruities, the term oversampling has become widespread and will be used herein as well.

The sample rate conversion process is illustrated in Figure 2. Let's start with a 48kHz sample. To change the rate to 96kHz requires that samples be taken twice as often. These extra samples must be computed from the existing samples. Since the value of the signal at the extra sample intervals is unknown, they are set to equal zero. The processor could be designed to assume that the new sample is the same as the previous sample, but this actually makes the processing more difficult. (See the sidebar "Digital Filters for Oversampling Applications.")

The square edges created by the sampling process introduce harmonics of the original signal. By low-pass-filtering the signal with a *digital* low-pass filter, the harmonics between 24kHz and 72kHz may be removed. When viewed in the time domain, the digital low-pass filter "rings" on every sample pulse from the original signal and fills in the missing sample value.

The block diagram of the digital-to-analog conversion system now looks like Figure 3. The digital signal is padded with zeros and fed to a digital low-pass filter. This drives a D/A converter that converts the digital words into analog samples. A deglitcher prevents transient errors while the bits going into the D/A are changing. The resulting signal is low-pass-filtered in the analog domain. This 2X oversampling system differs from a non-oversampled one in that some of the filtering is done digitally, making the job of the analog filter easier.

Digital filters may be made linear-phase very easily and, thanks to the low manufacturing cost for digital integrated circuits, are relatively inexpensive. However, this technique requires the D/A converter and deglitcher to operate at a much higher speed than the original 48kHz approach. For a four times oversampling system, the converter must be four times faster than the original.

How much is too much?

The laws of physics dictate that high speed and high accuracy are difficult to achieve in the same circuit. The more often the converter must output samples, the less time it has to settle to the true value of the sample. High-speed amplifier circuits tend to be less linear and to have higher noise. It is much the same problem as making a car handle well at high speed while making it ride smoothly.

Tradeoffs must be made, and these tradeoffs must keep reality in perspective. It does little good to make a highly oversampled system to save on the analog filtering if the necessary converter is grossly non-linear.

Unfortunately, the oversampling ratio has become a marketing buzz word and manufacturers have been making equipment with 8X or higher oversampling, which have linearity errors and noise modulation of 6dB or more. The

If the sampling rate is raised to 96kHz, the filter will only need to reach 96dB of attenuation at frequencies above 76kHz.

only inherent performance benefit of an oversampled system is the linear phase characteristic, and it may be achieved adequately by a 2X oversampler, and very well by a 4X oversampler. Once a 4X oversampling ratio is reached, the analog filter requirements may be filled by a single op-amp filter, making further increases unnecessary.

The other advantages of an oversampled system, lower cost of digital integrated circuits and tighter control of frequency response, also peak out around the 4X point. This is because the digital circuitry becomes more complex, and the required tolerances on the digital filter coefficients tighten as the sampling ratio goes up.

As explained in the sidebar "Digital Filters for Oversampling Applications," a digital filter uses many multiples of sample values by fractional constants and adds the results of the multiples to obtain the output samples. The results of these calculations create numbers much larger than the original word size (usually 16-bit) that is input to the filter.

Quantizing noise is produced when the precision of the result is rounded down to a smaller word size, which is usually the same as the input data. This quantization is just like that which occurs when converting the analog signal to a digital signal. To eliminate this source of distortion requires that the summation be dithered with a digital noise source at the LSB of the output word. Dithering the filtered signal makes the quantization noise independent of the audio signal. If dithering is not used, the output noise will contain half a bit of quantizing distortion.

However, whether dither is used or not,

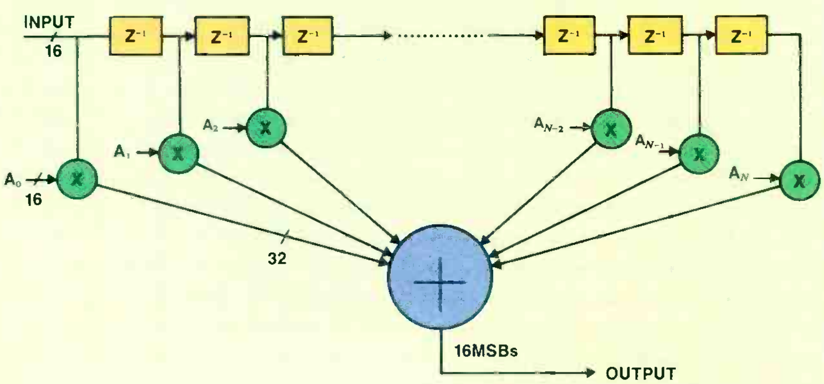
Digital Filters for Oversampling Applications

The topology of a digital low-pass filter commonly used for oversampling applications is called a *non-recursive* filter, or an FIR (finite impulse response) filter. The first name comes from the fact that previous filter output values never affect later output values. The second name is because an impulse at the input will cause the output to ring for a finite time. The figure illustrates the structure and operation of this type of filter. The input samples are fed into a long delay line composed of a series of single sample delays. The input and all of its delayed versions are multiplied by fractional constants and added together to obtain the output. By adjusting these constants, the response may be controlled down to a lower limiting frequency of approximately the sample rate divided by the number of stages of delay. At frequencies below this, there will be less than a full cycle of input signal in the delay line and the filter can no longer differentiate between input frequencies. In practice, the frequency limit is several times higher than this if reasonably accurate control of response is desired.

If a sample is equal to zero, its product is zero, and there was no need to compute the product in the first place. This fact can greatly simplify the design of low-pass filters for oversampling D/A applications. By assuming the unknown samples to be zero, they do not need to be used in computing the next output value. In a 2X oversampling application, for any given output sample, only half of the coefficient products need to be computed. Assuming the unknown output samples were the same as the previous samples would actually increase the computational overhead by two.

When used as a low-pass filter for oversampling A/D applications, the unused output samples do not need to be computed. This is because, unlike recursive or IIR digital filters, the output values are not used in any later computations. The major drawback to FIR filters is that the delay line length becomes excessive if low-frequency filtering is required. [For more information on digital filtering, see "True Digital Audio Mixers," February 1989, p. 28.]

The multiplication of two binary numbers results in a number with its width being equal to the sum of the two number sizes. For example, if the samples are 16-bit and the constants are 16-bit, their product will be 32 bits. If two of these products are added, the result will be 33 bits long. Four products summed results in 34 bits, eight summed gives 35 bits, and so forth. If the filter is 128 stages long, a common length in commercial ICs, an extra 7 bits will be required to handle the summation without introducing any roundoff errors. For the 16-bit data and 16-bit coefficient case, this gives 39 bits of data in the summation.



Digital FIR filter.

the noise floor will be increased. If the original data was 16-bit (with a one-half LSB noise floor) and the output is limited to 16-bit, the noise will be increased by 3dB. To maintain the noise floor at the same level as the input signal requires a larger word at the filter output than the input. This is one reason behind the 18-bit oversampling filters becoming popular on consumer CD players.

This effect is analogous to the increased noise floor that exists when analog filters are used that have the same signal-to-noise ratio as the input signal. If you have an analog signal with a 90dB signal-to-noise ratio and pass it through a filter that has a 90dB signal-to-noise ratio, the result will be a signal-to-noise ratio of 87dB. More specifically, a 16-bit filter output gives a 96dB noise floor. If it operates on an ideal 16-bit input signal, the result will be a signal with 93dB of dynamic range. Adding an extra bit of resolution (17-bit) to the output results in a filter with a signal-to-noise ratio of 102dB. Operating on an ideal 16-bit signal, this 17-bit output filter yields a −95dB noise floor.

Signal reconstruction

One aspect of the signal reconstruction process that is often overlooked by designers is shown in Figure 4. A low-pass-filtered square wave will overshoot on the leading corner, and a linear-phase low-pass will produce overshoot on both the leading and trailing corners. When a digital signal is reproduced using a D/A operating at the original sampling rate and an analog low-pass filter, all the overshoot occurs in the analog domain. The extra headroom in the analog filter is usually easy to obtain and no clipping occurs.

In an oversampled system, most of the filtering is done digitally. The overshoot will thus appear at the output of the digital filter. If the digital filter is designed to have unity gain for sine wave signals, it will clip on the overshoot it creates on square waves. To handle this increased dynamic range, an additional bit of headroom in the converter is required.

If a 16-bit signal is fed to the filter, a 17-bit D/A will be required to pass the output without clipping or attenuation. The filter gain must be arranged so that a full-scale sine wave into the filter only drives the D/A to one bit (6dB) below full scale. This extra bit becomes the headroom for signals from the filter. Unfortunately, commercially available 18-bit oversampling filters, with which the author is familiar, put both extra bits at the bottom of the digital word (to reduce the noise increase effect described earlier), not one at the bottom and one at the top (providing additional headroom) where they belong.

It is possible to use oversampling technology to reduce the required number of bits in the D/A converter—for a given number of bits of analog resolution at the output. This was done in some of the early CD players where 14-bit D/A converters were used in a 4X oversampling system to get 16-bit resolution. This technique is diagrammed in Figure 5. The idea is to use the bits below the ones fed to the D/A in a feedback loop to the input of the quantizer. These error bits are subtracted from the input signal to cause the error to average out to zero.

If a 4X oversampling is employed, the four 14-bit output samples can be designed into the filter to average out to a value in between the 14-bit resolution of the words by having the four samples be different values. If three of the output samples are at zero and one is at one, the average output will be 0.25. If two are at zero and two are at one, the average will be 0.5. The analog low-pass filter performs this averaging function since it cannot change at the 192kHz rate.

By appropriate digital filtering of the quantizing error signal, its energy may be pushed to high frequencies above the

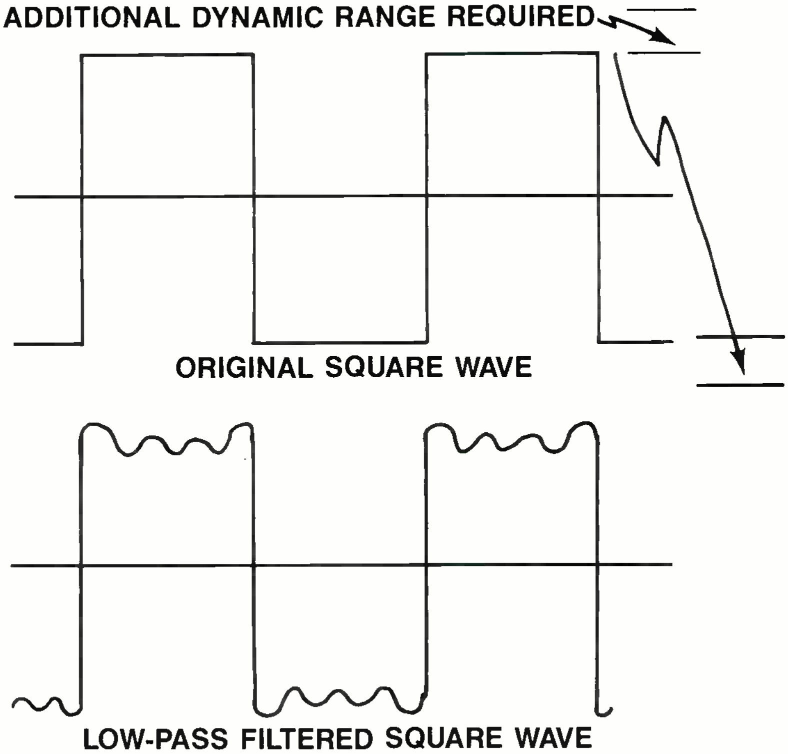


Figure 4. Overshoot in low-pass filters.

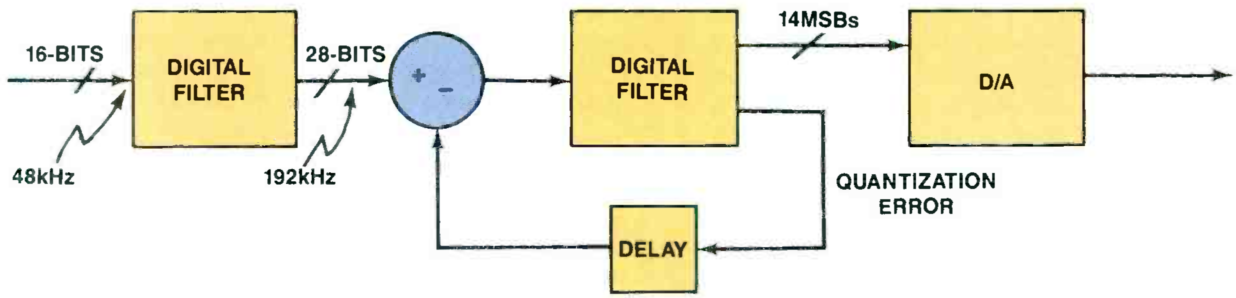


Figure 5. Noise-shaping D/A converter.

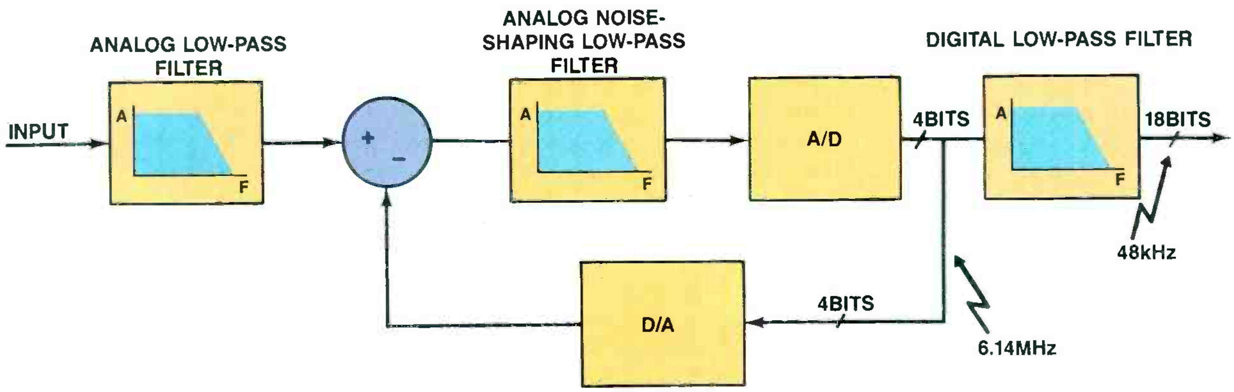


Figure 6. High-resolution oversampling A/D converter.

audio range. This allows the analog low-pass to remove it and gives lower noise in the audio band, if the path from the oversampling filter to the quantization filter can avoid the noise floor increase previously discussed.

Oversampling analog-to-digital conversion

The analog-to-digital conversion portion of the system has many similarities to the digital-to-analog portion. It is possible to trade off filtering in the analog domain for filtering in the digital domain. Recall that the function of the anti-alias filter is to prevent signals above the Nyquist frequency from entering the A/D converter and, ultimately, the audio band. Flat frequency response in the audio band requires that the filter pass all components below 20kHz with no attenuation. To prevent all alias components from entering a 16-bit digital signal, you would have to attenuate all signals above 24kHz by at least 96dB. However, if you restrict your concern to just the *audio band* in the digital signal, only signals above 28kHz have to be attenuated by 96dB or more. As with the reconstruction filter, these requirements are tough to meet with an analog low-pass filter.

By sampling the analog signal at 96kHz, for example, the anti-alias filter only needs to eliminate signals above 76kHz. If all we care about at the output are signals below 20kHz, this is a much easier analog filter to build because we have an additional 56kHz in which to reach the −96dB point. However, if the sample rate we want is 48kHz and we now have 96kHz, a sharp 20kHz low-pass that will roll off before 24kHz must be incorporated. Since the new sample rate is a fraction of the original sample rate (in this case, ½), all that is required is to throw away the unnecessary samples. This reduction of sample rates is commonly called *decimation*.

With the FIR digital filter design described in the sidebar, unnecessary samples do not even need to be computed. This lightens the computational load substantially. The filter complexity is therefore determined by the output sample rate and the sharpness (number of taps) of the design.

It should be noted that the term *oversampling* is quite appropriate for this case; the original signal is sampled at a rate much higher than dictated by the Nyquist criteria.

The sample and hold and the A/D must both function at a much higher rate in an oversampling system. As with the D/A portion, higher speed means lower accuracy. The trade-off again becomes one of anti-aliasing filter errors vs. conversion errors. The sample and hold must acquire the input signal in proportionately less time, and the A/D must convert this voltage to a binary word with a similar speed increase. This is much more difficult for the analog-to-digital converter than for the digital-to-analog converter.

To understand why, consider the operation of a conventional A/D. Most A/D converters use a D/A internally in a form of feedback loop to determine the binary output value. The D/A output is compared to the input signal and the bits are all set to zero. The most significant bit is then set to one and the input is again compared to the D/A output. If the signal is larger, the MSB is left at one. If the signal is smaller, the MSB is set back to zero. The second bit is set to one and the test repeated as with the MSB. This process is continued until all bits have been tested. The resulting bit pattern is the binary representation of the input.

This type of converter is called a *successive approximation converter* because the value is determined by successively testing each bit. For a 16-bit converter, this test must be performed 16 times, forcing the D/A to produce 16 new values during one input sample. Since the D/A is already running so fast, making it faster is very difficult. However, there is at least one A/D on the market, and used in audio equipment, that yields 14-bit accuracy at sampling rates compatible with 4X oversampling.

Oversampling A/D converters can benefit from the increased resolution offered by appropriate dither and filtering, as was discussed for the D/A. The performance benefits can be even more dramatic and much more economically justifiable. The concept is to run a low-resolution A/D converter (which can therefore be faster), at a very high sampling rate and to reduce the sampling rate with a digital filter. The extra resolution created by the large number of input samples averaged together for each output sample produces the needed resolution.

If the system merely raises the converter sample rate and low-pass-filters the data to obtain the desired sample rate, the improvement is not dramatic. Decimation by two will give an extra bit of resolution; by four will give two bits and so on. However, the noise improvement is proportional to the square root of the bandwidth reduction, which means the noise floor only improves by 3dB (or one-half bit) per factor of two reduction in sample rate. This can be improved by dithering the A/D converter with noise that has most of its energy at high frequencies in the range that will be filtered out. This will allow the 1-bit-per-factor-of-2 in decimation to be achieved. Non-linearities in the system cause the high-frequency noise to intermodulate and create low-frequency noise that will limit the ability to achieve even greater improvements in resolution for a given decimation ratio.

By making the A/D part of a feedback loop, even greater improvements in resolution may be achieved for a given decimation ratio. The basic scheme is diagrammed in Figure 6, and is used by one manufacturer in an 18-bit converter. The input has a feedback signal subtracted from it and the error signal is converted by the A/D. It runs at a very high sample rate, usually between 3MHz and 6MHz. The output of the A/D is converted back to analog voltage and subtracted from the input. Before driving the A/D, the error signal is filtered by a high-order low-pass filter. This has the effect of pushing the quantization noise up to high frequencies, where it will later be filtered out by the digital filter.

For small signals, less than 24dB below full scale, the converter is operating with only the least-significant bit. This makes the converter perfectly linear at low levels since there can be no mismatches between sizes of the A/D bits. These mismatches, commonly called *weighting errors*, are errors in the binary weighting of bits. The bits should each be larger than the last by a factor of 2, for example 1, 2, 4, 8 and so on. In practical A/D conversion, they might be 1, 1.995, 4.003, 8.011 and so on. As the signal level is increased, more bits are used, which causes distortion to appear. However, this distortion is largely masked by the higher program material level.

Converters of this type are often referred to as oversampling A/Ds, but the term *noise-shaping A/Ds* is more accurate. A simple oversampling technique cannot produce the dramatic accuracy improvements found in noise-shaping designs. Several other converters have appeared on the market recently using similar schemes but limiting the conversion hardware to one bit. This produces very good linearity at all levels but limits the ultimate resolution with today's technology to something around 16 bits. Given time, this should improve to yield higher resolution converters.

