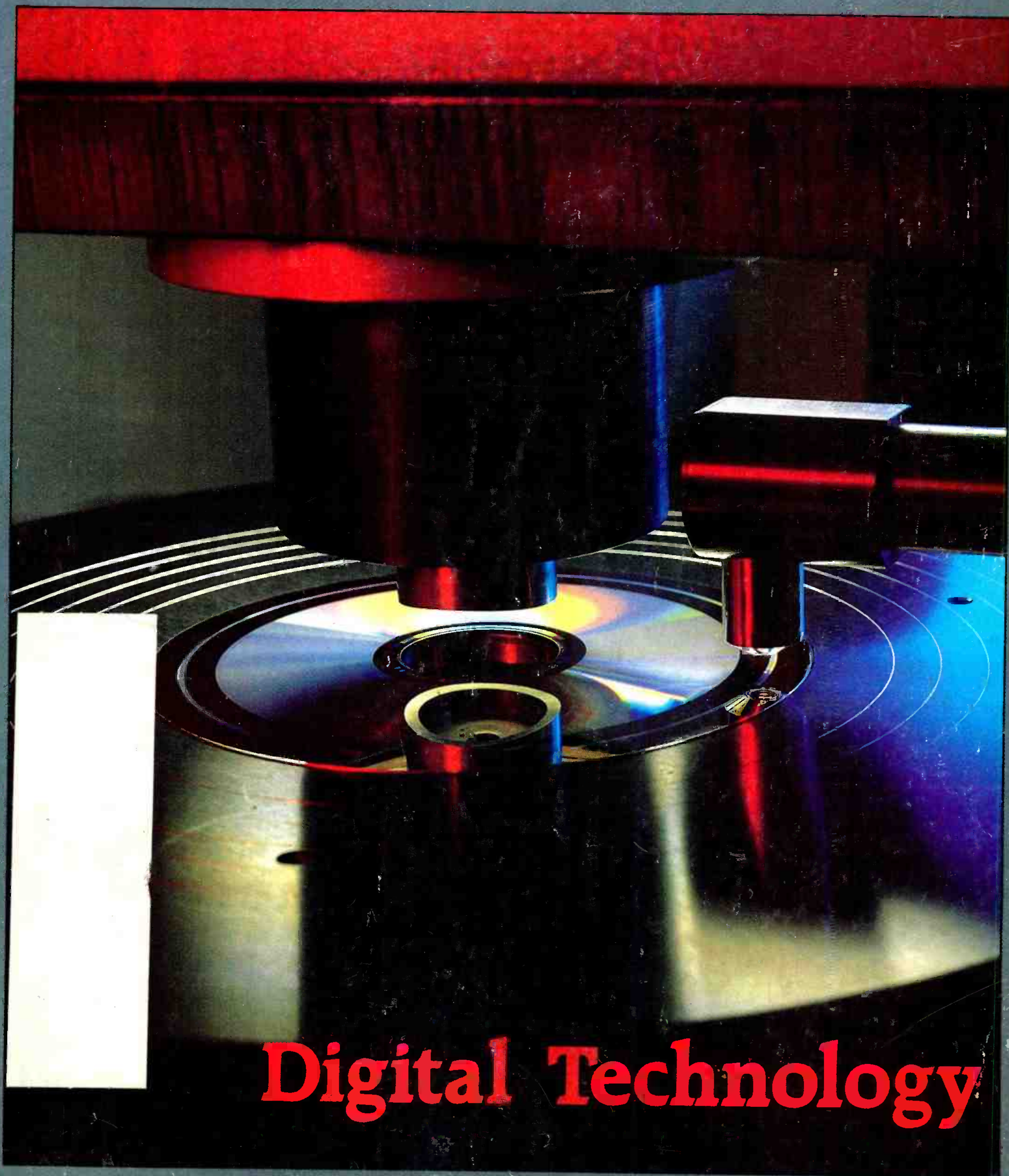
March 1988

Recording ENGINEER/PRODUCER

The Applications Magazine for Audio Professionals

$4.00

AN INTERTEC PUBLICATION



Digital Technology

Only one digita
puts a complete recordin

The Synclavier® Digital Audio System and the Direct-to-Disk™ Multitrack Recorder combine to form The Tapeless Studio, a complete computerized recording environment for effects, music and dialogue post-production. The heart of The Tapeless Studio is the proprietary high speed Synclavier computer, which integrates and controls the functions of a diverse array of hardware and software for the generation, manipulation and storage of sound.

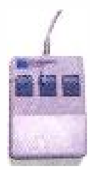
The Synclavier system's outstanding power, speed, and ease of use derives from the computer's unmatched ability to address massive amounts of digital sound data. A maximum configuration of 8 gigabytes of on-line storage offers the largest capacity of any system of its kind.

Synclavier memory can be configured to match your specific requirements.

Winchester hard disks provide on-line "workspace" for projects in progress and libraries of frequently used sounds. High density floppies are used for storing individual edit sequences, sound libraries and installing factory software updates, while 15 Mb streaming cartridges provide hard disk back-up. Optical Disks offer a

full 2 gigabytes of memory for mass on-line recording storage of sound data.

Random Access Memory is used for the recording, editing and playback of short instrumental sounds or sound effects: 32 Mb are available, again the largest of any audio system. Additional system RAM provides storage for the 200-track Memory Recorder — maximum size is 8 million events.



Synclavier user interfaces include the computer terminal, mouse controller, the 76-note velocity/pressure keyboard, and the optional Digital Guitar. System interfaces such as the MIDI, SMPTE, Multi-Channel Output Distributor and external timing modules may also be added.

Software updates keep the Synclavier system at a state-of-the-art level. Recent enhancements make available advanced features like cut-and-paste editing for the Synclavier Memory Recorder, mouse-based editing for sampled sounds, database organization of archived sounds and effects, and full SMPTE event and MIDI editing capabilities.

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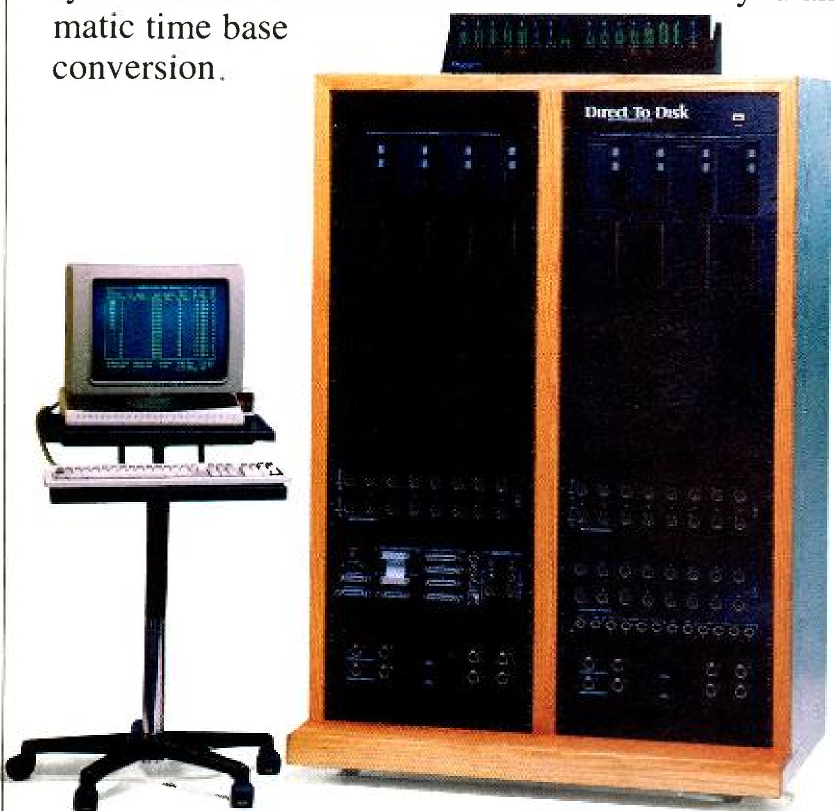
udio workstation studio at your fingertips

The Direct-to-Disk Multitrack Recorder, featuring unsurpassed 16-bit/100kHz fidelity, can now be configured in stand-alone 4, 8, and 16-track units. It records and plays back audio information from a dedicated network of winchester hard disks, and backs-up on convenient, reliable data cartridges. Direct-to-Disk also incorporates its own SMPTE synchronizer with automatic time base conversion.



Like the Synclavier, the Direct-to-Disk system is based on proprietary computer hardware and is software updateable. Current software implements advanced non-destructive cut-and-paste style editing. A maximum continuous recording time of over 3 hours is available at a sampling rate of 50 kHz with a 16-track system.

The Tapeless Studio has been designed as a modular, open architecture system, which can be expanded as your business grows and as new technology becomes available — while protecting your original investment in the system as well as the time spent learning its operating procedures. New England Digital provides comprehensive training for you and your staff.

Technical assistance and service are available on-site or by phone from any of our offices worldwide. There are company-owned sales, service and training offices in Los Angeles, New York, and Chicago, with branch offices in Nashville, Toronto, France, Great Britain, West Germany and Japan, in addition to our corporate headquarters in Vermont.

We recognize that when you make a significant capital investment in equipment, you are also, in part, investing in a company. New England Digital is a research-oriented, American computer company dedicated to a single goal: building the finest computer-based digital audio systems for the music, recording and post-production industries.

New England Digital ™

Circle (1) on Rapid Facts Card

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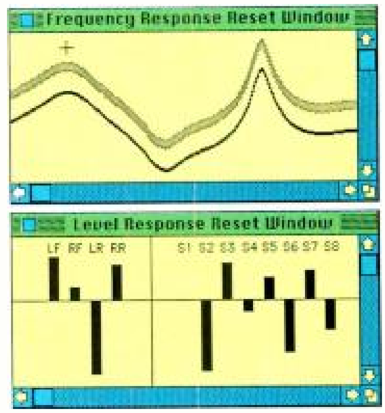

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Departments

On the Cover:
At Technetronic's new "second generation" CD manufacturing plant in West Chester, PA, a master-stamper is optically aligned prior to punching the center hole. Special effects photo courtesy of Mobay Corporation in Pittsburgh, PA.

Photo by Richard Holzer

RECORDING ENGINEER/PRODUCER—Volume 19, No. 3—(ISSN 0034-1673) is published monthly by Intertec Publishing Corporation, 9221 Quivira Road, P.O. Box 12901, Overland Park, KS 66212-9981. Second-class postage paid at Shawnee Mission, KS, and additional mailing offices. POSTMASTER: Send address changes to Intertec Publishing Corporation, P.O. Box 12901, Overland Park, KS 66212-9981.

Is What You Get, What You Need?

The descriptors "digital audio workstation" and "direct to hard disk recording" are not synonymous.

As described in Paul Tydelski's article on page 33, "A Historical Perspective of the Digital Audio Workstation," there is a classical definition of workstation that defines all the necessary components. Alternatively, direct to hard disk recording describes a mechanical process involving the input, storage, and retrieval of audio material using high-capacity magnetic-media computer disk drives.

As you will learn from our report, it's important to understand these differences. Buying a system, any system, is not necessarily going to fulfill all your needs. Your specific application should determine which type of system and configuration is most appropriate. There seems to be considerable confusion among potential consumers.

The primary questions seem to be:

- who are the players.
- what type of systems are offered.
- what are the systems capabilities.
- when are the systems going to be ready for the marketplace.
- how much do/will they cost.

I also sense a different type of confusion among manufacturers. Where do their products stand in relation to other workstation or hard disk recording systems? And equally important, how do they relate to conventional analog and digital products such as tape machines, audio consoles, and signal processors?

In an effort to clarify these points and to establish an operational basis for hard disk recording systems and digital audio workstation technology, *RE/P* has developed this "Digital Audio Workstation Report." Although the need for this type of information is obvious, the opportunity to provide it, in the developing stages of a trend, is unique.

Establishing a method of evaluation and developing a manageable means of implementation were but two the challenges we had to overcome. The method chosen was an operational report. Because there is already so much marketing *hype* in the pro audio industry, it was important that we produce as objective of a report as possible.

This questionnaire was developed with the cooperation of the manufacturers and was designed to challenge the capabilities of all the systems. Our representatives required a demonstration of the claimed capabilities. In this way, the units were evaluated based on their current operational status.

Going into this report, the only knowledge these companies had was the type of questions which might be asked. And although several manufacturers asked to review the questions in advance of the test, none were allowed the opportunity.

I would like to thank Scott Gershin, Jeff Largent, and Richard Elen for their efforts in the Los Angeles, Boston, and London areas respectively. They had the unenviable task of facing the manufacturers with these demanding questions. Additional thanks go to all the manufacturers whose representatives willingly and cooperatively gave of their time to assist *RE/P* in its search for some basic, but sorely needed information.

New columnist

On a separate note, it's my pleasure to welcome Jeff Burger to the *RE/P* masthead as a feature writer and columnist for our new "Understanding Computers" column. He joins us with a wealth of knowledge and experience, having dealt extensively with both audio and computers.

His credits include technical writing for several professional audio manufacturers and trade publications, and composition, engineering and production of electronic music. He is president of Creative Technologies in Los Angeles, and is the author of "The Murphy's Law MIDI Book," published by Alexander Publishing.

With this column, we are establishing a framework of computer knowledge and information for the pro audio industry. We will break the computer down to its fundamental elements and carefully explain terminology, applications and capabilities. This will help you make informed buying decisions and better use this technology. This column won't teach you how to write code, but it will help you better understand computers.

Certainly, there are many books and magazines covering computers, but we feel distinctive in offering the audio/computer combination in an ongoing format—written specifically for pro audio readers and their applications.

As we continue with this column we welcome your input and suggestions regarding specific information you would like to see covered. RE/P

Michael Fay
Editor

IN THE PAST WE HAD A BIG ADVANTAGE OVER THE COMPETITION. NOW WE'VE GOT A SMALL ONE.

Until UREI's 813 Time Align® Monitor entered the studio, speaker systems had become a "smear" on the industry. A "time smear," in which high and low frequencies subtly assaulted the ear because they arrived out of sync. The results were general listener fatigue and unrealistic sound, particularly on lead instruments and vocals.

The UREI 813 solved the "time smear" problem with Time Alignment™, unifying sound into a single point source. This dramatic breakthrough, along with other major technical advances, soon established the 813 as the industry standard.

Now UREI introduces less of a good thing: the 809 Time Align® Studio Monitor. The 809 delivers all the engineering depth of its big brother, but at a compact size and price that's ideal for small control rooms and near-field applications.

UREI's 809 features a remarkable, all-new 300mm (12") coaxial driver that achieves a true one-point sound source, superior stereo imaging, and tight bass. It incorporates a unique titanium diaphragm compression driver that unleashes unequalled high frequency response.

The 809 has exceptional power handling capabilities, high sound sensitivity, and low distortion. It accomplishes precise acoustic impedance matching and smooth out-of-band response with UREI's patented high-frequency horn with diffraction buffer. And its ferrite magnet structures assure the system's high sensitivity drivers will not degrade with time and use.

UREI's Model 809 Time Align® Studio Monitor. Smaller package. Smaller price. Same impeccable "813" sound quality. See how it fits into your studio today.

JBL Professional
8500 Balboa Boulevard
Northridge, CA 91329

UREI

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NEWS

Computer BBS geared toward pro audio community

A new bulletin board service, A-V Sync is operating in Atlanta, providing software, text files and conference-oriented message bases for people working in various segments of the industry.

The service is geared toward sound, recording, audio for video, film, graphics, RF and FCC items, and general production. Aside from public domain software and program demos, there is an extensive collection of MIDI files that is updated regularly.

The service operates on a non-profit basis, with a yearly charge of $25 for maintenance and utilities. Payment can be made by Visa, Mastercard or Discover Card, or by personal or company check.

The system can be reached by modem at 404-320-6202. Computer settings are: no parity, 8 data bits and 1 stop bit (N-8-1). Phone inquiries can be made at 404-438-5858.

Duplicator publishes master tape guide

Diskmakers Inc. has published the "Guide to Master Tape Preparation," a guide to the requirements for preparing and handling master tapes for pressing and duplication. The company, which is located in New York, Chicago, Philadelphia and Puerto Rico, created the book in response to clients' questions about master tape specifications and duplication and pressing requirements.

The free 16-page book is available by calling 800-268-9353 or 215-627-2277.

News notes

Rane Corporation has moved into new corporate headquarters at 10802 47th Ave. West, Everett, WA 98204-3400; 206-355-6000.

Audio Services Corporation has appointed the following rep firms: ASR Enterprises, covering Maryland, Virginia, eastern Pennsylvania, Delaware and New Jersey; Bencsik Associates, representing Florida and Puerto Rico; Kodo Associates, covering Minnesota, North and South Dakota, and northwest Wisconsin; and Piper Associates, covering Massachussetts, Maine, New Hampshire, Vermont, Connecticut and Rhode Island.

CORE Rents Sound, a new pro audio, video and lighting equipment rental company, has opened at 10440 Westpark, Houston, TX 77042; 713-977-9500.

Oberheim has moved to 2015 Davie Ave., Commerce, CA 90040; 213-725-7870.

AVS Systems, the audio equipment division of Vaughn Communications, has moved to 7901 Computer Ave. South, Minneapolis, MN 55435; 612-831-3435.

Aphex systems has reached a licensing agreement with Vestax for the use of Aphex's Aural Exciter circuit. Vestax will use the circuit in various products for the musical instrument industry.

Ampex Magnetic Tape Division has begun air-delivering product to European and Asian distribution centers. The division has also increased its North American ground fleet to 34 trailers, increasing the number of ground deliveries to its American and Canadian distribution centers.

SPARS On-Line

[**Editor's note:** The SPARS On-Line column will not appear this month, but will return in the April issue.]

NAB Convention: April 9-12

The 66th annual convention of the National Association of Broadcasters, promises to deal with a number of important issues facing the broadcast industry. The show has gained in importance to *RE/P* readers as more studios perform audio-for-video and broadcast-related work.

Equally important is the ability to see equipment that was introduced at the AES Paris show, without having to wait for the domestic show in November.

After two years in the Dallas Convention Center, the convention returns to Las Vegas, and will be split into three venues: the Las Vegas Hilton, the Las Vegas Con-

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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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Qualified:
United States (Domestic Only) **$24.00**
Foreign **$45.00**
Non-qualified:
United States (Domestic Only) **$30.00**
Foreign **$60.00**

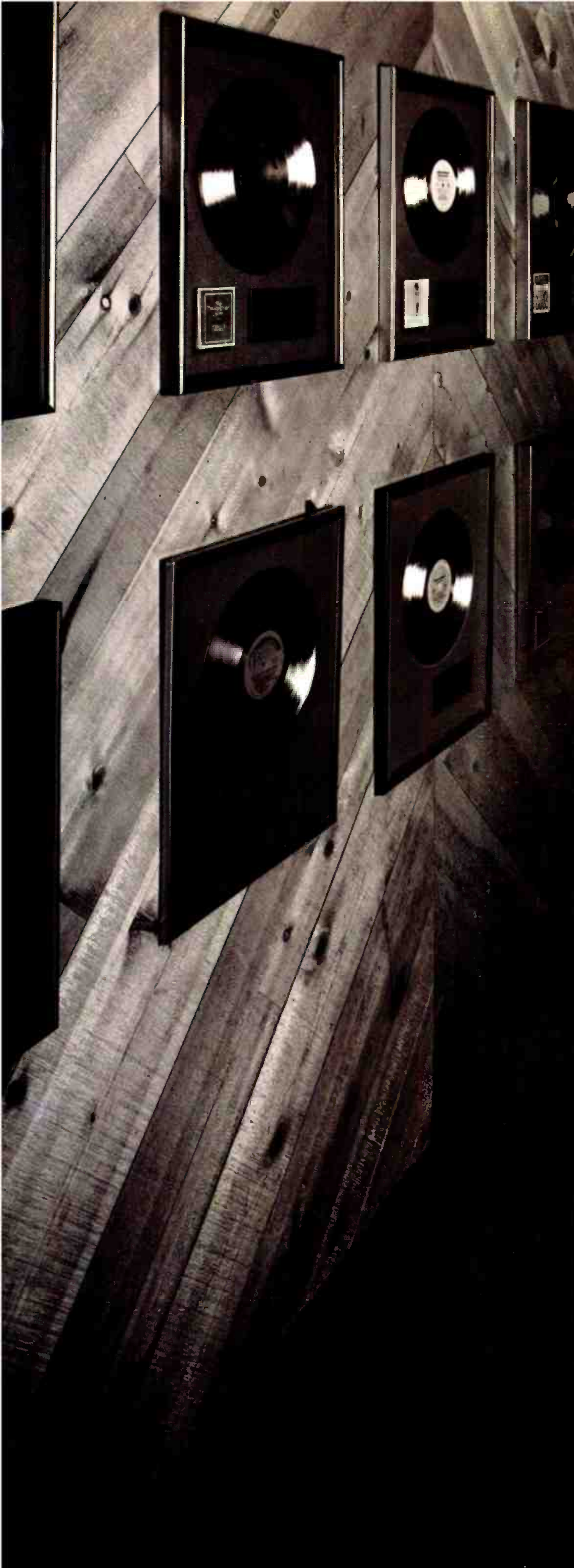
Optional airmail for non-qualified readers is also available for an additional $55.00 per year. Foreign subscriptions are payable in U.S. funds only by bank check or money order. Adjustments necessitated by subscription termination at single copy rate.

Recording Engineer/Producer is not responsible for any claim by any person based on the publication by **Recording Engineer/Producer** of material submitted for publication.

ISSN 0034-1673 $4.00 + $0.00.

Member, Business Publications Audit of Circulation

Sustaining member of Acoustical Society of America



WHAT YOU DO WITH THE M-600 MIXER IS YOUR BUSINESS.

That's why we've designed it to meet or exceed your most demanding requirements. And made it the easiest, most flexible professional mixing console you'll ever work with.

The M-600 is modular. Which means you can custom configure the console to *your* audio or video production needs. The M-600 lets you choose up to 32 input channels, or you can start with 16 or 24 input channels and expand the board as your needs change. Optional stereo modules can also be added to provide even more line inputs for MIDI instruments and video production convenience.

Installation and wiring is exceptionally easy. The M-600 is the only modular mixer that's available with all the necessary finished cables and installation hardware. And that can eliminate a lot of installation hassles and expense. At the same time, no other mixer at its price gives you multi-pin, computer-type connectors for quieter, more secure connections.

But the real pleasures of the M-600 will only be evident after it's in your studio. Up to 64 stereo or 128 mono inputs can be accessed directly from the top panel. A patch bay can be added for fast, flexible routing. That's convenience.

The M-600 has all the features you'd expect in a professional mixing console. Like balanced insert patch points on all inputs, PGM busses as well as the stereo master buss for increased signal processing capability. Plus sweep-type parametric EQ, balanced inputs and outputs, phantom power, talkback/slate channel and all the audio performance you'll ever need. Without the exorbitant price you don't need.

So check out the M-600 modular mixing console. It's ready for fame when you are.

TASCAM

Circle (6) on Rapid Facts Card

vention Center and an outdoor exhibit area. With a record number of exhibitors, 28 technical sessions with 150 papers, five panel discussions and four workshops, the show promises to be a busy one.

Show hours are: April 9-11, 8:30 a.m.-6 p.m.; April 12, 8:30 a.m.-4 p.m. For more information, contact NAB at 202-429-5350.

Selected exhibitors

Following is a selected list of exhibitors that would be of most interest to *RE/P* readers. As of press time, booth numbers for some exhibitors were not available; locations of some exhibitors may also be different when the show opens. For the final list of exhibitors and their booth numbers, consult the official program at the convention.

Acoustic Systems	5224
Adams-Smith	1513
ADC Telecommunications	3480
ADM Technology	4369
AEG Bayly	719
Agfa-Gevaert	3880
AKG Acoustics	1245
Alpha Audio	2256
Alpha Video & Electronics	2980
ALTA Group	5917
Amber Electro Design	1201
Amek/TAC	145
Ampex AVSD	3302
Ampex MTD	3302
AMS/Calrec	3372
Amtel Systems	2444
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ANT Telecommunications	170
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Cetec Vega	1127
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EELA Audio	159
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Eventide	871
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RE/P

MANAGING MIDI

By Paul D. Lehrman

Making MIDI More Human

Music notation is a wonderful language. To the trained reader, a square inch of paper can contain details of pitch, volume, timbre, expression, harmony, duration, rhythm, dynamic changes, articulation and many other items crucial to music performance.

It is a language designed by human beings, for human beings and has served as the chief means of communicating musical ideas between human beings (at least in Western civilization) for seven centuries. It's changed and developed over the years, but its constancy is remarkable: on paper, a Palestrina *Kyrie* means exactly the same thing today as it did the day it was published.

But like most human languages, although music notation can convey a lot of information in very subtle detail, it is not terribly precise. It can be open to interpretation. Just as different people will not make exactly the same sounds when they read a loud a piece of text on a page, different musicians can take the same piece of written music and perform it quite differently.

MIDI is a very different kind of language. It is precise, and is not open to interpretation. Assuming all other circumstances are identical, a performance of a MIDI sequence in one concert hall or studio will be exactly the same as a performance of the same sequence anywhere else. There are no two ways to play a MIDI sequence; if you change any aspect of the performance, you have to change the sequence itself.

MIDI is not a great human language. For humans to deal with it at all, it must be translated and displayed—as numerical data, in some higher-level alphanumeric or symbolic form, or graphically, using grids, "piano rolls," or line segments.

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

One of the ways to make MIDI more human would be to let musicians work with it in the form of music notation, by putting a notation "front end" on a sequencer program, for example. But getting that to work right is a lot harder than you might think.

Think of a chime patch on a synthesizer. How long is that note? Well, if you were notating it, it would be a dotted-half at least, but as far as the keyboard, and therefore MIDI, is concerned, it was a 16th note. Which is correct?

That's an extreme, but more subtle examples can easily be found: a long string patch with a slow attack, a harp patch in which the sustain pedal chokes the sound (many stock patches for FM synths to that) a drum machine that ignores duration completely, or a sampler in which velocity-switching chooses between two voices with very different envelopes.

Similarly, while music notation can tell you what you're *going* to play, through the use of articulation and dynamic symbols, it is terrible at describing what you've *just* played. When you see a fermata, you know to hold the note an extra beat. But can a computer, when it reads a note from a keyboard that's being held an extra beat, know to put a fermata over it? In the same vein, how does a computer differentiate between a staccato quarter-note and an ordinary eighth note?

Away from a computer, standard music notation is still the best way to communicate musical ideas.

No wonder every program designer has felt he has had to develop his own way of displaying MIDI data. But we can't ignore musical notation, either. Many musicians have spent years becoming fluent in it, and away from the computer, it is still the best way to communicate musical ideas among people.

Certainly, it can be used in a sequencer front end for some operations, such as transposing, scale changing, moving sections around, looping, generating rhythmic templates and even entering notes for parts where preserving minute differences between notes in onset or duration aren't important—like string pads, for example. And of course, it's indispensable when parts are going to be played (or doubled) by human performers.

Notation and accurate editing of MIDI data has so far been an either/or proposition. Some software programs have tried to provide both, but one side of the equation always suffered. A more successful approach has been to modularize the process and provide two different programs with two different orientations, which could exchange data with each other.

Unfortunately, the results so far have been programs that are slow, or not particularly accurate, or that are able to work only with other programs from the same manufacturer, thereby making them useless to anyone who would prefer to use different software along the way.

The ultimate solution is a program in which one has the best of both worlds: a powerful sequencer with a good notation module (with the emphasis on the former), or modular programs that are flexible, well-matched and fast. There are several reasons for optimism on this front. One is simply the falling price of computer memory, which allows even-greater amounts of RAM to be used in the average computer, which in turn allows programs to get larger and more powerful.

The development of MIDI Files (in which I talked about a few months back) as a standard format exchange of MIDI data, is a step in the right direction as well. For notation, however, in which graphic elements like spacing, beaming, text, etc., are significant aspects of a document, MIDI Files will ultimately prove too primitive and slow. Perhaps a new form of format will emerge so that notation programs will be able to exchange data.

That may sound far-fetched, but as I write this, Apple Computer has startled a lot of people by announcing a Sample File Standard, so who knows—it could happen.

In the meantime, there are at least six companies working on new high-level notation programs, and we should start seeing the fruits of their labors in the first half of this year. Me, I'll be happy when I just have a program that will let me assemble a sequence with my favorite sequencer program, quantize the daylights out of it and print out an accurate and uncluttered-looking score from it. RE/P

Neve—for the digital experience.