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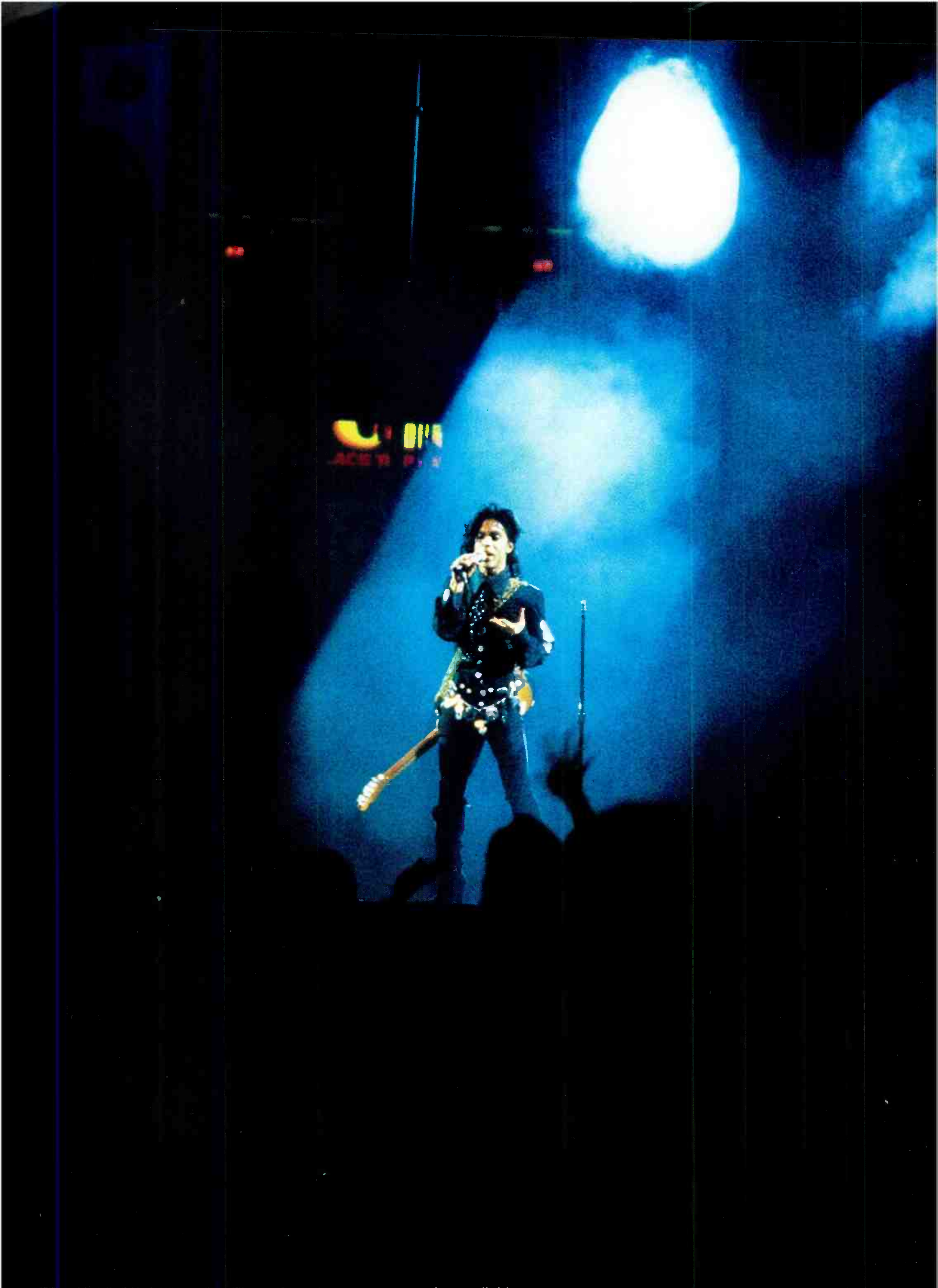

Prince on Tour

By David Scheirman

The sound system design from dB Sound features new E/V manifold loudspeakers and Crest amplifiers.

Touring concert sound companies and loudspeaker and amplifier manufacturers often work closely in the development of new, portable sound system packages. This is a welcome change from two decades ago, when high-performance, "roadable" amplifiers and speaker systems often had to be owner-built.

Today, all major professional sound companies rely on factory-built power amplifiers, with a few well-known manufacturers dominating this market. While many touring systems still feature speaker enclosures that are custom-built by the P.A. companies that use them, some firms feel that premanufactured

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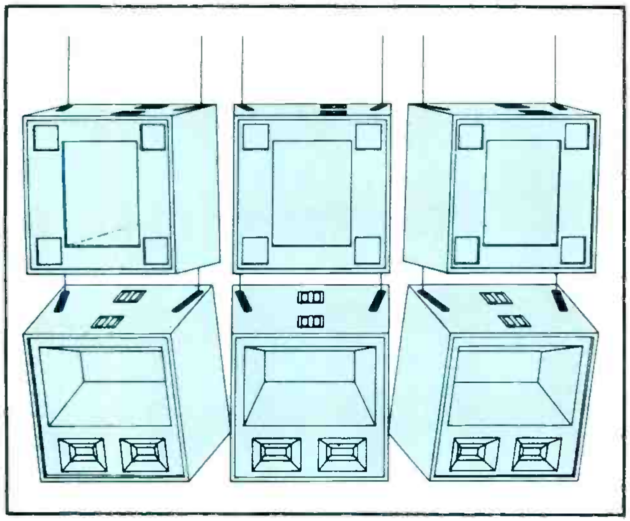

Figure 1. The E/V MT-4 loudspeaker system is designed to be used in multiple-cabinet hanging arrays.

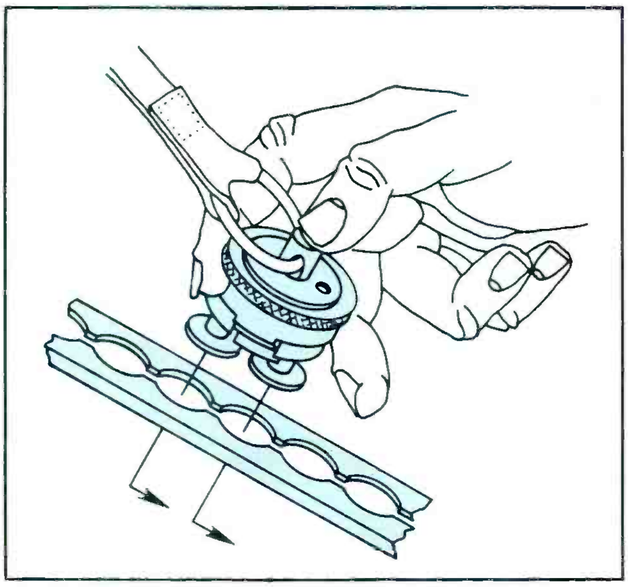

Figure 2. Aeroquip hardware fittings attach to the locking track on each enclosure.

Figure 3. House soundmixer Rob "Cubby" Colby, shown here at the house mix position.

systems are a good capital investment in today's competitive concert marketplace.

Embarking on his first American tour since "Purple Rain" in 1985, Prince fronted a show that was presented, in the round, to large audiences—and was one of 1988's major arena tours. Prior to the show's U.S. debut, a series of European concerts was held.

The concert sound system was designed and packaged by dB Sound of Des Plaines, IL. The veteran touring sound company had a considerable challenge to meet: how best to present high-fidelity sound for a very particular, production-conscious artist, while keeping the overall size of the system as small as possible. This was necessary both to keep trucking requirements to a minimum, and to cut setup and teardown times.

The tactics chosen by dB Sound included boosting the system's electrical headroom using several of Crest's new model 8001 amplifiers, the first large-scale use of E/V's MT-4 system and a relatively large and experienced sound crew. After checking out the entire system during production rehearsals at Prince's Paisley Park facility and field-testing it with major outdoor shows in Europe, the new dB system was shipped back to the United States to begin the American tour. The first dates were held in Chicago in mid-September. Concerts were staged across the country at venues such as New York's Madison Square Garden, Atlanta's Omni and Philadelphia's Spectrum. This article examines the dB system in action at the 19,500-seat Spectrum in Philadelphia on Oct. 18, 1988.

System development

dB Sound has relied on custom-built modular loudspeaker systems for many years. When the opportunity arose to work with the new Electro-Voice MT-4 system during 1986 and 1987, the company responded positively.

As dB Sound began to develop a speaker system several years ago, a dialogue began between the two companies' engineering departments. "We came to have a healthy respect for E/V's commitment to manufacturing better-sounding and more-reliable transducers," recalled dB Sound president Bruce Gordon. "One day, my partner, Harry Witz, and I realized that if they could really deliver a new generation of speaker products, and package them in a box that met our transportation and hanging needs, the result would be really powerful.

"We have found the 36"x36"x30" box size of the MT-4 system to be ideal for concert use," he explained. "Harry and I have always been interested in trying to get as much sound as possible out of a touring system that is based on building-block enclosures. In fact, dB has had several design evolutions of our own custom systems over the past decade or so."

Each MTL-4 enclosure houses four high-excursion 18-inch woofers, mounted magnet-to-magnet in what E/V calls a "radical manifold" format. The four speakers face the sides of the box, and each cone exits into a tuned chamber with porting at the corners of the cabinet. The MTL-4 bass system offers a stated frequency response of 40Hz to 225Hz and weighs 297 pounds.

The companion MTH-4 enclosure is the same size, but weighs 70 pounds more. This is because of the high number of transducers housed in each cabinet: four DL 10X 10-inch midbass woofers, four modified DH2 2-inch upper-midrange compression drivers exiting from a single flat-front horn and four DH2010 1.25-inch high-frequency compression drivers on a similar horn. The cabinet makes use of E/V's manifold technology by tightly grouping four drivers of each type through a single exit. This reportedly minimizes phasing and lobing aberrations. Recommended crossover points are 160Hz, 1,600Hz and 8kHz.

Identically sized, MTL-4 low-frequency boxes and MTH-4 low/mid/high-frequency boxes are designed for combination in multiple-cabinet arrays. (See Figure 1.)

E/V's hanging hardware option for each cabinet includes a pair of aluminum *L-tracks* (Aeroquip #34030-3) on each side. Double-stud ring fittings can be clipped into the L-tracks at different points, so that the hanging angle is adjustable. (See Figure 2). The cabinet can be suspended so that the 60° coverage angle is in either the horizontal or vertical plane.

System design for Prince

Initial field-testing of the new loudspeakers was conducted by dB Sound throughout 1987. The MT-4 system was considered as a viable option when initial sound designs were discussed in anticipation of Prince's 1988 tour. The ultimate decision to use the E/V enclosures rested with Rob "Cubby" Colby, Prince's house soundmixer. (See Figure 3.)

"When we first saw the artist's rendering of the proposed stage and set design for the 1988 tour, they showed a huge row of speaker cabinets going all the way around in a circle above the stage," recalled Colby. "Harry Witz and I wondered, 'Who thought of this?' We knew that they were going to want to do the tour in the round, but if the show had used the amount of speakers that were shown in the drawings, it would never have fit in the trucks.

"We needed to get our hands on a powerful, compact speaker system. When a show is to be presented in the round, there are often more stringent weight restrictions. Also, there is less room to hang a large system in the center of a building because of all the other production gear. We decided to bring the MT-4 system in for production rehearsals, and we based our sound design around it. With

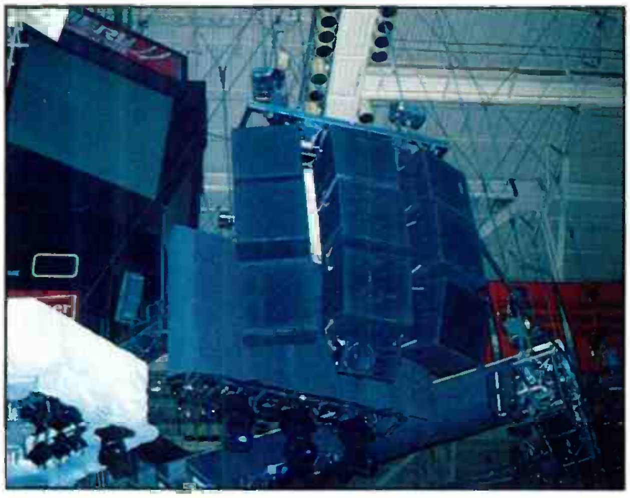

Figure 4. One of six flying arrays, each comprising a total of six MTH-4 enclosures and three MTL-4 enclosures. Additional MTL-4s were positioned on the floor and used as subwoofers.

Figure 6. MTL-4 enclosures, each housing four 18-inch speakers, were positioned on the floor around the stage and hidden with theatrical scrim.

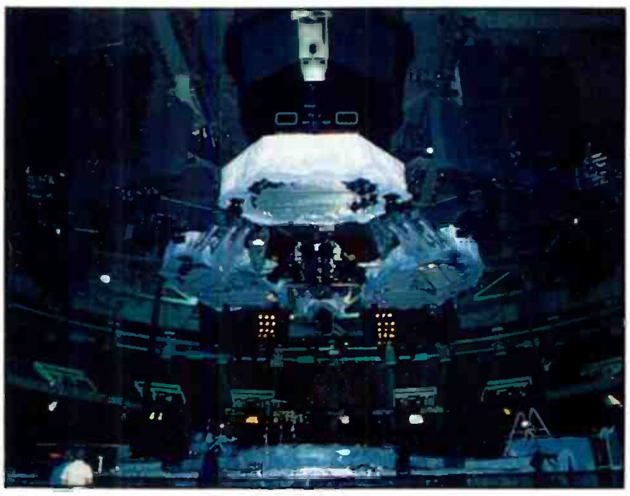

Figure 5. A custom-built octagonal stage was carried on the tour. The sound system was hung to offer 360° coverage.

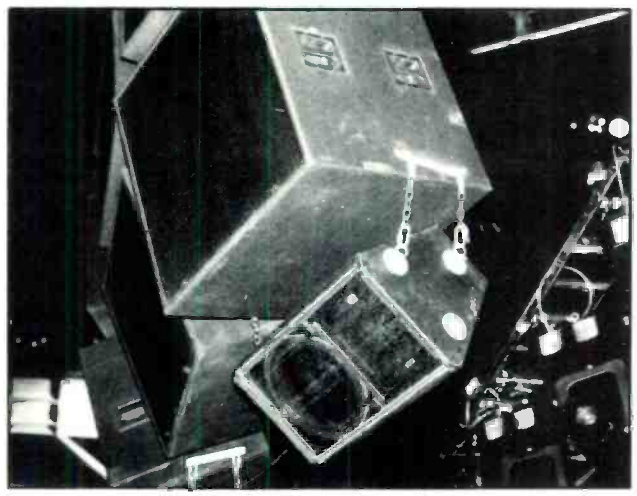

Figure 7. E/V's new Deltamax cabinets were suspended beneath the MTL arrays and used as downthrow devices.

E/V's help, we did computer modeling of the theoretical output of the system in different configurations and did polar-plot overlays to check the coverage. It looked like it would work."

The final system design for arena shows in the round included a total of six different flown arrays, each comprising nine enclosures (six MTH-4s and three MTL-4s). The three low cabinets were lined up in a center row, with a group of three mid-high cabinets on each end of the array. (See Figure 4.)

When the show's octagonal stage was set up in the center of a large arena, the six flown speaker arrays were rigged to form an approximate circle, with two arrays on the two sides of the stage that faced the ends of the building and consequently had the longest throws to seating areas. The input signals were configured so that a left/right audio image was apparent for most of the audience. (See Figure 5.)

To complement the 54 cabinets in the air (18 low and 36 mid/high), an additional 16 MTL-4 low cabinets were positioned on the floor in four different groups around the stage. These cabinets were used as subwoofers; a separate E/V crossover was fed a summed left/right input signal from an output matrix on the main mixing console. A Klark-Teknik DN360 graphic equalizer and a dbx 160x compressor/limiter were provided in-line, offering control flexibility for the MTL-4s on the floor. (See Figure 6.)

Adding another dimension to the sound of the show was a quad system positioned over the corners of the floor area. These loudspeaker cabinets were arrayed in a 360° coverage pattern and suspended from aluminum flying grids. Additional amp racks located beneath each of the four arrays powered this auxiliary system.

"We originally were putting the quad arrays all the way out in the corners of the arena," recalled Colby. "That arrangement sort of blew away the people sitting closest to them. We moved the arrays in so they were over the floor, right at the point where the stands began to rise. The sound of the show was much more consistent that way."

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Figure 8. *Deltamax cabinets were used for near-field coverage of the forward seating areas.*

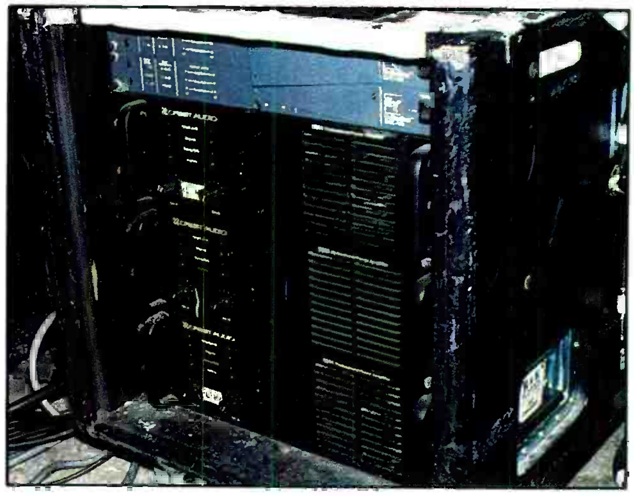


Figure 9. *Crest model 8001 amplifiers, housed in dB Sound's compact amp racks, were used to power the main sound system.*

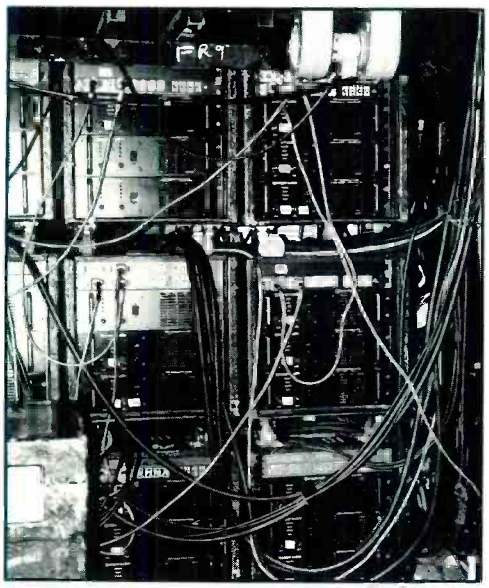


Figure 10. *Amplifier racks loaded with Crest 8001s were grouped together beneath the stage platform. Once the set was assembled and the show was underway, access was difficult.*

Figure 11. *A power distribution panel for the stage ac, including Juice Goose line conditioner/voltage regulator devices for each circuit.*

For near-stage sound coverage, dB selected the new DeltaMax DML-1152 full-range loudspeaker system from E/V. Sixteen units were employed, eight of them as downthrow devices suspended from the bottoms of the MT-4 speaker arrays. (See Figure 7.) Weighing 96 pounds and measuring approximately 30"x18"x16", the DML-1152s were hung beneath the MT-4s with nylon hanging straps.

Other DML-1152s were hidden behind theatrical scrim cloth and directed at the floor audience. (See Figure 8.) These trapezoidal enclosures were driven with the DeltaMax electronic controller. [See "Electronically Controlled Speaker Systems," January 1989, p. 22.]

Powering the system

dB Sound had chosen to supply a relatively compact speaker system for a show that required high sound-pressure levels. To keep distortion to a minimum, amplifiers that offered a high degree of electrical headroom were needed. Crest 8001 amplifiers were selected to energize the 70-cabinet MT-4 system and the supplemental close-coverage DeltaMax speaker units.

"The touring sound industry is very important to us," stated Craig Hannabury of Crest. "Our products face their toughest tests when put on the road with a major. The feedback we get from touring sound companies is a big help when finalizing the design of a new product like the model 8001."

Supplying a stated 1,400W per channel into 2Ω, the 8001 is one of the most powerful amplifiers available to professional users. The 8001s were housed three to a rack and operated into a 2Ω load for the bass speakers and 4Ω loads for the other components. Eighteen such racks were provided for the main system alone. (See Figure 9). Three amplifiers were available for every four speaker enclosures (two bass and two mid/high).

"The Crest 8001s are extremely powerful, and they have been very reliable," said Tom Nicks, a member of the dB Sound crew. "Due to set design, the amp racks were hidden beneath the stage in a pretty tight space that was hard to get at. We couldn't afford to have amplifiers that were constantly going down or giving us problems. The 8001s were trouble-free." (See Figure 10.)

There is increasing concern over the stability of the ac supply, now that digital electronic equipment has become commonplace both at the soundmixing positions and on the performance stage. For the stage equipment ac supply, a 3-phase power drop was hooked up to racks loaded with Juice Goose line conditioning and voltage stabilizing units. Each of eight separate stage circuits was supplied with a unit. (See Figure 11.)

System setup and operation

Several dB technicians worked with system crew chief Bruce "Slim" Judd to get the main system suspended and plugged in, while Jeff Nelson worked with the outer quad arrays. Tom Nicks assembled the front-of-house mix position.

"It may seem like a large crew, but on an elaborate show such as this, it's necessary to get everything done on time," advised Nicks.

Typically, the show loaded into a venue one day ahead of the first performance. Setup took about 10 hours. On show day, the call would be set at 10 a.m. "It's almost a 3-hour show with only a short intermission," noted Nicks. "There are a lot of mic and set changes during the show, so it takes lots of hands to keep everything moving. Sometimes, we don't find out what's going on during the second half until intermission—when Prince has decided what he wants to do for that particular crowd."

When preparing for each show, Colby found it advantageous to fire up the entire speaker system during the band's sound check. "I like to have the whole P.A. on during the afternoon, even though most of the system is pointing away from me into an empty room," he remarked. "Since in the round, the group covers the entire stage area, they need to have an accurate idea of the acoustical energy they'll be confronted with as the monitor levels are being set. It works out best that way."

House mixing equipment

Colby worked on a pair of Yamaha PM-3000-40 consoles, specially modified by dB Sound to have integral patchbays that allowed him access to all inputs and outputs. The patchbay assemblies were built to match the case styling of the PM-3000 and accept T.T. patchpoint cables. (See Figure 12.)

A wide range of signal processing gear was available at the house mix position. (See Figure 13.) A Klark-Teknik DN402 was channel-inserted on most vocal microphones, and Prince's mics received GML parametric equalizers. "That is the only outboard processing device on the input of his vocal mic," Colby said.

An Eventide Harmonizer and an AMS delay unit were available on auxiliary buses, as special effects for Prince's vocal mics. All effects units were returned in stereo with the exception of these two devices. Three dbx 900 racks and 12 channels of Drawmer DS201 noise gates were patched into many of the 57 mic lines returning from the stage.

Four Yamaha SPX-90IIs were used for effects on the horns, drums and background vocals. "There were no sequenced vocals on this show," explained Colby. "I got good flange and chorus effects from the SPX-90s. There was a drum machine used on about one-third of the show's tunes—acoustic drums being used to trigger digital sound samples. For the most part, though, what the audience heard was

Figure 12. A pair of Yamaha PM-3000 consoles were used at the house mix position. Each console was custom-modified by dB to include an attached patchbay unit.

Figure 14. The monitor mix position was equipped with both Soundcraft Series Four and Midas consoles. The Midas is pictured here. Note the Sennheiser wireless microphone receiver units on shelf above the console.

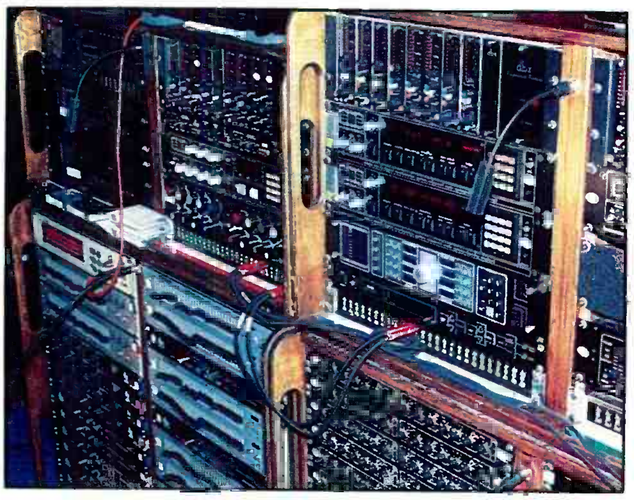


Figure 13. House electronics racks, located behind the consoles, held noise gates and compressors for channel-insertion and a wide variety of special effects, delay and reverberation units.

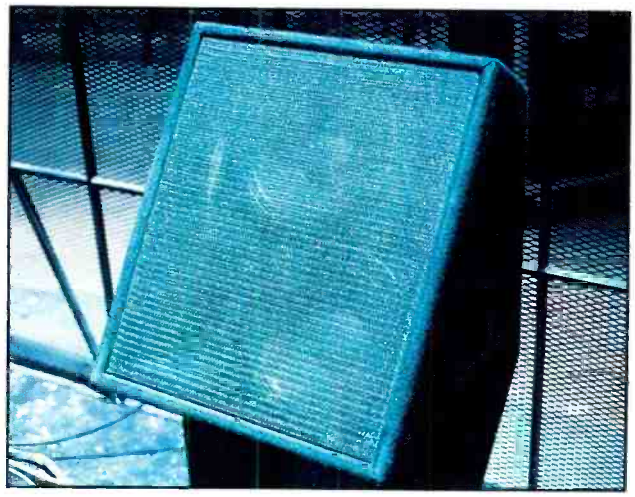


Figure 15. dB Sound's custom-built floor monitor enclosures were positioned around the octagonal stage to cover the performance area.

what was being created on stage."

The special quad system was fed prerecorded sounds stored on a Sony PCM 2500 R-DAT machine. A Midas console with eight inputs and four outputs, custom-packaged by dB Sound for the tour, gave Colby control over the signal going to the four outer speaker clusters. PR41A input modules were used in the short-frame console, along with Penny & Giles quad panners. Eight group outputs from the PM-3000 were fed to the eight Midas inputs.

"A lot of the spontaneous characteristics of the show came from the quad effects system," explained Colby. "For a particular show, we could really change the feel of a tune depending on what we mixed into it. And we had all sorts of special effects materials like street-corner noises and traffic sounds to work with."

Colby remarked that the jams used by the band during soundchecks and rehearsals often turned into material for upcoming album projects. For that reason, he recorded everything played either in the rehearsal studio or on stage.

To document live shows on the tour, Colby used VHS hi-fi videotape. A 6-channel mixer enabled him to blend the left and right board outputs, a pair of audience-area mics and the quad system effects mixes. Additionally, a Tascam 238 8-track cassette deck supplied tape copies.

Stage monitor system

dB Sound's Harald Danker was responsible for an extensive stage monitoring system that featured both Soundcraft Series Four and Midas Pro4 consoles. Danker oversaw 16 separate mixes, though there were never more than half that many performers on stage. (See Figure 14.) Although only 70 feet from the house mixing console, the monitor position was hidden from view by the stage set and Danker could not see Colby from his board position.

Different mixes were set up for the various stage "zones," including one overhead monitor mix. This featured one of dB's high-powered speaker enclosures pointing straight down from the center of the lighting grid. When doing keyboard work in the centerstage area, a hydraulic lift raised Prince 8 feet into the air, taking him into the sound field of the overhead monitor.

Most on-stage mixes featured a mirror-image pair of dB's custom slant monitors, with a left/right stereo program input. Loud, full-range music mixes were the rule. Other monitors included double 12-inch slants around the perimeter of the stage, providing a general vocal and music wash for performers working the edges of the stage. (See Figure 15.)

dB technician Peter Greenlund was responsible for stage cabling and microphone setup. All vocal mics were wireless, using Sennheisers operating in the 950MHz (UHF) band. Sennheiser mic elements and RF transmitters were combined with Yamaha headsets to make a hybrid, hands-off unit for some performers. The saxophone was equipped with an AKG mic and a Nady RF system, while the trumpet player used the Sennheiser system. Electric guitar and bass RF systems were by Nady.

Using the system

"In Europe there are a lot of speaker systems to choose from now," Colby noted. "We could have used anything. Some of the bigger P.A. firms in the United States are keeping systems over there now, but we felt it was worth it to take our whole system over and make sure everything was fine-tuned before the start of the American tour."

About 30 dates were done in Europe, some outdoors. "We had no real problems with individual components in the system," he recalled. "There were no damaged 10-inch speakers and no problems with the 2-inch drivers at all. We did find a few 18-inchers that were rubbing after all of those shows, and we replaced a few HF tweeters due to fluid loss *(ferro-fluid)*. All in all, the system held up just fine."

In the United States, Colby encountered a variety of concert venues. "The show sounded good at the Omni in Atlanta and at the Maple Leaf Garden in Toronto," he remembered. But the arena in Greensboro, NC, was a really tough room because of all the reflective surfaces.

When doing a show in the round, the sound system is closer to more people. And while it's easier to excite the room and have a good, clear sound, the challenge is to get even coverage of the whole space. Some of the venues can be difficult regardless of the speaker system used. This is because so many of the concert facilities are actually designed for sports events—it makes a difference acoustically.

Colby arranged the system so that matrix outputs on the PM-3000 fed the downstairs subwoofers, the E/V DeltaMax downthrow system, and the upstairs left and right array groupings. In each venue, a great deal of time was spent balancing the relative levels of these sends, and walking through the building to gauge the system's coverage.

The number of touring shows that are presented "in the round" has increased in recent years. And as trucking costs continue to rise along with audience expectations for production quality, systems such as the one assembled by dB Sound can be real problem-solvers for the touring concert industry.

RE/P

Circle (23) on Rapid Facts Card

Engineer/Producer Interview
Keith Olsen

By Bob Owsinski

Keith Olsen was one of the first Trident Di-An owners. He shares his views on engineering, production and technology, along with some of his experiences working with this unique, digitally controlled analog console design.

Heart, Pat Benatar, Ozzy Osbourne, Whitesnake, Santana, The Grateful Dead and Foreigner are among the platinum-selling acts produced by Keith Olsen. He has an interesting studio, complete with a molded-fiberglass front wall, dual Sony PCM-3324s, European-style gray decor and a Trident Di-An console. *RE/P* talked with Keith in the control room of his Goodnight LA studio the day after he finished putting the finishing touches on the new Kingdom Come album.

On having his own studio

***RE/P:** You've had your studio for about 10 years. Why did you decide to build your own studio in the first place?*

Bob Owsinski is a Los Angeles-based musician/producer/engineer and free-lance writer.

KO: The lost time, the trashed equipment from bands that were in over the weekend, the Hertz and Avis mentality of the commercial studios. It's human nature: you drive your own car or you drive a rental car, and you're always more comfortable in your own car. I didn't want to do my projects in a rent-a-car.

RE/P: *Who designed your studio?*
KO: I did.

RE/P: *Acoustics and all?*
KO: Yeah. When I built it, the main parameters were to make the room comfortable for the artist, and to make the control room comfortable for a producer, an engineer and an entire band, because it always seems that someone is stuck down in a corner somewhere and saying, "It doesn't sound right," because he's on a couch under the speakers.

I wanted to make it so I didn't have to do a lot of trapping and acoustic fixes. I wanted a really hard surface on the front wall of the control room, and something that wouldn't change size or shape because of temperature or humidity or the loud bass that control rooms are subjected to. So I decided to use a fiberglass mold for the whole front. And after 10 years, it doesn't rattle!

Many times I had gone into studios before I built mine, and you turn up the bass [guitar] and you hear a noise. What is it? Is it fret rattle? Is it the speaker cabinet you're micing rattling? Is it something else, or is it the control room? And you don't know if somebody blew a speaker during the session the night before. So I wanted to build it strong so it wouldn't change size or shape. Screws vibrate out; glue breaks. Welded steel was a good option, but it was a little costly and a little heavy. I found out that laminated fiberglass costs about the same as steel, and it looks a lot better.

RE/P: *What kind of monitors are you using?*
KO: Those are George Augspurger monitors. He designed the enclosures and selected the components. They're bi-amped, with absolutely no EQ on them. Any tuning that we've done has been at the speakers themselves. We bring in a B&K test set every couple of years and check the volume coming out of each individual speaker component, so if we need more midrange because of a slight dip, we just turn up the midrange [level at the speaker].

RE/P: *Do you have a MIDI room?*
KO: No. This is the MIDI room (motions his arm around the control room).

RE/P: *Are you heavily into sequencing or sampling?.*
KO: Oh, yeah. I've done entire albums that way.

RE/P: *I was under the impression that most of what you do involves live tracking.*
KO: It does. There's a reason why...in this business, it's called, "Let's go out and play music." That 4-letter word "play" was something that I liked to do when I was in the sandbox, and something I also like to do when I go to work. It's play! And the reason we're playing music is that it's fun. It's fun to play by yourself, but it's a lot more fun to play with other cats.

RE/P: *Have you ever considered more than one room so that you can work several projects at once?*
KO: You can only do one project at a time. I don't care who you talk to. You cannot think about Kingdom Come and Santana at the same time. The bands are staking their careers on your judgments, and they're paying you real well to produce, and so you have to give them 100%.

Day 1 was murder. Day 2 was a little uncomfortable. By Day 3, there was no problem.

RE/P: *You're mostly known as a producer, but you also have a reputation as being an engineer as well. Do you get to do much actual engineering these days?*
KO: About 50% of it. I do have an engineer (Gordon Fordyce) that I work with. It's wonderful to put on two hats and have one hat way off to the side so you can be a record producer and not have to worry about looking at the tape to see if it's distorted some place.

On the new console

RE/P: *What are the projects that you've done on the Di-An?*
KO: Let's see, I did Ozzy's [Ozzy Osborne], the new Kingdom Come record, the new Night Ranger, Fiona, a new band for Core Records [Olsen's label] called Time Gallery. I did two new songs that went on REO's "Greatest Hits," a Rick Springfield song for "Iron Eagle 2," and the new T'Pau album.

RE/P: *You've done a lot of work on it then?*
KO: Oh, yeah, from the moment they turned me loose on it. Thank God it works as well as it does because this place is going constantly; seven days a week. I've let them [Trident] have a day here and a day there to do their updates on it.

RE/P: *What made you purchase the Di-An?*
KO: I liked the concept. It really made sense to me. I've been twisting knobs all these years, using Neves and SSLs. Did you ever try to do a recall on an SSL and make it sound exactly the same? It doesn't. What if you did a mix a week ago and it was absolutely perfect except that you forgot something? With the Di-An, just insert the disk, and after about 15 minutes of resetting the program parameters on the outboard gear, it sounds exactly the same.

RE/P: *Which brings up the question that since there is usually so much outboard gear used in a mix, it must be tough to recall the parameters exactly. Have you found that to be a problem?*
KO: We never touch the input level controls on the outboard equipment. On the Di-An, all the output level controls, all the send and pan positions, and the master send positions can be memorized. Because there are no pots, everything is an absolute status, and everything, every parameter of the console, is resettable.

RE/P: *Twenty-four auxiliary sends offers a lot of flexibility. How many of these actually get used on average?*
KO: What I've done is dedicate aux sends to effects. Therefore, I don't have to patch, and all of my remixes are immediately resettable.

RE/P: *Besides the obvious differences, does it differ sonically from your previous console?*
KO: I can't believe the sonics on this console. The EQ sounds like a Trident A Range, and the punch sounds like a straight piece of wire. This console doesn't have any pots, so it doesn't have any caps in there to change the sound.

RE/P: *Since there are no knobs on the console, it seems like a radical departure for a veteran engineer. Was it difficult to get used to a new method of operation with the Di-An?*
KO: Day 1 was murder. Day 2 was a little uncomfortable. By Day 3, there was no problem. And on Day 3, I had to do something that I could never do before, and I was able to do it, and it made me love the sucker. I had to redo a mix because I forgot this little percussion part that was hidden in a track that happened only on the breakdown of the outro. I threw in the disk, punched in the part that I missed, and you cannot tell that it was a different mix. From that day on, I was in love with it because I had just saved


The Trident Di-An console.

myself half a day of my life. Now this was the third day that I had the console!

***RE/P:** Have any outside engineers used your Di-An?*
KO: There's been a whole bunch of engineers and artists that have spent some time on this console. Any artist that spends a day on this console would have a hard time going back to a knob console.

***RE/P:** Obviously, the repeatability of the digital control is nice. But does the digital control of the functions give sufficient resolution?*
KO: Oh yeah, I'm very happy with the resolution. The resolution was never, ever a problem. And the only time that you really need a lot of resolution is on the fader.

***RE/P:** What do you think of the moving faders?*
KO: The moving faders are great. The system is very easy to use and has features that I like a lot. A lot of times I get an artist who wants to ride his vocals during a mix because he knows how he sang it and where he put the inflection. So anybody can operate this.

***RE/P:** It seems helpful to have four banks of memory for the parameters. Do you use this feature much?*
KO: Yeah. The artist always says, "I know what I want to hear," so I say, "Show me." That's saved my life because it's a producer's dream. I can save my settings while the artist experiments with what he hears. That way, everybody's happy.

***RE/P:** This board is pretty long, and the various functions seem spread out across the board. Does that create any problems?*
KO: The layout of the new Di-An is better. They've moved some things around to make it easier. Mix center is truly mix center now. Actually, you don't have to go that far anyway. It's easier than a knob console, for instance, because you're always in the middle of the console when you're EQing. On Channel 1, which always seems to be the bass drum, you just hit the access button and you still stay in mix center to EQ.

On digital

***RE/P:** Are all your projects done digitally now?*
KO: Yes. I can find very little wrong with digital machines.

***RE/P:** You have a preference for the Sony machines. Why?*
KO: You need two machines for safeties. And the Sonys are cheaper. The Mitsubishis and the Sonys sound exactly alike. I had them all sitting in here with four producers, three engineers and four artists trying to figure out the difference between which bass drum was coming off which track, and which was coming directly in. They could all spot Direct In, but nobody could spot the differences in machines. I could definitely hear the difference between machines with and without Apogee filters, though.

***RE/P:** Do you have Apogees installed?*
KO: No. I don't know why, but I went for the first law of rock-and-roll: "If it works, don't screw with it." Everybody liked what the machine was doing and said that things sounded great. So I was a bit afraid to change anything.

***RE/P:** What are your feelings on the new Sony 48-track? Are you excited at the prospect of a new format?*
KO: No, and for the same reason that the 32-track Mitsubishi isn't appealing to me. You pay a quarter of a million dollars for the thing, or $175,000 for the Mitsubishi, and what do you do when you want to make a copy? You have to spend another quarter of a million for that inevitable time when you want to edit because you want to take out a chorus. Or you have to rent one at $1,000 a day. I have two 24-tracks that link up really well.

***RE/P:** What do you think about hard disk technology?*
KO: It takes too long to off-load and on-load data files. And if you're doing a proj-

Operation of the Di-An Console


The aux send routing panel.


A digitally controlled analog EQ panel.

Most functional controls used throughout the Di-An are similar in format and employ touch-responsive switches with positive tactile feedback characteristics. These are coupled to the appropriate read-out device. Momentary contact changes a setting by a single increment. Sustained contact varies a setting continuously until it reaches its operational limit. The clock speed of this variance is determined by the user.

Level is controlled using two keys arranged vertically and engraved with arrow symbols. When either key is pressed, the level changes in the direction of the key's arrow. The present level is read in decibels on the associated static bar-graph display. This system is used on all gain and level controls (including EQ boost and cut) except the channel input gain, which uses a numeric display.

Pan is controlled by two engraved arrow keys placed horizontally. These move the signal left and right of the stereo image in the direction of the arrow key pressed. Current pan position is displayed on the horizontal static bar-graph immediately above the button pair.

EQ frequency selection works in the same manner as the pan control (horizontally paired arrow keys), but is coupled with a numeric display of the frequency. The Di-An offers four EQ memories per channel. Complete alternative EQ setups can be stored in each of these memories and, like the console's other parameters, compared, recalled or changed at any instant in sync with time code, MIDI or manually by depressing one of the four memory keys.

Toggle switches are fitted as required with status LEDs showing their operative mode. (An illuminated LED indicates that the function is active.) These lights change to display the status of each active channel as it is called up. In addition to these "access dependent" displays, the Di-An incorporates a full routing indicator panel, eliminating the need to activate individual channels to ascertain their status.

Complete alternative setups (defined by the user) are maintained on disk and load automatically into memory for immediate console readiness on power-up. These can be programmed to change in sync with SMPTE time code, MIDI or manually.

Photo courtesy of Trident.

ect that doesn't have any sequencer information, then 16 tracks is not enough. And 32 tracks is extremely expensive and takes an enormous amount of time to back up and to on-load and off-load information. I've talked to some people who have used it and swear by it, and I'm sure that it sounds great. But when you get down to a cassette running at $1^7/_8$ips that's been duped off at 60-times speed, can you really tell the difference?

***RE/P:** Do you find DAT useful?*
KO: DAT's an interesting format. I like what they sound like, but I just wish that they would sound more like the original. I like the simplicity of it, the user ease, and the bells and whistles of it. I would not want to master off a DAT. I archive stuff on it. It's wonderful for archiving sampled sounds, and for rough mixes. I have a Sony 2500.

***RE/P:** Do you mix to a 1630?*
KO: No, I mix to two tracks on the same [PCM-3324] machine. That's another reason that remixes are so easy.

On preproduction

***RE/P:** I know that you're a great believer in preproduction. Have you ever had an artist write in the studio?*
KO: No. I used to do that. I gave that up after a few times. Why have a whole bunch of people wait around while you're trying to figure out the meaning of the song or writing one line of lyric? You can get a lot more done with your time.

***RE/P:** What's your method of preproduction?*
KO: The songs have to be done first. Then you go into a rehearsal studio with the band. The band really has to put it together by themselves first because if I put it together for them, it's going to sound like a different band. I want it to sound like the band sounds, then I'll go through and touch up this or that section, or throw some things out, or put some things in, and get it just so it's accessible in the marketplace but still sounds like the band.

***RE/P:** Do you have your artists do demos before they come into the studio?*
KO: With all of the home studios that artists have, and with all of the equipment available, the artists don't want to present a song unless it's been demoed. Sometimes it can lock you in, so you have to weed through it to make sure that the song is worthy and not just a whole bunch of production technique. So you have to go through and make sure that there is a melody there.

***RE/P:** Do you write with any of your artists?*
KO: Occasionally. Sometimes I find myself tearing up my own songs and giving artists a piece of this or that. There's a fine line between producing and writing.

***RE/P:** Do you ever play on any of the records that you produce?*
KO: Occasionally I play a part here or there. I try to keep my keyboard chops up. But if I really want a special keyboard part, there are some really amazing players out there. When you're competing with a lot of hot product out there, why not use the best!

On recording

***RE/P:** Do you usually record an entire band during basics?*
KO: Very, very rarely. You always think that you're going to cut the whole band but then you set up, and it's back to 3-piece basics.

***RE/P:** Here's a bit of a philosophical question. Do you enhance the sound that the artist gives you, or do you approach recording from a documentation standpoint?*
KO: It depends on the sound. If a guitar player has the most amazing sound on his amp, but it just isn't coming across on the speakers, then you've got to enhance. And most of the time you have to enhance the drums. In the studio, a snare drum might sound great, but in the control room, against the other tracks, it might sound pretty tacky. The microphone is just seeing this little piece of stretched plastic with some pieces of wire on the bottom that is being hit with a stick the size of a baseball bat—and it doesn't hear the concussion of the room. So you've got to enhance.

***RE/P:** Tell us how you plan your overdubs. Do you like to complete all overdubs on a given cut before moving on to the next song or do you try to cover all the ODs that have similar instrumentation?*
KO: I don't get to do this very often because a lot of bands don't understand this concept, but I love to start song No. 1, cut the basics for song No. 1, overdub song No. 1, vocal song No. 1 and mix song No. 1, then put it aside and start on song No. 2. I love having that focus of all your energy and all your ideas on that song. Every time I've done it, it's gone like that (snaps fingers). That's how we just did the Kingdom Come record, and we finished it in 39 days, which included a couple of breaks. This only works if you've got a band. If you don't have a band, then forget it, because it's hard enough getting all your players together to do one track at one time.

***RE/P:** What's your approach to mixing? How much outboard gear do you use?*
KO: I've got a bunch, but I end up using three reverbs, three sets of delays, and two harmonizers. I like having a few longer delays and a lot of tight delays to broaden the sounds out a bit. And I use the longer delays instead of reverb. Sometimes I time them out—sometimes there's no relationship. The medium one I always time. The snare always has its own chamber, a highly modified Echoplate that's gated.