Preparation of master tapes for Compact Disc is a highly exacting process, requiring precise and repeatable control of levels, filtering and equalisation without degrading the original quality.

To achieve these requirements when compiling from digital recordings it is essential to keep the processing in the digital domain, so that the signal remains digital throughout the whole recording and reproduction chain.

The Neve Digital Transfer Console – designed by the world leaders in digital audio processing – provides a digital stereo mixing and processing chain developed from proven Neve DSP technology, with the unique facility of 'snap shot' automation of all parameters under either manual or SMPTE time code control.

Circle (8) on Rapid Facts Card

The Neve DTC has two stereo digital inputs accepting either Sony PCM 1610/30 or AES/EBU formats with automatic sensing of pre-emphasis, and one stereo analogue input, all with individual gain and balance trim.

The mixed signal may be processed by the comprehensive Neve Dynamic Range Control and the unique Neve Formant Spectrum equaliser with peaking/shelving selection and variable Q; the EQ may also be used in the Dynamics side chain, and a delay facility is available to give 'zero attack time' dynamics.

Second-order high-pass and low pass filters are structured before the processing section.

Digital output metering is by high-resolution instantaneous-reading bargraphs; a separate digital bargraph provides metering of analogue signal levels and dynamics.

The stereo digital output may be either Sony PCM 1610/30 or AES/EBU, but at the same frequency as the input, with or without pre-emphasis.

A separate stereo analogue output provides monitoring facilities or a feed to analogue effects units etc.

The console is capable of automated operation of all parameters from SMPTE time-code using up to 200 'memories' which may also be manually accessed; the integral floppy disc system may be used for permanent storage of these 'snap shot' configurations.

Neve

A Siemens Company

PROCESSING SOUND AT ITS PUREST

NEVE ELECTRONICS INTERNATIONAL LIMITED, CAMBRIDGE HOUSE, MELBOURN, ROYSTON, HERTS. SG8 6AU.
TELEPHONE: 0763 60776. TELEX: 81381. FACSIMILE: 0763 61886. CABLES: NEVE CAMBRIDGE.
RUPERT NEVE INC, BERKSHIRE INDUSTRIAL PARK, BETHEL, CONNECTICUT 06801.

SOUND ON THE ROAD

By David Scheirman

The Right Stuff

Have you ever wondered who it is that decides why one mixing console is such a requested item that the manufacturer is always out of stock because every major P.A. company has plunked down a deposit for several consoles...and others can hardly be given away? Why is it that one of two brands of amplifiers seem to be proliferating like weeds and others cause touring sound technicians to turn up their noses and sneer?

Or perhaps wondered what secret group of bench technicians somewhere gives the thumbs-up sign on a pro sound product so that they become all the rage...and then you have to spend unbudgeted, hard-won dollars *modifying* the product to get it to even work in the field?

The answer to these questions like who decides such things is simple. The people who decide are you and I. Every time we suspend our own critical, common-sense view on sound equipment that has been fine-tuned over many years, and agree to cooperate with purchases of expensive gear that are made purely for me-too status reasons, we do ourselves a disservice.

"You too, can own the amplifier that rocked the Mega-Bang Festival." Or perhaps, "It's got to be good...Luftwaffe Parade, Europe's latest art-rock sensation, is pictured here, playing through a whole rack of the things."

Marketing is the word, sales is the game, and advancement of the true state of the art of live concert sound is often a very secondary consideration. Lest some of you manufacturers out there start bristling, let me commend the broad range of companies that have developed product lines over the years that do work well for touring sound...so well, perhaps, that some of the companies have been able to grow with the industry and diversify their efforts into other related markets.

The problem that creeps in is that, in an effort to sustain growth and maximize profit potential once the assembly line is tooled up and the market is targeted in the sales reps' cross-hairs, sincere engineering efforts can give way to crass commercialism. And, the plain truth is that many products must be custom-modified for professional use.

If I see a picture of all those tiny power amps packed into a small case, and go out and buy them for myself and put them in a case like the picture and then start doing major concert work, will everything be fine? Well, maybe. Or maybe I'll need to put special refrigerated-air cooling systems in those racks, or perhaps cut extra ventilating holes in the sheet metal frames to allow them to operate in such a densely-packed environment.

Where do I find these things out, and how do I know if I need to modify commercially available gear? And how do I know what to modify?

In the early days the right stuff just plain did not exist.

In the early days of live concert sound, the "right stuff" just plain did not exist. It was dreamed up, bread-boarded late at night in hotel rooms, patched together from refugee components, borrowed from radio stations and recording studios and built from parts ordered by mail.

The quest for the "right stuff" has been a long and arduous one, and it just doesn't seem right for the brass ring on the merry-go-round to turn out to be made from styrofoam, or for the Golden Fleece to be a promotional toy held aloft by a quota-bound sales manager.

Did the Holy Grail turn out to be made of plastic? Did Babe Ruth round third base, to thunderous applause, only to find that homeplate had been swiped? No. And the live concert sound business still has a wide variety of commercially available products to choose from that are designed and built with integrity.

While the marketing techniques are seemingly sprouting more quickly than weeds after a spring rain, there has never been a time that concert sound system designers had so many options to choose from for items like consoles, crossovers, power amps and loudspeaker systems. Separating the wheat from the chaff takes some time, but it is time well spent. Sometimes even the wheat can use a little doctoring. A critical eye toward how products under consideration might fit into your own idea of the ideal sound system can often point to simple fixes and effective improvements that make a system more reliable, safe or effective.

One enterprising sound company met this challenge by removing the electrical power supply section with transformers from each unit and mounting them across the floor of the rack; each individual amplifier had a high-tech quick-disconnect umbilical cable running up into the rack and neatly strapped in place.

The ability to change, modify and perform serious research is a valuable company skill, and not one that is cultivated overnight. Once learned, however, it can mean the difference between greatness and mediocrity. To locate the "right stuff" in today's pro sound equipment line-up, a live sound technician or sound reinforcement company must be willing to look beneath the skin, past the glitter and tinsel.

If those responsible for authorizing the purchase of expensive audio equipment are not capable of verifying a device's electrical and mechanical integrity, it should not be difficult to contract with someone who can. An aware and informed customer group can make a better watchdog over the entire audio manufacturing industry than any government agency, public policy or consumer protection bureau.

Looking for the "right stuff" can be both fun and educational, and the search shouldn't stop with the glossy sales brochures. Major concert sound companies don't operate that way, and neither should younger sound companies, traveling shows or live-music venues shopping for new gear. The occasional problem products will all but disappear when no one buys them. RE/P

David Scheirman is *RE/P*'s live-performance consulting editor and president of Concert Sound Consultants, Julian, CA.

UNDERSTANDING COMPUTERS

By Jeff Burger

Computers Aren't Just for the High-Priests of Silicon

Computers. They're everywhere. In the office. In the home. Yes, and in the studio. They now play such an integral role in the music making process (and the management thereof) that we felt the time has come to take an in-depth look at the inner workings of our electronic friends as well as their applications in an ongoing column.

Our first thought deals with overcoming fear on the part of newcomers. Despite the enormous complexity of the technology we deal with every day in the modern recording studio, computers are still often thought of as being reserved for the high-priests of silicon. In truth, the task of actually writing computer programs requires just as much specialized experience as do the jobs of producers, engineers and musicians in their own ways.

Fortunately, much of the work has been done for us. For the most part, we work with pre-programmed applications. The better the program is, the easier it is to use. The computer is then simply an environment in which to use these new tools.

So if you're even the least bit timid of the idea of sitting down in front of a computer, remember this: you no more have to know about programming to put computers to work for you than you need to know the intricacies of the circuits inside an equalizer in order to boost a frequency. Having said that let's pursue the more positive approach that, like any creative medium, the more one knows about the subject, the more one can get out of it.

We can break down the applications of the computer in today's recording studio into two basic categories: management and production.

Let's face it, in your office...the recording studio is many things, but it is still—at the bottom line a business with similar needs to most other businesses. Word processing software provides a quantum leap over the typewriter for many reasons.

The most important one is that completed documents are stored semi-permanently on disk and can be retrieved instantly for modification, correction and re-use without having to re-type the entire work. We're talking major time saving here, folks.

Word processing allows the user to concentrate first on the message itself without regard to formatting considerations like margins and style. Text is easily manipulated, copied, moved, deleted and/or corrected using metaphors like "cut-and-paste." When the words themselves are satisfactory, margins, page breaks and their formatting options can be easily invoked to create the proper presentation.

Other time-saving facilities such as "search-and-replace" and spell-checking can also make life in the fast lane much easier.

A digital revolution is taking place in the control room and computers are at the heart of it.

Databases serve as a way to keep track of similar pieces of information and often replace traditional storage methods such as index files and filing cabinets. Candidates for database storage include your address/phone book, tape library log, equipment inventory, client list, prospects, booking schedule and more.

Computers can easily perform a search for information that matches certain criteria across many files with blinding speed and report the results in a variety of useful ways. For example, a client database might be printed out in its entirety on gummed labels for a mailing announcing the grand opening of your new Studio A Taco Bar. You might also want the computer to do a more selective search, such as a list of all the rental companies who can supply you with that new XYZ rack-mount hexaphonic electro-zither for a session that's already started.

Databases are also invaluable for cataloging libraries of music cuts, video clips, sound effects, records, digital samples, etc.. Whether you have a client more prolific than Frank Zappa or you've just returned from Botswana with 200 hours of F1 samples, giving the computer a few key words will tell you that the elusive footage you seek is stored in Vault C, Isle 41, Shelf 5, Reel 10.

Computers can also help you manage your funds. At the simplest level, accounting packages simplify the "old in/out" game of Accounts Payable and Accounts Receivable. Reports can be generated that give you a better handle on your cashflow or provide a profit and loss statement for your banker. (Yes, there's usually a command that answers the musical question, "How Much Money Do I Have Today?")

Spreadsheet software allows you to have "what-if" sessions with numbers to project accurate forecasts, make competitive proposals and determine the profitability of new projects like the topless sauna in Studio D. A spreadsheet basically provides a series of rows and columns into which numbers or formulas can be inserted which interact continuously to provide a current "bottom line" which you can make informed decisions.

A digital revolution in the control room is taking place in the music-making process and computers are at the heart of it. Most digital or digitally-controlled studio gear shares many of the elements found in the computer — a microprocessor, memory, display, keypad and storage.

Indeed many types of modern equipment such as digital samplers, delays, and equalizers are more akin to computers with highly-specialized hardware and software than to musical instruments.

The single most influential aspect of all this is that the MIDI protocol has provided a path for standardizing communication between instruments, computers, processing gear, synchronizers and even mixing consoles. It's no great revelation that a good deal of today's music is made with the aid of MIDI sequencing. Computers act as a composing and arranging station to help us manage and control the electronic orchestra found in so many sessions.

The computer also serves as a programming extension for today's knobless generation of studio gear. One way manufacturers have cut costs is by replacing individual knobs with a handful of multi-function controls and a small display

Jeff Burger is owner of Creative Technologies, a computer consulting company in Los Angeles and a technical writer.

Simple Intelligence

Let's assume that results are what's important, not which tools you use.

Audio production is at least as much art as science; there will always be those who ascribe a magical aura to certain pieces of equipment. But if your client list is built on quality and consistency rather than techno-voodoo, the DCM 232 in-line console with CAT automation can give you more of both.

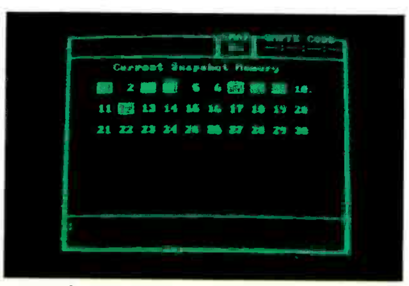
You need more console, not more headaches.

You're working for more demanding clients, on bigger projects, with tighter deadlines. You need greater flexibility, expanded features, enhanced performance. What you don't need is a "megastar" console—or the jumbo mortgage that goes along with it.

DDA's AMR 24 has already set new standards of audio performance and versatility in the "classic" split configuration. Now the in-line DCM 232 combines the accuracy of digital-quality audio, the flexibility of digital control and the capacity to handle a pair of synchronized digital 32 tracks.

CAT Central Automation Terminal: engineered to speed your work flow.

The DCM 232's Central Automation Terminal controls one of the most ingenious automation systems ever to shorten a mixing session. Along with the precise fader and muting control you'd expect, the CAT system includes advanced functions like *Channel Copy* that lets you duplicate a channel's signal flow as many times as you need to. The computer will recall a "snapshot" of most console switch settings manually or via SMPTE code.



With all of its convenience functions, this CAT won't leave footprints all over your tracks. The DCM 232 maintains an overall dynamic range of 100 dB with at least 22 dB headroom at each stage, thanks to exacting calculation of every circuit component.

Circle (10) on Rapid Facts Card

A console investment that instantly doubles your returns.

Each of the DCM 232's channels, *including the four band EQ section,* can be split during mixdown. So a 56 channel frame can handle as many as 112 inputs from samplers, synths and digital storage media. You'll probably run out of control room space before the DCM 232 runs out of inputs.

The advantages of the DCM 232 in-line console with CAT automation are explained more fully in our brochure. To get a copy, write on your letterhead to the appropriate address below.

DDA DCM 232 SPARS

Klark-Teknik Electronics Inc., 30B Banfi Plaza North
Farmingdale, NY 11735 (516) 249-3660

Unit #1, Inwood Business Pk., Whitton Rd.
Hounslow, Middlesex, UK TW3 2EB
01 570 7161

window which looks into a larger world behind the front panel. The computer can serve as an expanded control panel, a library of sounds or settings and a display for things like sampled waveforms.

Computers also provide other creative aids that were not possible until just recently. They can act as a semi-random compositional and/or sound generating device (juxtaposition inversions of Mozart's melodies against the reversed beat of The Bleeding Safetypins' latest hit using the sound of an Albanian aardvarks mating call grafted onto the attack of a car horn). They can calculate the best tempo to use to align eighth notes with the death blows in the new Kung Fu movie you're scoring or generate a SMPTE hit list for placement of sound effects.

The computer also serves as a programming extension for today's knobless generation of studio gear.

Several high-end digital workstations have combined all of these elements into a single package capable of scoring, orchestrating and creating the sounds for an entire piece of music or soundtrack. Fairlights, Synclaviers, Waveframe and others are examples of computers tailored to the recording process. Many others are appearing for the specialized purpose of automating the placement of sound effects and dialogue.

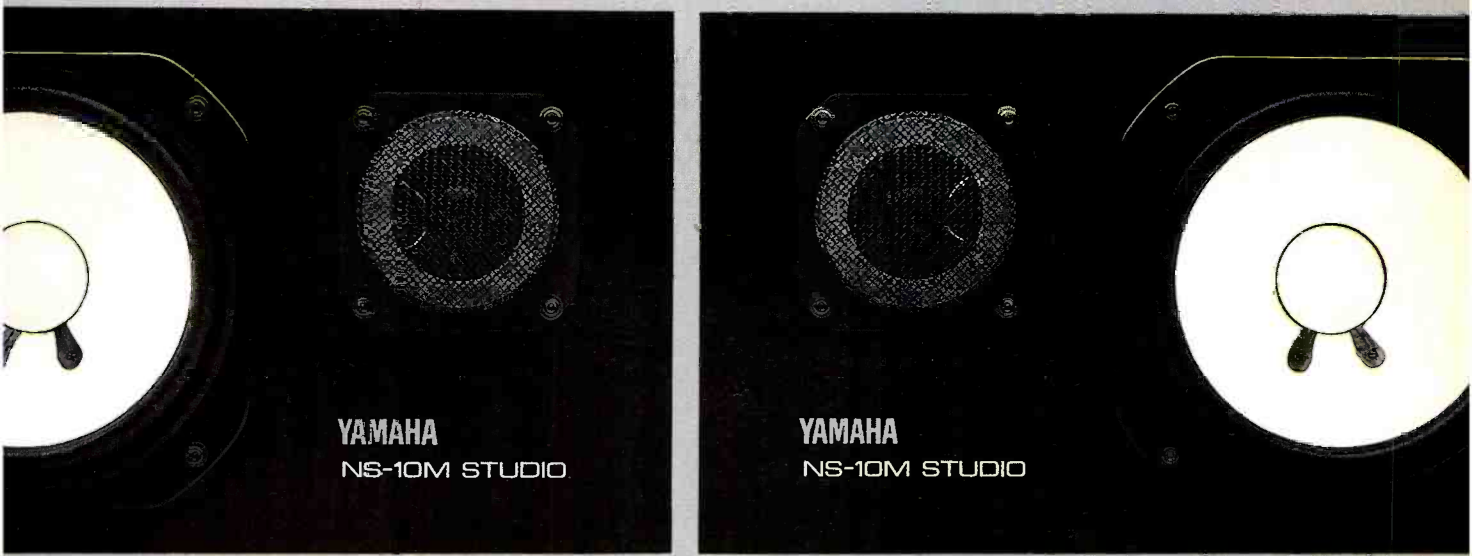
While the computer is a virtual microcosm, it can also be your key to the world without leaving your seat. Using modems, a studio in Los Angeles can download a synthesizer patch from an online sound library in New York at 3 a.m. and musicians can literally phone in their performances from different parts of the globe. Telecommunication also provides access to other valuable on-line services such as tour booking, travel information, professional forums, electronic mail, industry news and bulletin boards.

The computer is a virtual microcosm, and your key to the world without leaving your seat.

And the list goes on! We've gotten a bit silly here, largely to emphasize that computing doesn't have to be taken too *seriously* to get *significant* results. Next month we'll pop the hood and take a look at the basic elements that make most computers tick and in future columns we'll cover various types of hardware, software, applications and tips as we continue to explore the role of the computer in the studio.

RE/P

Listen to what engineers in 47% of all recording studios have already heard.* And what they haven't.



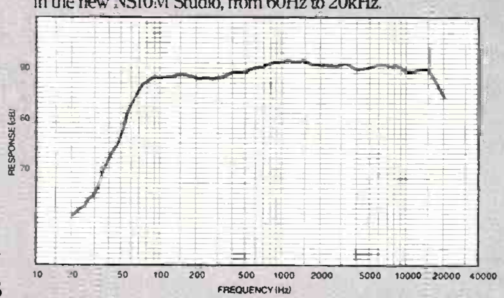
What we're going to tell you about the new NS10M Studio reference monitor may sound familiar, and for good reason.

The NS10M Studio is based on our legendary NS10M which, judging from its popularity in recording studios, delivers the near-field acoustic imaging that most engineers have demanded.

So rather than listen to competitive monitors to improve the NS10M, we listened to professionals like you.

And ended up retaining the best aspects of the NS10M's performance, while enhancing others.

Frequency response remains exceptionally smooth in the new NS10M Studio, from 60Hz to 20kHz.



That means you can expect the same smooth frequency response. The same high-power handling capability. And the same ability to take on the stresses of a longer duty cycle. All while maintaining accurate spatial definition without inducing listener fatigue.

Listening to what engineers needed also meant making refinements designed specifically for the studio environment.

Like connector terminals that accept large-diameter speaker cable for optimum signal quality. A 3.5cm dome tweeter with built-in acoustic damping tailored for near-field monitoring. And a horizontal configuration so the NS10M Studio never gets in your way.

And because it takes a pro to better service a pro, the NS10M Studio is sold exclusively through authorized Yamaha Professional Audio dealers.

The NS10M Studio.

Proof that at Yamaha, we listen to professionals as much as professionals listen to Yamaha.

Yamaha Music Corporation, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Avenue, Scarborough, Ontario M1S 3R1.

*1987 MIX Magazine Annual Recording Industry Directory

YAMAHA®
Engineering Imagination™

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Digital Audio Workstations: An Overview

By Richard Elen, Scott Gershin and Jeff Largent

An overview of digital workstation technology and an introduction to *RE/P*'s digital audio workstation report.

During the last decade, we've seen drastic and continual changes in technology. The only way to weather it, is to try your best to keep up with the developments and keep abreast of the basics in which that technology is based.

It is because of this that prompted us to create this report, an update on the technology of random access digital workstations, including hard disk recording and digitally sampled information.

In our pursuit for the truth (as best as we could search out and obtain), we met with each of the manufacturers and had them show us their most up-to-date systems. Our hard-and-fast rule was that if we couldn't see it, it still resided in that special zone called "Vaporware."

To be fair to the manufacturers, we did accept Beta versions or updates that will be released within a few months after this article is published. When this occurred, we made special note letting the public know that we didn't see it but were told that it would be made available shortly. (As to its true existence, we'll leave that up to you, but you can't say we didn't inform you.)

We tried to focus our questions on specifications and applications, thus trying to avoid those questions that lend themselves to opinion rather than facts. It is up to you to use the information given and, combined with information from other sources, decided as to which device or devices would best fit your needs.

One note: With the rapid growth in technology, mostly software-based, be prepared to deal with certain bugs that are inevitable with each product. This is a small price to pay for our addiction toward new technology.

Richard Elen is a recording engineer, producer and president of Creative Technology Associates in Somerset, England. Scott Gershin is sound designer and effects editor at Todd-AO/Glen Glenn in Los Angeles. Jeff Largent is sound designer at Target Productions, Boston.

Additional stories:
Digital Audio Workstation Report, pg. 38
Additional Questions, pg. 19
An Historical Perspective, pg. 33

Additional Questions

As we developed our master questionnaire, there were many good questions which didn't fit our format properly. These additional questions may be valuable when making purchase decisions.

1. Does the company maintain a network of full-time, factory trained service technicians nationwide and worldwide? If so, how many?
2. Does the company have an organized, ongoing training program, staffed by full time-training specialists?
3. Has the company established a track record of product enhancement through regular, reliable software updates?
4. Does the company have experience in the field servicing of studio equipment?
5. How many systems, that primarily use hard disks for recording/editing/playback, are currently being used by customers?
6. What is the cost-per-second, per-minute, of the system's storage medium?
7. Specifically, what is the cost-per-second of RAM? Of magnetic disk? Of optical storage?
8. How transportable is the system, and what is the actual size of various modules that complete a standard system?
9. How long would it take a sound engineer marginally familiar with computers to learn to do useful work on the system?

Editing

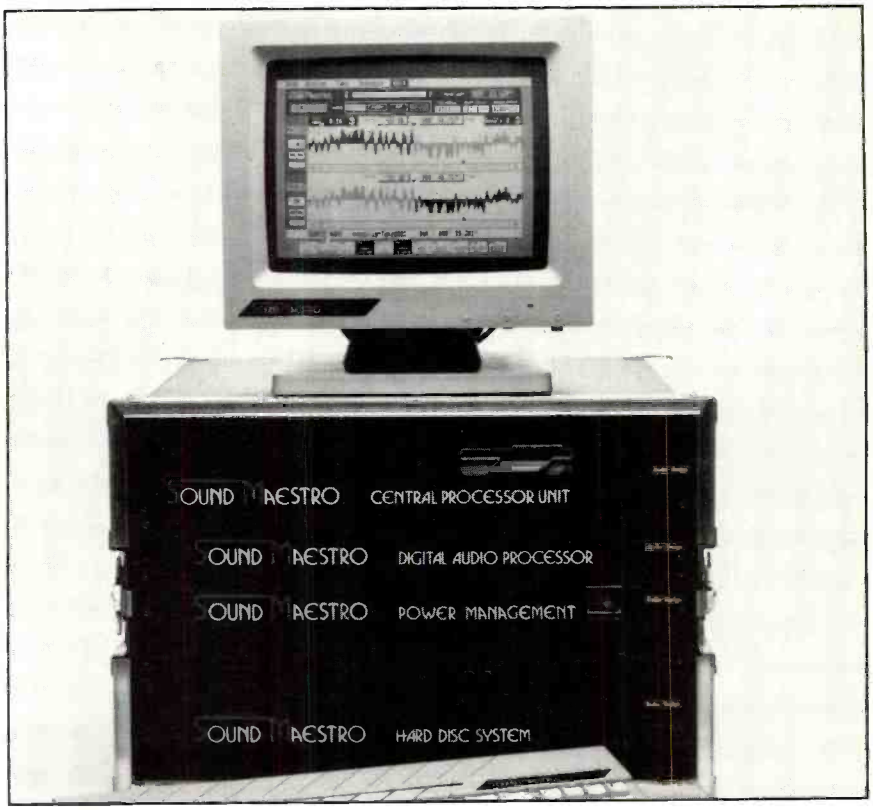
10. Describe the editing functions that can be performed on the audio material that is played in real time from disk.
- *Fade up/Fade down editing?*
- *Crossfade editing?*
- *Variable edit parameters?*
- *How many channels can be edited at the same time?*
- *Any limitations crossfade editing has on the number of channels of disk playback?*
- *Any limitations on the number of edits?*

Continued on page 33

Audio + Design Recording

The Sound Maestro is a program recording and editing system completely integrated in a rack-mount format. The unit features a mains management (filtering) unit, a central processing unit (Atari Mega ST and A+DR hardware), a digital audio processor, hard disk units, and VDU and keyboard/mouse control.

The standard hard disk units run under SCSI, which is capable of allowing up to 61 disk units to be configured. The hardware can be installed in a machine room with up to a 25 meter extensions to VDU and keyboard controls. Digital interface is via AES/EBU and one option of either PCM 701/601/501-es. Access to PCM 1610 via A+DR modified PCM 701 is available with ADMIX digital fader/mixer.

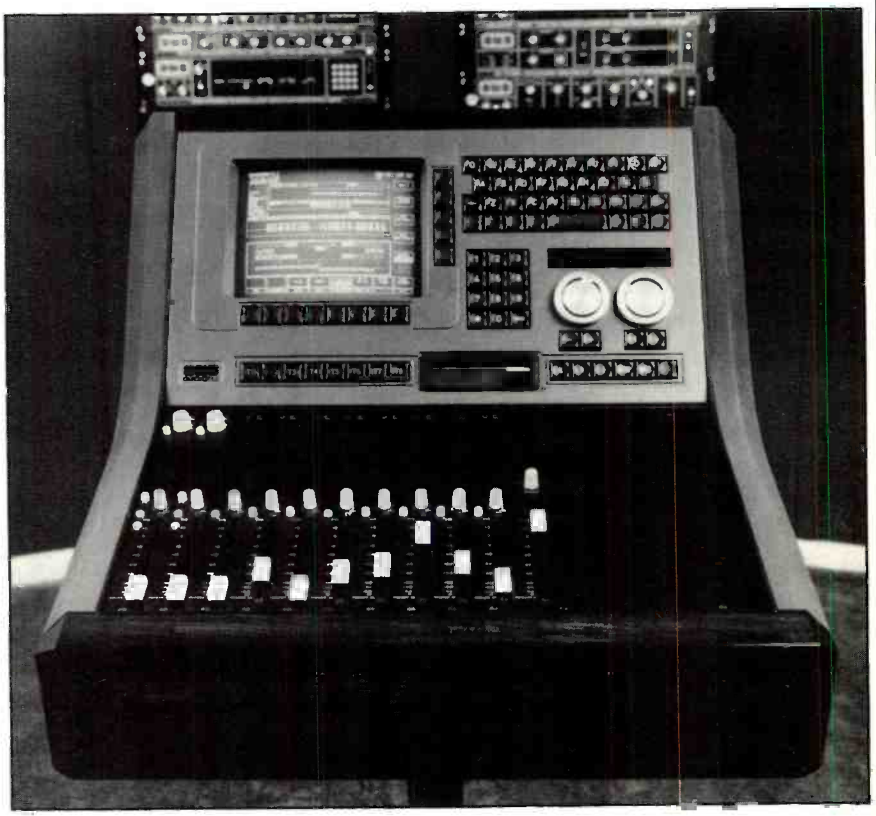


AD+R Sound Maestro system

Advanced Music Systems

The AudioFile by AMS is strictly an audio editor created for the needs of the television, film, and the music mastering industry. AudioFile is an 8-output, 16-bit, hard disk recorder with a sampling rate of 44.1kHz or 48kHz. It uses cut and paste techniques and places desired sections of recorded material to one of the eight outputs.

If you investigate the AudioFile, be aware of some of the differences in terminology. For example, a sampled section of audio is called a "cue." Once a cue is manipulated (fade in/out, cut and paste, etc.) it is called an "event" and a track is called an "output." Only an event can be placed onto an output. The AudioFile can hold between two and six hours of audio information. AMS hopes to incorporate soon a digital signal processing card to the system.