***RE/P:** How loud do you monitor? And do you monitor differently when tracking and mixing?*
KO: I like working on the NS-10s at a reasonable level throughout the project; about 85dB to 90dB.

***RE/P:** What type of system do you reference your mixes with outside the studio, for example, at home?*
KO: My studio is my reference. When it sounds right here, I know it will sound right outside of here.

***RE/P:** Do you have a favorite piece of gear?*
KO: Those two Sony PCM-3324s. They're the handiest things that I've ever had in my life.

In general

***RE/P:** Do you prefer working with established artists or new artists?*
KO: I like a good mix of both.

***RE/P:** After all these years, do you ever feel that you have a formula?*
KO: Sometimes. It seems I have a certain recognizable sound that I always end up with.

RE/P

Digital Signal Processing: Living up to the Promises?

By Gary Hardesty

Beyond the mystique of digital audio, what are manufacturers doing to improve sonic quality?

Not long ago, the words "digital audio" brought to mind images of the future, a future much better than the world of analog audio. The audio industry promised digital recording of incredible sonic purity. Envisioned were digital edit stations and digital mixers, all vastly superior to their analog precursors.

As more companies introduced digital products, however, it was found that while some promises were kept, some were not. The digital product was often noisy, and sounded different, maybe not as good as its analog counterpart. When it came to digital recording, a lot of people came to the conclusion that perhaps the old analog means were not so bad after all. And with the development of modern noise reduction, analog recording seemingly offered all that digital could.

The fact is that until now, digital designers have focused not so much on ways to raise sonic quality, but rather on ways to implement functions that could not be easily implemented in the purely analog world (such as reverberation simulation—one thing digital can usually do better than analog).

But as we approach the end of the decade, isn't it time that digital audio lives up to all of its promises? Currently, with respect to overall audio performance, there are no utilitarian digital audio products on the market that even come close to their analog counterparts. (By utilitarian, I'm speaking of equalizers, crossovers and compressor/limiters.)

Sixteen-bit digital audio can provide no more than 96dB of dynamic range. Contrast this with the average analog equipment, and you are short by at least 10dB.

Sixteen-bit digital audio can provide no more than 96dB of dynamic range.

If you are thinking in terms of an effects device, like reverb, then you probably don't need 18 bits. For this kind of application, in which apparent dynamic range is improved by mixing the reverb signal with the direct signal, 16-bit audio is usually fine. But this is not the case with an equalizer, crossover or signal delay. In these applications, the lack of performance cannot be buried in a mix.

Frequency response and phase linearity are two more important issues. The digital format clearly gives up a lot if phase and frequency response are not as good as they are with analog. (In all fairness to manufacturers, it is not yet entirely clear with which of these, and to what extent, we should concern ourselves.)

Certainly the lack of adequate technology in the area of A/D and D/A converters has not helped. But for their part, consumers have only recently begun to demand increased audio performance—perhaps because it has taken this long to get beyond the mystique of digital audio and settle down to realities.

Along with this, the digital audio manufacturing community is finally realizing that to stay in business, they must create products whose sonic performance rivals that of analog. As a result, I believe you will soon see digital products that offer the following:

1. Dynamic range of at least 104dB.
2. Excellent phase- and frequency-response linearity.
3. Greatly reduced degrees of low-level distortion.
4. Frequency response beyond 20kHz.
5. New levels of performance in functions such as limiting and EQ, which were previously restricted to the analog domain.
6. Interconnection standards throughout the industry.

Gary Hardesty is president and director of engineering at Audio/Digital in Eugene, OR.

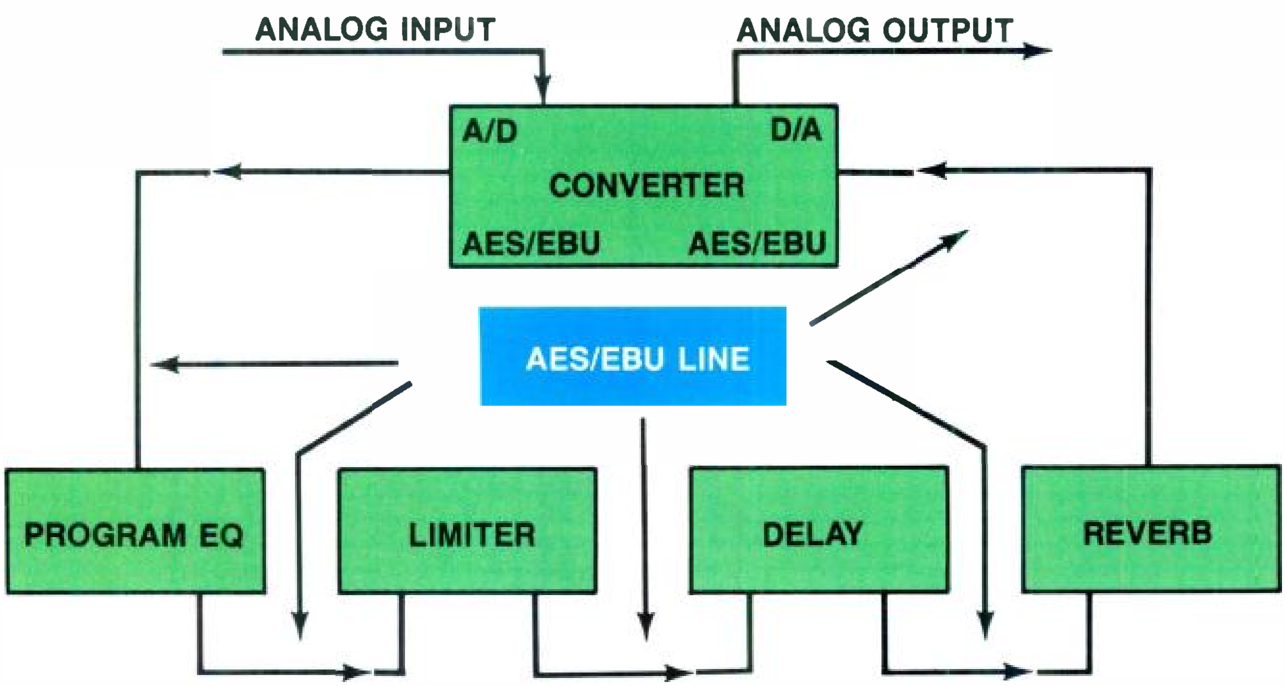


Figure 1. *Serial connection of several DSP devices using a single, stand-alone A/D/A converter.*

Until recently, there were several major problems to overcome when considering the introduction of a digital product, even for companies already producing digital products:

1. Lack of reasonably priced, high-resolution A/D and D/A converters suitable for audio use.
2. Lack of off-the-shelf DSP chips fast enough for implementing complex algorithms at full audio bandwidths.
3. Lack of industry-accepted interconnection schemes that would allow one to remain in the digital format.
4. Limited understanding of the various forms of degradation in the digital domain, such as quantizing distortion.

Sonic performance requirements

Many people now believe that digital audio systems used for "mainstream" processing must be at least 18-bit, or about 104dB in dynamic range, to be acceptable in professional applications.

The actual method of processing the digitized audio is generically called *digital signal processing*, or DSP. The type of DSP depends on the algorithms used. As might be expected, the hardware and software of such systems is quite complex and has prevented many companies from getting involved.

In the past, cost effective DSP for high-volume products was best done by custom DSP hardware. This is the method used by most of the companies that make digital reverbs. Of course, this is expensive and requires a great deal of understanding of the task at hand. But this should change in the near future with the availability of fast, off-the-shelf DSP chips that are inexpensive enough for general use.

With the recent introduction of the dbx 18-bit A/D converter, followed by introductions of 18-bit D/A chips from companies such as Burr-Brown and Analog Devices, we now have A/D and D/A systems at our disposal that can approach the best analog quality. Cost, however, is still a major concern. These devices are still expensive, but we can expect prices to drop as use increases.

The Motorola 56001 seems destined to become the de facto DSP chip of the industry.

DSP chips

Some semiconductor manufacturers have introduced DSP chips that seem to be suitable (fast enough, with an instruction set that lends itself to easy audio use) for full-bandwidth audio. In particular, Motorola, Yamaha and AT&T make such chips that are currently being used. Present at the most recent AES show in Los Angeles were companies exhibiting DSP development boards—all using the Motorola 56001 chip. And, Motorola announced a lower-priced version of the chip, as well.

The 56001 has been mentioned in a number of AES papers, and by simple virtue of the number of companies using it and the support software behind it, the 56001 seems destined to become the de facto DSP chip of the industry.

It seems clear that soon someone will offer a general-purpose, high-level, audio-based developers' software package to support the chip in depth. Rank Corporation has announced such a package, but it is not available yet.

Digital interfacing

If we are going to have a digital world, all manufacturers of such equipment need to follow some type of control and interconnection standards. This, too, is beginning to be understood. Some time ago, the AES introduced the AES/EBU protocol for the interconnection of digital audio devices. This appears to be the most likely protocol to take effect, because many major manufacturers have decided on it as their format, despite such other attempts as MADI.

At the time of this writing, several companies have made available AES/EBU chip sets to allow easy implementation of the standard. This will allow the introduction of digital audio products that have no on-board A/D or D/A converters, and will certainly help bring down costs. You will simply serial-connect as many DSP devices as desired, then plug into your audio converter. (See Figure 1.) This is similar to today's methods of interconnecting audio equipment, except you will stay in the digital domain and, theoretically, have no signal degradation.

When it comes to controlling such equipment, salvation may also be close at hand. This year, several companies are planning to introduce MIDI-controlled industrial products. (By industrial, I'm speaking of mainstream products, such as distribution amps and program equalizers.) Several companies met at the Los Angeles AES convention for the purpose of detailing a control protocol that is currently RS-422- based, but may swing to MIDI, as well.

Based on the large number of articles that have appeared in the last two years addressing the concerns of degradation in digital audio, it might appear that preventing such degradation is a simple task. (For more information, see "Performance Aspects of Digital Oversampling", p. 20.) However, as many companies have discovered, defining the problems and correcting them are not the same thing.

Crystal Semiconductor, dbx, Sony, Burr-Brown and Analog Devices, to name a few, all have audio-oriented A/D and/or D/A chips that are becoming, more and more, full subsystems, thereby greatly easing the task of engineering these critical sections. A full subsystem incorporates on-board sample and hold, voltage references, comparitors and D/A circuitry. Before long, someone will introduce a small, high-resolution A/D/A set with built-in anti-alias and reconstruction filtering.

Making the digital transition

A traditionally analog company that has recently introduced a full digital product is Valley International. President Norm Baker says, "Building a digital product is something we have wanted to do for some time. But we did not want to get involved in the traditional areas of delay and reverb, in which there is already so much competition.

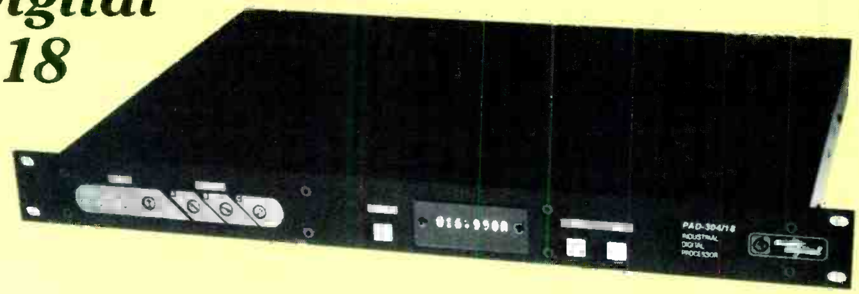
"What we've done is develop algorithms that would allow us to do signal processing in areas like compression, expansion, noise gates, even equalization. We felt we had to impart the same quality in digital that our analog schemes had delivered. We are able to do things digitally that we could not do in analog."

Baker says that the real difficulty for Valley was to implement its established analog algorithms with DSP. He believes 16-bit audio is sufficient for now, but realizes that, ultimately, more resolution is important. Valley has chosen to work with the AT&T DSP16 chip.

The Valley digital product is the Digital Dynamics Processor (DDP), and a companion serial A/D/A box. This product performs multiband spectral modification and compression for tailoring broadcast or recorded audio.

White Instruments, a long-time manufacturer of analog equalizers, is also interested in the digital domain. Emory Strauss, director of marketing, says White is interested in digital because of the versatility of digital products. Strauss sees a close parallel to the personal computer

The Audio/Digital PAD-300/18



Audio/Digital has gained a wealth of information because of our early development of digital products for the professional market. One of our products is the industry's first 18-bit delay for alignment and synchronization applications. The PAD-300/18 features a dynamic range of at least 105dB. With a high sampling rate and simple analog filtering, phase and frequency are very linear—producing analog-like sound quality.

The PAD-300/18 has one audio input and three audio outputs. The delay range is 0ms to 650ms, in 10μs steps. Distortion is less than 0.01% at 1kHz, and the frequency response is 10Hz to 22kHz $\pm$0.5dB.

Its RS-422 ports follow the proposed IED (Innovative Electronic Designs) standard for complete remote control, including gain, delay, mute/bypass and page control. A rear-mounted barrier strip allows page control for up to 18 pages of settings; pages are also accessible via RS-422 ports.

An 8-character LED dot-matrix display shows all menu prompts, as well as diagnostics. There is a front panel bypass control via gold contact relays. A rear-mounted security enable/disable means there are no code sequences to memorize.

A failure-detection and relay-controlled reset/bypass detects and reacts to either the loss of ac power or the partial loss of the dc power supply. (In the latter case, a failure message will appear on the display.) When power resumes, the processor is reset and the delay re-activated.

The processor continually monitors the quality of the incoming ac power and reacts, if necessary, with either a display indication or a mute/bypass.

world, with software being the primary difference in various DSP machines. How soon they get involved depends on performance.

"The industry is going to go through an interim stage where it compromises performance for flexibility," he says, adding that the real hang-ups are the A/D and D/A sections. "Sixteen-bit would be a big step backward.

"The most important thing that our customers have to do is demand that we manufacturers get together and establish some standards, because in the forseeable future we are going to be stuck with 16 and 18 bits. Our customers should demand that we, the manufacturers, get together to figure out our interconnections. And, we need to do this in a very, very big hurry."

Dane Butcher, president of Symetrix, says, "The primary impetus for getting into the digital domain is the availability of a new generation of DSP processors. It is something we just couldn't pass up anymore. There are certain things that we can still do in the analog domain more inexpensively, and perhaps to even higher levels of performance, particularly in the area of dynamic range. But bottom line, the technology is getting more accessible, and there are more and more exciting, powerful things that can be done with the new DSP parts."

In November, Symetrix introduced a prototype DSP machine in the form of a multitrack digital recorder. This is the second digital product for Symetrix (the digitally controlled ambient noise sensor being the first). Will the Symetrix DSP machine be 16- or 18-bit A/D/A? "We have designed for 18-bit data paths, but at present we are using 16-bit," says Butcher.

What kind of risks are involved?

According to Butcher, "There is a great deal of up-front investment; manufacturers are essentially buying into the technology. The industry is building a new technology base and we are paying out huge sums of money to do so. If there is a given function you are trying to accomplish with a product, you have to look at the choices on a case-by-case basis, and make the decision to go analog or digital. The moment you say you want to do time-domain-based processing, the choices are very limited in analog, and you have to turn to digital technology."

Symetrix has chosen the Motorola 56001 as its DSP chip because "at the time it came out, it had more speed per dollar than anything else on the market."

This is a representative sample of manufacturers only, and does not include all the companies considering entering the market with digital products.

Although digital can offer a lot of excitement, we are still some time away from that excitement being translated into real "mainstream"-type products. But it is clear that the manufacturers are finally listening to the customer and not allowing the digital excitement to replace the quality of analog products. Over the next two to three years, you can expect to see innovative companies introduce all-digital products that not only rival analog quality, but offer even more performance.

RE/P

Sampling vs. Hard Disk Recording

By Dave Erb, Dave Frederick and
Steve Cunningham

Applications, strengths, weaknesses and capabilities of digital sampling synthesizers and multitrack hard disk recording systems.

Amid the constant deluge of new professional audio products, new systems continually emerge that, by virtue of their capabilities, enhance our productivity. The digital revolution, in particular, has brought with it many leaps in audio technology. Two of the more significant innovations are digital sampling and hard disk recording. Both processes allow recording and editing of digital audio, either in random-access memory (RAM) or on hard disk. But which technique is most useful for a given job? How can a given digital product be applied? Can existing technology integrate smoothly with new processes? Valid questions, all.

During the past several years, sampling synthesizers have become common in many different audio applications, including musical composition and performance, sound effects recording and editing, and vocal or instrumental overdubbing. And soon, it is anticipated, digital hard disk recording technology will become more affordable and accessible to many in professional audio production. Samplers and hard disk recorders share many functions and applications, but they have different strengths and weaknesses, and the combination of the two can sometimes exceed the sum of the parts.

Dave Erb is director of software engineering, Dave Frederick is a product specialist, and Steve Cunningham is vice president of marketing for WaveFrame Corporation, Boulder, CO.

Digital sampling

Using internal RAM, a sampling synthesizer allows for the storage of digitally recorded sounds and the triggered playback of these sounds at various pitches and amplitudes—usually under the control of a MIDI keyboard or similar device. Most samplers include functions traditionally found on synthesizers, such as low-frequency oscillators, envelope generators, and low-pass filters. Unlike other types of synthesizers, samplers create sound only from the recorded material in memory, although modifications can be made with the synthesis portion of the instrument. During playback, typical modifications to a signal include pitch-shifting, pitch or amplitude modulation, and looping (like a tape loop) to sustain the sound beyond the original sample's length.

What goes in, must come out

Samplers really only have two aspects: recording and playback. For most musical applications, you either use sound samples that are available from the manufacturer and third-party vendors, or prepare your own samples. In live performance, you treat the sampler strictly as a musical instrument, ignoring the recording functions.

However, samplers are not limited to musical applications. The fact that they can instantly play sound samples in response to external signals, such as MIDI or time code cues, makes them useful in other areas as well. For example, if you're spotting sound effects for video, you can load the effects for a given segment into the sampler and set up a trigger-time for each effect in an edit decision list (EDL). By triggering the effects from a time code track on the video, the sampler will synchronize the sound effects track with the video on playback. And you can quickly check the accuracy of your effects spotting and edit the placement of each effect by adjusting a value on the EDL.

Current sampling technology provides high-quality audio and flexible synthesis. Most of the 16-bit samplers on the market today use linear encoding schemes that usually provide 96dB of dynamic range, and many support stereo signals. Some systems provide a graphics interface to permit easy and intuitive programming of signal parameters, such as envelopes and loop points. A few of the high-end sampling synthesizers even permit waveform display, cut-and-paste audio editing, precision (microtonal) tuning, graphic envelope editing and reel-rocking, among other advanced functions. (See Photo 1.)

All of these features make the job of creating and editing sounds quicker and easier than ever before, and allow for the radical tailoring of a sound without degradation in audio quality. These

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Marvin Caesar, President

At Aphex we have a problem with the President. Marvin Caesar wants everything the company makes to be the "best." Marvin is *not* an engineer, he is an audio zealot who doesn't understand the word "impossible."

When engineering approached him with a design for a gate, he wasn't satisfied. He wanted the ultimate gate, an *expander/gate* that was extremely fast, perfectly click-free, loaded with features, with traditional Aphex audio quality.

Impossible with available technology they said. Marvin wasn't satisfied.

So, the engineers developed a new VCA, the Aphex VCA 1001. Then they created the Aphex 612 Expander/Gate. Marvin was finally satisfied.

In fact, he is so confident that the Aphex 612 is the world's best expander/gate he is offering $2500 to the person who finds a better one. If you think you know of a commercially available expander/gate that can begin to compete with the Aphex 612, write for full details of this offer.

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Here are a few highlights of the world's best expander/gate:

- Attack time—variable from less than 4μSec (with no clicks!) to 100mSec
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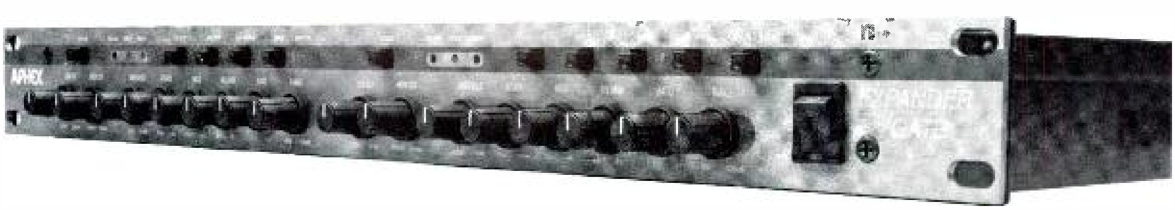
- Range—0 to 100dB
- Bandwidth—5Hz to 100kHz, + 0, − 0.2dB
- THD @ + 20dBm—0.006%
- IMD (SMPTE) @ + 10dBm—0.006%
- Noise and hum—fully attenuated − 94dBm
- Servo-Balanced Transformerless Inputs and Outputs

Can you beat that?