AMS AudioFile

Track team for the long run.

Will the digital multitrack you buy today stand up to the challenges of tomorrow? It will if it's the Sony. In addition to 24 superb digital audio tracks, the PCM-3324 supplies a wealth of creative recording options.

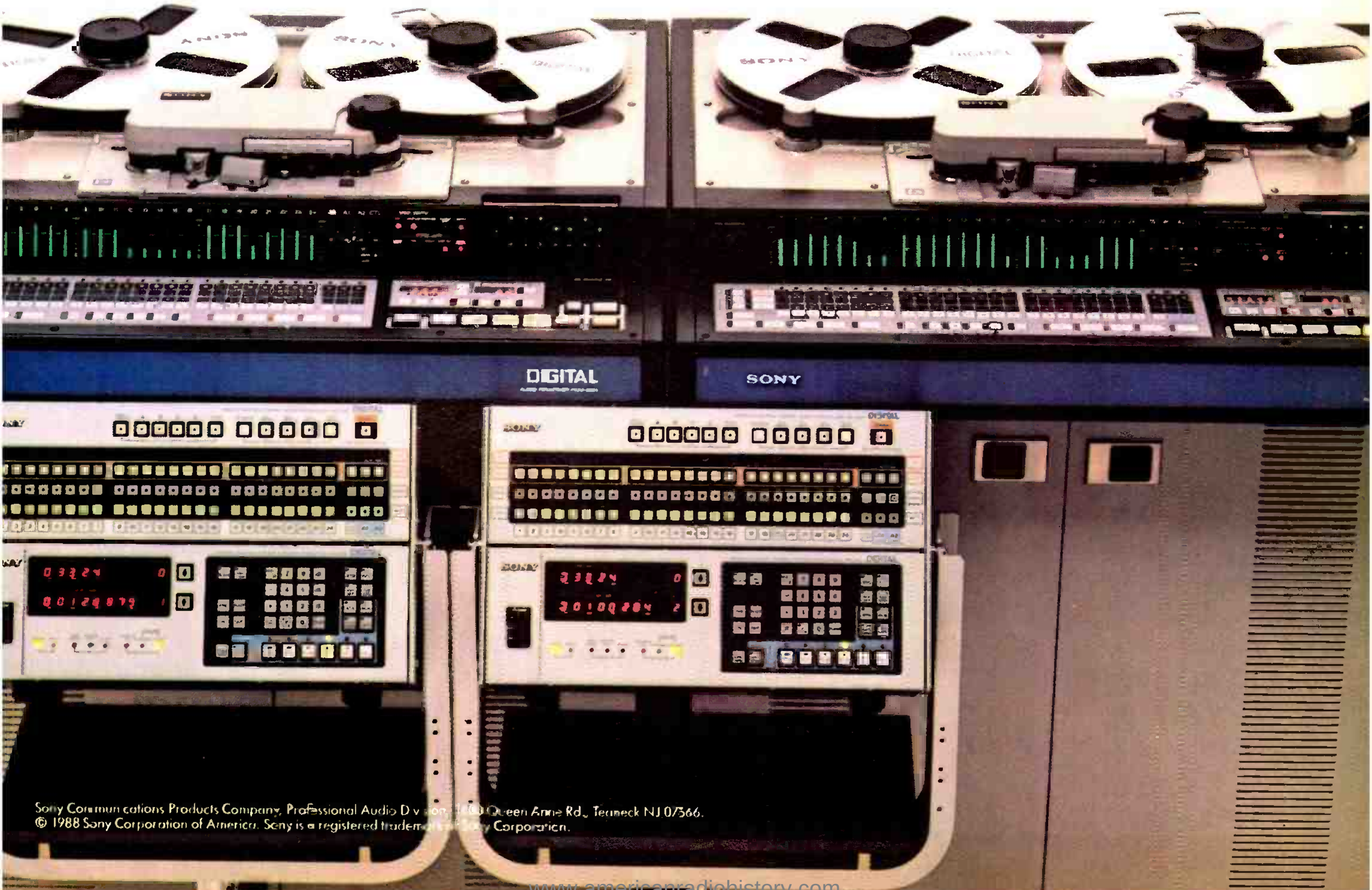
Simply cable two machines together and you've expanded the system. Unlike other recorders, a pair of PCM-3324s can maintain word-sync lock, even in varispeed and editing situations. The result? 48 perfectly synched tracks, zero audio phase shift.

Press a few buttons on the machine or the remote and you can perform electronic editing feats, such as single or multiple-track fly-ins and variable crossfades from 1 to 370 milliseconds. Of course, you can also edit with a razor blade. And built-in resolving capabilities let you lock the PCM-3324 to NTSC, PAL or 24Hz sync signals.

Sony's published APIB protocol ensures that the PCM-3324 of today can interface with tomorrow's technology. No other system expands so easily or does so much. Get the full rundown from your Sony Professional Audio representative. Or call us at 800-635-SONY.

Circle (13) on Rapid Facts Card

SONY®
Professional Audio



A new level of performance is just a touch away

The new Ensoniq EPS Performance Sampler and SQ-80 Cross Wave Synthesizer with Poly-Key™ pressure sensitivity

Discover a new level of performance in the new Ensoniq EPS Performance Sampler and SQ-80 Cross Wave Synthesizer. With Poly-Key pressure sensitivity you'll find more expression than in any other sampler or synth.

Playing an instrument with Poly-Key pressure is a musical treat. Each individual note responds vividly to your touch. You can control the modulation of pitch, vibrato, brightness or loudness—even the mix between two different sounds—all by varying the pressure of individual keys. So, instead of just playing a chord, you can command an entire string section. Or give horns real individuality. Or play dozens of other expressive effects you never could before.

The Ensoniq EPS — The only sampler that can play and load at the same time

It's hard to be expressive when your keyboard is silent, so the EPS lets you load sounds from the disk *while you are playing.* No other sampler—regardless of price—has this important performance feature.

Another new means of expression—Instant Patch Select—lets you choose alternate wavesamples or programs instantly in real time. With two patch select buttons located near the pitch wheel, you can instantly add expressive variations to a sound as the spirit of the moment moves you.

In addition, the EPS has 20 dynamically assigned voices, 20Hz to 20KHz audio bandwidth, 16 bit data format, 13 bit sample converter, 24 bit internal processing, floating point output with 96dB dynamic range and a built-in 8-track MIDI sequencer. And since the EPS can convert and play Mirage sounds, there's a ready library of over 2500 sounds available right now.

The Ensoniq SQ-80 — Studio technology with the performance touch

In addition to expressiveness, your instrument needs sounds that can cut through a stage full of amplified instruments. The Ensoniq SQ-80 Cross Wave Synthesizer cuts like a sharp knife.

Cross wave technology involves grafting the transient attack characteristics of one sound onto the beginning of another. The SQ-80 has a total of 75 sampled and synthesized waves on board, including multi-sampled bow, pick, breath and hammer attack transients, as well as inharmonic loops and sampled and synthesized sustain waves. So you can create thousands of sounds that not only cut, but sing and soar as well.

There's also an 8-track MIDI sequencer and built-in 880K disk drive for program, sequence and MIDI system exclusive storage. Each disk can store up to 1728 different programs and 10 full sequencer or MIDI system exclusive blocks. With one disk, you can be set up and ready to play before the guitar player tunes up.

Discover a new level of performance. Step up to an Ensoniq EPS Performance Sampler or SQ-80 Cross Wave Synthesizer at your authorized Ensoniq dealer, today. For the name of your nearest US dealer call toll free: 1-800-553-5151.

ENSONIQ Corp, Dept. R, 155 Great Valley Parkway, Malvern PA 19355 □ Canada: 6969 Trans Canada Hwy., Suite 123, St. Laurent, QUE. H4T 1V8 □ ENSONIQ Europe BV, Domplein 1, 3512 JC Utrecht, Holland □ Australia: Electric Factory, 188 Plenty Rd., Preston, Vic. 3072 □ ENSONIQ Japan, Ochanomizu KS Bldg., 2nd Floor #201, 3-3-1 Hongou Bunkyo Ku, Tokyo

ensoniq®

THE TECHNOLOGY THAT PERFORMS

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ensoniq
EPS
SQ80
ensoniq

Digital Audio Research

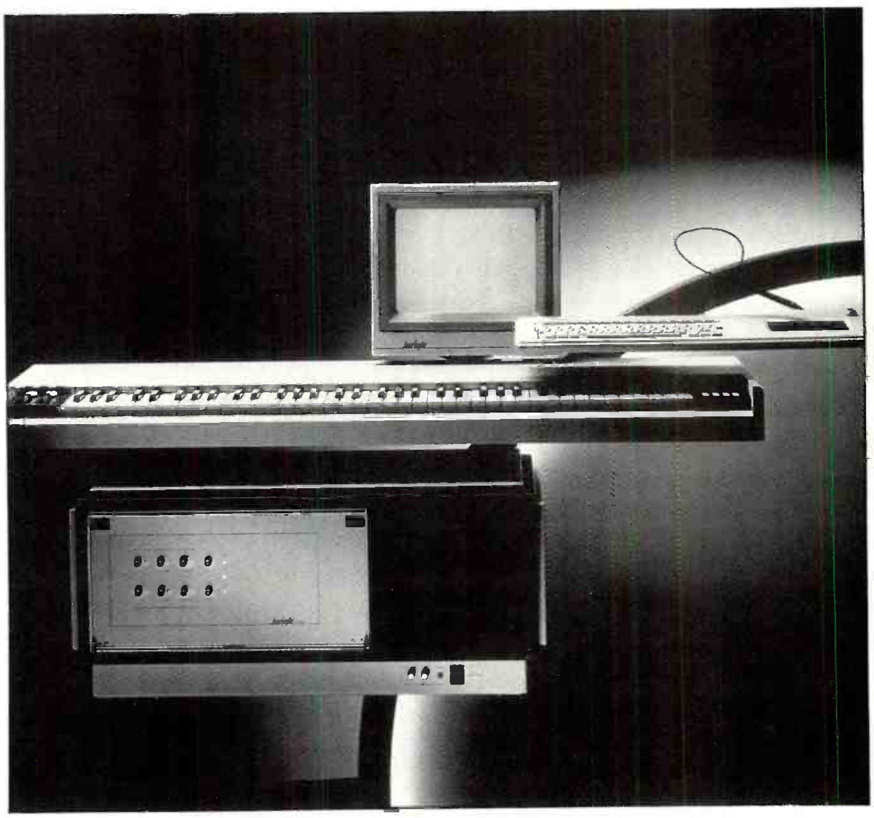
The Soundstation II is billed as a "second generation" production center, and features a touch screen in addition to pushbuttons, allowing for more user flexibility. The processor and storage unit contains the signal processing hardware, the disk drive, application software, power supplies and system interconnections. The basic system uses two Winchester drives, with a storage capacity of two track-hours.

The system uses 18-bit A/D conversion, 20-bit storage and 24-bit processing accuracy. Soundstation II can be used for a variety of applications, from compact disc masters to film soundtracks to audio for video projects.

Digital Audio Research Soundstation II

Fairlight

The Series III is a 16-bit digital sampler with a sampling rate of 98kHz that excels in advance screen techniques and audio manipulation with the use of icons and graphic representation of commands and functions. The Series III, being a dedicated 16-voice monophonic sampler, contains four to 14 megs of internal RAM and has 16 outputs. During the next year Fairlight is becoming aggressive and plans to introduce a stand alone terminal called the MFX that will control external machines as well as the complete functions of the Series III. It is the company's answer to replacing the musical keyboard with a control terminal that is designed toward the video/film industry. During the demonstration of the Series III, Fairlight showed us a Beta version of a hard disk recorder that is incorporated with the system.

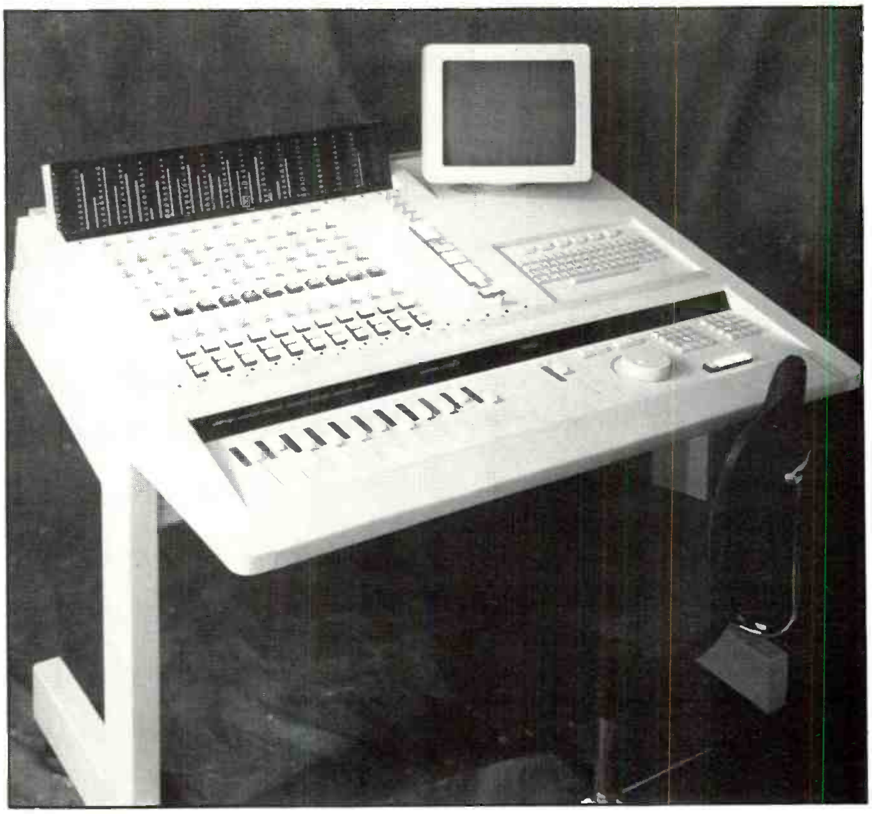


Fairlight Series III

Lexicon

The Opus is a hard disk based, integrated digital audio production system. It provides familiar hands-on operation with a console-like workstation, while using CRT displays, QWERTY keyboard and a "Scrub/Shuttle" knob for "non-destructive" editing in the digital domain.

A standard 800Mb hard disk provides 120 minutes in an 8-track format. As many as four disks may be added to the system for a total capacity of 480 minutes. The 6'x4" electronics cabinet weighs 670lbs and can be located up to 1,000 feet away from the workstation.



Lexicon Opus

New England Digital

New England Digital, makers of the Synclavier and the Direct-to-Disk system, has created a digital work environment that includes both the technologies of sampling (waveform manipulation) and hard disk recording into one package or as stand alone devices. Because of this, we decided to treat each of the units as a separate device.

The Synclavier, being a 16-bit digital sampler with a stereo 100kHz sampling rate, is a keyboard/terminal based system controlled by a mouse that has a RAM capability between four and 64 megs, contains a 200-track sequencer, and accesses optical disk, floppy, various sizes of hard disk and Kennedy tape drive as storage mediums.

The Synclavier also incorporates visual waveform editing and resynthesis techniques for processing sampled information. Also, a waveform can be heard, for editing capabilities in various speeds, by controlling the speed of the mouse while dragged across the waveform (a version of reel rocking or scrubbing).

New England Digital Synclavier system

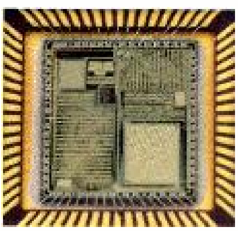
The price you have to pay to be free.

As a musician, you know you've got to be free. Free to express the music that's inside you. But, sadly, buying an "affordable" keyboard or sound module often means compromising your expressive freedom by accepting second-rate sounds and limited capabilities.

At Kurzweil, we don't think the instrument you play should build fences around your imagination. So we developed our revolutionary 1000 Series to help turn it loose. The 1000 Series delivers authentic Kurzweil sounds with more voices, more programming power, more creative freedom than you ever thought possible at prices you never thought possible.

Freedom from Financial Oppression.

Thanks to our new VLSI microchip, Kurzweil 1000 Series modules start at just under $2000. VLSI has enabled us to pack



each Series 1000 instrument with up to 120 of the same impeccable, 16-bit floating point, digitally-sampled sounds found on the legendary Kurzweil 250.® Choose from the 1000 SX String Expander Module, 1000 HX Horn Expander, 1000 GX Guitar Expander, 1000 PX Professional Expander (which provides a varied collection of our most famous samples) or the K1000 (the keyboard version of the 1000 PX).

Freedom of Expression.

The 1000 Series' generous polyphonic capabilities free you from the expressive limitations of 12-voice or 16-voice systems. The 1000 PX and K1000 offer the power of 24 voices while the 1000 SX, 1000 GX and 1000 HX each have 20 voices. And, if you need more voices, you can combine all four 1000 Series modules to create an 84 voice, 8-output super-system. So go ahead. Stack voices to your heart's content. Build complex, multi-voiced sequences. Go for those big, two-fisted chord shapes. The 1000 Series modules give you miles of sonic territory to roam at will.

Freedom of Choice.

The 1000 Series is truly democratic too. Three different operating modes let everyone—from novices to advanced programmers—benefit from the 1000 Series' bountiful capabilities. In Play Mode, you can use those great Kurzweil sound programs just as they are. Simply select a program and play. The Compiled Effects mode lets you apply a variety of popular effects to any of the preset sound programs. And the Modular Editing mode takes you deep inside the 1000 Series' voice architecture.

So stand up for your rights. You owe it to yourself to check out the 1000 Series. For more information, visit your nearby Kurzweil dealer. Or write to us at Kurzweil Music Systems, Inc., 411 Waverley Oaks Road, Waltham, MA 02154, (617) 893-5900. In Canada, write to Heinl Electronics Inc., 16 Mary Street, Aurora, Ontario L4G 3W8, (416) 727-1951.

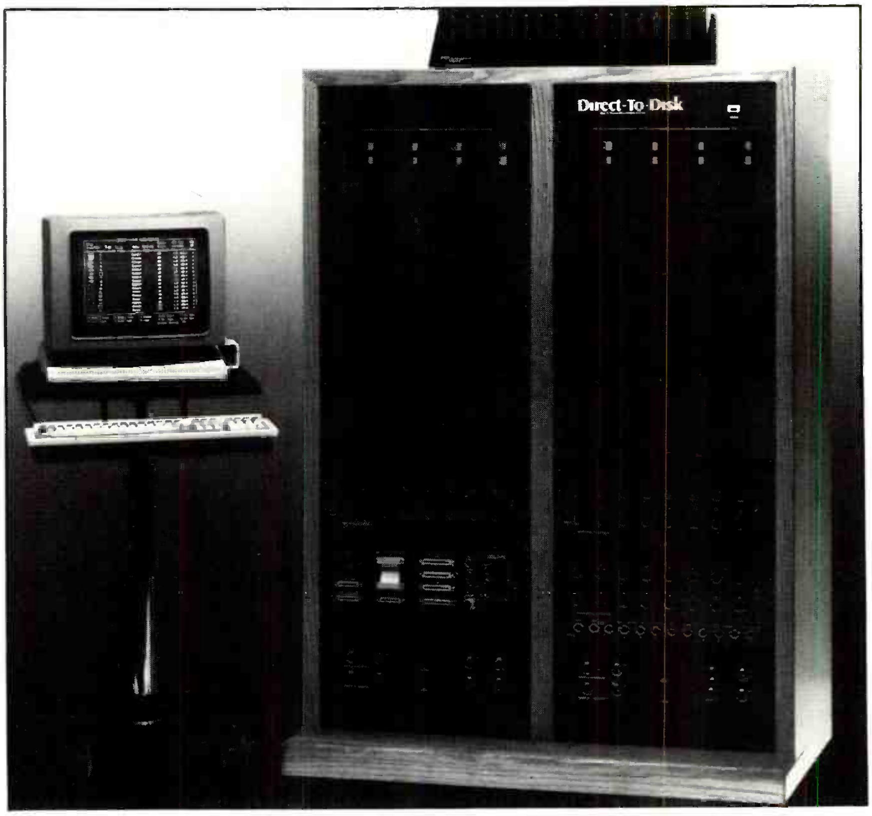
KURZWEIL™
Music Systems

* $1995 suggested retail price for 1000 SX and 1000 HX. $2,395 suggested retail price for 1000 GX and 1000 PX. $2,595 suggested retail price for K1000. All specifications and prices subject to change without notice.

Circle (38) on Rapid Facts Card

New England Digital

The Synclavier's newer brother, the Direct-to-Disk, is a multitrack digital recorder that stores information directly onto hard disk and is configured so that each hard disk contains two tracks which are combined with other hard drives to equal the number of configured tracks. The Direct-to-Disk system uses cut and paste features and can communicate and transfer information to and from the Synclavier. Both devices can sync up to external time code signals.



New England Digital Direct-to-Disk system

Polyphonic FX

The Optical Transfer Station (OTS) is an IBM-based system that uses the Akai S-900 (12-bit sampler with a 40kHz sampling rate) and a combination of optical disk and 20 meg removable hard disk (Bernoulli) to create a polyphonic sound FX device that enables the user to fire off samples from the computers EDL using the polyphony to offset and pitch change each of the eight voices.

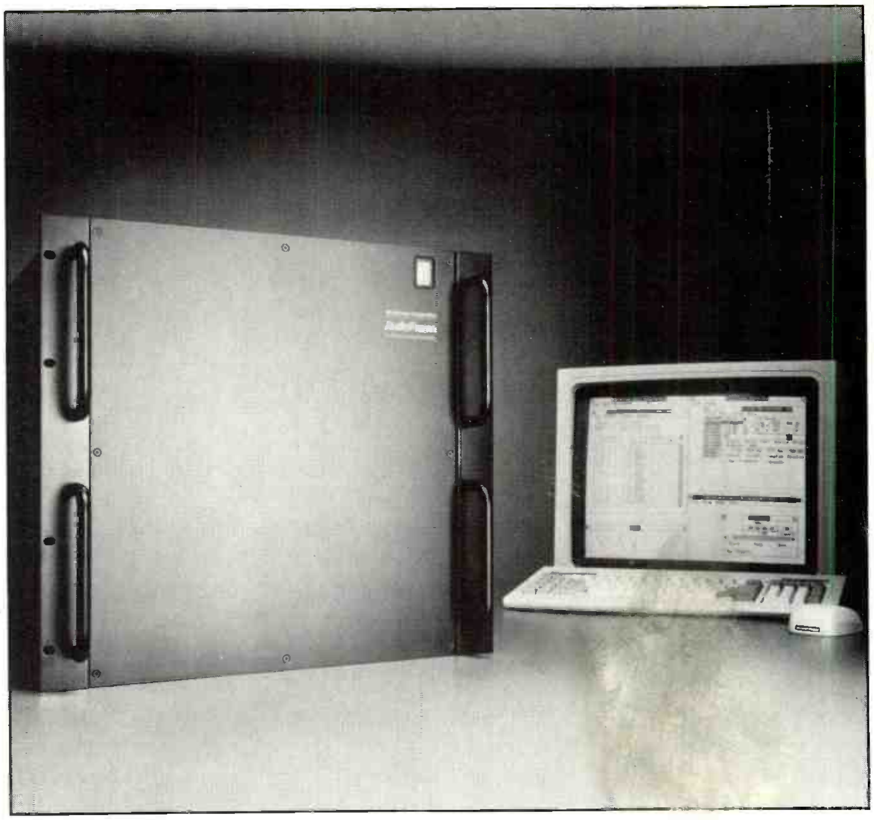
Samples are stored on the optical disk and downloaded into the S-900 via the systems edit decision list and contains a data management program to store samples in a tree structure environment. Polyphonic FX hopes to incorporate the soon-to-be-released stereo 16-bit sampler by Akai, the S-1600 and the optical jukebox, which can hold up to 66 optical disks. PFX does not use a musical keyboard to create polyphony but instead accesses it through the main screen.

Polyphonic FX Systems Optical Transfer Station

WaveFrame

The AudioFrame is a 16-bit multi-tasking sampler with a fixed sampling rate of 44.1kHz that uses interpolation to deal with the problem of alliasing when playing back a sample at a different sampling rate. AudioFrame consists of an IBM compatible (AT or better) that controls the AudioFrame rack. The rack is expandable through the insertion of cards, and comes in groups of 16 outputs per card containing two megs of RAM for each card. The cards can be expanded to accommodate larger amounts of RAM when combined with another card. Reel rocking can be achieved by using control data from a MIDI keyboard's modulation wheel. Waveframe plans on disclosing developer kits to other manufacturers so that third party devices will be able to interface with the AudioFrame.

Waveframe can be used in the music industry as well as the video/film industry by being able to sync to external time code. Because the AudioFrame uses the IBM-compatible technology, the system can use any of the third-party peripherals for that system.



Waveframe AudioFrame

These are just a few highlights of each device. We feel that this article in no way is an end-all or that it tries to answer all of our questions. Instead, it is intended to create a platform to aid your own investigations concerning how to fulfill your own technical requirements. [For additional questions to ask when you conduct your own investigations, see the sidebar on page 19—Editor.]

Also, it is easy to get confused with what is readily available and what is manufacturers' pre-press hype, so we hope that for the next few months we've set the record straight as to what is available and what we can look forward to.

The data contained in this report were compiled during the Christmas holidays, and we would like to thank those individuals who lasted through our three- to four-hour interrogation and product trial. We hope that we've invoked and inspired more questions while answering a whole lot more.

For more information

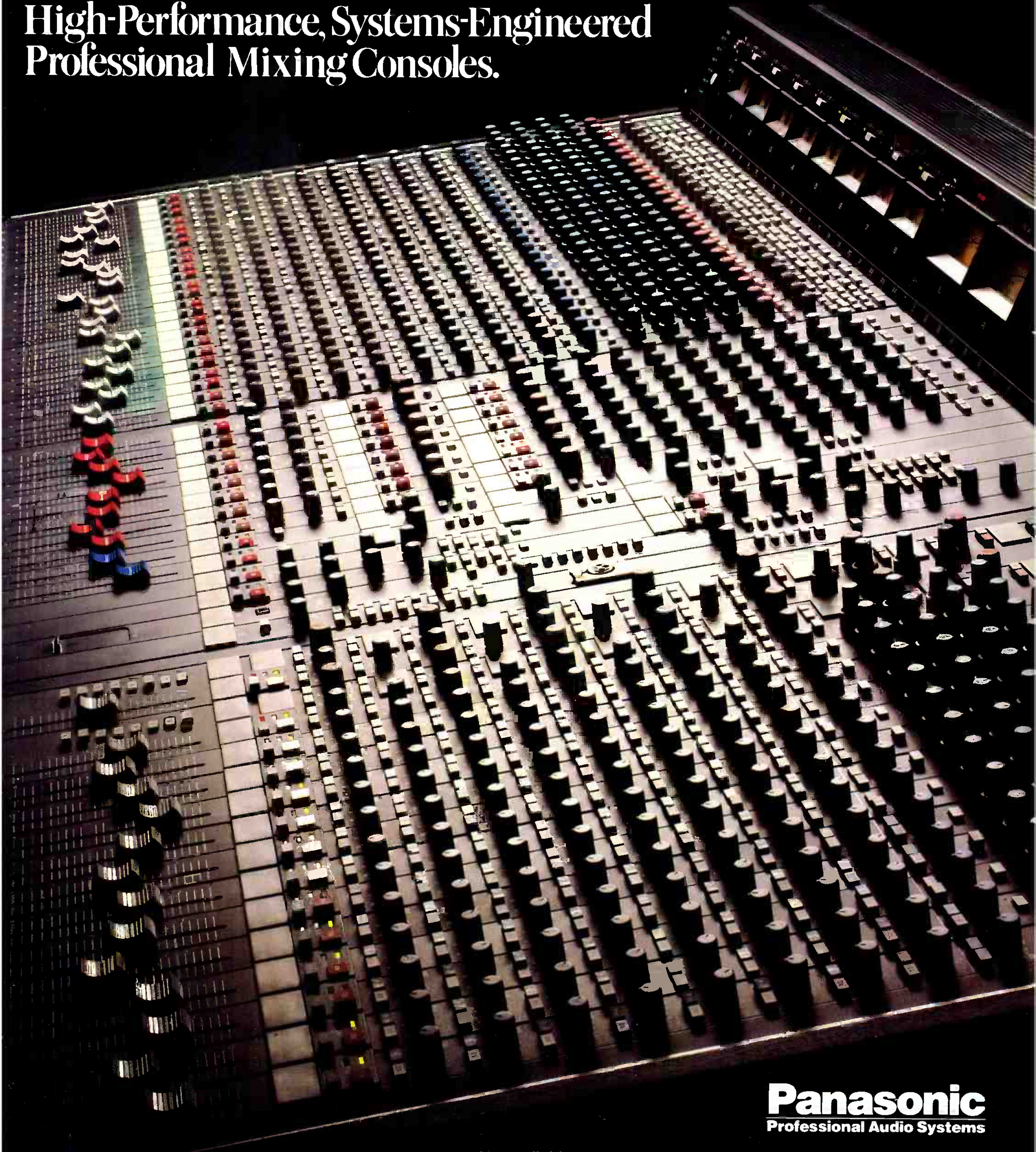
For more information on the following manufacturers and products, circle the appropriate circle number on the Rapid Facts Card.

Audio + Design Recording: *101*
Advanced Music Systems: *102*
Digital Audio Research: *103*
Fairlight: *104*
For-A Corporation: *111*
Hybrid Arts: *105*
Lexicon: *106*
New England Digital
  Direct-to-Disk: *107*
  Synclavier: *108*
Polyphonics: *109*
WaveFrame: *110*

RAMSA

S840 SERIES

High-Performance, Systems-Engineered Professional Mixing Consoles.



Panasonic
Professional Audio Systems

Introducing the Ramsa WR-S840 Series Professional Audio Mixing Consoles.

Why did Ramsa design this console?

How many times have you compared the specifications for two different sound mixing consoles and wondered why, though their specs were very similar, they sounded so differently? It happens all the time.

One of the primary causes for this apparent discrepancy between specs and actual performance is related to the way in which specifications are measured. Typically, a spec is measured in a laboratory, with highly controlled conditions. The console is driven from a single, well-behaved source (an oscillator or noise generator), and terminated by a resistive pad and some sort of analyzer. Under these conditions, you may see very impressive specs for noise and distortion. Yet take that same console into a studio, an arena, or a theatre and consider what happens. You'll connect dozens of inputs from a variety of locations (some carrying noise modulation from dimmers, fluorescent light ballasts, wireless mics, or the security crew's walkie-talkies). You'll hook up the outputs to loads through hundreds of feet of reactive cable. You'll patch in signal processors that may be fed from different AC systems (with hum and noise current riding on the difference in ground potential). Suddenly, the performance is orders of magnitude less impressive.

In fact, many consoles become unusable under these conditions. This is one of the areas we sought to correct in designing the WR-S840 series.

Our market research indicated there was a genuine need for a truly professional sound mixing console without a lot of costly gimmicks. The basic function of such a console was already well-defined, yet we wanted to design a console that would become the benchmark which future consoles must emulate. One that would exhibit comparable performance and stability in the field to that exhibited in the lab. And our experience in sound reinforcement, and especially in the broadcast industry (where the demands are extraordinary), gave us the know-how to do the job. The result is the WR-S840 series of consoles.

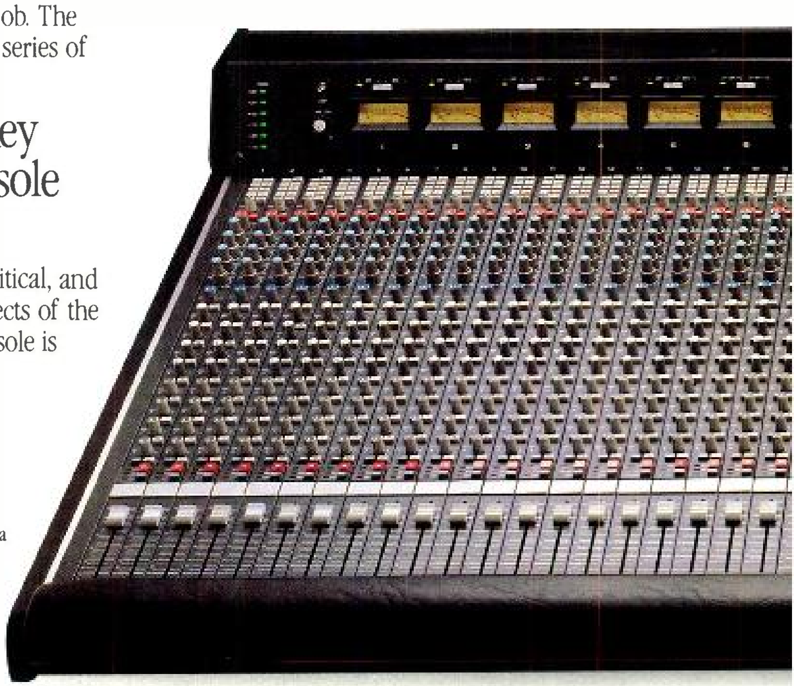
Crosstalk: a key factor in console performance.

One of the most critical, and least understood, aspects of the performance of a console is crosstalk, or leakage. When you put a signal into the console, and it comes out everywhere, that's crosstalk. Crosstalk is usually present even in the finest consoles. For example, the "infinity" marking on a fader really doesn't turn the signal off—it just attenuates it. Crosstalk also applies to all volume controls (aux sends, pan pots, etc.) and to switches (pre/post, assign, on/off, etc.). In other words, it affects everything in the console. The real question becomes, "how much has the crosstalk been attenuated, and what is its frequency response?"

Crosstalk is as important an attribute of a console's performance as is distortion or noise. Yet typically you will see only one crosstalk figure (adjacent inputs to adjacent outputs). No wonder you cannot determine much about the real system performance from looking at a typical crosstalk spec!

Consider that a typical crosstalk figure of −55 dB at 1 kHz can degrade to −40 dB at 10 kHz due to the capacitive nature of high frequency crosstalk. It doesn't do much good to achieve a low distortion figure, like 0.01% (which is 80 dB below signal level), when the crosstalk is −40 dB at 10 kHz. That level of crosstalk is roughly the equivalent of 1% distortion at 10 kHz!

Ramsa uses several approaches to reducing crosstalk. In our physical bus structure, we pair a ground conductor with each signal conductor so that adjacent channels of audio are always isolated by a ground. Even with all those grounds, it would still be possible for crosstalk components to modulate the poten



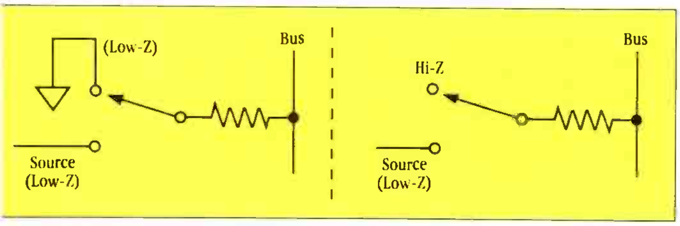
WR-S840 Configured as a 52 Input House Console.

tial on the cumulative ground bus—resulting in low frequency crosstalk, and a muddy sound. Therefore, we also run a heavy (⅛″ x ¾″) chrome-plated solid copper ground bus the length of the console, to which each module's audio common is tied. You won't see evidence of this difference in the specifications, but you will sure hear the difference in the studio or in a live performance.

Another way we combat crosstalk is to back-ground all assignment switches. This means that when a bus is not assigned through a given switch, that feed to the bus is shunted to ground. This prevents the signal brought to the switch from leaking onto the bus.

In a non-back-grounded (that is, a conventionally grounded) system, you can see very low noise figures with only one input assigned to a bus, yet the noise comes up higher and higher as more inputs are assigned. Every time the number of inputs assigned is doubled, the white noise rises 3 dB and the hum increases 6 dB. That's why the specs can look good in the lab, and not be very impressive in the field.

You don't get something for nothing, and unfortunately, back-grounding does cause the summing amps to run at maximum noise gain. In order to keep the overall noise low in the WR-S840 series, while avoiding crosstalk by using the back-grounding scheme, we designed a discrete summing amp circuit for the console. This new amp yields an improvement of better than 12 dB in noise performance over conventional monolithic IC amps. A very pleasant side benefit is that the system noise is not only lowered, it doesn't change as more inputs are assigned (see specs).



amsa back-grounded switch (left) versus conventional bus assign switch (right).

A proven method of professional console grounding.

Grounding is the most misunderstood aspect of system design. Improper console grounding can cause odd noises and unwanted sounds that are difficult or impossible to eliminate regardless of how carefully the rest of the system is assembled.

There are really just two basic types of grounding practices: single-point and distributed. Single-point grounding can work well within a small sound system. Distributed grounding lends itself to large systems, including those up to the size of your telephone company! Clearly, we selected a distributed system as being the appropriate technique for the WR-S840 series. We didn't do anything new, we just did our homework.

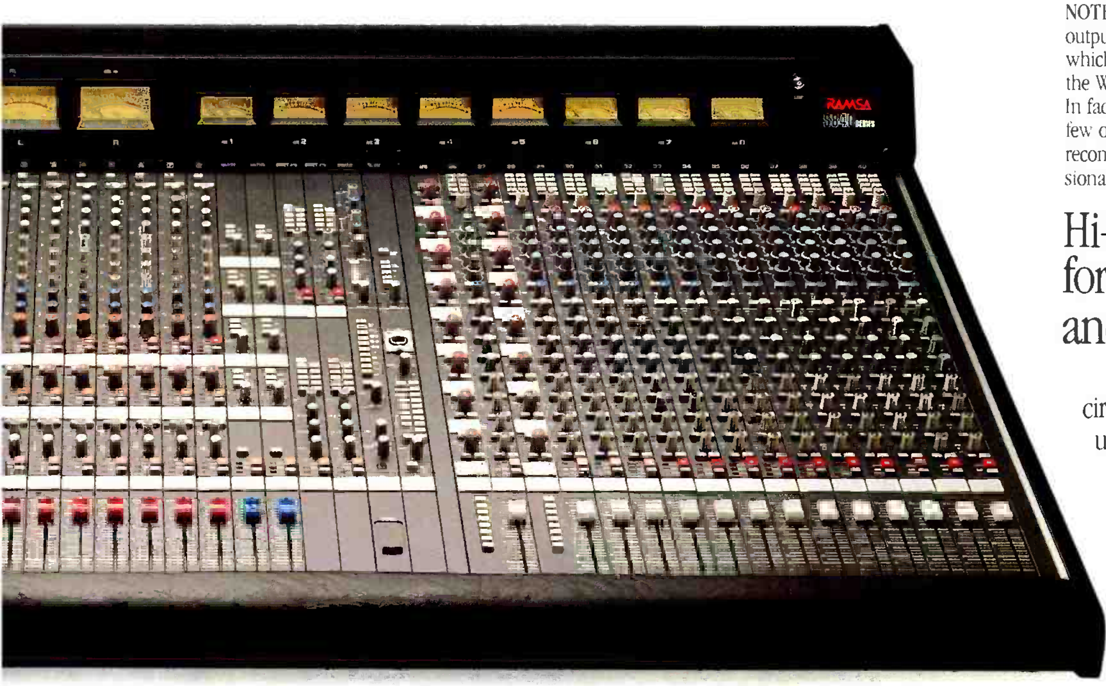
As far back as the 1950's, the Journal of the Society of Motion Picture and Television Engineers (SMPTE) characterized, in detail, a distributed grounding technique which they recommended for use by all equipment manufacturers. Film studios generally have large installations—sometimes spanning many acres, with up to a dozen sound stages all fed from a central tape machine/film dubber room. Similarly, TV network facilities invariably distribute audio through multiple floors of high-rise buildings,

"It's no great engineering achievement to build a console that yields good specs on the bench. But getting it to perform to those same specs under 'real-world' conditions... That's quite another matter."

John Windt, consulting engineer

with a lot of RF present. Hence, the SMPTE practices have to work for the toughest of grounding situations. We followed SMPTE recommendations in the WR-S840 series because they comprise a truly professional methodology that eliminates the vast majority of hum, buzz and other ground-related problems.

NOTE: SMPTE also stated that input and output connections must be balanced, which is why all inputs and outputs on the WR-S840 series consoles are balanced. In fact, these consoles are among a very few on the market that comply with all recommended SMPTE practices for professional sound equipment.



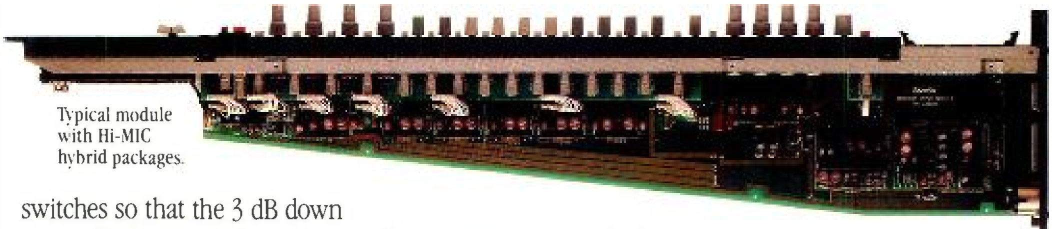
Hi-MIC circuitry for precision and reliability.

A Hi-MIC is a cost-effective circuit package that makes use of surface-mount technology. Traces are plated on a substrate, and then all resistors, capacitors, transistors and ICs are surface mounted to that substrate. The package is then potted for stability and thermal conductivity. Resistor tolerances in our

Hi-MICs can be held to within a very low 0.5%, resulting in excellent uniformity. Hi-MICs, which are essentially self-contained circuit blocks, also simplify troubleshooting.

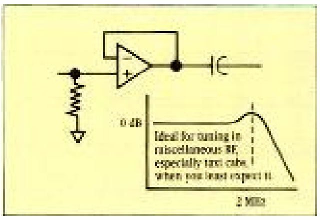
Good, stable circuit design— It doesn't cost you any more for us to do it right!

Consider the buffer amps. Consoles invariably have many circuits to buffer one stage from the next, serving as an impedance converter without changing the gain. For example, there's always a buffer between each pan pot and its assign



Typical module with Hi-MIC hybrid packages.

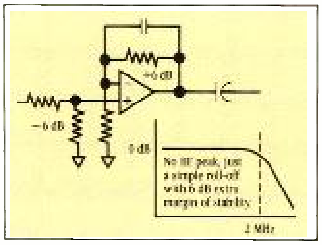
switches so that the 3 dB down point of the pan pot does not shift, no matter how many assign switches are selected. Unfortunately, the typical cir-



Typical single-IC, unity gain buffer stage and its response.

cuit that performs this function ends up creating problems of another sort. That's because the simple, expedient circuit will create a peak in the response at a very high frequency, somewhere around 2 MegaHertz. The designers assume this will not cause problems because, after all, the audio range is limited to about 20 kHz, and this is 100 times higher in frequency.

At Ramsa, we know better. We realize that a 2 MHz peak is an invitation to radio frequency interference. It turns the console into a great business radio receiver so that a passing taxi cab, for example, is likely to contribute to your program. For this reason, we



Ramsa unity gain buffer stage and its response.

designed the unity gain buffer for the WR-S840 series. It now has more parts, but the circuit is much better behaved, exhibiting a simple, non-peaked single-pole response and providing some 6 dB greater margin of stability than the typical designs in other consoles. What's more, all capacitors in the audio path are Elna™ "Carafine" grade low-ESR type for superior transient response.

The universal input circuit.

Our engineers developed a unique input preamp circuit. Its 41-position control can be continuously adjusted, with 1.5 dB detented resolution, to accept any source from −60 dBu to +4 dBu nominal level while maintaining a full 20 dB headroom throughout this range. The circuit uses discrete transistors for low noise, and a bipolar 25 volt supply for the headroom.

Long-life MRP™ controls.

MRP (Matsushita Resistive Plastic) describes a proprietary RAMSA process for building high quality, high precision, long life controls. For example, the faders and input attenuators on the WR-S840 consoles are rated at 300,000 operational cycles. This is some 20 times the life of a typical carbon fader!

Fiberglas-Epoxy circuit boards throughout.

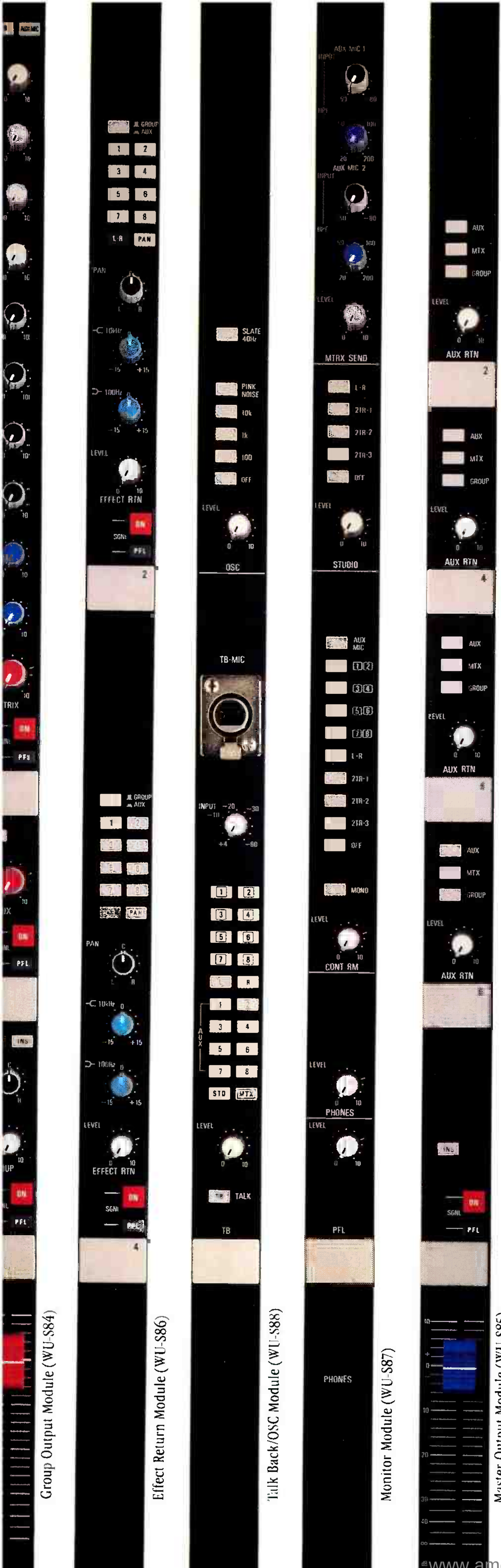
Fiberglas-epoxy boards may cost more, but they offer numerous advantages. They don't warp, they are much stronger than phenolic boards, they can be fabricated with double-sided traces and plated through holes. That's why every circuit board in the WR-S840 consoles is of this design.

Comprehensive 4-band equalization.

Each input module is equipped with a powerful 4-band, state-variable, sweep-frequency equalizer with high speed (20 volt per microsecond) bi-FET ICs. This circuit complements the engineer's acoustic and

Mono Mic/Line Input Module (WU-S81)

Sub Input Module (WU-S82)



Group Output Module (WU-S84)

Effect Return Module (WU-S86)

Talk Back/OSC Module (WU-S88)

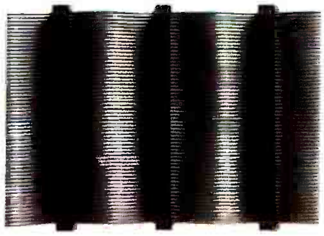
Monitor Module (WU-S87)

Master Output Module (WU-S85)

musical tastes at typical boost/cut levels where the controls are normally used for subtle shaping and enhancement. At higher levels of boost or cut, the 'Q' of the filters compounds and becomes higher, for more effective shaping of sound. Center detents on the gain controls facilitate rapid neutralization of each band.

Ribbon wire bussing and gold contacts for reliability and low noise.

All power and audio connections between modules is carried via flexible ribbon cables. These busses mate with the module PC boards through gold-plated contacts. Thus,



without any large motherboard, there is no strain on the connectors as the console travels and flexes, and contact resistance remains uniformly low for the entire life of the console.

Sweep-frequency high-pass filters.

Each equalizer is accompanied by a high pass filter with 12 dB/octave slope and a knee frequency sweepable from 20 Hz to 200 Hz. This affords precise control of wind noise, stage rumble, vocal plosives ("pops"), etc. without cutting desired program frequencies.

Submix modules expand your input capacity by a factor of four.

The Submix module is a double-width input module that has 8 mic/line preamp circuits, each equipped with its own phantom power, phase reverse, assign on/off, direct out and PFL switches. That's four times the input density of standard input modules. The Submix module also has the same overall 4-band state-variable EQ, variable high pass filter, PFL and channel ON switches as the standard input modules. It also has the same 18 sends, including: 8 aux bus assigns, stereo pan, and 8 primary mix busses. Just three of these function-packed modules can handle a 24 track tape mix. Stuffing a full sized mainframe with 20 modules accommodates 160 inputs, though it is unlikely you'll use such a configuration. Realistically, you may wish to fill 16 slots with 8 of these Submix modules (thus handling 64 inputs), and the remaining 24 slots with standard input modules, for a total capacity of 88 mic/line inputs.

Stage monitor modules turn the WR-S840 into a 40 x 18 stage monitor board.

By inserting Stage Monitor Input modules into the mainframe, a 40 x 18 stage monitor mixing console is created—with the features and flexibility of monitor boards costing more than twice as much. Each module includes a 100 mm fader, plus 10 mono bus level controls and 4 pair of concentric stereo bus level controls. The stereo controls are internally switchable so they can serve as 8

Monitor Input Module for Stage Monitor Console (WU-S83)

WR-S840 Configured as a 40 x 18 Stage Monitor Console.

independent assigns, for a total of 18 discrete mixes on each module. Each mono assign, and each pair of stereo assigns, has its own pre/post fader switch, as well as an assign on/off switch. The Monitor module also has the same 4-band state-variable EQ, variable high pass filter PFL, overall channel on, phantom power and phase reverse switching as the standard input modules. This is the perfect complement to a WR-S840 house console.

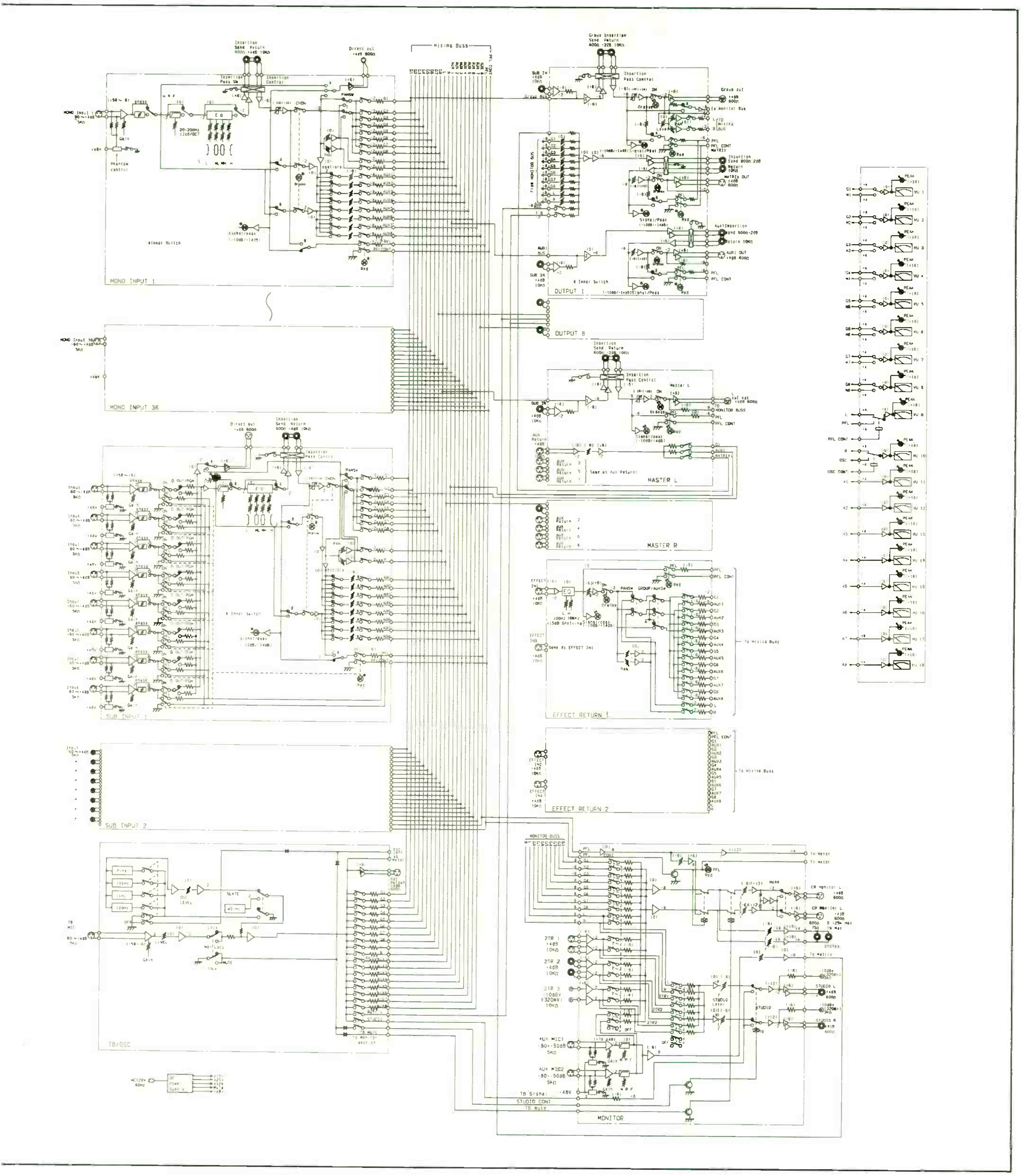
A new look and feel—a whole new console.

The WR-S840 series not only represents a major step forward in performance, it also embodies a new look and feel. Elegant, neutral styling is housed in a low-profile (12″ high) package that makes it easy to peer over the meter panel. All volume controls are detented, and all knobs are subtly color coded so they don't assault the eye—yet they maintain good contrast under a variety of lighting levels and hues. The top of the meter panel is horizontal and flat, so you can stack it with small monitors or other equipment. And it's even covered with a non-skid rubber surface to help keep these objects in place. Internally, there are many pre/post switches; the standard post fader & EQ Direct Output can be made pre-EQ to feed a house console, or pre-fader & EQ to provide a split off the mic pre without double-terminating a mic. There is a built-in dimmer circuit and 3 sockets for standard LittLites. In short, we paid attention to the little details that make your mixing pleasurable and efficient.

Uncompromising audio performance.

In keeping with the overall design integrity of this series, the mechanical and electrical performance are exemplary. High-speed 5532 opamps are used extensively. Numerous power-supply decoupling caps (7 on each input module alone) keep the supply voltage very "stiff" to avoid muddiness and interaction between circuits. The distortion is so low that at nominal operating levels it is immeasurable because it is below the noise floor (less than 0.01% THD). Even at +24 dBm output into 600 ohms, the distortion rises only to 0.05%.

The −3 dB points are 10 Hz and 140 kHz. At the same time the console is very stable and has no tendency to oscillate even when all controls on the console are at maximum level! The overall signal-to-noise rati (with all inputs at nominal, all channels on and assigned to a bus) is better than 80 dB. Add the 20 dB output headroom, and you've got 100 dB dymani range in a worst-case, real-world scenario. That's quiet. In fact, the output noise approaches that of an 18 bit digital system. The common mode rejection ratio (CMRR) fo the universal mic/line inputs i an impressive 75 dB at 1 kHz (65 dB broadband, 20 Hz to 20 kHz). The internal gain structure is conservative, with an extra 6 dB of headroom on th summing amps, so that as channels are added to the mix the mixing busses do not rapidly overload and require that inputs be backed off.