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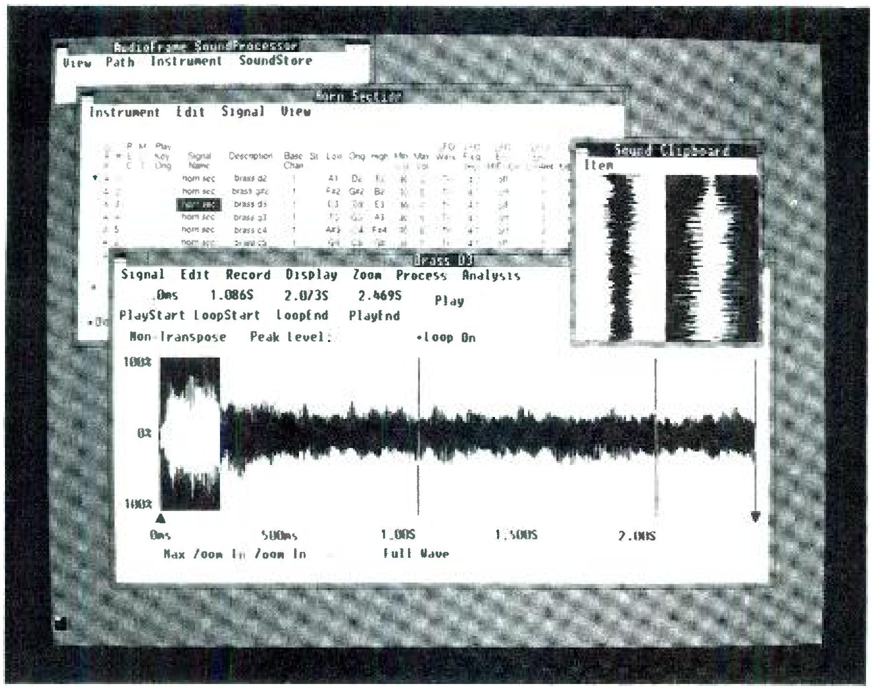

Photo 1. A sound file with the on-screen waveform editing feature.

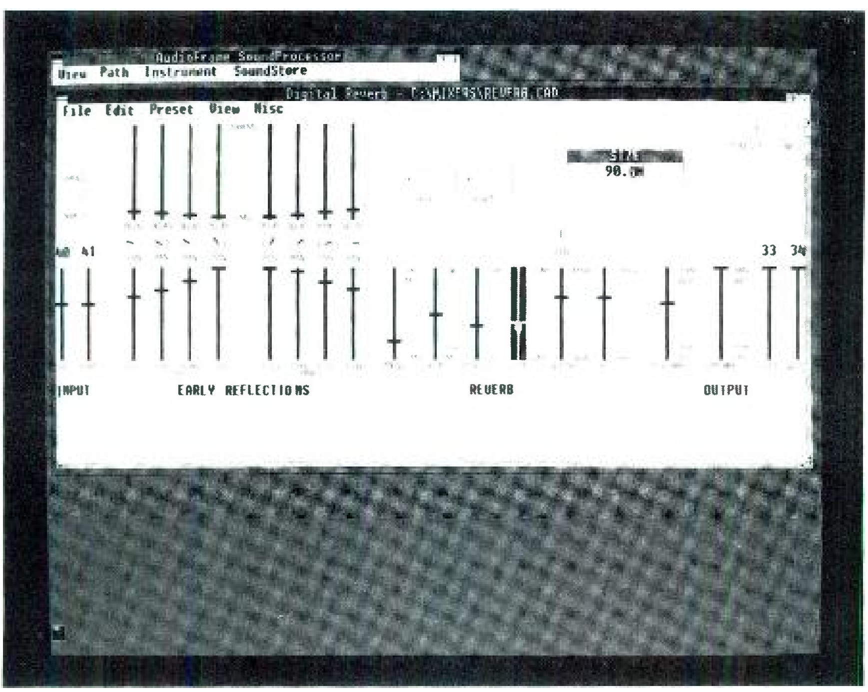

***Photo 2.** Digital signal processing effects such as EQ, delay and reverb can be accessed while remaining in the digital domain.*

capabilities are valuable in creating modified samples of musical instruments, in editing dialogue, and especially in sound effects design and editing, where the object may be to tailor an effect for a particular purpose or to create a totally new sound effect that doesn't exist in nature.

Unfortunately, all of these functions don't come cheaply. While RAM provides an extremely fast and flexible environment in which to work with audio data, it is a somewhat costly commodity. And 16-bit recordings use up sample memory quickly: at a sampling rate of 44.1kHz, one minute of mono sound occupies about 5Mbytes of RAM. So, while it might be possible to purchase a sampler with enough memory to record and edit an entire album or film, the cost would be prohibitive.

Clearly, samplers are best suited for working with multisampled musical instruments, individual sound effects and sound bites. For the professional who wants the flexibility and the digital recording quality of RAM, but needs to work with large passages of audio, the alternative is hard disk recording.

Hard disks

Hard disk recording systems support real-time storage and retrieval of one or more tracks of digital audio—on a magnetic storage device. Most are designed to emulate 2-track or multitrack tape recorders, and most offer some of the basic editing functions found on samplers.

Hard disks offer access times that are only slightly longer than RAM at a far more economical price per track-minute of recording time. For 16-bit, 44.1kHz digital audio recording, hard disk memory cost is about 1/100th the price of equivalent RAM ($50/minute vs. $5,000/minute), and is considerably less expensive than digital audio tape. Accessing a sound on a hard disk recorder takes only a fraction of a second. And though they don't access as fast as RAM-based samplers, they are many times faster than accessing sounds from a reel of tape.

Although the basic purpose of a hard disk recorder is to emulate a traditional tape transport, it can offer several advantages over tape. The recorded tracks are independent, and the operator can quickly and easily cut, slip or bounce any track or group of tracks without affecting neighboring tracks or waiting for any audio copying to occur. Because hard disks are random-access devices, the operator can locate or edit any portion of any track in a fraction of a second. Some systems also permit non-destructive editing, allowing the operator to audition a set of edits and start over if the results are not acceptable.

What's the hitch?

While hard disk recorders are especially good at recording and playing long passages of audio, most do not include editing functions more sophisticated than cut-and-paste, and simply cannot carry out the kind of radical sound modification possible on a sampler. And because it has no sound generation capabilities of its own, a hard disk recording system cannot easily be used to compose or perform music.

Another problem presents itself when a project is finished and it's time to begin the next one. This is easy with tape recorders; you simply remove the reel of used tape and put on a fresh one. With hard disk recording, however, this type of swap would require spare disk drives and a complete reconfiguration of the system.

Think of a hard disk as a reel of tape that cannot be removed. To save your work and move on to the next project, you have to re-record the sound to a removable tape before erasing the hard disk. Hard disk recorders will usually incorporate a streaming tape backup system, but it takes time—depending on the system, anywhere from 30% to 200% of real time.

There's every indication that erasable optical disk drives, with performance comparable to magnetic hard disks, will become available in the near future. Because they use removable media, backup (storage) and restore (restore to on-line conditions) procedures will only be necessary for long-term archival purposes.

Over the next few years, media manufacturers plan to increase the capacity of optical disk drives, so that a single removable optical disk could hold 600Mbytes (about two track-hours of 16-bit, 44.1kHz digital audio).

Digital audio format compatibility presents a more difficult challenge to the transfer of audio between digital devices. In the past, transferring digital audio between machines with different sampling rates and formats required that the audio be converted to analog at the output of one device, sent via audio cable, then redigitized at the input of the next device. This process introduces two additional D/A/D conversion steps that add noise and distortion to the signal.

Let's assume you're using a sampler and a hard disk recorder to produce your projects. How do you get these machines to communicate digitally while preserving the sound you've worked hard to create? There are only two paths. First, if your digital devices can communicate in one of the various accepted formats, or if you have a separate format converter (which can be expensive), you can transfer sound in the digital domain. The number of in-

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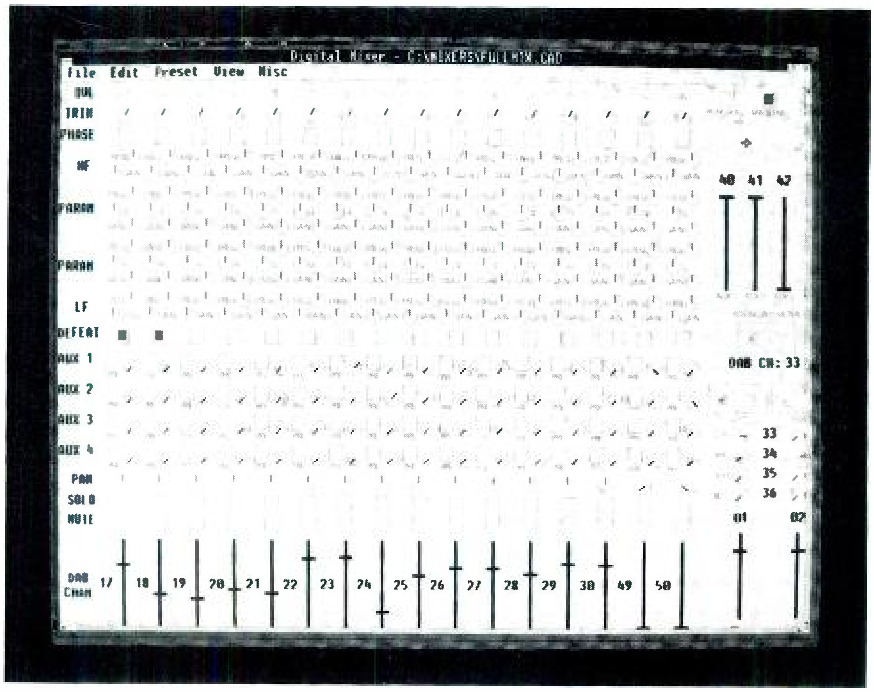
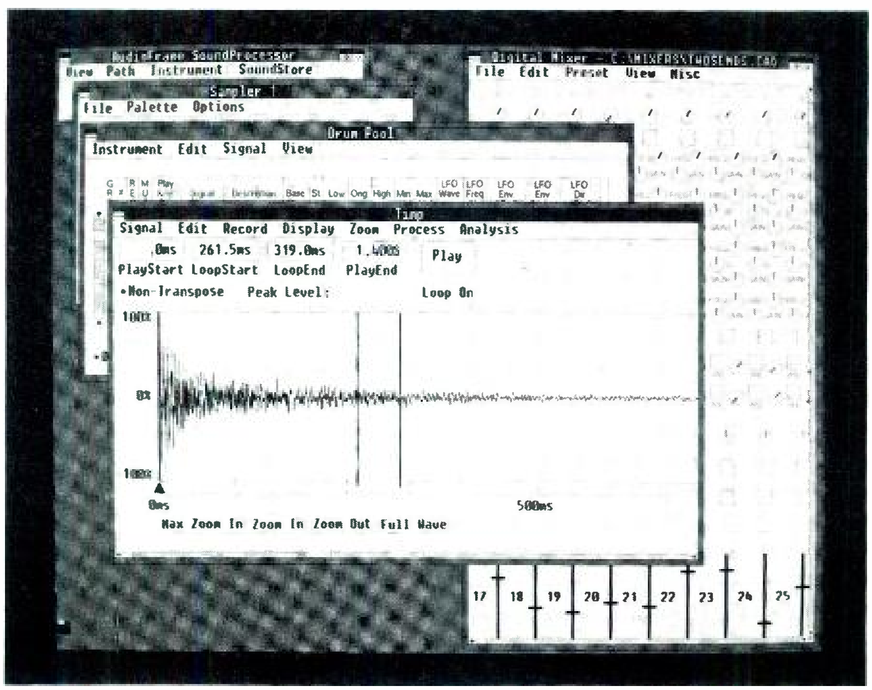


Photo 3. Virtual mixers allow the audio signal to be sent to, and returned from, various processors without leaving the digital domain.



WaveFrame's AudioFrame is an example of an integrated system combining sampling, editing, mixing and signal processing facilities.

dependent channels you can transfer simultaneously will depend on the format and the capabilities of your systems.

The other option is to use an integrated system, which allows the sampler and the hard disk recorder to communicate directly to one another. (Only very recently have systems become available that can do this.) These systems circumvent the problem of transferring audio via standard formats by using an internal protocol.

The movement of audio throughout the system is far easier and more efficient. It's more like working in the analog world, and 32, 48 or 64 channels of simultaneous transfer is well within the realm of possibility. These systems generally have provisions for transferring the finished audio output to a target medium digitally.

So many tracks, so little control

Let's assume you have a project that is nearly complete. The instruments and sound effects were generated and altered in a sampler, then recorded and edited digitally on tracks of a hard disk recorder. Those tracks must now be mixed down. As discussed, if you have a digital mixer that can communicate audio to and from your hard disk recorder, you can get the job done without leaving the digital domain. But if you don't have a digital mixer, then 16 or more tracks of digital audio have to be run through 16 or more digital-to-analog converters, over 16 or more cables, into 16 or more analog inputs, through 16 or more sets of analog faders, pan pots, equalization circuits, trim pots, auxiliary sends, analog processors and buses. All of this analog circuitry may add noise and distortion to the sound.

Integrated systems are now available that have built-in digital mixing, equalization, and effects capabilities. (See Photo 2.) Because the digital mixer can be integrated into the same system as the sampler and hard disk recorder, audio can be sent from the sampler, through the mixer, to the hard disk recorder, back through the mixer, and onto the final outputs without ever leaving the digital domain. Because the process is all digital, no additional noise or distortion should be added. (See Photo 3.) [For more information on integrated systems, see February, pg. 28, "True Digital Audio Mixers."]

Coping with complexity

Audio engineering has made tremendous strides since the days of purely mechanical sound reproduction, and with these strides has come a variety of devices, media, formats and new technologies. With the introduction of digital recording, digital sound synthesis, digital mixing and effects processing, and now random-access digital editing, the choices and technologies open to the audio professional have advanced geometrically. The pace of innovation can outstrip the ability of just about any audio professional trying to keep pace with all the new developments.

Most digital audio devices today imitate the functionality and interface of analog equipment, allowing users to become productive with the new technology quickly and easily. However, the surface similarities may mask the real power of digital systems. For example, a hard disk recorder that can record or play four tracks at a time can be used just like a 4-track tape recorder that happens to have very fast transport. But it's possible to do things on a hard disk recorder that are beyond the capabilities of a tape recorder, such as assigning different outputs to a given track at different times, permitting the 4-track hard disk recorder to work like a tape recorder with more than four tracks.

Integrated systems address other important concerns as well; hardware interface schemes are simplified and exact, and user interfaces have common elements for various applications. At the same time, these systems tend to cost less, since the various modules can usually share common components like D/A and A/D converters, power supplies and enclosures.

What's in store?

In the future, we can expect to see samplers and hard disk recorders with greater storage capacity and better audio quality. Digital audio storage capacity for samplers will improve as the RAM standard, which is today making the transition from 256-bit chips to one megabit, increases to four and 16 megabits on a single chip.

Hard disk storage devices with capacities of thousands of megabytes, capable of storing several hours of CD-quality digital audio, will be readily available. Magneto-optical disk drives will provide performance comparable to standard magnetic hard disks with the tremendous advantage of being a removable media. Twenty- or 24-bit storage and processing will prevent the introduction of any audible distortion during the mixing and mastering process.

In addition to this evolutionary change, we can expect revolutionary changes, although they're a little harder to predict. Samplers may be coupled with new sound-generation techniques, such as synthesis based on mathematical models of musical instruments and other acoustic devices. Hard disk recorders will create new editing methodologies, which will take advantage of the full power of random-access editing.

Rather than expecting hard disk recording and sampling technology to lose their separate identities and merge into equipment that performs both functions, we expect samplers to remain specialized for sound generation (possibly coupled with sound synthesis), and hard disk recorders to become increasingly powerful sound editing engines. The coupling of the two technologies offers users unprecedented quality, power and productivity in generating and manipulating sound.

RE/P

Live Recording: The Boston Symphony Orchestra

By E. Brad Meyer

This recently recorded performance of Richard Strauss's "Elektra" was digitally edited and mixed on Lexicon's Opus.

It's definitely a high-pressure situation. The conventional wisdom is that opera recordings are too expensive to make in the United States, yet here are the men from Philips Classics Productions, camped out at Soundmirror recording in Boston, editing a live opera recording from Boston Symphony Hall.

Their schedule allows just three days to edit and mix the entire project on a system that they've never used before—that, in fact, no one has ever used for a major classical production. Even the room where they are working is brand-new, with squares of Sonex and Tube Traps standing out in sharp contrast to bare concrete walls. The Philips crew has plane reservations for Amsterdam at 9 p.m. It's now 1:30. The editing isn't finished, which means the final mixdown hasn't even begun. And on top of that, the computer that runs the editing/mixing system is down. So why are they willing to let a reporter in the front door at a time like this?

One reason that Wilhelm Hellweg and Onno Scholtze, the producer/balance engineer team, seem so calm is that they don't have to worry about fixing the computer. They are using Lexicon's Opus digital editing and mixing facility, and if you have trouble with an Opus, you don't have to hold a telephone receiver with your shoulder and work the controls while someone talks you through a bunch of diagnostic procedures. Instead, you connect the Opus to the main office and let them control the system directly, via modem.

E. Brad Meyer is a consultant, recording engineer, audio writer and the owner of Point One Audio in Lincoln, MA.

Photo 1. *The stage configuration for orchestra and soloists at Boston Symphony Hall. Note microphone placements.*

As an engineer in Lexicon's Waltham, MA, headquarters checks out the problem, ghostly commands and responses appear on the screen in front of an empty operator's chair. After a couple of minutes Everett Porter, Soundmirror engineer and Opus operator for the opera project, notices that the contents of the screen are starting to look familiar. The typed command "DIR" appears; at last, something that makes sense to a mere PC user. Sure enough, this is followed by what is obviously a directory of files. In Waltham, the engineer inspects the list and decides which file probably contains the current data.

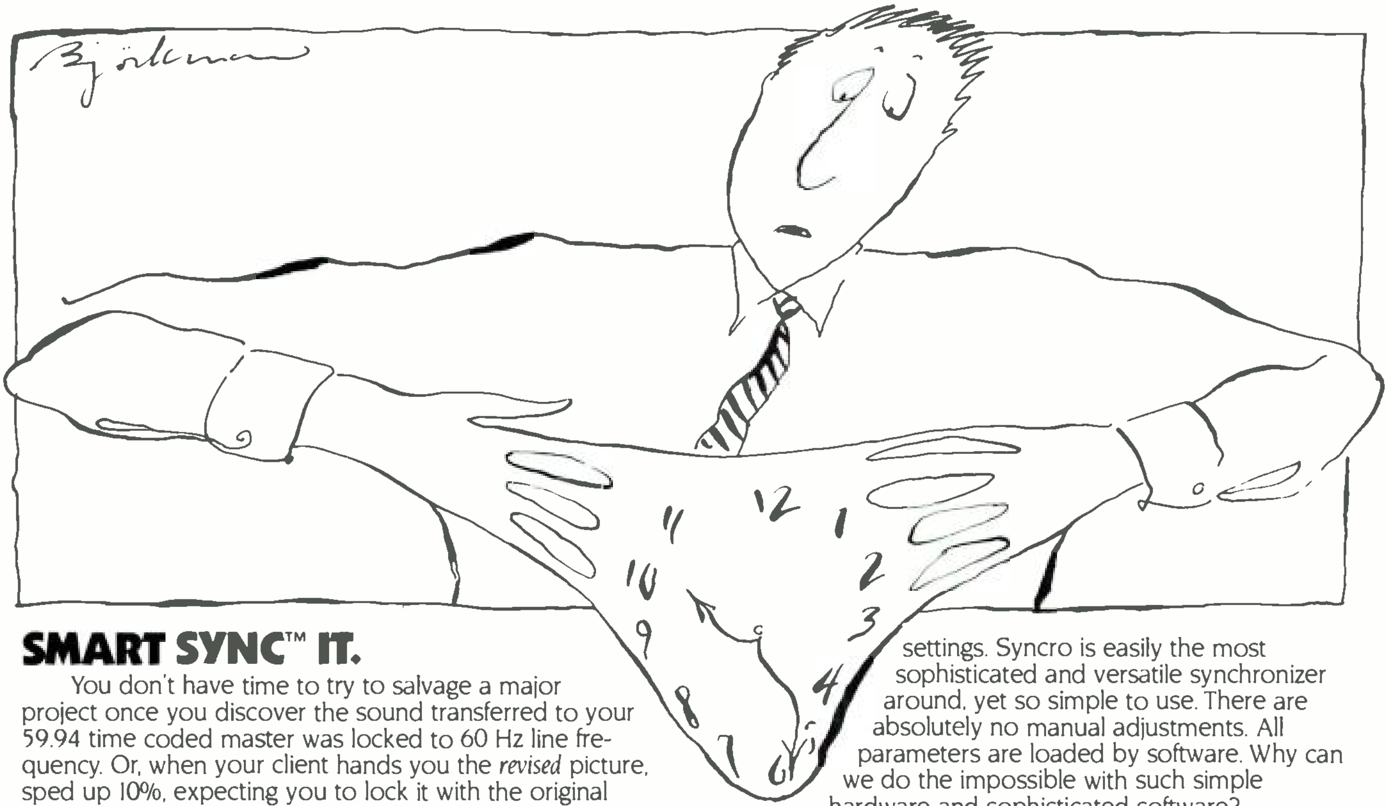
"Is PH883 your project?" he types.

Everett taps back a "yes" on his own keyboard, and the sequence of instructions to restart the system and load PH883 appears, followed by the words: "OK, you're all set."

The problem is solved. They are up and running again, having lost only the most recent edit.

This particular Opus installation is just a few miles from Lexicon's home office, but it could be almost any place that has a phone. This troubleshooting method has already worked from as far away as Italy, and with studios being built in increasingly rural and exotic locations, the capacity for truly remote control could turn out to be an important advantage.

DON'T GET STUCK WITH TIME ON YOUR HANDS.



SMART SYNC™ IT.

You don't have time to try to salvage a major project once you discover the sound transferred to your 59.94 time coded master was locked to 60 Hz line frequency. Or, when your client hands you the *revised* picture, sped up 10%, expecting you to lock it with the original multi-track. With our exclusive, patent pending, Smart Sync feature (SoundMaster Auto Restored Timebase SYNChronization™), set your varispeed by entering four digits, establish your offset *anywhere* in the show, roll and record. Your Soundmaster™ Integrated Editing System will treat that tape *exactly* as any other. How is the miracle performed?

IT'S THE SOFTWARE.

That's right. As the time code is read from tape, everything is handled for you automatically by some sophisticated software. *Everything*. Your machine is synchronized while running off its standard speed. And, here's the clincher. Even though the time code on your tapes is divergent, with the real world difference changing every frame, Smart Sync maintains a constant system offset. So you can start and stop anywhere in the show, and pick-up locked with confidence. How does Smart Sync do it?

IT'S THE HARDWARE.

Our Syncro™ machine controller can synchronize, not just chase, from a fraction of, to several times play speed. With Smart Sync, *you* control literally thousands of varispeed settings. Syncro is easily the most sophisticated and versatile synchronizer around, yet so simple to use. There are absolutely no manual adjustments. All parameters are loaded by software. Why can we do the impossible with such simple hardware and sophisticated software?

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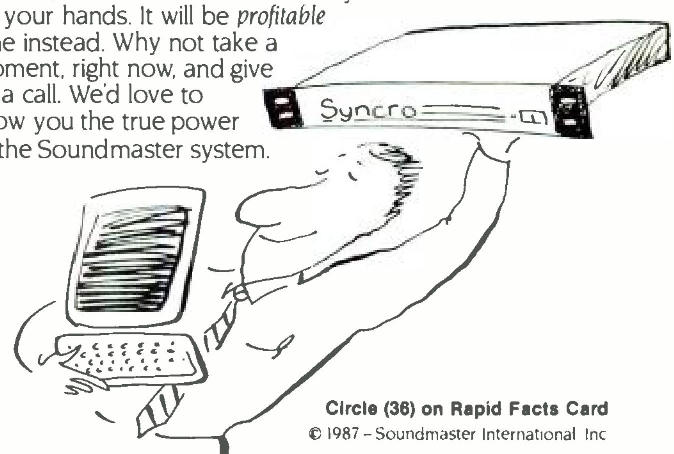

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The recording

But we've jumped into this story in the middle. To get a clearer picture of what's being edited and how, let's begin with the recording process.

As part of its 1987-88 season, the Boston Symphony, under music director Seiji Ozawa, performed Richard Strauss's "Elektra" in a series of concerts using an expanded 120-piece orchestra and singers arrayed over a specially built 3-tiered stage.

In June 1988, Ozawa entered into an agreement with Philips to make a live recording of the opera in the fall. Philips asked John Newton, president of Soundmirror, to make the original recording and provide facilities and technical help to edit and mix the tapes immediately afterward. Newton accepted the challenge, largely because he knew that an Opus could be installed in time to do the post-production.

If successful, the abbreviated production schedule promised to save money and produce a final product in time for the soloists to approve the recording before their schedules scattered them all over the world. It would, of course, also provide an excellent test for the Opus system.

The concerts took place Nov. 12, 18 and 19. Craggy, somewhat relentless, and hardly what you would call tuneful, "Elektra" is, nevertheless, a powerful work, and the habitually critical and staid Boston audiences exploded in tumultuous applause at the end of every concert. Producer Hellweg said, "The biggest advantage of live performance is that you capture emotions that are generated nowhere else."

After the concerts, two makeup sessions were scheduled to fix rough spots and provide breaks in the music. "Elektra" is 105 minutes long, without a single complete stop—too long for LP, cassette or CD. A stop and restart around the halfway point were recorded for all three formats, then two more near the one-quarter and three-quarter marks for the LP and cassette versions.

Microphone placement

The stage construction with both musicians and microphones in place can be seen in Photo 1. Engineer Scholtze likes to use, as his main array, an enhanced version of the well-known 3-spaced-omni setup: instead of a single central pickup there are two microphones faced away from each other, panned hard left and right. (The B&K 4006 capsules become directional at high frequencies, like all condenser microphones with large enough capsules to have the appropriately low noise-floor needed for classical recording.)

The stage was fitted with its largest extension (putting the lip about 11 feet out into the hall), and the four main

Photo 2. *Editing "Elektra" at Soundmirror are Philips staff engineer Martha deFrancisco and Soundmirror engineer Everett Porter.*

microphones were mounted just behind the conductor's head and suspended from above on a pipe positioned 3 meters above the stage. Scholtze and Soundmirror engineer Henk Kooistra also placed a quasi-X/Y pair of mics roughly 3 feet from and 5 feet above the front edge of each tier of the singing structure. The Schoeps MK41 hypercardioids were spaced about 10 inches apart with an included angle of 90°. (See Photo 1.)

One other pair of B&K 4011 cardioids was placed at the lip of the lowest level of the singers' stage; pointed upward, these acted almost as floor mics, and were occasionally used in place of the X/Y pair.

Various accent microphones, including Schoeps MK4s, MK6s, MK41s and Neumann TLM170s, were placed throughout the orchestra and attached to parts of the scaffolding for "off stage" singers who were hidden from the audience beneath the stage structure. An additional pair of Schoeps MK41s was mounted on the ends of the main boom. They were used to pick up two choruses that sang briefly from two side balconies late in the piece. (This arrangement proved unsatisfactory, and the choral sections had to be taken from the makeup sessions—with the two groups on the main floor, each recorded with a pair of Sanken 41s.) In all, 27 microphones were in the concert setup, though most of the time far fewer were in actual use.

During the sessions, all microphones were mixed down to 12 tracks and recorded on a Sony 3324. The four omnis on the main stereo bar plus the hypercardioids for the choir went to Tracks 1 and 2; the four X/Y stereo pairs for the vocals went to Tracks 3-10; Tracks 11 and 12 received a stereo mix of the remaining accent microphones.

Editing

The machine at Soundmirror has two 800Mbyte disk drives (up to four are possible), giving it a capacity of 254 track-minutes. It can hold 127 minutes of stereo, about 32 minutes of eight-track or even 2.5 minutes of 99-track material.

Opus has 12 digital input channels and can keep track of up to 99 channels at once internally, but its disk I/O is limited to eight tracks at one time. (You can edit 24-track material, but you must perform each edit three times.)

Though Opus has the memory capacity for entire short selections and most normal orchestral movements, it won't hold three complete performances of the 105-minute work, even in 2-track form. To take advantage of the system's flexibility, the producers wanted to work with the maximum number of channels that could be manipulated with a single operation. So after each concert, the digital 24-track machine went back to Soundmirror, where Tracks 1-8 were fed into Opus, mixed to four tracks, and then laid back to the original 3324 master, on Channels 13-16.

The first time the engineers listened to the 8-channel "Opus" mix—Channels 13-16 and 9-12 together—they realized that something was amiss. The four tracks that were going through Opus were delayed by about 250μs—enough to produce audible effects. After the editing was completed, the eight tracks were copied back to a new 3324 tape, with the original tracks (9-12) delayed an appropriate amount during the transfer.

Opus has a tape-streamer system that can back up or restore the entire contents of a 2-disk-drive machine in 112 minutes, but there wasn't enough time between sessions to use it. For this project, a second 3324 tape served as backup.

Each cue point is identified in the computer's memory by hour, minute, second, frame and subframe, and there is also a space for the user's identifier, in this case, the page and measure numbers. Edit points chosen by ear at the push of a button are accurate to a single sample, while the minimum overall step size for points that have been moved is one-hundredth of a frame.

With Opus, you can give each track its own level, EQ, reverb, cross-fade time or edit point while leaving the other tracks unchanged. You can also slip a single track in time. And everything is done in the digital domain, with word lengths great enough to round off the numbers correctly and maintain proper dithering. (See "Performance Aspects of Digital Oversampling," p. 20.) Some of these features turned out to be crucial for the "Elektra" project.

Three performances and two makeup sessions would seem to offer plenty of choices for editing any given spot in the music. But where someone made a mistake on one night, it was not always possible to just drop in a measure or two from another. Mismatches occurred for several reasons.

The fact that other events were scheduled at Symphony Hall between the three "Elektra" performances meant that the entire setup—including the stage extension, scaffolding and even the microphones—had to be dismantled after each concert and between the makeup sessions.

Another important factor was the use of paired microphones for vocal pickup. While a turn of only a few degrees can cause a noticeable shift in apparent placement, no singers can be expected to stand in exactly the same position throughout every performance. And in some cases, after listening to the day's playback, soloists decided to try different tone colors for the next performance.

Other variables producing differences in the sound included temperature and humidity (which affect reverberation time at high frequencies), and the mood and physical condition of the musicians. In addition, there was the acoustical variation between three live performances recorded in front of packed houses, and two makeup sessions recorded without audiences.

Theoretically, given time, all of these problems are solvable, but time was in short supply. The finished product had to be good enough to satisfy the artists, some of whom had the power to veto its release.

With a conventional digital editing system, mismatches in stereo placement, reverberation or tone color require that you move the edit point. With Opus, if the singer moves to the left just before the cut, you can change the balance in the stereo pair after the cut, then slowly restore it. If a singer takes a deeper breath on one night, resulting in a different vocal color, a little EQ may do the trick.

Two Lexicon 480L reverb units were connected through the Opus's digital patchbay to help adjust the apparent distance of a singer from a single pair of microphones—without affecting the sound of the orchestra. The parametric EQ software in the 480Ls also came in handy because the Opus's parametric EQ module was not yet installed.

The Soundmirror team matched the hall sound between live performances and patch sessions the old-fashioned way—by adding absorption to the hall. "For the makeup sessions, we treated the hall with around 2,300 sabins of absorption," said Newton. "We used blankets and squares of Sonex to kill flutter echos, and 25 Acoustic Sciences absorbers, of different sizes, to control the low frequencies. I defy anyone to tell the difference between the live performance and the makeup sessions from the reverberation."

Apart from a couple of brief interruptions, editing went smoothly. But no one can work as rapidly as possible the first time around with a new system. One 10-minute section (the so-called monologue) took nearly as much editing time as the rest of the piece put together. In the end, however, the consensus was that the edits in the finished product were cleaner than would have been possible with a more conventional digital system. (The Philips engineers did report one minor oversight in Opus's design: There was no convenient place on the console to put the music score.)

Mixing

The three days originally allotted for post-production were almost over when the editing was finally completed. At this point, five 8-track segments of about 20 minutes each were stored on the 3324. These were to be mixed down into stereo in Opus, and then stitched together twice—once for the CD master and once for the LP and cassette—with different sets of beginnings and endings. In addition, there would be a continuous version for transfer to DAT for audition purposes, and possibly for eventual release in that format.

During the final mixdown, the producers also had to incorporate one last element: the sound effects. The score called for various Roman street sounds including bulls bellowing, carts rumbling and crowds walking or marching. These effects were recorded in Symphony Hall on the final day, with the cooperation of a small horde of enthusiastic students from the nearby Berklee School of Music. Once the effects were transferred into the Opus, it was a simple matter to incorporate them into the final mix.

The final step in the process was to be the conversion of the stereo mix from the 48kHz sampling rate, used for recording and editing, to 44.1kHz. The airline reservations were changed after all, and Hellweg and Scholtze stayed an extra day. But even that was not enough to complete more than the first CD's worth of the 2-track mix.

This meant that in addition to the half-finished 2-track version, the Europeans would take home a 3324 tape containing the complete 8-track version of the edited mix. Philips's current digital mixing system runs only at 44.1kHz, so the eight tracks had to be converted individually to the lower sampling rate. This was done in two passes using two stereo sampling rate converters.

Without Opus, this particular project would probably never have happened, although Philips does have another way of doing this kind of editing. They mix the multitrack master to a digital 2-track and edit the mixdown. If an edit requires a different level or EQ, the mix is redone to improve the match between the two halves. Although Scholtze was heard grumbling about the holdup during one of the computers brief down-time periods, completing a project of this size has typically taken closer to a year than three weeks.

The future for this kind of project may depend on how well the CD sells, which may depend on how well the emotional and sonic impact of the original performances comes through in the reproduction of the recording. In the end, the challenge for the engineer and producer was to make the recording sound as though nothing has been done to manipulate the original performance, while, at the same time, using all the capacity for manipulation that is inherent in the systems hardware and software. Judging from the sound of the preliminary mixdown, this recording will be one to watch. RE/P

ENGINEER/PRODUCER INDEX

A

Phil Adler (E/P): 105 Murdock St., Brighton, MA 02135; 617-787-9125. **Credits:** WSBK-TV 38; NESN; NBC (CP-TV), Boston Red Sox remote broadcasts; New England Sports Network (CP-TV), Boston Bruins hockey remote broadcasts; The Souls (MP-A), "The Light In You"; Electric Toys (MP-A), "Electric Toys."

Richard Adler (E/P): Box 21272, Nashville, TN 37221; 615-646-4900. **Credits:** John Prine (MP-AR), "John Prine Live"; The Kentucky Network (KET), (MP/PP-MV), "The Lonesome Pine Specials"; Johnny Cash/Polygram Records (MP-A), "Johnny Cash is Coming to Town"; The Blair String Quartet (MP-A), "American Chamber Music"; John Hartford (MP-AR), "Annual Waltz."

Ray Alcazar (P): 4309 Shadyglade Ave., Studio City, CA 91604; 818-508-9045. **Credits:** Powers of Persuasion (MP-A), "America on $5,000 a Day"; Marc Woodson (MP-AT), "Rolling with the Times"; Cynthia Riddle (PP-F), "Fear Itself"; First Quincy (CP-R), "A New Dawn"; Michael Ceroli (MP-A), "Powercuts."

John Anthony (E/P): Box 25, Shenorock, NY 10587-0025; 212-713-5757. **Credits:** Staples-Strom (MP-AT), "Homeless Lament"; Staples-Strom (MP-AT), "Caroline."

B

Philip Barrett (E/P): Omega Audio, 8036 Aviation, Dallas, TX 75235; 214-350-9066. **Credits:** MCA/Nanci Griffith (MP-A), "One Summer's Evening"; U.S. Holocaust Museum (PP-F), museum film; Bola Films/Coca Cola (CP-TV), "Max Headroom"; Maroch/McDonalds (CP-TV/R), "McDonalds"; Sugar Hill/Robert Earl Keen (MP-A), "Live at Sons of Herman Hall."

Editor's note: This is the first installment of the Engineer/Producer Index, RE/P's monthly department that spotlights engineers/producers and the work they are creating. To be listed in the Index, fill out the card located in the back of this issue.

Brian Basilico (E/P): Box 1491, Bolingbrook, IL 60439; 312-759-9311. **Credits:** AT&T Network Systems (PP-MV), "We all Win"; Video Network (CP-TV), Ed Napelton dealerships; Matt Sisson (MP-A), "Ain't Comin' Home"; Ed Dunn (MP-S), "You put the Merry in my Christmas"; St. Margaret's Hospital (CP-R), "Radio Spots '89."

Craig R. Bishop (E/P): 27 W. 75th St., #4B, New York, NY 10023; 212-580-0974. **Credits:** HBO (CP-TV), "Next On"/"Best On"/"Winter '89"; CBS Sports (CP-TV), "Winterfest"; Old Milwaukee Beer (CP-TV), "Bluefish"; Burger King (CP-TV), "Batteries Not Included"; GRP Records/Billy Cobham (MP-A/S), "Picture This" (album)/"Same Ole Love" (single).

Dean Burt (E/P): Box 18401, Encino, CA 91416-8401; 818-705-3342. **Credits:** Whitney Houston/Arista (MP-A), "Whitney"; CBS/Sony/Nancy Wilson (MP-A), "Nancy Now"; MCA/Oingo Boingo (MP-A), "Boingo Alive"; Epic/Tease (MP-A), "Remember..."; A&M/Jeffrey Osborne (MP-A), "Emotional."

C

Richard Cannata (E/P): 2119 Fidler Ave., Long Beach, CA 90815; 213-498-6492. **Credits:** EMS Systems (CP-R), "Let us Help You"; Mission Softs (CP-R), "Childs Fast Learning"; Pace Security (CP-R), "We Make you Feel Safe"; Ricky Cannata (MP-A), "On Tape."

Jeffery N. Ceja (E/P): 19139 Seminole, Redford, MI 48240; 313-537-1281. **Credits:** B-Flat Blo Records (MP-A/MV), "Harlet: Virgin Wings"; David Danielle (CP-TV), "Last Sunday"; Bob Stumpmeir (MP-A/MV), "Halloween at Harpo's"; Mario Resto (MP-MV/A), "Smiling Boy"; Logical Emotion (MP-AT/MV), "Communication Breakdown."

Jan Celt (P): 4026 NE 12th Ave., Portland, OR 97212; 503-287-8045. **Credits:** Napalm Beach (MP-A), "Liquid Love"; Janice Scroggins (MP-A), "Janice S. Plays Scott Joplin"; The Esquires (MP-A), "Play"; Obo Addy (MP-A), "African American"; Tom McFarland (MP-A), "Just got in From Portland."

David Chornow (E/P): 911 N. Laurel Ave., W. Hollywood, CA 90046; 213-650-7302. **Credits:** Metropolis Pictures (F), "Street Asylum"; In Productions (F), "Dance Spirit of Ghana"; Metropolis Pictures (PP-F), "Squad"; Empire Entertainment (F), "Night Crew"; Action Int'l Pictures (F), "Bounty Hunter."

E

Steve Ett (E/P): c/o Chung King House of Metal, 247 Centre St., New York, NY 10013; 212-925-4356. **Credits:** Def Jam/L.L. Cool J (MP-A) "Radio," "Bigger and Deffer," third album in production; Def Jam/Beastie Boys (MP-A), "Licensed to Ill"; "Profile Records/Run DMC (MP-A), "Raising Hell," "Tougher Than Leather"; Def Jam/Public Enemy (MP-A), "It Takes a Nation of Millions"; Def Jam/Danzig (MP-A), "Danzig."

F

David D. Fihn (E/P): 4239 Forest Plaza Drive, Hixson, TN 37342; 615-870-3446. **Credits:** Jim Burke Buick Nissan (CP-TV),

Key

Name (Title Code): Address; Phone Number. **Credits:** Client/Artist (Credit Code-Subcode), Project Title.

Title Codes:
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"Year-End Closeout"; Economy Honda (CP-TV), "Closeout"; Calfee Cadillac (CP-TV/R) "Not all bad News"; Midfield Auto Sales (CP-TV), "The Dealmaker"; McGee Dodge, (CP-TV), "Easy to Buy."

S. Fitzstephens (E/P): 235 W. 76th St., New York, NY 10023-8214. **Credits:** A&M Records/Rosie Vela (MP-A/S), second album demos; Billy Mulligan (MP-S) "Traditional Tunes"; Billy Mulligan (MP-A), "Billy Mulligan."

Ron Freiheit (E): 1520 Ninth Ave. N.E., Owatonna, MN 55060; 507-451-1503. **Credits:** Jeff Wilday/Rosemary Tolzmann (MP), demo; OTC Division, SPX Corporation (PP-CI), "Right to Know."

G

Adam Green (E/P): 425 S. Pine St., Richmond, VA 23220; 804-344-8151. **Credits:** SST/Alter-Natives (MP-A), "Group Therapy"; SST/Always August (MP-A), "Geography"; GWAR (MP-MV), "Stupid White Chick"; SST/Alter-Natives (MP-A), "Throw Down."

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H

Dann E. Haworth (E/P): 3580 Rainbow Blvd., Suite 816, Kansas City, KS 66103; 913-362-2242. **Credits:** Tony Glise (MP-A), "Overview"; Bon Ton Soul Accordion Band (MP-A), "We Only in it for de Monkey"; Tracy Creach (MP-A), "From the Heart"; The Verandas (MP-A), "American Tradition"; Contemporary Commercial Concepts (CP-R), Eslinger's Mens Wear.

Dennis Hetzendorfer (E/P): 8795 SW 57th St., Cooper City, FL 33328; 305-434-1377. **Credits:** Hope Enterprises (CP-TV), "Bob Hope Christmas Special"; Tel-Air Interests (CP-TV), "Frankie Avalon Presents"; Tel-Air Interests (CP-R), "Gulfstream '89 Supertrack"; Independent Records (CP-AT); Amy Dee Music (CP-TV), "King Orange Parade."

David W. Holt Sr. (E/P): 314 Sumner St., Elyria, OH 44035; 216-322-5560. **Credits:** E.M. Rollins (MP-A) "Originals and Others"; E.M. Rollins (MP-S) "The Edge of Dawn"/"I'm Down"; On Trak Productions D.J. Service (CP-R) "Now Booking for '89"; Backstreet Noise (PP-MV), "Every Star in Heaven"; Jesse Youngblood (MP-S), "River of Love"/"Forced Love."

Keidi Howard (P): Box 40222, Long Beach, CA 90804; 213-433-6195. **Credits:** Ti Marie (MP-S), "Who Needs Boys?"; Jumbalaya (MP-A), "Walking on Sunshine"; The Superiors (MP-A), "The Superiors"; Keidi (MP-S), "New Love"; Veoni Edwards (MP-S), "Jealousy."