Mixing Buss
MONO INPUT 1
MONO INPUT 36
OUTPUT 1
OUTPUT 8
MASTER L
MASTER R
SUB INPUT 1
SUB INPUT 2
EFFECT RETURN 1
EFFECT RETURN 2
TB/OSC
MONITOR


RAMSA

WR-S840 Test Data

Frequency Response	+0, -1 dB, 20 Hz to 20 kHz; -3 dB at 10 Hz and 140 kHz; any input to any output bus, at any level control setting
Total Harmonic Distortion	
At Nominal Level	0.01%*(+4 dBu in & out)
10 dB Above Nominal	0.006% (+14 dBu in & out)
20 dB Above Nominal	0.05% (+24 dBu in & out)
Intermodulation Distortion	
At Nominal Level	0.01%*(+4 dBu in & out)
10 dB Above Nominal	0.01% (+14 dBu in & out)
20 dB Above Nominal	0.1% (+24 dBu in & out)
Dynamic IM Distortion	
At Nominal Level	0.01%*(+4 dBu in & out)
10 dB Above Nominal	0.005% (+14 dBu in & out)
20 dB Above Nominal	0.02% (+24 dBu in & out)
Phase Response	+60° at 20 Hz, -20 ° at 20 kHz
Noise	
Equivalent Input Noise	-127 dBm (150 Ω source, DIN audio bandwidth 22 Hz - 22 kHz)
Output Signal-to-Noise Ratio	86 dB (all faders off, all channel switches off, all modules assigned to the bus) 80 dB (all faders at nominal, all channels on, all modules assigned to the bus, input gain controls at +4)
Maximum Voltage Gain	
Input to Group Out	84 dB (±2 dB)
Input to Aux Out	86 dB (±2 dB)
Input to Matrix Out	90 dB (±2 dB)

* There is no measurable distortion at nominal level; this value is actually the noise floor of the console, which is why distortion drops at 10 dB above nominal level.

† Crosstalk values are the difference between the level fed into the driven circuit, and the level measured in the non-driven circuit.

Crosstalk†	
Adjacent Input to Adjacent Output	85 dB at 1 kHz; 72 dB at 10 kHz
Fader (Maximum Kill)	80 dB at 1 kHz; 60 dB at 10 kHz
Channel On/Off Switch	95 dB at 1 kHz; 75 dB at 10 kHz
Aux Send Pot	75 dB at 1 kHz; 60 dB at 10 kHz
Output Module (All Combinations of Group-Aux-Matrix)	80 dB at 1 kHz; 65 dB at 10 kHz
Common Mode Rejection Ratio	
Standard Channel Input	70 dB 20 Hz to 20 kHz at any input gain control setting
All Other Inputs	40 dB at 1 kHz; 35 dB at 10 kHz
Equalization	
Low (shelving)	±15 dB, sweepable 40 Hz to 400 Hz
Low-Mid (peaking)	±15 dB, sweepable 160 Hz to 1.6 kHz
High-Mid (peaking)	±15 dB, sweepable 800 Hz to 8 kHz
High (shelving)	±15 dB, sweepable 1.6 kHz to 16 kHz
High Pass Filter	12 dB/octave, sweepable 20 to 200 Hz
Mixing Bus Assigns	
Standard Input Modules	8 group, 2 stereo & 8 aux send
Monitor Module	10 mono & 4 stereo (stereo outputs are switchable to dual mono in pairs, for a total of 18 busses)
Meters	18 VU meters with LED Peak indicators: 8 switchable for Matrix or Group, 2 Stereo, 8 Aux
Headroom	20 dB minimum throughout the entire console
Power Requirements	120 V AC, 60 Hz, 600 VA (Ramsa WU-PS80 Supply)
Overall Dimensions (WxHxD)	72-7/8" x 12" x 39-3/8" 1851 mm x 305 mm x 999 mm
Weight	293 pounds (133 kg) approx.

Input Characteristics

	Actual Impedance	Nominal Source Impedance	Nominal Level	Max Level Before Clipping	Connector
Standard Channel Inputs	5kΩ	150Ω ~ 600Ω	-60 ~ + 4 dBu	-40 ~ +24 dBu	XLR-3
Aux In, 2-Track 1 In, Efx In	10kΩ	600Ω	+4 dBu	+24 dBu	XLR-3
Input Module Insert Return, 2-Track 2 Input, Sub Input	10kΩ	600Ω	+4 dBu	+24 dBu	T/R/S Phone Jack
2-Track 3 Input	10kΩ	10kΩ	-10 dBV	+12 dBV	RCA Jack
Output Bus Insert Return	10kΩ	600Ω	-2 dBu	+24 dBu	T/R/S Phone Jack
Aux Mic Input	5kΩ	150Ω ~ 600Ω	-80 ~ -50 dBu	-60 ~ -40 dBu	XLR-3

Output Characteristics

	Actual Impedance	Nominal Load Impedance	Nominal Level	Max Level Before Clipping	Connector
Group, Aux, Matrix, Stereo, Control Room & Osc. Out	75Ω	600Ω	+ 4 dBu	+24 dBu	XLR-3
Input Insert Send, Direct Out, Studio Out	75Ω	600Ω	+4 dBu	+24 dBu	T/R/S Phone Jack
Group, Aux, Matrix & Stereo Insert Send	75Ω	600Ω	-2 dBu	+24 dBu	T/R/S Phone Jack
Studio Out	500Ω	10kΩ	-10 dBV	+12 dBV	RCA Jack

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Data measured on an Audio Precision test set is available written request. Elna is a trademark of Elna America, Inc Carson, CA. MRP is a trademark of Matsushita Electric Corporation of America.

R-840

A historical perspective of the digital audio workstation

By Paul Tydelski

A new buzzword has been circulating among audio professionals lately: workstation. The roots of this word are buried in the past computer music experiments of Lejaren Hiller, Max Mathews, John Chowning and others.

These early users may not have had the convenience of today's operator—having dealt with rooms full of "warm" equipment, software based on simple single-line statements and storage on punch cards or tape—as opposed to the contemporary desktop hacker using structured languages, floppy or hard-disk storage and any number of controllers. Only recently has the vision of these pioneers become a reality.

Although the means and methods differ, the former and current techniques of sound manipulation share a common ground and a realization that a workstation should include at least the following capabilities: input derived from analog or digital sources, digital signal processing, editing and a programmable output.

Advanced electronic technology is now allowing faster and more complex implementation of historical workstation concepts. The current model for the digital workstation was proposed by F. Richard Moore during International Computer Music Conferences in 1985 and 1986. Some of the basic elements of the system are as follows.

Paul Tydelski is a technical consultant, engineer and producer in San Diego.

The requirements of a modern computer workstation are: a central host; a commercial computer; custom interface hardware and software that would allow general purpose computing; signal processing; real-time synthesis and interaction; and various commercial or customized input and output devices.

In further detail, the proposed version of the workstation includes two separate computers, by definition as well as use. These two computer subsystems are a general purpose processor and a real-time processing section. Each of the subsystems may be used as an independent stand-alone system or integrated into one larger functional unit. There exists a bus link between the two machines, which includes both data and control signal transfer, and inter-system interrupt handling. Each subsystem also includes an independent display and storage capability.

A current workstation implemented at the Computer Audio Research Laboratory (CARL) at the University of California at San Diego is based on a SUN-3/160, running Berkeley UNIX as the operating system for the general purpose processor.

The real time portion is comprised of a DY-4 multiprocessing network, running the Harmony operating system (see W.M. Gentleman's publication for the National Research Council of Canada, 1983-85). Both of these systems are based on the VMEbus (an international design standard) and offer flexible growth and expandability. Audio conversion is implemented through a TMS320 processor, and the output is directed to a Sony PCM-701.

The real time processing is accomplished through a combination of 68020-68881-based single board computers. Multichannel analog I/O and MIDI interfacing are obtained through other VMEbus devices, allowing such things as a Roland MPU-401 and other custom signal processing devices to be used.

A system such as the one at CARL has a price tag comparable to many of today's complex commercial units. It may even be a justifiable alternative because of the nature of the customized application. General and specific software is available along with some source code, allowing the composer/programmer to meet nearly any compositional need.

With a system such as this, a data manipulator has access to a powerful computing station, where he can compose, record, edit, modify and apply various tasks to a sound file, allowing complex compositional decisions to be acted upon in the general purpose or real time section of the instrument.

In many ways, workstations like the one at CARL are the forerunners of the current trend toward commercially available systems as detailed in the article on page 38.

Continued from page 19

- *Any limitation on how close edits can be?*
- *How are edit points located and marked?*
- *Time alignment functions?*
- *Any limitation on how close two sounds can be on a given channel of disk playback?*

11. Describe the editing functions that can be performed on samples played out of RAM.

Audio material organization

12. Describe how audio material is organized within the system. How do you keep track of projects or elements of projects?

13. Describe the back-up procedure for the hard disks in the system.

14. Does the system have any position reference other than time code?

15. How is material to back-up identified?

16. Is incremental back-up possible?

17. What new hardware features are available as add-on upgrades?

18. What is the layout of the EDL?

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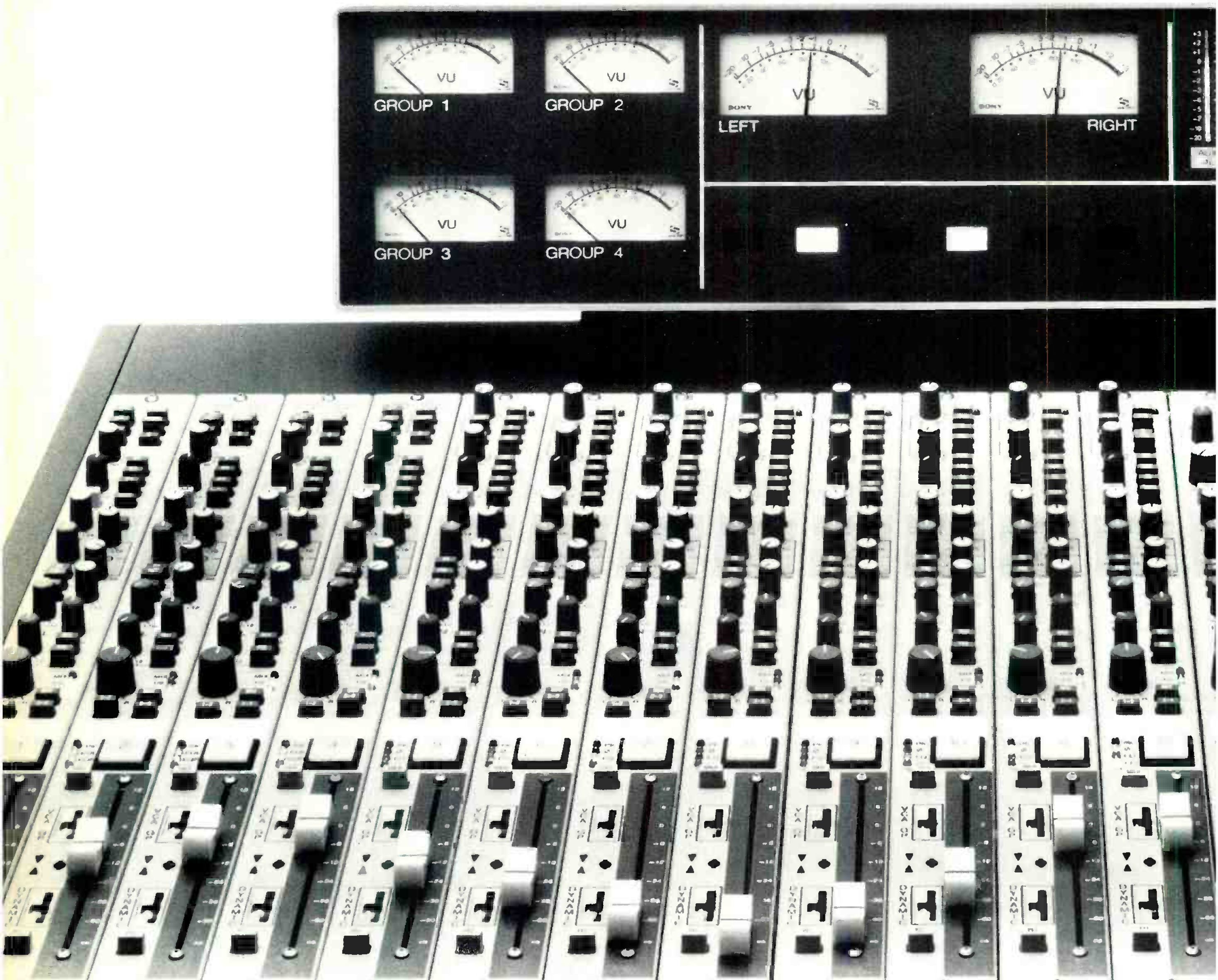

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Professional Audio

Digital Audio Workstations: Operational Report

Compiled by Michael Fay

Thinking of buying a workstation?
This exclusive report details the configurations of 11 systems.

Legend:
Abbreviation: Company name (Product name).
A+D/R: Audio + Design Recording (Sound Maestro System).
AMS: Advanced Music Systems (AudioFile).
DAR: Digital Audio Research (Soundstation II).
FAIR: Fairlight (Series III).
FOR.A: For-A Corporation (Sirius 100).
HA: Hybrid Arts (ADAP).
LEX: Lexicon (Opus).
NED/D: New England Digital (Direct-to-Disk).
NED/S: New England Digital (Synclavier).
PFX: Polyphonics (Optical Transfer Station).
WAV: WaveFrame (AudioFrame).

Other footnotes:
?: Data have not been verified or is not available.
N/A: Not applicable.
(*): Update available within six months.

1. What is the systems target market? Sound effects, Foley, music mastering, multitrack recording, music composition and production, audio processing, dialogue editing or background effects.
A+DR: Music recording, editing, mastering, post-production, broadcast.
AMS: Mastering, SFX, Foley editing, dialogue, ADR, background effects.
DAR: All the above.
FAIR: SFX, background effects, Foley editing, audio processing,hard-disk recording, mastering, dialogue editing and ADR.
FOR.A: Broadcast radio and television, as a 2-track digital audio replacement for cart machine systems.
HA: ADAP = Multitrack recording, SFX, audio processing, mastering, background FX, music composition/production (ADAP II: Foley, dialogue, ADR, multitrack mixing).
LEX: Video/film post-production, all digital editing applications, SFX, mastering, dialogue editing, BG FX.
NED/D: Mastering, dialogue, background FX, multitrack digital recorder.
NED/S: SFX, dialogue (w/ lg RAM units), Foley, music comp/prod., background FX, music publishing/transcription, audio processing.
PFX: SFX, Foley creation.
WAV: SFX, background FX, Foley creation, audio processing, music composition and production.

Hardware configuration

2. How many outputs/inputs does the system have?
A+DR: 2 outputs, 2 inputs (stereo). Analog or digital from any AES/EBU source.
AMS: 8 outputs, 2 inputs.
DAR: Up to 8 channels analog and digital I/O for each processor/storage unit.
FAIR: 16 outputs, 2 inputs.
FOR.A: 8 channels.
HA: 2 outputs, 2 inputs.
LEX: Ins = 12 channel strips, 12 patch points, 4 aux out = 12 direct, 12 patch points, 4 sends, 2 mix, 2 monitor mix.
NED/D: 8-16 outputs, 4-16 inputs.
NED/S: 8-36 outputs, 2 inputs.
PFX: 8 outputs, 1 input.
WAV: 8-64 outputs, 2 inputs.

3. Are the outputs software-assignable to different channels/tracks?
A+DR: N/A.
AMS: Yes.
DAR: Yes.
FAIR: Yes.
FOR.A: Yes.
HA: N/A.
LEX: Yes.
NED/D: No.
NED/S: Yes.
PFX: N/A.
WAV: Yes.

4. Is this a 12-, 16-, or 32-bit system?
A+DR: 16-bit, expandable to 24-bit.
AMS: 16-bit.
DAR: 18-bit converters, 24-bit storage, minimum 24-bit processing, 56-bit accumulator used in processing.
FAIR: 16-bit audio, 13-bit processing.
FOR.A: 16-bit.
HA: 16-bit.
LEX: 16-bit audio, I/O. Internal paths are larger.

Michael Fay is the editor of *RE/P*.

NED/D: 16-bit.
NED/S: 16-bit.
PFX: CPU = 16-bit; Akai S900 = 12-bit.
WAV: 16-bit.

5. What processor is used?
A+DR: 68000 (Atari Mega ST). AMS: 6800, 8086, 8085, Z-80.
DAR: 80286 (*"Multibus 2-inch allows 386 upgrades).
FAIR: 68020, 6800, 6809.
FOR.A: 280.
HA: DSP, 6800.
LEX: 6 families of processors are used.
NED/D NED proprietary.
NED/S: NED proprietary.
PFX: 8286 (Akai 8086).
WAV: Intel, proprietary 8016.

6. How much RAM does the system have (minimum/maximum)?
A+DR: 4 meg.
AMS: 0.
DAR: 2 meg RAM per 4 channels (1 processor = 8 channels, depending on application).
FAIR: 2 meg to 14 meg per track (*28 meg).
FOR.A: 6.8 seconds of RAM. Up to 8 controllers can be tied to the system.
HA: 4 meg (*8 meg).
LEX: Minimum, system accesses material from disk.
NED/D: N/A.
NED/S: 4 meg to 64 meg.
PFX: 1 meg.
WAV: 2 meg to 48 meg (30 meg = 16 voices).

7. What data dump standards are supported?
A+DR: AES/EBU to 1610/1630, F1.
AMS: Interfaces to 1610-1630 Sony (*AES/EBU and PD).
DAR: User-dependent; 1610, optical, tape streamers.
FAIR: None (*AES/EBU).
FOR.A: SMD interface.
HA: SDS, Digidesign compatible (*ADAP II AES/EBU, SDIF).
LEX: Analog, Sony SDIF (*AES/EBU).
NED/D: AES/EBU/PD.
NED/S: None.
PFX: None.
WAV: MIDI, SDS, AES/EBU = output only.

8. Is the system open-ended?
A+DR: Yes.
AMS: Yes.
DAR: Yes.
FAIR: Yes.
FOR.A: Yes.
HA: Yes.
LEX: Yes.
NED/D: Yes.
NED/S: Yes.
PFX: CPU = yes.
WAV: Yes.

9. Does the system have serial and/or parallel communication ports?
A+DR: Yes, one each.
AMS: RS-422.
DAR: Serial = 1-RS-232/-422, more as option. Parallel = Option.
FAIR: RS-232, RS-422, SCSI, four MIDI.
FOR.A: Serial RS-232.
HA: Yes.
LEX: Yes.
NED/D: RS-232.
NED/S: RS-232.
PFX: Yes.
WAV: Yes.

10. Is a printer port available for hard copy printout?
A+DR: Yes, includes PQ subcode list.
AMS: Yes.
DAR: Yes, as option.
FAIR: Yes.
FOR.A: Yes.
HA: Yes, through Atari peripherals.
LEX: Yes.
NED/D: Yes.
NED/S: Yes.
PFX: Yes.
WAV: Yes.

11. Can the system be accessed through a modem? At what baud rate?
A+DR: No.
AMS: No.
DAR: Yes, via RS-232 at 38.4k baud rate.
FAIR: Yes, maximum 9,600.
FOR.A: Yes, 1,200 to 9,600 selectable.
HA: Yes, third-party modem, Digidesign-compatible.
LEX: Yes, 2,400.
NED/D: Yes, 2,400 and 9,600.
NED/S: Yes, 2,400 and 9,600.
PFX: No.
WAV: Yes, baud-dependent on third-party modem.

12. What controllers come with the system?
A+DR: Terminal, mouse.
AMS: Terminal.
DAR: Soundstation II control console.
FAIR: Interactive graphics terminal w/ light pen (*MFX keyboard w/ jogger and mouse).
FOR.A: Up to 8 record/play workstations.
HA: Terminal, mouse.
LEX: Opus workstation console.

NED/D: Terminal, remote, mouse.
NED/S: Keyboard, terminal, mouse, guitar.
PFX: Terminal.
WAV: Terminal, mouse.

13. What other controllers can be used?
A+DR: ?
AMS: None.
DAR: N/A.
FAIR: MIDI.
FOR.A: RS-232 devices.
HA: MIDI.
LEX: N/A.
NED/D: MIDI, terminal, mouse, remote box.
NED/S: MIDI, foot pedal, touch ribbon, breath.
PFX: None.
WAV: MIDI.

Software

14. Is the operating system stored in ROM or on hard disk?
A+DR: Both.
AMS: Both.
DAR: Mostly on hard disk.
FAIR: Hard disk.
FOR.A: ROM.
HA: Both (has to access per screen).
LEX: ROM.
NED/D: Stored on hard disk, resides in RAM.
NED/S: Stored on hard disk, resides in RAM.
PFX: Stored on hard disk, resides in RAM.
WAV: Stored on hard disk, resides in RAM.

15. Can the system access multiple windows?
A+DR: Yes.
AMS: Yes.
DAR: Yes.
FAIR: Yes.
FOR.A: No.
HA: Yes.
LEX: Yes.
NED/D: Yes.
NED/S: No.
PFX: Yes.
WAV: Yes.

16. Does the system offer Help screens?
A+DR: No.
AMS: No.
DAR: Yes.
FAIR: Yes.
FOR.A: "Prompt" messages, yes; help screens, no.
HA: No.
LEX: No.
NED/D: No.
NED/S: Yes, through monitor section only. Not in real time mode.
PFX: Yes.
WAV: No (*Yes).

17. Can the system remember multiple commands with one keystroke, ie., softkey?
A+DR: Yes.
AMS: No (* Yes).
DAR: Yes, user-defined.
FAIR: No.
FOR.A: Yes.
HA: Yes, with third-party (Macro Mouse) Antic catalog.
LEX: No.
NED/D: Yes.
NED/S: Yes.
PFX: No.
WAV: No.

18. Is the source code user-accessible?
A+DR: No.
AMS: No.
DAR: Software calls are available to support user-algorhythm development.
FAIR: No.
FOR.A: Yes.
HA: Developer kits available.
LEX: No.
NED/D: No.
NED/S: No.
PFX: No.
WAV: No.

Hard disk recording

19. How much direct to hard disk storage (meg/seconds min/max.) does the system have?
A+DR: 30+ mins./320meg; up to 64 disk devices may be used.
AMS: 2 hours/760 meg (8 track), 6 hours/1500 meg (8 track).
DAR: Basic=(4)5.25 SCSI plug-in ports. Requires one 5.25-inch Winchester for each 4-channel group; 1 hour/380mb disk at 16-bit/48kHz; up to 64 drives.
FAIR: 19 min/190 meg. Up to 13 hours/4.8gig.
FOR.A: Minimum track time = 40 minutes; mono/20 minutes stereo w/170 meg; maximum track time = 2,000 minutes mono/1,000 minutes stereo w/ 1.1gig.
HA: N/A. (* ADAP II = 45 minutes stereo max.).
LEX: Minimum, 120 minutes/800meg; maximum, 480 mins/3.2gig.
NED/D: 25 minutes per track/160 meg.
NED/S: None.
PFX: N/A.
WAV: None.

20. On direct to hard disk recorders, can the system record and playback, simultaneously, in real time, and in external lock?
A+DR: ?
AMS: Yes.
DAR: Yes.
FAIR: No (*Yes).
FOR.A: Yes, simultaneous record and playback. (External lock not available at this time.)
HA: Yes.
LEX: Yes.
NED/D: Yes.
NED/S: N/A.
PFX: N/A.
WAV: N/A.

21. Can the hard disk-based recorder accommodate multiple recording and playback channels simultaneously, i.e., are there A/Ds and D/As for each channel?
A+DR: Yes.
AMS: Yes.
DAR: Yes.
FAIR: No (*Yes, D/A for each).
FOR.A: Yes.
HA: ?
LEX: Yes.
NED/D: Yes.
NED/S: N/A.
PFX: N/A.
WAV: N/A.

22. Does the hard disk recorder system offer cut, splice and paste features?
A+DR: Yes.
AMS: Yes.
DAR: Yes.
FAIR: Yes.
FOR.A: Yes, cut, paste and loop.
HA: N/A (*Yes, ADAP II).
LEX: Yes, with crossfade editing.
NED/D: Yes.
NED/S: N/A.
PFX: N/A.
WAV: No (*Yes).

Legend:
Abbreviation: Company name (Product name).
A+D/R: Audio + Design Recording (Sound Maestro System).
AMS: Advanced Music Systems (AudioFile).
DAR: Digital Audio Research (Soundstation II).
FAIR: Fairlight (Series III).
FOR.A: For-A Corporation (Sirius 100).
HA: Hybrid Arts (ADAP).
LEX: Lexicon (Opus).
NED/D: New England Digital (Direct-to-Disk).
NED/S: New England Digital (Synclavier).
PFX: Polyphonics (Optical Transfer Station).
WAV: WaveFrame (AudioFrame).

Other footnotes:
?: Data have not been verified or is not available.
N/A: Not applicable.
(*): Update available within six months.

23. Can a section of hard disk material be looped?
A+DR: Yes.
AMS: No.
DAR: Yes.
FAIR: Yes.
FOR.A: No.
HA: N/A (*Yes, ADAP II).
LEX: No (*Yes).
NED/D: Yes.
NED/S: N/A.
PFX: N/A.
WAV: N/A.

Sampling/synthesis

24. Does the system have any sound generating capabilities?
A+DR: No.
AMS: No.
DAR: No.
FAIR: Yes.
FOR.A: No.
HA: Yes.
LEX: No.
NED/D: No.
NED/S: Yes.
PFX: No.
WAV: No.

25. What type of synthesis does the system have?
A+DR: None.
AMS: None.
DAR: None.
FAIR: Additive with digital cross-modulation (255 harmonic 16-bit).
FOR.A: None.
HA: Draw wave, function generator.
LEX: None.
NED/D: None.
NED/S: Additive, resynthesis, timbre frame (FM wave table, 8-bit).
PFX: None.
WAV: None.

26. Does the playback sampling rate change with note changes?
A+DR: N/A.
AMS: N/A. Not a keyboard system.
DAR: N/A.
FAIR: Yes.
FOR.A: N/A. Not a keyboard system. Sampling rate fixed at 33kHz.
HA: ?
LEX: N/A. Not a keyboard system. Sampling rate fixed at 48kHz. (*Varispeed capability.)
NED/D: N/A. Not a keyboard system.
NED/S: Yes.
PFX: Yes.
WAV: No, interpolated.

27. What is the maximum sampling rate of the system?
A+DR: Variable, input-dependent.
AMS: 44.1kHz and 48kHz.
DAR: 200kHz, 24 bits.
FAIR: Input = 96kHz; output = 192kHz.
FOR.A: 33kHz.
HA: 44.1kHz.
LEX: 48kHz.
NED/D: 100kHz.
NED/S: 100kHz.
PFX: 40kHz (using the Akai S900).
WAV: 44.1kHz.

28. Is the sampling rate fixed, variable or selectable?
A+DR: Variable (32kHz, 44.056kHz, 44.1kHz, 48kHz. Will lock to whatever is supplied).
AMS: Selectable.
DAR: Selectable: 32kHz, 44.056kHz, 44.1kHz, 48kHz or external.
FAIR: Variable and fixed.
FOR.A: Two rates selectable.
HA: Selectable.
LEX: Selectable: 48kHz, 44.1kHz, 44.056kHz.
NED/D: Variable.
NED/S: Variable.
PFX: Variable.
WAV: Fixed.

29. Can you start in the middle of a sound and hear the sound?
A+DR: Yes.
AMS: Yes.
DAR: Yes.
FAIR: Yes.
FOR.A: Yes, through the edit mode.
HA: Yes.
LEX: Yes.
NED/D: Yes.
NED/S: No.
PFX: Yes.
WAV: Yes.

30. How many polyphonic voices are available?
A+DR: N/A.
AMS: N/A.
DAR: N/A.
FAIR: 16 per rack.
FOR.A: N/A.
HA: 6.
LEX: N/A.
NED/D: N/A.
NED/S: 64 samples, 32 FM synthesis.
PFX: 8.
WAV: 16 to 256.

31. How many sounds/voices can be accessed or played at once?
A+DR: N/A.
AMS: 8 tracks.
DAR: Dependent upon output channel configuration.
FAIR: Dependent on voice configuration; 1008 sounds in memory, 16 playable simultaneously.
FOR.A: 8 channels (requires 4 controllers, 2 channels each).
HA: 6 voices.
LEX: 8 channels.
NED/D: Dependent on track configuration.
NED/S: 96.
PFX: 8 voice.
WAV: Dependent on voice configuration.

32. Can the system be used on-line, off-line or both?
A+DR: ?
AMS: Both.
DAR: Both.
FAIR: Both.
FOR.A: Both.
HA: Off-line (*ADAP II, on-line).
LEX: Both.
NED/D: Both.
NED/S: Both.
PFX: Off-line.
WAV: Both.

33. Does the system have multi-user capability? Give an example.
A+DR: No.
AMS: No.
DAR: Yes, at least 2 control consoles on a real time network for each processor/storage unit.
FAIR: No.
FOR.A: Yes, up to 8 users on-line at once. Also, PC-compatible.
HA: No.
LEX: No.
NED/D: No.
NED/S: No.
PFX: No (*Yes, networking).
WAV: Yes. Multiple racks can be shared between multiple terminals.