J

Danny Jensen (P): 523 Whitecloud Drive, Boise, ID 83709; 208-323-0986. **Credits:** Onati Restaurant (CP-TV), TV commercial; Idaho Children For Youth Commission (CP-TV/R), "Juveniles In Jail"; Christ Faith Ministries (CP-TV), TV commercial—revival; Hewlett-Packard Corp. (PP-CI), "Integrated Circuit Assembly Line" video; Whiteman Manufacturing Co. (PP-CI), "Getting The Most From Your Power Trowel" video.

L

Fritz Lang (E/P): 1 Margo Place, Huntington, NY 11743; 516-549-8851. **Credits:** A&M/DIR/Iggy Pop (MP-AR), "Live at the Channel"; Rocker Arm/Torn Pockets (MP-A), "Trouble Again"; Strange Way/Uncontrollable (MP-A), "Uncontrollable."

Norton Lawellin (E/P): 1561 Sherburne Ave., St. Paul, MN 55104; 612-645-9281. **Credits:** St. Paul Winter Carnival (PP-CI), "Vulcan Opening"; Marilyn Sellars Radio Show (CP-R), holiday shows; Cheap Carpet (CP-R), holiday jingle; Kevin Kay (MP-AT), "I was Made for You"; Lue Yang (MP-A), "Lue Yang."

Bill Leavenworth (E): 5004 Pendelton St., San Diego, CA 92109; 619-274-3566. **Credits:** SBK Productions (CP-TV), Jiffy Lube spots; Eve/Screen Gems (CP-TV), Combat Roach Control spots; Eve/Screen Gems (CP-TV), Tylenol spots; The Film Tree (CP-TV) Arm & Hammer spots; Elliot Sluhan Productions (PP-F), "Air Force One"/PBS.

Paul D. Lehrman (E/P): 184 Palmer St., Arlington, MA 02174; 617-643-2700. **Credits:** Jewish National Fund (CP-TV), "Barren Land"; Lexicon Inc. (PP-CI), "The CP-1 Cinema Processor"; Themes/KPM Music (MP-A), "On Stream"; Jewish National Fund (PP-CI), "Land and Life."

M

David W. Mathes (E/P): Box 22653, Nashville, TN 37202; 615-822-6119. **Credits:** The Music Machine TV Corp. (CP-TV), "The Music Machine"; Sapphire Records/Silver Eagle Band (MP-A), "Silver Eagle Band"; Star Image Records/Johnny Newman (MP-A), "I'm in Ohio now (With Georgia on my Mind)"; Rising Star Records/DeAnna (MP-A/S) "Simple Love Song"; NRS Records/David Mathes (MP-A), "Sounds of a Saved Soul."

Edward R. Minnich (E/P): 127 Huron St., Elyria, OH 44035-3491; 216-322-3984. **Credits:** Backstreet Noise (MP-A/AR)

"The Best of Backstreet Noise"; Jim Matthews (MP-A) "The Jim Matthews Catalog"; Per-Min Productions (CP-R) "It's Your Wedding"; Long-Gone Records (CP-A), local spot—local talent; The Palace of Rock (MP-A), "Music of the '60s and '70s."

O

Jim O'Neal (P): 232 Sunflower Ave., Clarksdale, MS 38614; 601-627-2209. **Credits:** Otis Rush (MP-S), "Rooster Blues" session; Booba Barnes & the Playboys (MP-A); Lonnie Shields (MP-A); Wild Child Butler (MP-AR), "Lickin' Gravy."

P

Michael Panepento (E/P): 225 Oxmoor Circle, #811, Birmingham, AL 35209; 205-942-3222, 205-985-9336. **Credits:** Jazzharbor Recorders/Gary Motley (MP-A), "Stepping Out"; South Central Bell/Concept Co. (PP-CI), "Opera"; Metro Area Express/Kirkpatrick Agency (CP-R), "MAXX, Ride MAXX"; Foxxy Fatts & Company (MP-A), "Foxxy."

R

Al Reiners (E): 1431 N.W. 71st Ave., Plantation, FL 33313; 305-587-8258. **Credits:** IATSE, Local 545, Miami (house mix), Theatre of Performing Arts, Florida; IATSE, Local 477, South Florida (CP-F), Ford commercial; Inter-American Film Production Corp (CP-TV), "Key West"; Audio Facilities Inc. (CP-CI), Cabriner Inc. industrials; Tele/Film Presentations (CP-TV), Orlando Videotape Special.

Ray Rettig (E): c/o Cotton Hill Studios Inc., 18 Walker Way, Albany, NY 12205; 518-869-1968. **Credits:** Rich Goodhart (MP-A), "A Little Closer to One"; Ed Degen Hart (MP-A), "Seems"; Bill Krauss (MP-A), album not yet titled; Avis (CP-R/TV), advertising; SEFCU (CP-R), jingle.

Paul J. Rich (E/P): 1721 Olean Road, South Wales, NY 14139; 716-652-3750. **Credits:** New Friday's Children (MP-A), "It can Happen!"; Sandy Janice (MP-S), "Only our Eyes"/"'68 Summer"; Nine Pines Productions (CP-R), "Digital Demos and More"; Paul Richiusa (MP-S) "I'll be Missing You"/"Hopeless Romantic"; Poopchin (MP-A), "We will do it All."

Curtis Roads (E/P): E39-346, MIT, Cambridge, MA 02139; 617-628-2358. **Credits:** Morton Subotnick (MP-A), "Touch/Jacob's Room"; Various artists (MP-A), "New Computer Music."

Marc J. Rose (E/P): 11300 4th St. N., Suite 140, St. Petersburg, FL 33716; 813-576-4594. **Credits:** Shreekshow Partners Ltd (CP-R), "Shreekshow"; L. Berkley/Rich Little (PP), "Phonies"; Evatone Soundsheets, "Spin That Soundsheet"; St. Petersburg Grand Prix (MP), "Grand Prix Theme."

S

David Schwartz (E/P): 111 Park St., New Haven, CT 06511; 203-782-0273. **Credits:** New England Storytelling Center (CP-R), "Tellabration"; Guitar Roberts (MP-A), "Blues Master"; City Spirit Artists (CP-R), "Reaping the Harvest"; Heritage/WYBC FM (CP-R), "Live Radio Concerts."

Chuck Simone (E/P): 620 E. Bleeker St., Aspen, CO 81611; 303-925-2296. **Credits:** Showtime/Dabney Coleman (TV) "Showtime Comedy Club"; Ohlmeyer Comm./John Denver (TV), "John Denver: Christmas in Aspen"; Rainshadow Studios/Various Artists (MP-A), "Aspen Tracks."

Jack Stang (E/P): 168 Buckingham St., Hartford, CT 06106; 203-524-5656. **Credits:** Savage Bros. Band (MP-S/A), "Hey Little Lady"; Little Wing Band (MP-S/A); Deserts Direct (CP-R/TV, F/CI), "44 Cakes"; WLUH (CP-R), new jingle; Bill Chapin (MP-A), "Billy Chapin."

Bruno Strapko (E/P): 40 Fillmore Lane, Streamwood, IL 60107; 312-830-0701. **Credits:** Illinois Bell (CP-R), "Ceiling Fan :60"; Illinois Bell (PP-CI), "A Network of Protection"; Kraft (PP-CI), "Let's Talk" audiocassette; Illinois Bell (PP-CI), "Call ID Service" audiocassette.

Andrew Strauber (E): 420 Brookside Lane, Somerville, NJ 08876; 201-359-6014. **Credits:** David Frost/King World (PP-TV) "Inside Edition"; HBO/Andrew "Dice" Clay (PP-TV), "Andrew 'Dice' Clay Special"; Cinemax/Les Paul, (PP-TV) "Les Paul: He Made the Music"; DIR Broadcast/Billy Idol (PP-SR) "King Biscuit Flower Hour"; Merv Griffin Productions (PP-TV), "Wheel of Fortune at Radio City Music Hall."

T

Scott A. Terhark (E): 7276 S. Independence St., Littleton, CO 80123; 303-979-1772. **Credits:** ABC/"20/20" (CP-TV), "Steroids" episode; ABC/"20/20," (CP-TV), "Halcion" episode; Drummond Productions for Disney (CP-TV), "Going Home"/Judy Collins; NFL/United Way (CP-TV), players spots for United Way-1988; National Geographic Explorer (CP-TV), "Explorer" TV series, "Mardi Gras Indians" segment.

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Subcodes: F (Film); MV (Music Video); CI (Corporate/Industrial).

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Stephen Toback (E/P): 15150 Sherman Way, #108, Van Nuys, CA 91405; 818-997-8026. **Credits:** Nipper Productions/Violator (MP-A), "Violator"; Talk Radio Productions/Oliver Stone (PP-F), "Talk Radio"; Tomatrey Smiers Music/Whitewash (MP-SR), "Listen to Your Dreams."

Travis Turk (E/P): 2072 Whitney Ave, Nashville, TN 37210; 615-259-4299. **Credits:** Joan Jett (PP-MV), "Little Liar"; WTBS Superstation (CP-TV), "At the Hoop"; National Public TV (CP-TV), "TV Worth Watching"; Ronnie McDowell (MP-A), "I'm Still Missing You"; Viacom (PP-F).

W

Cal Walker (E/P): 1301 Briar Creek Road, Charlotte, NC 28208; 704-376-2949. **Credits:** Sunday Night Jazz (CP-R), "Sunday Night Jazz"; As It Was (CP-R), "As It Was 1969."

Richard Werner (E): 7878 Kingsbury Drive, Hanover Park, IL 60103; 312-289-5909. **Credits:** Chicago Brass Quintet (MP-A), "Virtuoso Brass"; Oriana Singers (MP-A), "10th Anniversary Collection."

Brian Wickham (E): 15-06 145 Place, Whitestone, NY 11357; 212-664-6360. **Credits:** NBC (CP-TV), "NFL Live!"; NBC (CP-TV), "Sportsworld."

Bruce Wildstein (E): 15425 Sherman Way, #261, Van Nuys, CA 91406; 818-785-7219. **Credits:** Bette Midler (MP-A) "Beaches"/Touchstone Pictures; Amii Ozaki (MP-A), album, Japanese release; Derek Ian (MP-SR), EP, European release.

Steven J. Wytas (E/P): 165 Linden St., New Britain, CT 06051-2413; 203-224-1811. **Credits:** CCSU Symphony (MP-A), "Holiday Performance"; Swanson/MacBeth (MP-AT), "The Blade Project"; Free World (MP-A), "The Tyranny of Culture"; Action!/TOTC (PP-CI), "Auto Crash"; Communication 480 (PP-F), "The Moodus Mystery."

RE/P

STUDIO UPDATE

Northeast

Greene Street Recording (New York) has upgraded its Amek APC 1000 console with syncronous reset, enabling users to automate, via SMPTE code, up to 99 complete console reconfigurations during a single mix. Data can be off-loaded to floppy disk for later recall. New equipment includes a Studer 820 multitrack, a pair of TC Electronik Spatial Expanders, an Eventide H3000 Harmonizer and the Circuit Design Polyframe compressor/limiter package. *112 Greene St., New York, NY 10012; 212-226-4278.*

Southeast

Real to Reel Recording Studio (Stockbridge, GA) has upgraded from 16-track to 24-track with the addition of a Trident 24 Series console and an Ampex MM1200 recorder. Other new equipment includes a Yamaha SPX-90II, Lexicon PCM 60 and -70, two Symetrix stereo 525 compressor/limiters and a Sony F1 digital processor. The control room has also been enlarged and renovated to accommodate the new equipment and to improve the acoustics. *4911 N. Henry Blvd., Stockbridge, GA 30281; 404-474-4776.*

Midwest

Hedquist Productions (Fairfield, IA) recently took first place in the London International Advertising Awards for "Radio: Best Use of Sound." The winning commercial, "Waltz," was produced for the Stamford, CT, Downtown Shuttle. The competition included more than 4,000 entries from 37 countries. *1007 E. Madison Ave., Fairfield, IA 52556; 515-472-6708.*

Chicago Trax Recording (Chicago) and its in-house production company, Music Chicago, received the Award of Distinction in the 1988 Communications Excellence to Black Audiences awards, for its production of "Coca-Cola USA." *3347 N. Halsted, Chicago, IL 60657; 312-525-6565.*

Southwest

Arlyn Studios (Austin, TX) has installed an 8-track studio to complement its 24-track room, designed to meet the needs of clients with smaller budgets. *200 Academy, Suite A, Austin, TX 78704; 512-467-2247.*

The Production Block (Austin, TX) has upgraded its Studio B to a 24-track MCI/TAC Matchless room with JBL 4330 monitors. Additional equipment includes an Ensoniq EPS sampler, a MIDI'd Linn-Drum, Yamaha DX-7, Roland Juno 60 and an Apple Macintosh with Performer software. *906 E. Fifth St., Austin, TX 78702; 512-472-8975.*

Southern California

Ameraycan Studios (North Hollywood) has refurbished Studio A and has installed a Solid State Logic 4000 G Series console with 56 inputs and Total Recall. *5719 Lankershim Blvd., North Hollywood, CA 91601; 818-760-8733; fax 818-760-2524.*

South Coast Recording Studio (Santa Ana) has purchased several new pieces of equipment, including an Otari MX5050 2-track, Roland DEP-5 effects processor, a second dbx 165A compressor/limiter, an Alesis Midiverb II, two AKG C414 mics, an Alesis drum machine and Roland keyboard/samplers. *1818½ N. Main St., Santa Ana, CA 92706; 714-541-2397.*

Alpha Studios (Burbank) has installed the first AMS/Calrec automated console on the West Coast, and the second in the U.S. The board features 56 channel inputs and 112 in remix. Also new is a Sony 3324A digital 24-track. *4720 W. Magnolia Blvd., Burbank, CA 91505; 818-506-7443; fax 818-506-4369.*

Canada

Comfort Sound (Toronto) has added a second 24-track and a Sola ac power isolator. The dual-multitrack system allows the mobile studio continuous recording without breaks for reel changes. *26 Soho St., Suite 390, Toronto, Ontario M5T 1Z7 Canada; 416-593-7992.*

England

Moles Studio (Bath) has installed a variety of equipment, including an Otari MTR 90 MkII 24-track, MTR-12 MkII ¼-inch 2-track, 64 channels of Optifile II hard disk storage for the DDA AMR 24 console, an Akai digital patchbay system, a Quantec QRS Room Simulator, AMS RMX 16, Neumann U67 mic, dbx 160 RM, two Teletronics LS-2As and two EAR 822Qs. *14 George St., Bath, Avon BA1 2EN England; 0225 333448.*

Italy

Medicina Studio (Bologna) has installed a New England Digital Synclavier and PostPro Direct-to-Disk system, said to be the largest of its kind installed in Italy. The system incorporates eight tracks of Direct-to-Disk recording on an optical disk system for more than 2GBytes of on-line sound storage.

Manufacturer announcements

Europadisk has installed a **Neve** Digital Transfer Console for preparing CD master tapes.

Audio Kinetics has installed three ES 1.11 synchronizers at Abbey Road's Studio 3 in London.

Solid State Logic has received its largest order to date from Disney-MGM Studios in Lake Buena Vista, FL, for an SL 56124 console with a 124-channel mainframe, 96 channels, a G Series Studio Computer with Total Recall and Instant Reset, moving faders and LCD metering. SSL has also sold more than 100 consoles in the Far East.

Soundmaster International has installed Integrated Audio Editing Systems at these facilities: ABC TV, Hollywood; SRO Recording, San Francisco; Gaffney Productions, New York; Buzzy's Recording, Los Angeles; VTR Productions, Toronto; Studio Place Royale, Montreal; Deschamps Recording, Toronto; and Master's Workshop, Toronto.

Total Audio Concepts has delivered a TAC SR9000 console to ATT Audio Controls, Australia, to be used in its rental stock.

WaveFrame has installed AudioFrames at eight locations: Wonderland Studios, Los Angeles, Stevie Wonder's personal-use studio; the home studio of Rob Arbittier, Wonder's programmer; Sheffield Recordings, Phoenix, MD; Sound Associates, New York; Processing Recording Studios, Greensboro, NC; Trax Sounds, Toronto; the electronic music department at the University of California at Santa Barbara; and West Productions, Burbank, CA.

RE/P

THE CUTTING EDGE

By Laurel Cash

Hot Stuff at January NAMM

Aphex Aural Exciter Type III

The Type III is the latest generation of Aphex's original Aural Exciter. The dual-channel Aural Exciter offers several new processing features and improved circuitry.

Among the newly added features are adjustable harmonic mixing (between odd and even), servo balancing of the I/Os, XLR connectors and the new Aphex VCA 1001.

There are two other new features worth noting. The first is the ability to use two modes of noise reduction. Mode A operates as a linear side-chain downward expander with a variable threshold. The expansion ratio of 2.5:1 permits the side-chain "mix" to follow the signal level below the threshold. Therefore, when the higher frequencies of the input drop below the threshold, the "mix" will drop at a proportional 2.5:1 ratio. This way, the signal-to-noise is not affected even with a great deal of enhancement.

Mode B is a new technique that operates to reduce source audio noise while the Aural Exciter Type III is enhancing the signal. This is said to allow the restoration of brightness and intelligibility to any noisy source, while improving the signal-to-noise ratio. Both modes apply to recording, broadcast and live sound applications.

Another new feature is what Aphex calls SPR, or Spectral Phase Refraction. It is suggested that through the many phases of recording, mastering, duplicating and reproduction, the phase of the low frequencies becomes delayed compared to mid and high frequencies. When this happens, the bass loses definition and the high end seems to get duller. The SPR is said to correct the bass delay anomaly to restore both clarity and openness. They say it significantly increases the apparent bass energy level without adding any EQ.

Circle (155) on Rapid Facts Card

Aphex Studio Clock

The Studio Clock is a SMPTE-to-MIDI converter. The difference in theirs is that it allows the user to create "tempo maps" from live musicians in real time. This is apparently possible because the Studio Clock contains the Human Clock Algorithm, plus a proprietary Aphex Rise Time Detection Circuit.

The algorithm lets the user create maps from MIDI clocks, MIDI note-ons, Quarter Note conductor click, live drummers or any other percussive sound source. This allows various players to write the map during different parts of a performance. The Studio Clock can also track and create a Map from a full audio mix.

Circle (156) on Rapid Facts Card

Aphex Feel Factory

The Feel Factory is an algorithmic feel composer, which allows the user to manipulate MIDI timing and velocity information from any existing sequence or drum machine pattern, intuitively and in real time. With the Feel Factory, a musician/engineer can assign one instrument or group of instruments to one of eight faders, move them forward or backward in time, while listening to the music. Therefore, the "feel" of the instruments and music can be adjusted. Algorithms can be created for unique rhythmic feels and recalled as presets via MIDI.

Circle (157) on Rapid Facts Card

Yamaha MIDI remote

The RTC1 is a MIDI-based control unit that adds new features and advanced capabilities to the DMP7, DMP7D and DMP11 digital mixing processors.

The RTC1 has a number of rotary encoders that give the user analog-like control of EQ, pan and special effects settings—just like the rotary pots in conventional mixing consoles. Separate Q, frequency and gain controls are provided for the high, mid and low EQ bands. The RTC1 retains the precision and reset capability of digital equalization. The assignable rotary controls extend this innovation to every parameter found in the DMPs.

The RTC1 can control up to four DMP mixers—four individual switches assign to RTC1 panel controls to the selected mixer. The 4-line LCD shows the user several effects parameters and settings simultaneously. Four rotary controls may be assigned to any parameter in the mix. On-board memory stores up to 50 sets of control assignments.

The RTC1 also provides new features for the DMP family such as fader grouping, single cross fade, and linked stereo input channel operation. Separate titles can also be assigned to each individual channel, fader group, cross fade group and mixer memory location.

Circle (158) on Rapid Facts Card

Aries Apollo recording console

Aries America has entered into the professional mid-range recording console market with the introduction of the Apollo.

The Apollo is a fully modular console, available in 16-, 24- and 32-input mainframes. An 8-channel expander unit is also available. Standard on all channels are balanced mic and line inputs with a new tape/line switching feature.

The output section is said to have full 16-track monitoring with 2-band EQ on 8 of the 16 returns. Access to all eight auxiliary sends on the 16 monitor returns adds additional flexibility. Also included with the Apollo consoles is full bar graph metering on the bus outputs, VU metering for the left and right output, PFL and an external power supply.

Circle (159) on Rapid Facts Card

Aries Astrid S/R console

Aries America enters another section of the marketplace with its new Astrid—a mid-priced sound reinforcement console.

A full-feature, fully modular console, the Astrid is available in both 24- and 32-input mainframes. As with Apollo, an 8-channel expander is also available.

Standard on the Astrid are balanced mic and line-ins, eight aux sends and 4-band EQ with selectable shelving points and sweepable midranges.

The output section offers eight effects returns and a 3-band EQ assignable to the group outputs or effects returns. An external power supply and VU metering are also standard features. Both consoles are available immediately.

Circle (160) on Rapid Facts Card

RE/P

Laurel Cash is *RE/P*'s executive consultant and is a freelance writer based in Los Angeles.

For more detailed information on the NAMM show, watch for the "NAMM Show Wrap-up" feature next month.

NEW PRODUCTS

TimeLine Lynx System Supervisor

System Supervisor is designed to provide integrated control over a wide variety of peripherals common to audio post through real-time software management. The 2U system includes two serial controller ports to allow control from one or two operator control devices. Additionally, four serial ports are available to control up to eight tape or film transports, using Lynx time code or film modules. Selectable system clock references, serial system diagnostics and five hardware expansion slots are included.