34. Does the system controller have multitasking capability, i.e., can it handle more than one application at a time?
A+DR: No.
AMS: No.
DAR: Yes.
FAIR: Yes.
FOR.A: No.
HA: No.
LEX: Yes.
NED/D: Yes, when combined with Synclavier.
NED/S: No.
PFX: No.
WAV: Yes.

35. Can the system be used as a master transport controller? How many slaves can it handle?
A+DR: No (*Yes).
AMS: Yes, one.
DAR: Yes, via RS-232. Depends on external synchronizer system.

FAIR: No (*Yes, 6 slaves).
FOR.A: The system provides one GPI trigger.
HA: No (*Yes, ADAP II).
LEX: No.
NED/D: No.
NED/S: No.
PFX: No.
WAV: No.

36. In real time, can the system combine sequencer and direct to hard-disk information?
A+DR: N/A.
AMS: N/A.
DAR: Yes, w/ external sequencer/ synthesizer.
FAIR: Yes.
FOR.A: No.
HA: No (*ADAP II, SMPTE page with hard disk recorder).
LEX: N/A.
NED/D: Yes.
NED/S: Yes.
PFX: N/A.
WAV: N/A.

37. Can the system be a slave to an external sequencer?
A+DR: No.
AMS: N/A.
DAR: Yes.
FAIR: Yes.
FOR.A: Yes.
HA: Yes.
LEX: Yes.
NED/D: Yes.
NED/S: Yes.
PFX: No.
WAV: Yes.

38. Is visual editing possible?
A+DR: Yes.
AMS: Track editing only (no waveform).
DAR: Yes, uses touch-screen as interface.
FAIR: Yes.
FOR.A: No.
HA: Yes.
LEX: Waveform = no; edit relationship "bar" display = yes.
NED/D: Yes.
NED/S: Yes.
PFX: No.
WAV: Yes.

39. Can the system emulate reel-rocking as performed on analog tape?
A+DR: No, mouse-based system.
AMS: Yes.
DAR: Yes.
FAIR: Yes.
FOR.A: Yes, ¾-, ½-, ¼-speed, either forward or reverse.
HA: No (*Yes, ADAP II).
LEX: Yes.
NED/D: Yes.
NED/S: Yes.
PFX: No.
WAV: Yes.

EDL

40. What list management features does the system have? Tree structure: catalog, sub-catalog, ripple, name search, time search?
A+DR: Directory/catalog.
AMS: Alphabetical.
DAR: All of the above.
FAIR: Catalog tree structure with all, except sub-catalog and time search.
FOR.A: Tree structure with catalogs and sub-catalogs. (*Name search and time search).
HA: None.
LEX: Tree structure: job, reel, track, segment.
NED/D: Alphabetical.
NED/S: Tree structure: name search, ripple, insert/delete, comments.
PFX: User-definable: insert/delete, BLK move, name/time search, ripple, comments.
WAV: Tree structure.

41. Can cues in an event list be bounced from one channel to another without recording, and in real time?
A+DR: N/A.
AMS: Yes.
DAR: Yes, instantaneous.
FAIR: Yes.
FOR.A: Yes.
HA: N/A.
LEX: Yes.
NED/D: Yes.
NED/S: Yes.
PFX: No.
WAV: Yes.

42. Can a group of events in an edit decision list be block-erased, then rippled or offset?
A+DR: No.
AMS: Yes.
DAR: Yes.
FAIR: Yes.
FOR.A: ?
HA: ?
LEX: No.
NED/D: Yes.
NED/S: Yes.
PFX: Yes.
WAV: No (*Yes).

43. Is the EDL compatible with any other systems?
A+DR: No.
AMS: No.
DAR: Supports established EDL protocol standards.
FAIR: No.
FOR.A: No (*expected compatibility with CMX).
HA: No (*Yes).
LEX: No.
NED/D: No.
NED/S: No.
PFX: No. Output is CMX and Cue Sheet style.
WAV: No.

Sequencing

44. If the system has a sequencer, how many tracks are available?
A+DR: N/A.
AMS: N/A.
DAR: N/A.
FAIR: 16. (*80). Also, cue list sequencer w/maximum 80 simultaneous events.
FOR.A: N/A.
HA: N/A.
LEX: N/A.
NED/D: 200.
NED/S: 200.
PFX: 16-channel random-access playback.
WAV: 200.

45. Does the sequencer store/recall all real time and preset information?
A+DR: ?
AMS: N/A.
DAR: N/A.
FAIR: Yes.
FOR.A: N/A (*future software will store the EDL).
HA: ?
LEX: N/A.
NED/D: Yes.
NED/S: Yes.
PFX: N/A.
WAV: Yes, in two files: 1) sequence data,

Legend:
Abbreviation: Company name (Product name).
A+D/R: Audio + Design Recording (Sound Maestro System).
AMS: Advanced Music Systems (AudioFile).
DAR: Digital Audio Research (Soundstation II).
FAIR: Fairlight (Series III).
FOR.A: For-A Corporation (Sirius 100).
HA: Hybrid Arts (ADAP).
LEX: Lexicon (Opus).
NED/D: New England Digital (Direct-to-Disk).
NED/S: New England Digital (Synclavier).
PFX: Polyphonics (Optical Transfer Station).
WAV: WaveFrame (AudioFrame).

Other footnotes:
?: Data have not been verified or is not available.
N/A: Not applicable.
(*): Update available within six months.

2) separate sound sample and routing data.

46. Can sequence tracks be offset globally or within a section of an existing track?
A+DR: No.
AMS: N/A.
DAR: N/A.
FAIR: Yes.
FOR.A: N/A.
HA: N/A.
LEX: N/A.
NED/D: Yes.
NED/S: Yes.
PFX: Yes.
WAV: Only globally.

47. Can individually sequenced notes be edited?
A+DR: N/A.
AMS: N/A.
DAR: N/A.
FAIR: Yes.
FOR.A: N/A.
HA: N/A.
LEX: N/A.
NED/D: Yes.
NED/S: Yes.
PFX: N/A.
WAV: No (*Yes).

48. Does the sequencer remember patch changes and sample information?
A+DR: N/A.
AMS: N/A.
DAR: N/A.
FAIR: Yes, both.
FOR.A: N/A
HA: No.
LEX: N/A.
NED/D: Yes, both.
NED/S: Yes, both.
PFX: N/A.
WAV: No.

Third-party

49. Do upgrades exist from third-party vendors?
A+DR: No.
AMS: No.
DAR: No.
FAIR: No.
FOR.A: Yes, Polyphonics FX system.
HA: No.
LEX: No.
NED/D: No.
NED/S: No.
PFX: Yes, Akai S900 and For-A system (*Akai 1600).
WAV: Yes, through computer vendors.

50. Does the system include a basic library of sounds or support libraries from third-party vendors?
A+DR: N/A.
AMS: No.
DAR: N/A.
FAIR: Yes. Sound Genesis (music) and quarterly subscription through Fairlight.
FOR.A: No.
HA: Music and FX.
LEX: No.
NED/D: No.
NED/S: Yes, music.
PFX: Yes, PFX custom.
WAV: Yes, through WaveFrame (*third-party).

51. What third-party hardware is necessary to configure the system?
A+DR: Sony PCM 701ES with additional ADR interface available with the system or user supplied.
AMS: None. Manatron monitor optional.
DAR: None.
FAIR: None.
FOR.A: None.
HA: Atari CPU, hard disk, other peripherals.
LEX: None (system comes complete, but includes several third-party devices).
NED/D: None.
NED/S: None.
PFX: System is self-contained and includes a custom interface to Akai samplers.

WAV: IBM AT compatible or better, 386 is recommended.

52. Does the system have mixing capabilities? Static or dynamic?
A+DR: None.
AMS: Static.
DAR: Not as such, but independent level control over each channel or group of channels. Four DSP expansion slots.
FAIR: Dynamic, through MIDI volume and velocity.
FOR.A: No.
HA: Static, SMPTE page (*ADAP II, indefinite number of tracks in non-real time).
LEX: Full digital mixing of 12 inputs. Dynamic.
NED/D: Static.
NED/S: Both.
PFX: Static.
WAV: Static (*Dynamic).

53. Is the mix level controlled in the digital or analog domain?
A+DR: N/A.
AMS: N/A (analog mixing is optional).
DAR: Digital.
FAIR: Analog.
FOR.A: ?
HA: Digital.
LEX: Digital.
NED/D: Digital.
NED/S: Digital.
PFX: Digital.
WAV: Digital.

54. Does the system have signal processing capabilities?
A+DR: No.
AMS: Varispeed playback.
DAR: Yes, varispeed, time compression/expansion, internal support for multi-DSP cards.
FAIR: Delay families (not including reverb).
FOR.A: No.
HA: Digital delay.
LEX: No (*EQ, dynamic processing)
NED/D: No.
NED/S: Panning, vibrato, arpeggiator, delay family except reverb, EQ.
PFX: ADSR, LFO, dynamic level control.
WAV: No.

55. Does the system have any facility for patching to external signal processing?
A+DR: Yes, via AES/EBU interfaces.
AMS: No.
DAR: Yes, via AES/EBU.
FAIR: No.
FOR.A: No.
HA: N/A.
LEX: Yes, through patch points and a send/return network.
NED/D: No.
NED/S: No.
PFX: Yes.
WAV: No (*Yes).

56. What type of filtering is available?
A+DR: None.
AMS: None.
DAR: None.
FAIR: Low-pass: tracking, fixed and time variant.
FOR.A: None.
HA: None.
LEX: None.
NED/D: None.
NED/S: SFM mode, frequency-selective (number crunching).
PFX: Low-pass.
WAV: Low-pass.

57. Does the system have any sound modification capabilities, i.e., looping, user-defined crossfade looping, reverse, merge, mix, redraw waveshapes, maximize volume, invert?
A+DR: Not a sampler, but has loop and repeat.
AMS: No.
DAR: Yes.
FAIR: All of the above, plus Fourier analysis and resynthesis.
FOR.A: N/A.
HA: Yes.
LEX: Crossfade parameters of duration, gain, slope, mark point.
NED/D: N/A.
NED/S: Looping, user-defined crossfade looping, reverse, merge, mix, maximize volume.
PFX: Looping, reverse, merge, LFO.
WAV: Looping, crossfade looping, reverse, maximize volume (*merge, mix, inverse).

Mixing/signal processing

58. Does the system have solo and/or muting capabilities? Static or dynamic?
A+DR: Mute/enable of source and master "reels" = dynamic; mute/enable during editing = static.
AMS: Solo = static; muting = static.
DAR: Dynamic on each channel for record, playback and editing.
FAIR: Solo = both; muting = both via software switch.
FOR.A: None.
HA: N/A (*Yes, ADAP II).
LEX: Solo and muting = dynamic.
NED/D: Solo = both; muting = static.
NED/S: Solo = both; muting = both.
PFX: Static.
WAV: Solo = static; muting = static (*Both, dynamic).

Storage/backup/protection

59. Can all edits and events be stored with the direct to hard disk information?
A+DR: Yes.
AMS: Edit info on floppy, cue info on hard disk, event list on floppy.
DAR: Yes.
FAIR: Yes.
FOR.A: Yes.
HA: N/A (*Yes, ADAP II).
LEX: Yes.
NED/D: Yes.
NED/S: N/A.
PFX: N/A.
WAV: N/A.

60. What is the system used to back-up data?
A+DR: F1 or R-DAT (*tape streamer).
AMS: Sony 701/601—¾-inch, VHS, Beta.
DAR: Customer selected: AES/EBU, SCSI for 1610, optical WORM, tape streamer, R-DAT, floppy disk.
FAIR: Optical disk, tape streamer.
FOR.A: None (*tape streamer).
HA: Tape streamer, optical (through third-party vendors).
LEX: One or two tape streamer drives.
NED/D: Digital tape streamer.
NED/S: Kennedy, optical disk, floppy.
PFX: WORM optical disk, Bernoulli.
WAV: Tape streamer (*Optical WORM).

61. What provisions or protection does the system have for main power fail interruption? Is any EDL or audio material lost?
A+DR: Only current edit lost.
AMS: None (*battery backup).
DAR: No data lost of power fails, except operations performed 10 seconds prior to loss of power.

Legend:
Abbreviation: Company name (Product name).
A+D/R: Audio + Design Recording (Sound Maestro System).
AMS: Advanced Music Systems (AudioFile).
DAR: Digital Audio Research (Soundstation II).
FAIR: Fairlight (Series III).
FOR.A: For-A Corporation (Sirius 100).
HA: Hybrid Arts (ADAP).
LEX: Lexicon (Opus).
NED/D: New England Digital (Direct-to-Disk).
NED/S: New England Digital (Synclavier).
PFX: Polyphonics (Optical Transfer Station).
WAV: WaveFrame (AudioFrame).

Other footnotes:
?: Data have not been verified or is not available.
N/A: Not applicable.
(*): Update available within six months.

FAIR: Sequences saved to hard disk as updated.
FOR.A: Can sustain a power loss of up to 80%. All information saved.
HA: None.
LEX: Continual log of controller events and periodical storage to disk. All info saved with the possible exception of edit in progress.
NED/D: None. Material lost if not saved to disk.
NED/S: None. Material lost if not saved to disk.
PFX: Optional power converter.
WAV: None.

62. Do the hard disks have any protection?
A+DR: Power supply filtering.
AMS: None.
DAR: Write protection.
FAIR: Yes.
FOR.A: N/A.
HA: None.
LEX: Yes, if power loss, hard disk will shut down with no loss of material.
NED/D: Yes.
NED/S: Yes.
PFX: Optical and Bernoulli disks are removeable.
WAV: N/A.

SMPTE

63. Does the system read and write longitudinal SMPTE formats?
A+DR: Reads = yes; writes = depends on external synchronizer.
AMS: Read only.
DAR: Reads = yes; writes = depends on external synchronizer.
FAIR: Yes.
FOR.A: N/A.
HA: Yes, also user bits.
LEX: Read only on system demonstrated (*delivered systems will also generate all standards).
NED/D: Yes.
NED/S: Yes.
PFX: Read only.
WAV: Yes.

64. Can the system read VITC?
A+DR: No.
AMS: No.
DAR: External synchronizer dependent.
FAIR: No.
FOR.A: N/A.
HA: No.
LEX: No.
NED/D: No.
NED/S: No.
PFX: No.
WAV: Yes (not on system shown).

65. Are you able to lock to SMPTE in slow mode (1/5 speed) and play to picture with audio reproduction?
A+DR: No.
AMS: No.
DAR: Yes.
FAIR: No (*Yes).
FOR.A: No.
HA: No.
LEX: No.
NED/D: Yes.
NED/S: Yes.
PFX: Yes, 1 frame to 40x play speed.
WAV: Yes.

66. Can you mark within a sample to set a SMPTE trigger offset?
A+DR: Yes.
AMS: Yes (hard disk recorder).
DAR: N/A. Not a sampler.
FAIR: No.
FOR.A: No. Unit does not provide for SMPTE interface.
HA: No.
LEX: Yes.
NED/D: Yes.
NED/S: Yes.
PFX: Yes.
WAV: No (*Yes).

67. Can the system stay in sync if there is a dropout in the time code?
A+DR: ?
AMS: No.
DAR: Yes.
FAIR: Yes.
FOR.A: N/A.
HA: Yes.
LEX: Yes.
NED/D: Yes.
NED/S: Yes.
PFX: No.
WAV: Yes.

68. What sequencer timing formats are supported? SMPTE, measures and beats, beats, patters?
A+DR: None.
AMS: N/A.
DAR: N/A.
FAIR: All of the above.
FOR.A: N/A.
HA: N/A.
LEX: N/A.
NED/D: SMPTE, measures and beats, beats, feet and frames.
NED/S: Same as above.
PFX: N/A.
WAV: SMPTE, beats (*feet and frames).

69. Can EDL/event lists be converted from one time code to another, i.e., from 24 frames to 30 frames?
A+DR: No.
AMS: 24 or 25 to 30 (not inverse).
DAR: EDL independent of time code format.
FAIR: Yes.
FOR.A: No.
HA: No.
LEX: No (*Yes, 24, 25, DR, 30).
NED/D: Yes.
NED/S: Yes.
PFX: Yes.
WAV: Yes.

MIDI

70. Does the system support the MIDI standard?
A+DR: External controllers.
AMS: No.
DAR: Yes, for synchronization and triggering.
FAIR: Yes.
FOR.A: No.
HA: Yes.
LEX: Only if converted to SMPTE time code.
NED/D: Can drive external MIDI devices.
NED/S: Yes.
PFX: Yes.
WAV: Yes.

71. Can it utilize the incoming MIDI data (aftertouch, velocity, pressure, no. of real time controllers, positional crossfade, velocity crossfades, release velocity)?
A+DR: No.
AMS: No.
DAR: No.
FAIR: Yes.
FOR.A: N/A.
HA: All but release velocity.
LEX: N/A.
NED/D: Yes.
NED/S: Yes.
PFX: No.
WAV: All but positional crossfade and release velocity (*all).

Legend:
Abbreviation: Company name (Product name).
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NED/S: New England Digital (Synclavier).
PFX: Polyphonics (Optical Transfer Station).
WAV: WaveFrame (AudioFrame).

Other footnotes:
?: Data have not been verified or is not available.
N/A: Not applicable.
(*): Update available within six months.

72. Can the system control external MIDI devices?
A+DR: External controllers.
AMS: N/A.
DAR: Yes.
FAIR: Yes.
FOR.A: ?
HA: Yes.
LEX: ?
NED/D: Yes.
NED/S: Yes.
PFX: Yes.
WAV: Yes.

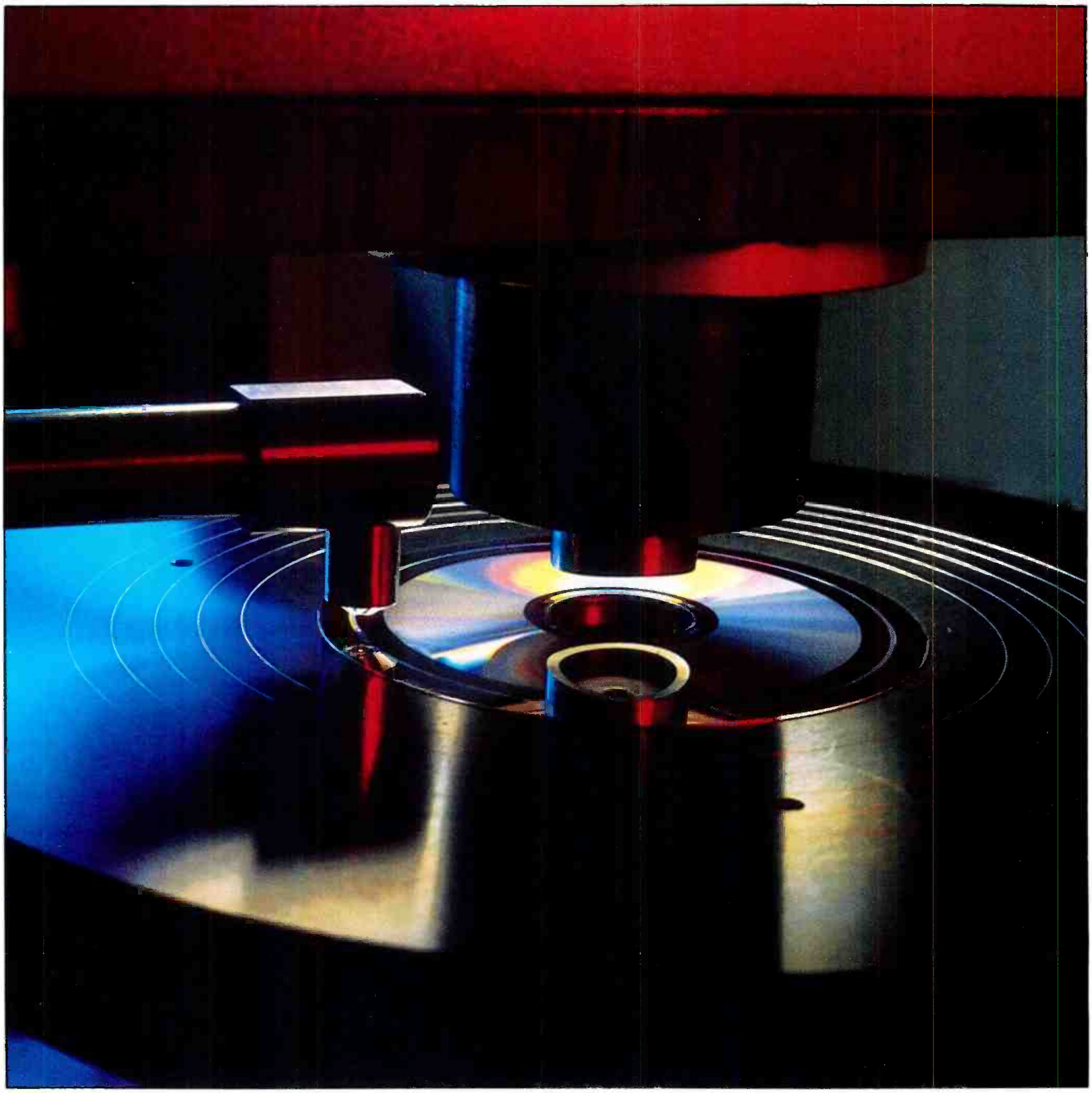
Other

73. If a purchase order were placed on March 1, 1988, what would be the delivery time of the system as demonstrated?
A+DR: ?
AMS: Immediate to 3 weeks.
DAR: Approximately 6 weeks.
FAIR: 14 to 60 days.
FOR.A: 14 to 30 days.
HA: Immediate (*ADAP II, April 30, 1988).
LEX: 90 days.
NED/D: Immediately to 30 days.
NED/S: Same as above.
PFX: Immediate to 60 days.
WAV: Immediate to 30 days.

74. What is the cost of the system as demonstrated?
A+DR: 20,000£ approximate = 1 hour stereo.
AMS: $85,000 = 2 hours mono, 1 hour stereo, DSP card, reel rocking, varispeed. Option = mixer plus stand, $85,000 to $125,000.
DAR: $79,500 = 4 channels/2 track hours.
FAIR: $39,950 = 8 voice/4 meg; range = $39,950 to $120,000; average = $75,000.
FOR.A: $35,000 = basic system with 170Mb drive, 1 record/PB remote, 1 mainframe w/ 2 channels, disk controller, power supply, cables.
HA: $1,995 = ADAP software (*$2,995 = ADAP II software w/ hard disk recorder. Atari CPU and hard disk not included).
LEX: $160,000 = 1 workstation, 2 A/D, 1 D/A, 1 LDI, 1 disk, 12 I/0 modules.
NED/D: Average = $115,000 for 8-track stand alone w/ terminal, meter and mouse (*remote box).
NED/S: Average = $150,000; range = $100,000 to $400,000; $290,000 = 32meg RAM, 8-track Direct-to-Disk system, 8 MIDI outputs.
PFX: Range = $10,000 to $32,000; $32,000 = rack-mount CPU, 400 meg WORM.
WAV: Range = $45,590 to $88,290; $45,590 = 16 voice, 16 meg/RAM, 2 ins, 8 outputs.

75. What additional features may we look forward to seeing in the near future?
A+DR: PQ subcode editing; SMPTE synchronization; MIDI.
AMS: Video editor control with Ampex Ace; looping; time compression; cut and paste of actual cue (sample).
DAR: Updates are noted within specific answers.
FAIR: 6820 waveform supervisor; MFX control panel; multitrack hard disk recorder; on-board digital mixer; 100kHz stereo sampling; expanded RAM and voice capabilities; AES/EBU standard supported.
FOR.A: Updates are noted within specific answers.
HA: Direct to hard disk recording; optical storage; EDL, CMX compatible.
LEX: Four-band, full range, user defined EQ; automation, as soon as possible; additional on-board signal processing; pre-dedicated application software.
NED/D: Updates are noted within specific answers.
NED/S: Updates are noted within specific answers.
PFX: Optical jukebox (66 optical disks); Akai S1600 stereo sampler.
WAV: 16-input digital mixing; hard disk recording; full MIDI editing; SFX database; third-party developers kit; signal processing; DSP card. RE/P

Digital Storage

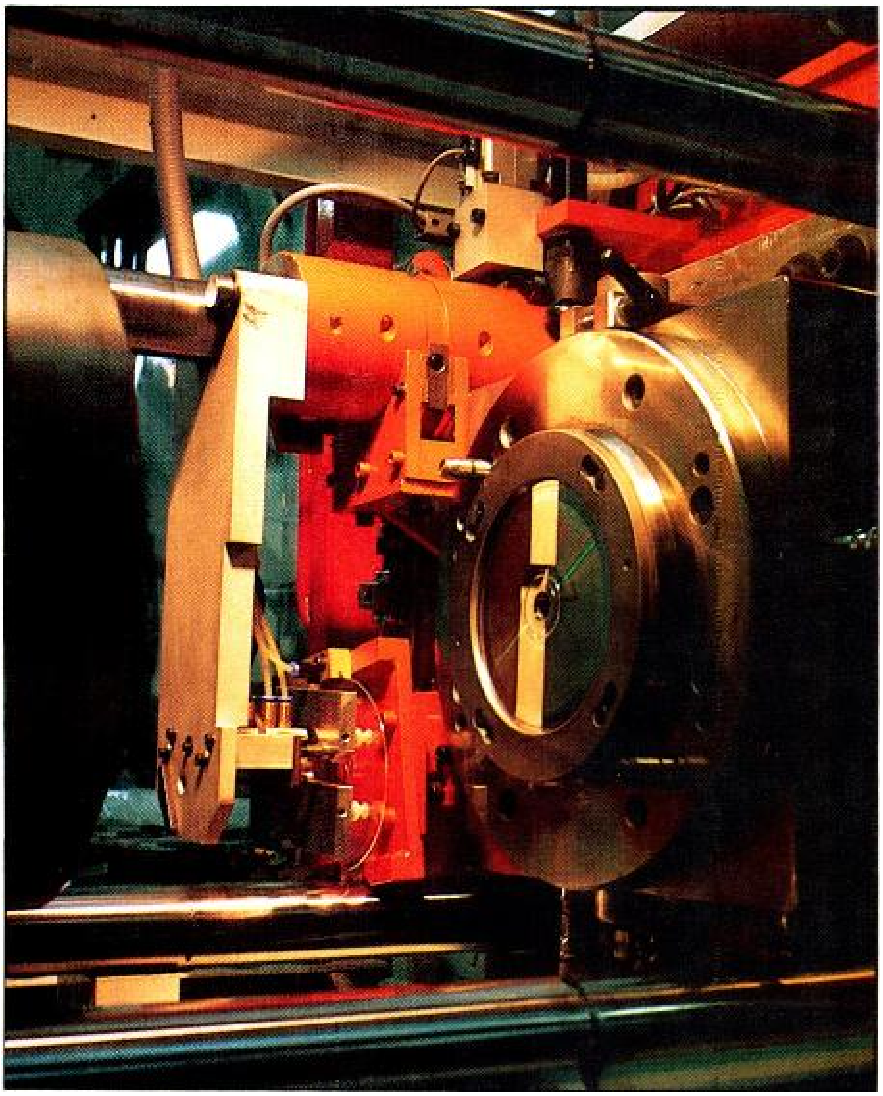


Removing a recently manufactured disc from the stamper.

Technology

By Paul D. Lehrman

An advanced, modular, in-line production system is making possible the delivery of high-quality compact discs, while allowing faster production cycles and lower costs.



The injection-molding chamber.

When a new technology matures, there are many signs. Prices go down. Supplies no longer become scarce, manufacturers start taking more competitive stances, "second generation" products and production methods emerge, and business people and investors from outside the industry start to become interested. That maturity is evident in the current state of the compact disc industry.

Until very recently, CDs were a luxury, for consumers, engineers and artists. If you wanted to hear what your work sounded like on CD, you had to be working on a major label contract because there was simply not enough pressing capacity for everyone. And to get time in a plant, you either had to wait in long lines or come up with huge orders. Pressing CDs was also expensive—an order of magnitude

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Applying the lacquer on a metallized disc.

Robot arm transferring the lacquered disc to the printer.

higher than the cost of vinyl—which made it difficult for a smaller label or production house that couldn't afford the risk.

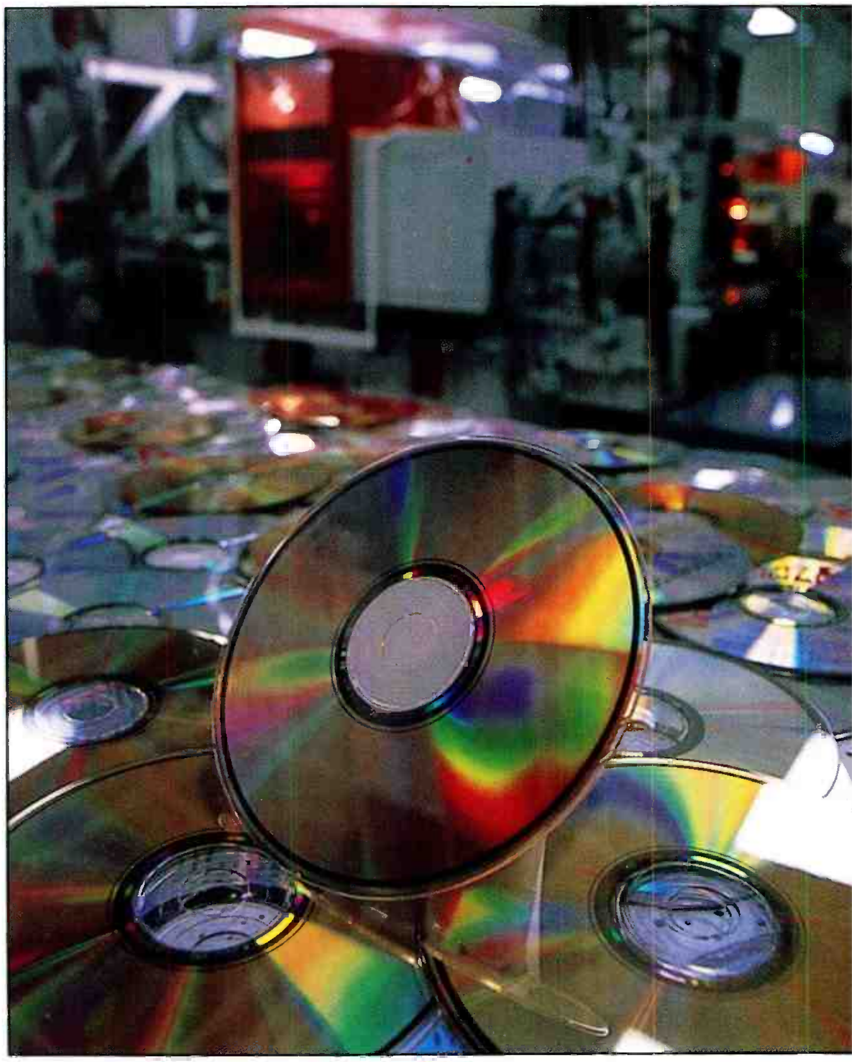
CD manufacturing

In the past year, however, that's all changed, as the capacity of CD manufacturing plants has risen to meet and even exceed demand. CDs are fast becoming the democratic medium everyone always hoped they would be. With nearly two dozen U.S. plants opening in the past two years, the smallest independent label can now get products quickly. Prices are also decreasing to the cassette level, and even vinyl in some cases.

Nowhere are these signs of industry maturity more visible than at Technetronics. a new CD plant in West Chester, PA, that opened last May. Technetronics positions itself to large and small clients as an alternative to more established CD plants by offering faster service, the ability to take on small runs and competitive pricing policies.

The company employs a new production process called Monoline that does not require the traditional clean room as in other CD plants.

When you walk out on the manufacturing floor at Technetronics, you don't have to cover your hair or your clothes. It's not a good idea to smoke inside the plant, but otherwise you could be in an ordinary plastics factory. Instead of the traditional method of manufacturing discs one stage at a time in batches and waiting between stages, the Monoline process works with only one disc at a time, pushing it through



The injection-molding chamber and a finished disc.

from start to finish in a single fully automated operation that lasts only two minutes, with a new disc emerging every 8.5 seconds.

The manufacturing process

Each assembly line is called a "cell," and requires only 24 square feet of clean area, which is provided by a plastic shroud that uses positive air pressure to keep contaminants out. The 8.5-second cycle time for each disc means that each cell is capable of producing nearly 7,000 discs in a 2-shift day. Technetronics currently has two cells fully operational, two more cell installations in progress and room for 16 more on the manufacturing floor.

The Monoline technique was developed by Ronald Kok, a Dutch consultant to the plastics industry who was involved with the CD's initial development.

"He came over on a 1-day consult," says Gary Kauffman, president of the company, "and we hit it off. I proposed that he become part of the company, and a week later he agreed."

Kok now owns a minority interest in Technetronics, and operates his own company, Rokoma B.V. He also owns the European licensing rights to the process.

Kauffman is a newcomer to the entertainment business, having made his mark as a successful builder of residential communities in the Philadelphia area, and on any given day is as likely to be found atop a front-loader as he is behind his office desk at Technetronics.

"It had all the right ingredients as a business opportunity," he says about starting the company. "It's a little more controllable than the building industry, where you're at the mercies of the weather and interest rates, and homeowners are never satisfied.

"Of course," he adds, "that may be a little naive on my part."

From the digital master tape (like all CD plants, the Sony 1610/1630 format is standard), a glass master is made first. A metallic substrate is etched onto a glass disc and cut with a laser. The next step is similar to traditional LP lacquer processing, with the resulting master having a shelf life of only three days, so a nickel-plated mother is made immediately using an underwater electroforming process.

Clean room vs. clean shroud

The water bath has been filtered for particulate matter as small as two microns.

"It filters out viruses," says public relations director David McQuade. "You have 5 billion pits on a disc, each one containing a digital bit, and 23 miles of track. The eyelash of an ant could cover dozens of pits. A few microns of oil from your hand will screw up the bath for a week."

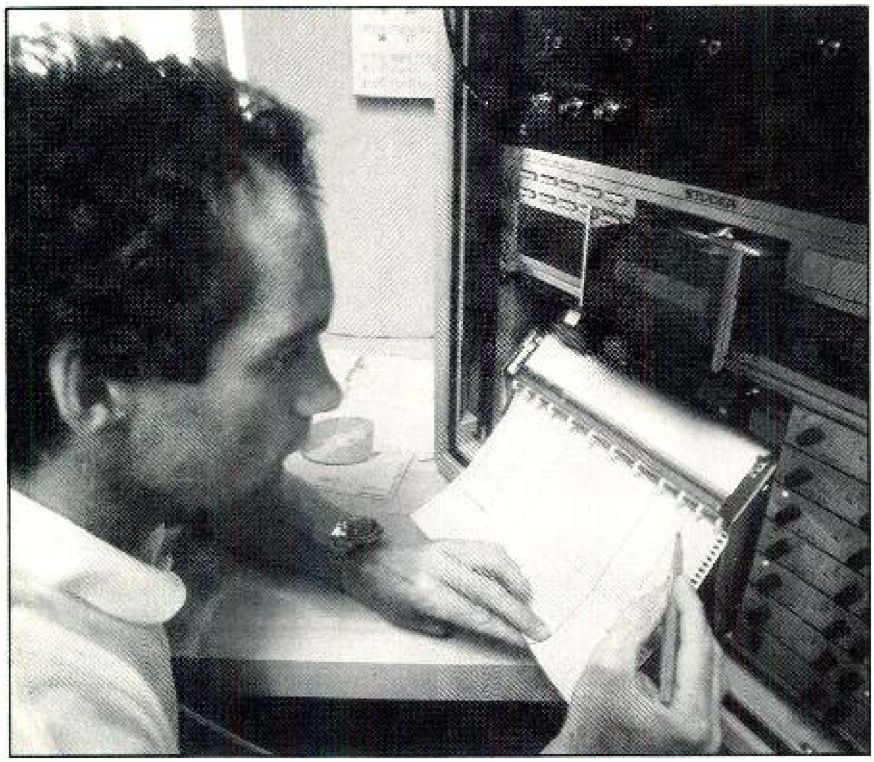
Like all of the clean areas at Technetronics plant, the electroforming is not done in a true clean room, but rather, under a "clean shroud," which provides what the industry calls Class 100 purity.

From the mother, one or more nickel-plated fathers are made, and then nickel-plated stampers are made from the father. The mothers, fathers and stampers all have a healthy shelf life, but it's hard to say exactly what it is—"the industry is too young," says McQuade.

The stampers are fitted into an injection-molding machine, which lives under another clean shroud.

"We use straight injection, not a combination of injection and compression like most plants," says McQuade, "to minimize internal stresses on the plastic resin, which can disturb the pickup laser."

The polycarbonate resin is heated so that it becomes a liquid and is then injected into the stampers. It's at this stage that the pit information is impressed into the disc by the stampers—not, as many think,



Block-error checking the finished disc.

at the metallization process. The metallization, which comes next, is necessary because the laser cannot reflect off of the clear plastic and read the data. The metal layer consists of a couple of microns of aluminum alloy (for videodiscs, a little gold is mixed in), and it is electrostatically "sputtered" on the side of the disc opposite the pit layer, so that it is behind the pits from the point of view of the laser.

Although the stamper only uses one side for data, the other side must be carefully prepared as well, and must have a perfect mirror finish. If not, the metal will not lie flat on the resin, and the laser reflections will be distorted. Contamination in the metallization process, known as pinholes, must be carefully guarded against.

"Pinholes," McQuade explains, "are essentially little explosions that occur when the disc is placed into the metallization chamber, which is a vacuum. In audio, this makes no difference except in cosmetic appearance, but as we move into CD-ROM, it makes a big difference."

Robotics in production

A robot arm grabs the disc and puts it into a lacquering machine that in a few seconds spins on a layer of lacquer.

"We mask the edge of the disc," says McQuade, "leaving more room between the edge of the metal and the edge of the disc. We don't use a 'sandwich' configuration where the edges of the metallization are left exposed. This way we get fewer problems with flaking and chipping."

Imperfections in the lacquer must also be kept to a minimum, so as not to disturb the laser beam.

After the lacquering, the disc is picked up by another robot arm and it leaves the clean shroud. It goes into a printing machine, where a silkscreen-like process is used to print one or two color labels.

In a batch-style CD plant, the through-time for a disc is typically several hours, while at Technetronics it is two minutes.

"The critical stage in the manufacture is between the injection molding and the metallization," says McQuade, "which is where contaminants can cause pinholes. We limit that stage to a second or so, in a very small area, and we closely protect the disc at that stage.

"But you don't need to protect the whole plant that way—the clean area for each line is only 4′ x6′. You could theoretically set this up in your garage. Why have people walking around in bunny suits, breathing their own air, when you don't have to?"

In fact, the robot operation means that humans never have to enter the clean area while assembly is taking place. The first time the disc is touched by a human being is after it comes out of the labeler, when it is visually inspected and then sent on to be packaged. The masters and stampers themselves, which would otherwise have to be stored in clean areas while awaiting use, are instead spun coated with a lacquer that is peeled off just before they are placed into the injection molder.

The Monoline technique

Using the Monoline technique has allowed Technetronics to operate with one-tenth the square-footage and one-tenth the personnel of a conventional plant with the same capacity. It requires only one person to monitor each cell, and there is a technical service support team on the floor at all times in case of problems. When any part of a line has trouble, the whole line does not have to shut down—there are storage points all along the way where the discs can collect until the trouble is fixed. Another advantage to the Monoline system is quality control. Discs are checked as soon as they come off the line, so any problems can be spotted *quickly* and there is far less chance of the disc being mislabeled or mispackaged. [This writer recently acquired a CD of Handel's "Water Music" and found in the jewel box, along with the Handel, a disc of Springsteen's "Darkness On the Edge of Town."]

"None of the components we're using—the lacquer coater, the metallizer, or the injector—were designed for one-on-one manufacturing," says Gary Kauffman. "They were designed for batch processing. We had to either change the design or get them custom built. We think we ended up with a system that is maneuverable, flexible and won't lock us into an older way of manufacturing."

The Monoline technique, besides lowering costs, improves quality, according to McQuade.

"Phillips allows a block error rate of 220 (errors per second)," he says. "Our typical error rate is less than 10."

A Studer CD player is in constant use, with a special error monitor, to look at samples of the finished product.

"Of course," he says, "the major record labels are not as interested in quality as they are in speed, service and price. But I figure, if you can drive a Cadillac for less than a Chevy, why not?"

Economics of the Monoline technique

When using this type of technology, the prices available to clients are unusual. While competition is forcing older CD plants to drop their prices below $2 for the is currently the subject of intense debate.

Experiments have shown that a signal passed through several of these filters will suffer clear audible degradation, but one or two well-designed units used in the signal chain are of questionable audibility.

Another problem with steep low-pass filters is high frequency distortion. The sharp cutoff response requires *high-Q* sections in the filter. These high-Q sections must handle large swings of signal current and voltage when the signal frequency approaches the filter cutoff. If the filter is a passive LC filter, the large current swings drive the inductors into non-linear behavior, resulting in distortion. In the case of an active filter, the op-amp slew rate and output current limitations cause distortion near the cutoff frequency. This distortion does not always show up with a THD test because the harmonics are *above* the cutoff frequency and never reach the filter output.

After filtering, the next stage in the signal path is the sample and hold. This circuit looks at the signal for a short instant in time and holds this value long enough for the A/D converter to convert the voltage into a binary pattern. The audio signal is constantly changing, requiring the sample and hold to grab the value in a very short amount of time. Otherwise, it will introduce an artificial smoothing of the signal over the sample time.

This also results in a high-frequency response rolloff. The time required to grab the signal is called the *aperture time,* and is a critical specification of a sample and hold circuit. A 1kHz signal must be sampled with less than 2.4ns aperture time for less than one-half least significant bit (LSB) error on a 16-bit conversion.

When a sample and hold is driven with a large high-frequency signal, it must be able to change from the maximum positive signal voltage to the maximum negative signal voltage in one sample interval. This process often imposes severe slew rate and settling time limitations.

The error rate on the new sample value depends on the last sample value. If it was very close to the existing value, there will be little error because the adjustment to a new voltage was an easy change to make. Deficiencies, however, generally show up as a high-frequency distortion similar to that introduced by the antialias filter.

Sampling intervals are set by a master clock in the digital circuitry. This clock must be extremely stable, or the resulting samples will not be equally spaced in time. The instability of the samples is called *jitter,* and results in an error in the amplitude of the sample. This error is larger if the slope of the signal is steep in the vicinity of the sample. Figure 2 illustrates this effect.

The error waveform may be thought of as a signal-dependent noise source, much the same as modulation noise in tape, but at much smaller magnitudes. The amplitude of the noise will increase with rising signal frequency. Jitter may also be introduced in the sample and hold itself, but the effect is the same. A 100dB S/N on the 1kHz tone described earlier would require a spec of less than 1.6ns jitter.

The A/D converter must faithfully translate the signal's voltage signature into binary words. Most A/D converters are limited to fewer bits of accuracy than they have resolution. This will show up as a higher noise and distortion floor than the resolution would imply. For example, in a system with 16 bits of resolution, the noise and distortion should be approximately 96dB below the amplitude of a full-scale sine wave. A converter with poorer accuracy will result in a higher noise and distortion floor than specs would indicate.

Datasheet claims of 16-bit conversion are to be taken with a grain of salt (maybe a pound of salt). A/D converters that are accurate to 16 bits cost several hundred dollars and are the current state of the art.

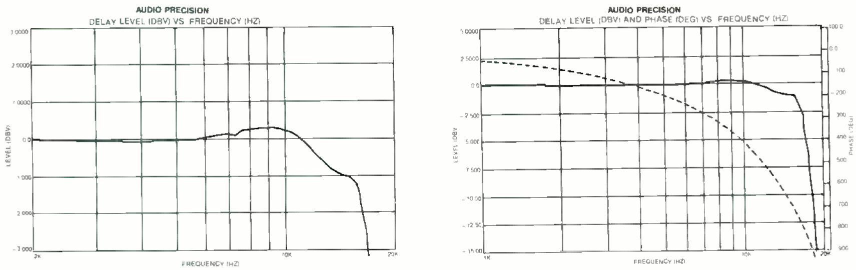


Figure 3. Amplitude and phase response of a commercial delay.

Figure 4. A 1kHz THD+N test as a function of signal level.

The typical accuracy of *commercial* 16-bit converters used in digital audio is about 14 bits. Substantial progress is being made in the technology involved, which is beginning to yield true 16-bit converters at affordable prices.

Delay line

In delay lines, the digital values representing each sample are placed in a memory and retried the appropriate amount of time later. The delayed samples are fed to a D/A converter to make an analog signal. When the binary input to the converter changes, the analog output will not transition smoothly from one state to the next. The bits will all change at slightly different times, and the switches inside the D/A will change at different rates, causing the converter to produce spikes at the edges of each sample.

These spikes are called *glitches*, and require a circuit to remove them, called a *de-glitcher*. It is essentially a sample and hold that is switched into the hold mode when the changes are sent to the D/A. These circuits have much the same problems as the input sample and holds.

To remove the staircase-type edges from the signal, a filter is placed at the output of the sample and hold. This is essentially the same as the antialias filter at the input, but is called a *reconstruction filter* because it reconstructs the analog waveform from the staircase output of the D/A. The same comments on distortion, overload behavior and phase response for the antialias filter apply to the reconstruction filter.

However, harmonic distortion in the reconstruction filter above its corner frequency can appear in the output signal. This is because there is no sampling process afterward to alias them down to lower frequencies.

Because the signal passes through two sharp low-pass filters, the phase and amplitude responses described earlier are magnified by two. The response of a commercial delay is shown in Figure 3. The total system is down 1dB at the high end and peaks up ½dB an octave below this. The phase shift is similarly doubled to more than 700° at cutoff.

Many of the distortion mechanisms discussed can be seen on THD+N tests. A 1kHz THD+N test as a function of level was run on the unit mentioned above. The data are given in Figure 4. An interesting bump can be seen in the measurements, which is most likely due to non-linearity in the converter. These data show not only the distortion in the classic sense, but the change in noise floor as a function of signal level because of ranging in the converter.

Another test for the presence of any compander circuits and for noise modulation due to ranging in the converter was developed by the British Broadcasting Corporation. It involves applying a low-frequency signal to the device and filtering it off at the output with a high-pass filter. This removes the signal and its significant harmonics, and allows measuring the noise level in the presence of signal. The signal level may then be varied and the noise level plotted.

This measurement was performed using a 10Hz tone, a 100Hz high-pass and a 10Hz notch on the delay tested above. The results are plotted in Figure 5. Note the broad rise in noise level below 0.1V. This is indicative of a ranging converter or a compressor circuit. The fallacy of measuring this unit's noise level with no signal is obvious.

Distortion tests

High frequency distortion tests are an interesting prospect in digital systems. Measuring harmonic in bandlimited systems such as tape machines has always been a problem at high frequencies because the harmonics are filtered off when they fall outside the system bandwidth. This limits THD measurements in a 20kHz bandwidth to approximately 7kHz for valid results.

This problem does not occur in digital systems because harmonics above one-half the sampling rate will alias down into the passband. They will not be at harmonic multiples of the fundamental frequency, but they will still appear. By measuring THD+N, a figure of merit may be obtained, but care must be exercised not to mistake inadequately filtered sampling glitches for distortion.

THD+N as a function of frequency was measured on a commercial digital delay and is shown in Figure 6. The two curves were run at levels at 1V and 0.5V input. To the extent that the two curves track each other, this portion of the rise in distortion at high frequencies may be sampling leakage.

Intermodulation tests will show distortion behavior very well, and are indicative of the audible side-effects of the filter. Figure 7 shows a delay excited with a pair of test tones close to its cutoff frequency. Distortion products will be generated at

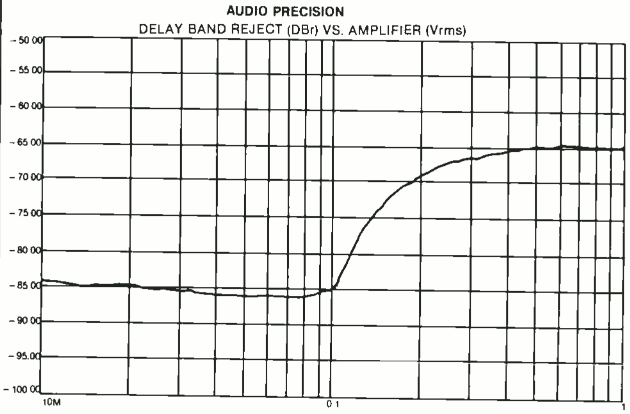


Figure 5. Quantization noise as a function of signal level.

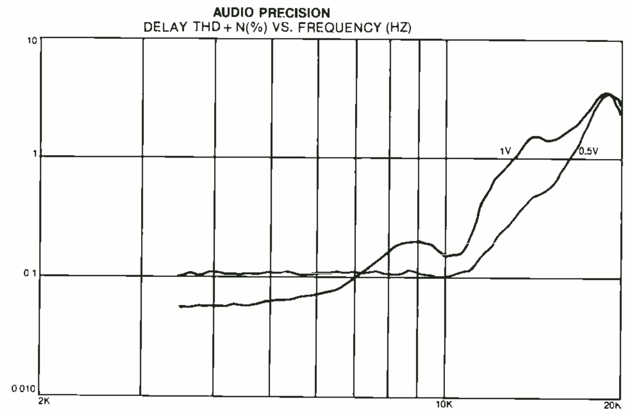


Figure 6. THD+N as a function of frequency at 0.5V and 1V levels as measured on a commercial digital delay.

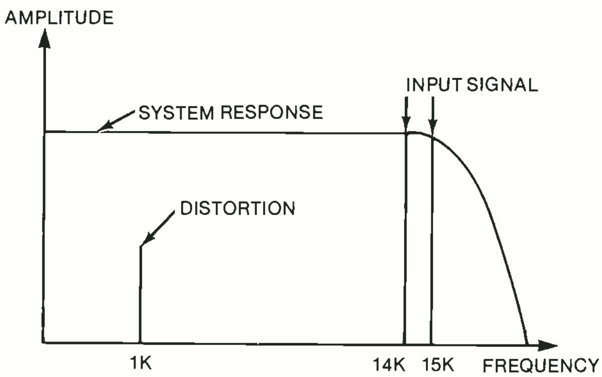


Figure 7. Difference frequency (CCIF IM) distortion measurements.

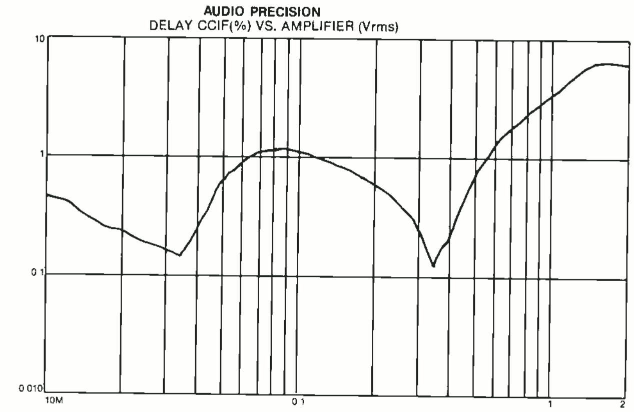


Figure 8. Difference frequency distortion vs. level on a 14kHz and 15kHz tone pair as a function of amplitude.

the difference frequency between the tones and at the offset frequency from the two high-frequency tones. A plot of the difference frequency distortion on a 14kHz and 15kHz tone pair as a function of amplitude is shown in Figure 8. The broad rise at higher amplitudes stems from a bona fide distortion mechanisms in the converters and low-pass filters.

Digital reverberators differ from delays in that they contain a digital arithmetic unit, which combines samples according to a formula. This mathematical processing adds noise and distortion to the signal, often as a function of the signal characteristics and the reverb algorithm.

This has two main causes. When samples are scaled (multiplied by a constant) and added, the processor used has a finite number of bits or precision in its calculation. The round-off errors can accumulate over several calculations and become significant contributors to the signal noise and distortion.

The second reason is more subtle. Suppose the arithmetic unit has 32 bits and the A/D converter is supplying it with a properly *dithered* 16-bit word. (*Dither* is a process by which quantization errors are reduced or mitigated by the use of random noise signals.) The processor would be capable of performing many thousands of calculations before it ran into round-off error limitations. However, more likely than not, the results of this high precision calculation must be truncated to 16 bits to make it through the output D/A converter.

Consider the effect of this process on a signal reflection path, such as a sample which is delayed and scaled down in amplitude. If the original signal was properly dithered for the 16-bit resolution of the converter, this dither will be lost when the samples are attenuated. For the case of 12dB of attenuation, we have taken out 16-bit properly dithered signal and turned it into a 14-bit undithered one.

The solution lies in digitally dithering the results of the calucations to the resolution of the next step in the signal path. This resolution limitation is usually imposed by the D/A. However, it may be the size of the memory words used to store the results for use in another calculation.

Commercial units exist that delay digitally and perform the combining in the analog domain. These suffer from a related set of problems. The delayed signals are output through a D/A converter and combined into an op-amp at the input to they delay. Each simulated reflection is passed through the A/D and D/A process once for each simulated wall in the room. A single late arriving reflection may have passed through the conversion process hundreds of times. Although the signal degradation can be enormous, the signal is attenuated each time through and contributes a small amount of the total reverberant energy. The average degradation of the reverberation quality is related to the average number of times reflections pass through the chain. .

There are no easy ways of testing the sound quality of the reverberant energy. This problem is analogous to testing the distortion on signals in a room. There are sharp peaks and dips in the response curve because of the calculations caused by reflections mixing with the direct sound and with each other.

Seldom are performance specs of commercial units presented to the level of detail shown here. The safest way to evaluate the performance of delays and reverbs is to get your hands on it and measure it yourself. Barring this, try to get distortion specs as a function of frequency *and* as a function of level.
Noise should be specified over a range of signal levels, not just with no signal present. Frequency response should be checked in the mode you want to use it, because it is often different for different operational modes of the device. You should *listen to* the unit you are considering.

RE/P

Hard Disk Recording Technology

Advancements in hard disk drives have provided new creative tools for audio professionals.

By Jerry Whitaker

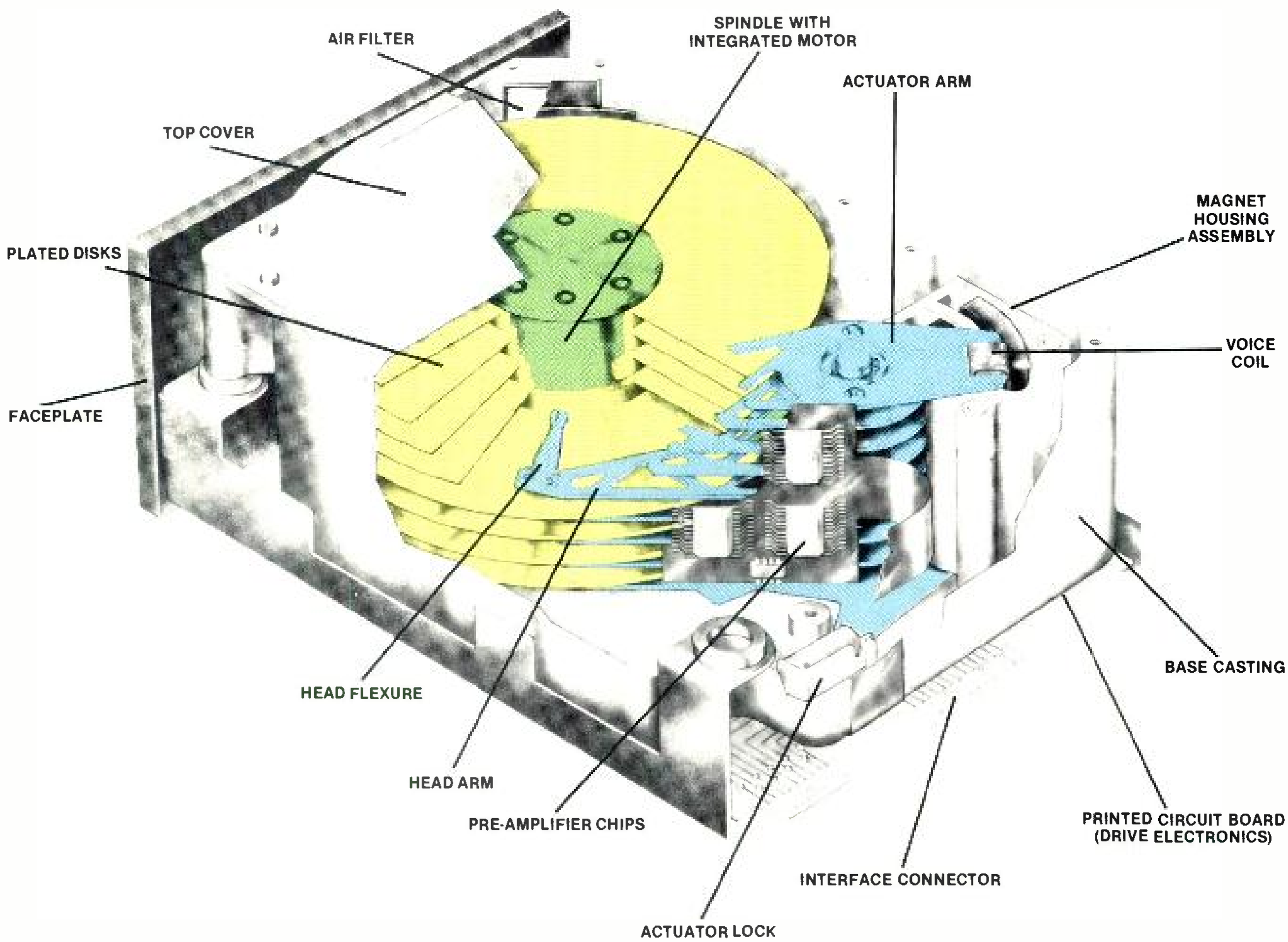


Figure 1. *Construction of a typical hard disk drive. Note that each disk is locked together by an integrated spindle motor.*

Data is data. Right? Well, when you're talking about data that represents video or audio signals, the similarities stop at the ones and zeros. The recording and reproduction of data for common computer applications is not *time dependent.* It is true that the access time for a storage device has a significant effect on the execution time of any given program. However, the only problem for the computer operator is a few wasted seconds or tenths of a second.