Circle (100) on Rapid Facts Card

Akai S950 sampler

Designed to succeed the S900, the 950 retains the 900's features and includes a number of new features. The sampling rate has been increased from 40kHz to 44.1kHz, yielding a usable bandwidth beyond 17kHz. Memory can be expanded to 2.25Mbytes, and a high density disk drive facilitates storage. Also included is a CD/DAT interface, allowing users to record samples directly into the unit. For CDs, 44.1kHz sampling is used; 32kHz or 48kHz is available for DAT.

Circle (101) on Rapid Facts Card

Texture software for WaveFrame AudioFrame

Texture is a MIDI music sequencer that runs in a graphic environment along with other AudioFrame applications, allowing easy composition sessions. The program uses a modular recording architecture of songs, patterns and tracks. Each pattern is 32 tracks deep, and up to 96 patterns may be recorded and linked to create a complete song. Step-time recording is supported, and multiple MIDI channels can be recorded on a single track. The program can be synchronized to all time code formats, VITC, MIDI clock and MIDI Time Code.

Circle (103) on Rapid Facts Card

Nady 501 VR portable wireless system

The 501 VR is designed for field production and provides five VHF frequencies between 170MHz and 216MHz. Companding Circuitry provides a dynamic range of 120dB and a frequency response of 25Hz to 20kHz. The receiver measures 2.85"x6", allowing it to be placed into equipment pockets or clipped on a belt. Available transmitters include a miniature lavalier or a lightweight hand-held. List price is $850.

Circle (104) on Rapid Facts Card

Ortofon PPA600 power amplifier

The two-channel amp delivers 225W of continuous output into 8Ω, or 350W into 4Ω with both channels driven. In the

Circle (38) on Rapid Facts Card

Circle (39) on Rapid Facts Card

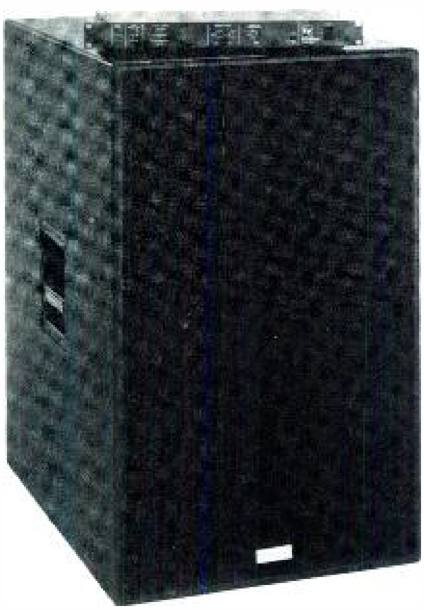
switchable mono bridge mode, the unit delivers 650W into 8Ω. If a dc voltage appears at the output terminals, the speakers will be disconnected. When the heat sink temperature reaches 70°C, the input signal will be attenuated; when the temperature falls to 60°C, the speakers will be reconnected and the gain will gradually increase to the previous level. Suggested retail price is $2,000.

Circle (105) on Rapid Facts Card

Electro-Voice DeltaMax DML-2181

The system uses the double 18-inch Manifold Technology subwoofer, which develops more acoustic output in a smaller package than in a traditional 18-inch system, according to the company. The enclosure holds two high-power, long-excursion woofers back-to-back in a central enclosed chamber. The system can be flown using an optional three-point flying system with aircraft-type pan fitting hardware.

Circle (102) on Rapid Facts Card



Sellmark CPA plastic attenuators

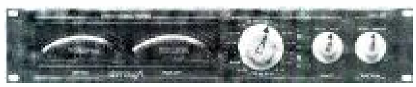
Manufactured in a controlled environment to ensure reliability, the attenuators feature a snap-in case design to facilitate rapid service checking and eliminates removal of the fader unit. Other features include low electrical noise, smooth operation, low torque and a working life approximately five times that of a carbon track-type fader, the company says.

Circle (107) on Rapid Facts Card

Dorrough stereo signal test set

The Dorrough 1200 is a modern version of the classic "Gain Set," allowing measurement down to -75dB. According to the company, it is the only test set available with the capability of measuring the stereo program signal in both the left and right or sum and difference formats. The test unit uses two Dorrough loudness meters, which indicate peak amplitude and the average on a single simultaneous display. Price is $1,650.

Circle (106) on Rapid Facts Card



Opcode software packages

The company has released four editor/librarian packages for the Macintosh, to be used with the Roland Multi D-Series, Roland MT-32, Yamaha REV-5 and the Oberheim Matrix-1000. (The latter is an upgrade of the Oberheim Matrix-6 program, while the rest are new.) All programs feature visual editing of all parameters on the screen, special editors for sections like drum setups, fully implemented patch librarians and "Bundles" for storing voices and performances together. Retail costs are $199 for the Multi D-Series, $149 for the MT-32, $199 for the REV-5 and $199 for the Matrix-1000. Owners of the Matrix-6 software can update for a fee.

Circle (108) on Rapid Facts Card

CD Saver from JBR Laboratories

CD Saver is a compact disc scratch remover and cleaner. It is simple to use, the company says, and restores discs to their original fidelity.

Circle (112) on Rapid Facts Card

Loudspeaker stand from Atlas/Soundolier

Weighing nine pounds, model SS100X is capable of supporting 100 pounds and can be adjusted from 66 inches to 90 inches high. It is made of aircraft aluminum and steel and is finished in a non-reflective ebony epoxy. Other features include a safety collar, tripod configuration base and five top adapters to accommodate a variety of speaker sizes and configurations.

Circle (113) on Rapid Facts Card

Rane programmable EQs

The company has introduced three programmable/MIDI equalizers: the MPE 28, a single-channel ⅓-octave; the MPE 14, a 2-channel 2/3-octave; and the MPE 47, a 4-channel, 7-band unit. All feature 128 internal memory settings accessible via the front panel controls or MIDI. Two curves can be modified or combined with a built-in program. Other features include Constant Q interpolating filters, programmable gain control, full mapping and sysex capability, and phantom power for remote control devices.

Circle (114) on Rapid Facts Card

E-mu Gold Series Library for Emax

The Gold Series contains 20 disks with sounds sampled and processed with the EIII and then digitally transferred to the Emax to preserve sonic clarity. The result is a significantly improved sample with increased fidelity and dynamic range, the company says. All diskettes are provided on pro quality TDK media.

Circle (110) on Rapid Facts Card

Ampex 478 low-print mastering tape

The tape line is intended for film and video post houses and studios where low-print mixdown is designed. A high-speed backcoating process provides exceptional packing, the company says, and reduces edge damage, pop strands and the need for slow rewind speed.

Circle (123) on Rapid Facts Card

dbx 563X Hiss Reducer

The latest addition to the half-rack Performer Series, the 563X combines an intelligent low-pass filter with dbx's True RMS detection. When musical or program high-frequencies are present, the dynamic filter opens, allowing the frequencies to pass. When the material is no longer there, the filter attenuates the HF energy, eliminating hiss. Suggested retail price is $219.

Circle (115) on Rapid Facts Card

GREAT COMPANY

Since the early seventies, Keith Olsen has been creating for the best. He's produced for the likes of Fleetwood Mac, Foreigner, Pat Benatar, and Whitesnake. His efforts have sold over *65 Million records.*

After working on every console imaginable, his response to the DI-AN was "It's the best sounding and most advanced console ever." And for his own facility, he bought one.

After all, one must consider the company they keep...

The DI-AN from TRIDENT

TRIDENT AUDIO USA
2720 MONTEREY ST. #403
TORRANCE, CA 90503
(213)533-8900 / Fax (213)533-7072

TRIDENT

GREAT COMPANY

Photo Courtesy of Goodnight LA Studios
Van Nuys, CA

Circle (41) on Rapid Facts Card

NEW PRODUCTS

Peavey Gatekeeper noise gate

The Gatekeeper is a 5-channel noise gate with one channel dedicated to vocal use. The vocal channel automatically opens when absence of signal is detected by the other four channels, which allows conversational level announcements to be made without readjusting thresholds from live performance levels. Release time is adjustable from 10ms to 600ms, with the threshold adjustable from 10dBV to constantly on. Suggested list price is $299.99.

Circle (111) on Rapid Facts Card

Publications

Trident Di-An brochure

Trident has released a 32-page brochure detailing the Di-An's design. The console provides the capability to store and reset every console function up to 512 times per mix.

Circle (148) on Rapid Facts Card

SSL publications

Solid State Logic has published a full color, annotated poster illustrating the modules that form the basis of the SL 4000 G Series Master Studio System. It is designed to give a complete system overview of the functions of each module and the available facilities. SSL has also published an operator's manual for the G Series, intended for new G Series users and existing users of the 4000 E and 6000 E. Sections include basic routing, signal flow, the patch, signal processor routing and an applications guide.

Circle (149) on Rapid Facts Card

Hal Leonard catalog

The latest Hal Leonard High Tech Catalog features publications on a variety of technical and musical subjects, including MIDI, sequencing, sampling, recording and synthesis.

Circle (150) on Rapid Facts Card

Valentino catalog

Valentino has released a four-color catalog detailing all the music and sound effects available from the company. A demo CD is also available.

Circle (151) on Rapid Facts Card

Fluke/Philips catalog

The 1989 test and measurement catalog from John Fluke Mfg. integrates the Fluke and Philips product lines into 19 major categories. Nineteen new products are featured, as are 19 new service programs. Ordering and warranty information is also included.

Circle (152) on Rapid Facts Card

Flex series processors from Rane

The Flex series products are modular signal processors. Each module is a self-contained processing function packaged in the half-rack format, and connects directly to pro equipment via 3-pin, ¼-inch and DIN connectors. The modular design allows a system to be custom-designed, with users purchasing only the functions they need. Basic processors will include a variety of mixing functions, splitters, crossovers, EQs, filters, compressors, limiters, program meters, headphone amps, output transformers and power supplies.

Circle (116) on Rapid Facts Card

Pro-Wipes for cleaning CDs

Available from KMD labs, Pro-Wipes are lint-free and absorb contaminants such as fingerprints and oils without damaging a compact disc. The wipes are cut in the shape of a disc to facilitate cleaning. Retail cost is $5 for a pack of 12.

Circle (117) on Rapid Facts Card

Wheatstone SDA-82 stereo distribution amp

The 8-channel unit may be used in a single-input/16-output or stereo input/eight stereo output configuration. All inputs and outputs have 3-pin gold connectors to allow load and source changes after installation. Each unit is individually active-balanced, and has 16 individual output gain controls to compensate for load-dependent gain and destinations.

Circle (118) on Rapid Facts Card

Prime Solutions hard disk software

Disk Technician Advanced software predicts and prevents hard disk problems, including hardware failure, for IBM PC/XT/AT, PS/2 and true clone computers. The software uses artificial intelligence to recognize telltale problem signs and determines testing and repair routines. A database stores test history, allowing the AI to refine its knowledge and eliminate random and intermittent problems. Retail price is $189.95.

Circle (124) on Rapid Facts Card

Jan-Al disk case

The Data FX case holds up to 115 3.5-inch disks and has a paper pocket for music and manuals. Custom sizes, colors and designs are available.

Circle (109) on Rapid Facts Card

Renkus-Heinz subwoofer system

Model SUB-152 is designed to be used as a sub-bass support for full-range speaker systems such as the company's FRS Dyna-Gard series. By restricting bandwidth, the bandpass design increases efficiency to 106dB. The system contains two 15-inch woofers in a specially tuned, 6-cubic-feet enclosure, providing frequency response to 45Hz. Standard connectors are dual banana and Neutrik NL4MP, and a terminal strip option is available. Suggested resale price is $1,700.

Circle (119) on Rapid Facts Card

Ampex 472 professional audiocassette

Designed specifically for studio use, the cassette is intended for such uses as dubs and client copies. It is available in Type I normal bias and Type II high bias. Available lengths are five, 10, 15, 30, 45, 60 and 90 minutes.

Circle (120) on Rapid Facts Card

Cuedos software for Yamaha DMP7

The software runs on Atari ST computers with a minimum of 1Meg in RAM, and provides full control of all level, muting, auxiliary EQ, pan and outboard effects programs on the DMP7. All parameters can be saved as a sequence of preset snapshot mixes, and entire sequences can be saved and reloaded onto floppy disk and the DMP7's own internal memory.

Circle (127) on Rapid Facts Card

New releases of TRF production library

TRF has released more than 70 CDs of new production music in various styles, including AV/industrial, New Age, electronic, sports, opening and closing themes, rock and classical. Various versions and lengths are included, and CDs are indexed at cutting points within each selection. Prices are $20 per CD. Libraries are also available on album at $9 each. Either format can be received on approval.

Circle (128) on Rapid Facts Card

Commander Electronics Lynex stereo sampler

Available from the Russ Jones Marketing Group, the Lynex is a 16-bit stereo sampler for the Atari ST. Housed in a 1U package, the device comes with 1Mbyte of internal RAM, which provides 10.5 seconds of mono and 5.25 seconds of 16-bit sampling at 50kHz. Memory can be expanded to 32Mbytes by adding a second box that houses expansion cards. The controlling software resides in the Lynex's memory, allowing the ST to run other software simultaneously. Suggested retail price is $3,300.

Circle (126) on Rapid Facts Card

Stantron Fan Tray

The cabinet cooling system takes up 1.75 inches of rack space in a standard rack or console, and can be ordered with three, six or nine fans. Features include an on-off switch for each group of three fans and circuit breakers for easy resetting if an overload occurs. Each fan delivers 89 CFM (free air).

Circle (121) on Rapid Facts Card

Hybrid Arts switching program

HybriSwitch allows users to load as many programs as their computers' memories will allow, the maximum being 16. Programs can then be switched back and forth without quitting and rebooting, turning the programs into one program. Data can also be switched between compatible programs without leaving the environment. All Hybrid Arts MIDI products, with the exception of the DX and CZ Android series, are compatible with the program. It runs on the Atari ST, and the retail price is $29.95.

Circle (129) on Rapid Facts Card

Belden multi-pair snake cable

The new line of cable features individually jacketed and shielded pairs for protection against signal loss. Eight different pair constructions are available, ranging from 4- to 32-conductor pairs. The cable is available in 100-, 250-, 500- and 1,000-foot put-ups; suggested retail is $640 for 1,000 feet of 4-pair cable.

Circle (130) on Rapid Facts Card

NEW PRODUCTS

Software and hardware updates

Sony expands MXP-3000 consoles

The 3000 is now available in a 56-input frame size, known as the MXP-3056VF, and includes a hard disk automation terminal and display boards placed in the center control section. Additional new features include stereo echo returns and improved cuing facilities. Traditional features from the 3000, such as low noise hybrid amplifiers in all mic pres, differential input/output stages and modular I/O channel strips, have been retained. Suggested list price is $190,000.

Circle (132) on Rapid Facts Card

Storage system for E-mu EIII

To be used with the Emulator III, the RM45 is a rack-mountable storage and backup system that uses 45Mbyte removable hard drive cartridges. By using a high-speed SCSI interface, the unit will load a 4Mbyte bank of sounds in less than nine seconds. Suggested list price is $2,500 and includes one 45Mbyte cartridge.

Circle (133) on Rapid Facts Card

Digital interface for AudioFrame

The WaveFrame Universal Digital Interface Module (UDI-4) allows the AudioFrame to directly receive and send digital audio information from a variety of digital sources with different sampling rates and formats. A plug-in module that occupies one slot in the system's Audio Rack, the UID-4 supports most professional and consumer digital formats. Price is $8,950.

Circle (134) on Rapid Facts Card

Expansion of Yamaha DMP7D mixing processor

An "All Digital" version of the DMP7, the DMP7D is designed for users who need a mixing processor with digital audio I/O. It supports Sony, Mitsubishi and R-DAT recorders and the AES/EBU, Sony SDIF-2 and S/PDIF formats. The unit is an 8-input, 2-output processor with multifunction motorized faders and three built-in digital signal processors. Price is $5,995.

Circle (138) on Rapid Facts Card

Digital interfaces for Yamaha DMP7D

Yamaha has released four IFU digital interface units that conform and distribute the digital audio data signal between a digital audio recorder and the DMP7D, allowing users the configure the unit into almost any digital system. The IFU1 is for Mitsubishi X850s, the IFU2 is for the Sony 3324 and the IFU3 is for Sony 1610/30 recorders.

Circle (139) on Rapid Facts Card

New chassis size for Amek BCII mixer

A new chassis size of 24 modules is available for the BCII, allowing such desk configurations as 16/4/2 and 18/4 to be produced in a minimum of space. The chassis can be supplied with attached meters or with meter hoods to house from two to seven VU or PPM meters. Additionally, meter hoods have been enlarged to provide for three full rack units in height for easier mounting.

Circle (135) on Rapid Facts Card

Enhancements to GML fader automation

The Series 2000 Automation Environment adds intelligent master machine control, Ethernet networking capability, an optional graphics display and extended editing capability. It is compatible with GML's Version 5.xx software and the standard interface hardware.

Circle (136) on Rapid Facts Card

New version of Opcode film software

Version 2.1 of Cue, the software for music editors and film and television composers, now includes the capability to import and play back MIDI Files. Other features include grabbing SMPTE cue points directly from video, cue sheets, tempo search, automated streamers and punches, custom music sketch paper and MIDI-triggered sound effects. The software runs on the Macintosh and retails for $595. Owners of previous versions can upgrade for a nominal fee.

Circle (137) on Rapid Facts Card

Optional interfaces for Akai S1000

Akai has introduced digital and hard drive interfaces for the S1000, S1000HD and S1000PB samplers. The IB102 connects an Atari or Supra hard drive to the sampler. Retail price is $99.95. The IB103 is an SCSI interface that allows standard SCSI-compatible drives to be used for high-speed data transfers. Retail price is $159.95. The IB104 is an AES/EBU stereo digital I/O port, allowing the sampler to be linked to other digital devices. Retail price will be announced.

Circle (140) on Rapid Facts Card

Memory expansion for Akai S1000

The EXM005 Memory Expansion board expands the S1000's memory by 2Mbytes. Each board yields about 12 seconds of stereo sampling at 44.1kHz. Because the S1000 is provided with 2Mbytes of RAM and has a capacity for three boards, a fully configured unit can offer 48 seconds of stereo sampling at full bandwidth. Retail price is $1,349.95.

Circle (141) on Rapid Facts Card

New version of Alchemy software

Version 1.2 of Blank Software's Alchemy brings true 16-bit stereo sound editing, processing and network support for the E-mu EIII and the Roland S-50 and S-330. It will also allow EIII and Ensoniq EPS owners with an SCSI port option immediate access to SCSI sample dump capability. The update is available to registered owners for a $25 handling and materials fee.

Circle (142) on Rapid Facts Card

New options for Soundcraft 6000 console

New options for the 6000 include stereo input modules and MIDI mute implementation. The stereo input modules are designed to increase the flexibility of the console. The MIDI Mute system stores up to eight songs, each containing 100 patches of complete mute settings. Each song and patch can be individually named using the keypad and 10 numeric buttons.

Circle (143) on Rapid Facts Card

Twister automation upgrade

Soundcraft's Twister automation system is now incorporated with SA software, a SMPTE-based system that offers real-time noise gates on each channel, 16 VCA subgroups, graphic editing of fader data and a comprehensive cue list incorporating MIDI trigger events.

Circle (144) on Rapid Facts Card

TAC Bullet console

The compact-format mixing desk is housed in a free-standing chassis that can be rack-mounted in a 19-inch bay. The meter hood is separate and contains seven 15-segment LED meters. Basic configuration is 10/4/2, and the console contains four stereo aux returns, one on each of the subgroup modules and two on the master/stereo module. Talkback, oscillator and monitor speaker outputs are provided with a speaker selection switch. An AFV interface will be available later in the year.

Circle (122) on Rapid Facts Card

Polydax midrange driver

The PR 17 TX 100 is a 6-inch driver that features a flat-edge 1-inch voice coil wound on a Kapton former, TPX cone material and treated flat surround. The magnet assembly has been anodized black, allowing for more efficient heat transfer from the voice coil for increased power handling. The company says that because of its high internal losses, TPX cones resolve the problem of stationary standing waves along the diaphragm, resulting in high efficiency coupled with high quality midband reproduction.

Circle (125) on Rapid Facts Card

Capture! software

From Imagine Music Group, Capture! is an editor/programmer/librarian that runs on the Atari ST. Features include icon control windows, standard GEM interface, sizable multiple windows and programmable randomizing. It is available for the Roland D-10/110/20, MT-32 and D-50/550 and the Ensoniq ESQ-1, ESQ-M and SQ-80. Price is $99.95.

Circle (131) on Rapid Facts Card

New generation of BGW 750 power amps

The 750F and -G are the fourth generation of BGW's 750. IHF output power has been boosted to 2,000W, yielding an overall dynamic range of 113dB. Other features include switchable subsonic filters, input mode switching and internal crossover card capability. The 750G includes an LED level display and LED status indicators for heat sink and transformer overtemperature. The 750F has LED modulation and true clip indicators.

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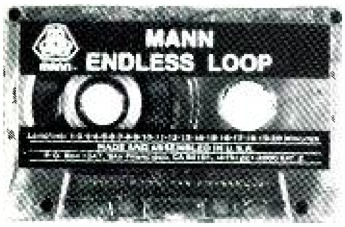

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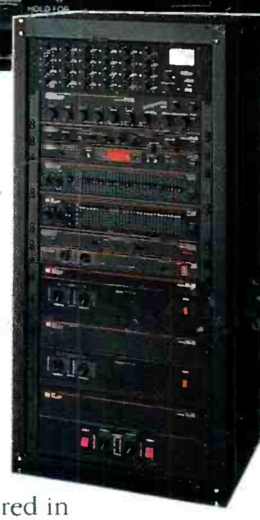

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