When you start recording video or audio, however, a tenth of a second is a big deal. A disk-based recorder must be capable of keeping up with the sampled input, and then reproducing the signal in *real time.*

Winchester disk

A Winchester disk drive is a historic name commonly used to describe a class of hard-drive systems. The exact origin of the name is a bit unclear. However, most accept the term to represent a wide variety of hard disk storage units, where the head floats over the medium on a thin film of air.

The design of a Winchester disk is elegant in its simplicity. There are few moving parts to wear out or lose their designed tolerances. In nearly all disk drives, one read/write head is associated with each side of each platter. (See Figure 1.) Each head is connected to an arm that is mechanically linked to the other read/write head arms to form a single mobile assembly. This assembly is moved across the disk by the *head actuator*, a special solenoid or motor.

Unlike floppy disk drives, hard disk platters constantly spin, at least while powered up. The mass of a large hard disk platter assembly can take significant time to achieve its designed rotational speed of about 3,600rpm. Start-up time for a hard disk from a cold stop can range from 10 to 30 seconds.

Most newer read/write head actuators use a *voice coil* assembly that provides fast and reliable operation. The voice coil approach is a substantial improvement over the jerking action of mechanical band positioner designs used previously. The voice coil mechanism operates in a fashion similar to a loudspeaker voice coil, hence its name.

A voice coil head actuator operates in a closed-loop environment. The disk controller always knows where the head is by reading data from a special, dedicated platter in the disk pack. This platter, the *servo surface*, includes a special pattern that identifies each disk location. The voice coil pulls the head mechanism against the force of a spring and, by varying the current through the coil, each part of the disk can be accessed.

The platters in a hard drive are generally constructed of thin rigid aluminum disks covered with a magnetic material, such as ferric oxide compound coated over the aluminum substrate and held in place by a binding material. The process is not unlike that used to produce magnetic tape.

Improvements in the medium itself have provided users with greater storage capabilities. Thin-film magnetic coatings have been applied to platters to yield disks with greater packing density.

Data tracks can be placed closer together on thin-film disks, meaning that the drive's read/write heads need not move as far between random bites. This improves, to a small extent, the *data access time* of the drive.

An additional benefit of thin-film technology is the durability it gives to the disks themselves. Thin-film disk packs are less vulnerable to some forms of head crashes, that are usually fatal to an oxide-coated disk. When a head crashes on an oxide-coated platter, it actually plows a small channel in the soft oxide coating, destroying data as it goes. When the harder thin-film media is used, however, the head may merely bounce off the disk, preserving the recorded data. The benefits to users are obvious.

Data format

When the drive reads or writes data, the head actuator must stop its lateral motion across the disk. Each time the platter completes one revolution, the head traces a full circle across its surface, defined as a *data track*. Each head traces out a separate track simultaneously across its associated platter.

The combination of all tracks traced out during one revolution at any given position on the disk is defined as a *cylinder.* This description comes from the shape that is traced by the heads from the bottom to the top of the disk pack.

Most hard disk drives divide each track into short arcs, or *sectors*, usually 17. The division of sectors on a disk is determined by the formatting of the system.

Construction

The construction and assembly of a hard disk drive is a delicate process. A typical drive must position the head in relation to the platter with an accuracy of 20μm or less while the platter spins at 3,600rpm, and these tolerances must be maintained over the life of the product. The intricacy of the assembly process of a hard disk is, in fact, comparable to that of an integrated circuit.

There is basically no maintenance required for a hard disk storage system. The units are assembled in a clean environment and evacuated to protect the media from contamination. Barring a break in the vacuum envelope of the recorder, the only contamination threat involves a head crash, which is usually catastrophic anyway.

To increase the security of the data recorded on a hard disk, better drives offer a *park and lock* feature that withdraws the read/write head from the active data portions of the disk when the drive motor is shut down. This minimizes the possibility of damage to active data during transportation of the drive.

Disk features

Winchester drives come in a wide variety of options and capabilities. The 5¼-inch hard disk drives available today can be supplied with upward of eight platters installed, and two heads per platter (front and back sides). Disk units can be supplied with a wide variety of configurations, stor-

MEDIA	STORAGE CAPABILITY	COST
5¼-INCH FLOPPY	360 kb	$ 1.50
5¼-INCH FLOPPY	1.2 Mb	3.50
3½-INCH CARTRIDGE	750 kb	1.50
3½-INCH CARTRIDGE	1.5 Mb	3.00
BERNOULLI DISK	20 Mb	100.00
OPTICAL DRAW DISK (12-INCH)	2500 Mb (double-sided)	450.00
WINCHESTER REMOVABLE	20 Mb	100.00
WINCHESTER DISK DRIVE	320 Mb	2,000.00

Table 1. *Comparison of the storage capabilities of various media and the approximate cost of the media. Note that the cost of a 320Mb Winchester drive applies to the hardware itself. (Source: reference 1.)*

Jerry Whitaker is the editorial director for Broadcast Engineering and Video Systems magazines.

age capabilities and data access times.

Disk drives provide data output in one of two modes: parallel or serial. Parallel drives permit faster writing and reading speeds because the entire byte of data (all 8 bits in an 8-bit system) are available simultaneously. Parallel disk units are more complicated and, not surprisingly, more expensive.

Advancements continue to be made in packing densities for hard disk units. Each year brings new improvements in disk technology, driven primarily by the computer industry's growing needs for mass storage and rapid, random access.

Error correction

Although digital recording is, theoretically error free, much discussion has been given to *error correction* for various digital recording formats. In point of fact, there is no such thing as "error correction." More correctly, it is "error concealment."

Error correction, however, is primarily a concern only with media that is exposed to the operating environment. Dust, dirt and other contaminants can make portions of any magnetic media unreadable, as illustrated in Figure 2. It is the function of error correction to allow reconstruction of the lost data so that the fault is not apparent to the user.

The types of media most susceptible to errors caused by media contamination are reel-to-reel or cassette/cartridge tape and floppy disks. High-quality hard disk drives are evacuated and hermetically sealed to prevent contamination from affecting the storage media. As a result, error correction schemes are rarely required for a hard disk-based system. The result is a system that is simpler to implement in hardware and software.

Even with the best hard disk media, however, some contamination can occur during the manufacturing process. The solution to this problem is implemented during assembly of the disk-based recording system. A software routine examines all storage addresses on a hard disk unit prior to shipment to look for media errors. After those bad addresses are identified, the test fixture burns a PROM that will reside in the host data-management computer. The PROM identifies all of the addresses to which data should not be written or read. The end result is that no errors are generated in the recording and reproduction processes. With no errors to correct, there is no need for error correction.

This is not to say, however, that error correction could not or has not been implemented in disk recorders for audio or video. Error correction codes have been developed by the computer industry to permit reconstruction of lost data from the remaining data, provided the errors are not too profound. Codes used in equipment today are virtually media-independent. For example, the Reed-Solomon code can be found implemented in systems using Winchester disks, compact discs and magnetic tape.

With the continual drive in the computer industry to pack greater amounts of data onto a given medium, the importance of error correction codes increases. With higher packing density, minute media imperfections can result in large data losses. Higher packing densities require higher quality media and, in some applications, more powerful error correction schemes.

How many bits?

The resolution to which an analog audio or video signal is coded during the analog-to-digital conversion process involves a number of critical trade-offs. Additional bits mean greater dynamic range, which translates into the primary system specifications of concern to users. Additional bits also mean a more complicated machine, and a more expensive machine.

Each additional bit in the data word doubles the resolution of the recorded analog signal. An 8-bit system can have a theoretical dynamic range of 49.76dB. A 16-bit system's theoretical dynamic range is 97.76dB. A shorthand formula for determining the theoretical dynamic range of a digital system is as follows:

Dynamic range (in decibels) = (6 x N) + 1.76, where N represents the number of bits in the system. [Editor's note: See Jeff Burger's "Understanding Computers"

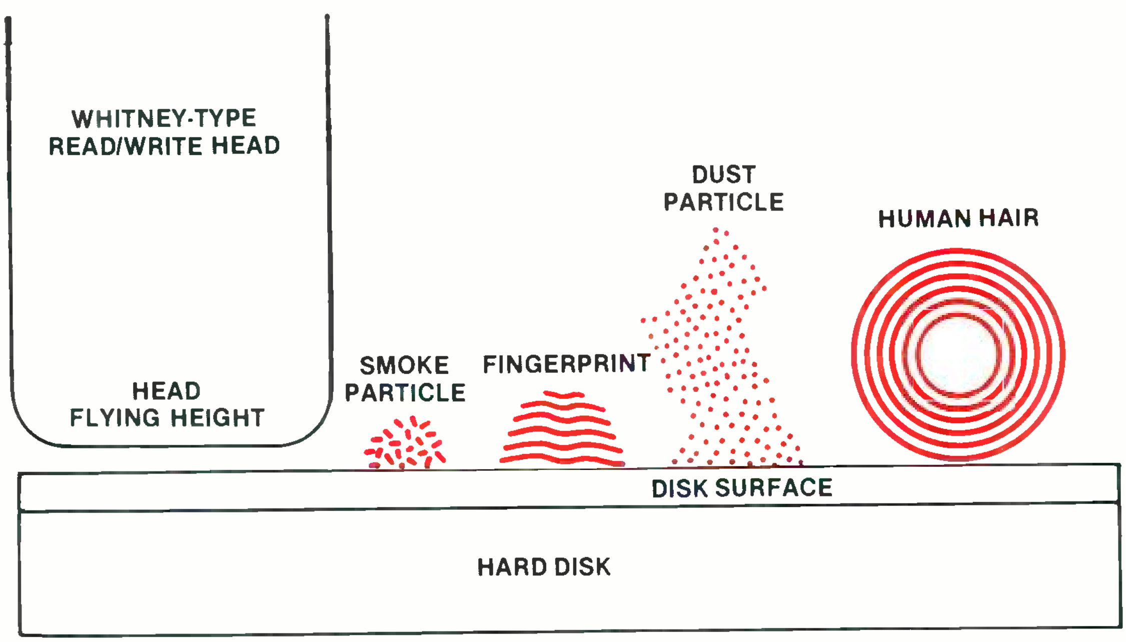


Figure 2. *A dramatic illustration of why hard disk drives are evacuated and sealed. Many commonly-occurring contaminants can destroy data, or even the disk drive itself. (Source: reference 6.)*

column in the April issue for further explanation of bits as they relate to audio.]

In audio recording using hard disk systems, a number of variations can be found. The current practical limitation is 16 bits, although higher-bit-rate systems have been produced. Because the bandwidth requirements of audio are small compared to that of a video signal, disk-based audio recorders offer users much longer recording times and, in some cases, user-selectable sampling rates. Systems are available that allow the user to trade-off recording time for audio bandwidth.

For example, one system might provide 26 minutes of stereo recording time on a group of Winchester drives with a sampling rate of 50kHz. At the option of the user, however, the sampling rate may be switched to 100kHz, but at the penalty of only 13 minutes of stereo recording time available. It may be important to the user to have the wider bandwidth when recording synthesized sounds that exhibit extended high-frequency information.

The bandwidth (frequency response) of a digital audio or video system is limited by the sampling rate of the A/D converter and its associated filters. The *Nyquist* criteria states that any sampled continuous waveform can be reproduced faithfully, provided the sampling rate is at least twice the highest frequency present in the sampled waveform. For the disk-based audio recording system described previously, the sampling rate of 50kHz would provide frequency response out to 25kHz. At the 100kHz sampling rate, response would extend in theory to 50kHz.

Anti-aliasing filters are used in A/D converters to prevent sampling of input signals that are higher than the frequency limits set by the Nyquist theorem. Figure 3 shows a representative filter used in the A/D stage of an audio recorder. The filter is designed to exhibit zero loss at 20kHz, and 86dB loss at 22.5kHz. Not surprisingly, the design of anti-aliasing filters is a difficult proposition.

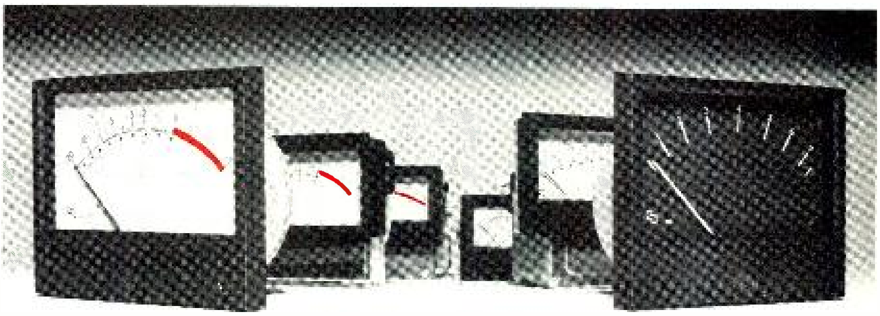
Disk-based recording systems offer the user virtually infinite layering capabilities.

Distortion in a digital system is the result of inaccuracies in the quantization process of the A/D converter. It is directly related to the bit resolution. The greater the number of bits used to define the input signal, the more faithful the digital representation will be. For video systems, distortion components show up as errors in differential gain and phase.

Hard disks are byte-organized devices. Data buffers, which serve to smooth the data flow from the disk, can be used to process the output data to provide special functions or effects. The data may also be modified in place by providing access to an audio or video signal processor.

Disk-based recording systems offer the user virtually infinite layering capabilities. Because sealed hard disk units do not use error correction schemes, the full benefits of digital recording can be realized. In any practical digital video or audio reel-to-reel or cassette/cartridge recorder, there is a finite limit to the number of generations that any section of program can be recorded before media-induced errors become visible or audible. The major drawback to a hard disk system is its limited storage capabilities.

Within the time frame of the recording limit of a disk-based system, access to a



program segment is fast, sometimes instantaneous. To improve access time, the methods by which data is written to the disk must be managed carefully. For example, if one segment of a program spot on a video recorder is stored on cylinder 1 and the next segment is stored on cylinder 330, only 1 field access time is available to move from one point to the other if the recording is to be played back in real time.

Access time

The factors determining the net data-transfer rate of a disk drive are numerous and complex. Software designers must use ingenuity when designing for random access. One approach involves allocation of temporary storage positions on disk or in RAM for duplicate portions of data to accommodate the access times that are required for the audio recorder. These operations are performed automatically without intervention by the operator.

The mechanical design of the hard disk system and components has a major effect on the time it takes for a disk drive to go from one data address to another. For real time recording and reproduction of audio, the heads must be at a given address at a specific time.

Some of the mystery and misunderstanding that occur when broadcasters shop for a disk-based recording system is that users look at their PC disk drives and say, "well this Winchester disk only cost me $250," and then look at a disk-based audio recorder and see that the drives cost much more. That extra cost goes into the electrical design and mechanical hardware needed to provide a system that will meet the necessary specifications for access time and other critical parameters.

Most disk drives have a servo system that allows the head to lock onto the proper storage address. There is with any system, however, some amount of delay in the actual movement of the head carriage. If a command is given for the head to move to track 10, the head will generally not land directly on track 10, but will overshoot slightly. When the system corrects for the overshoot, the head may overshoot slightly in the other direction. Rapid settling of the head into a specific track is a function of the weight of the head and carriage and the power and reaction time of the head-moving mechanism.

The effective data-transfer time of a drive is determined by the time it takes for the heads to seek a new sector, and includes the *latency* time (a period of one revolution of the media), track-to-track access time and full sweep time (the maximum *seek time* for a new track).

The disk-controller hardware and software play a key role in improving access time. Important elements include the ability of the controller to transfer a track of corrected data at a single pass and to minimize the gap in data transfer at track boundaries by *spiraling*.

Spiraling represents the degree to which data can be transferred without interruption. Several modes of spiraling can be identified. The most rapid mode occurs when the *sector seek* at a track boundary is sufficiently fast to permit continuous data transfer across the boundary without incurring a track latency delay. This is possible for multihead drives when the heads can be electrically switched within cylinders.

Under normal circumstances, however, when a seek to an adjacent cylinder is initiated, the access time caused by head movement (typically 2ms-10ms) will incur a latency (typically 16ms). *Skew sectoring* is a technique by which the first sector in each cylinder is offset to anticipate this delay and can be used to advantage in disk systems with many tracks of low capacity.

Most disk drives have a servo system that allows the head to lock onto the proper storage address.

Interchanging media

One drawback to disk recorders is the difficulty of interchanging media. In the computer industry, disk packs are available that permit moving files from one machine to another, or one facility to another. The costs, however, for such systems are high. Furthermore, removal of the disk pack from a sealed environment opens the door to possible contamination and subsequent data errors.

Because of the inherent problems in exchanging disks, some form of digital off-line storage is used. Several variations of hardware and formats are available for both video and audio recorders, most centering around streaming linear magnetic tape. There is, however, a time penalty in streaming tape off-line storage. Most systems cannot run real time. Again, the cost of the off-line system is a major concern. Off-line storage using optical disks holds promise for future systems.

Optical disk drives

The use of an optical storage medium is attractive because it permits exchange of programming between facilities. Most optical drives offer error specifications that compete well with Winchester drives. Most WORM (write once, read many) optical drives currently available, however, offer low data-transfer rates and data access times that are significantly longer than their magnetic counterparts.

For audio applications, the data-transfer rate and seek time are critical performance factors. High transfer rates provide the excess data needed for buffer memory

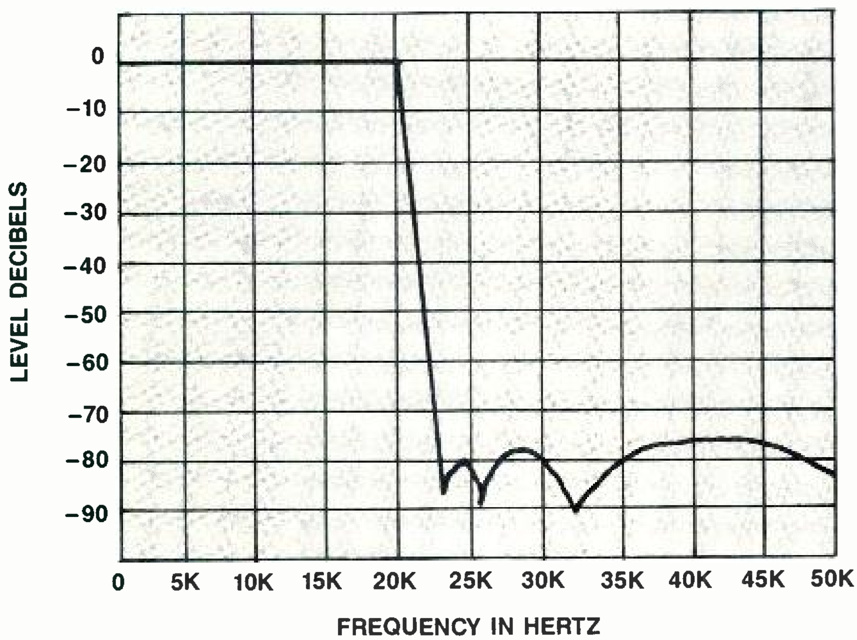


Figure 3. *A typical frequency response plot of an 11th order elliptical anti-aliasing filter used in a digital audio recorder. (Source: reference 1.)*

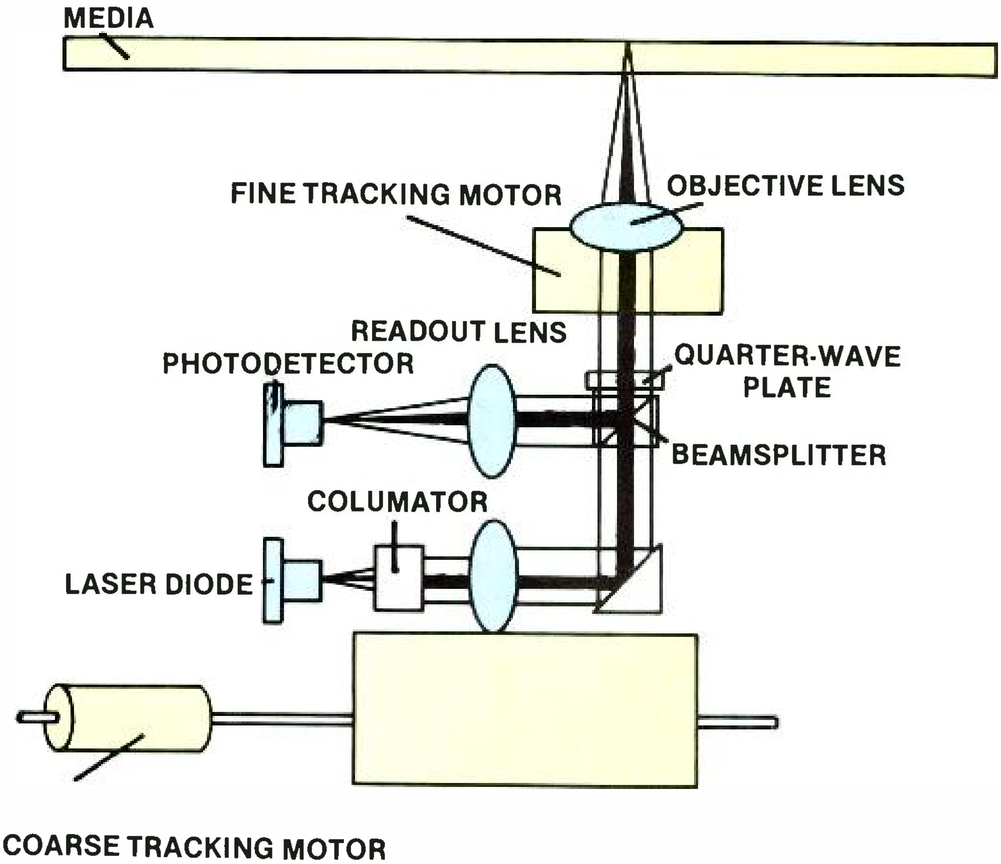


Figure 4. *Major components of a WORM disk drive. Note that the fine tracking motor is positioned to permit reading several tracks without movement of the coarse tracking system. (Source: reference 4.)*

during seek periods when no data can be read.

Improvements will continue to be made in the key specifications for optical disks. At present, several equipment manufacturers have introduced products based on WORM drives that provide non-real-time storage of images and sounds.

During the recording process for a WORM disk, a laser is used to etch a hole in a thin metal film near the surface of the disk. Surface tension on the film itself causes the hole to enlarge to its final form. Figure 4 shows a simplified diagram of a WORM drive. The system uses an integrated focusing scheme for reading data from the disk. A coarse positioning mechanism and *fine tracking* motor work together to allow direct access to a number of tracks on the disk without the need for additional movement of the coarse tracking assembly.

The future

What does the future hold for media storage? Optical storage of various types is emerging and will continue to be applied for various applications, first to functions that do not require real time audio playback. Streaming tape technology will also improve, and new off-line storage methods based on video technology will become practical.

One concern with any new storage technology is volatility in the marketplace. In an area of rapidly developing technology, the lifetime of a product can be less than a year. Audio professionals do not want to purchase a product that may be made obsolete by a new development before the unit is paid for. Audio professionals demand consistency in the hardware they buy.

Established technologies generally provide the best return on investment for the end-user. RE/P

Editor's note: The author wishes to thank the following individuals for providing input for this article: Mark Pinkel, Abekas Video Systems; Kevin Dauphinee, Digital Audio Research; and Mark Terry, New England Digital.

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The Capital Reinvestment Zone

By Tom Martin

The type of equipment purchases you make has an important effect on the future growth of your studio.

Case A: Bob Sparks slowly amassed the necessary equipment to put together his own 8-track "garage" recording studio. Although it was only 8-track, he had an impressive collection of mics, digital signal processing equipment, monitors, amps and headphones.

Bob's clientele consisted of local bands, friends and an occasional jingle. His hourly rate averaged about $15. For the most part, Bob's studio was generating enough capital to be self-sufficient. That is, Bob didn't have to sink cash from his regular job into the studio to keep it running.

Case B: Like Case A, Empire Inc. also amassed the necessary accoutrements, but at a faster pace and on a larger scale. Empire is located in a large city and is a full-scale automated 24-track facility with some of the newest toys available. Empire's clientele consists of commercial post-production, an occasional national recording act and tape duplication.

What do the two cases have in common? They are about to take a trip into a land where time and history have little meaning, where the dollar in your pocket can lead you down the yellow brick road or become a ticket for a trip through the looking glass of nightmares. Both are about to take a trip into the Capital Reinvestment Zone, an ethereal state where a recording facility must eventually reinvest its hard-earned money for such purposes as expansion or replacing old, unreliable equipment.

At some point in the life of a recording facility, and for any number of reasons, it becomes desirable and/or necessary to purchase some additional equipment. For the most part, equipment purchases fall into three categories: major, sustaining and replacement.

A *major* purchase will cost a lot of money and usually results in a dramatic upgrade of the facility's capabilities. A *sustaining* purchase represents the acquisition of a less expensive piece of equipment to address a specific operational need (such as a better reverb or new microphones). A *replacement* purchase is simply the replacement of equipment that doesn't perform any longer or is unreliable and is beginning to give the facility a bad name.

Tom Martin is a consultant with Audio Image Consulting in Seattle.

How, when, why this is done and what is purchased become the central issues to a decision that must eventually be made. This decision can put a facility out of business or possibly catapult it into a higher income bracket. For this reason, some guidelines should be established to help the studio owner make a wise decision and thus minimize any loss.

Technology decisions

With regard to any purchase, technology should first be addressed. One of the questions studio owners have to decide when making a major purchase decision is whether to go digital or stay analog.

If the answer is digital, what type of digital system should you purchase? Should you buy from a new company? Or should you consider investing in a new type of technology, such as a random access hard disk recorder, instead of digital tape?

Unfortunately, there is no easy answer. It can be difficult to get enough accurate information to make a clear buying decision. There is also the whole issue of format compatibility. When you consider that a digital 2-track recorder can cost upward of $25,000, a digital multitrack (any format) can cost upward of $195,000 and random access digital recording equipment often starts at $50,000, it becomes obvious that whatever your choice, if any, at best it may be as reliable as a roll of the dice. So what do you do?

The first thing is to decide if it is time to purchase. This can be a simple process of looking at your books or at your existing equipment. Obviously, if your multitrack is constantly stopping in the middle of a session and doesn't seem to respond to any repair efforts, it may be time to give it the boot.

Likewise, if you have been trucking along with your "good ol'" reliable 16-track and have a loyal client base, it may be a good time to look at upgrading to 24-track and possibly an automated console. It may also be just the right time to purchase another digital reverb or sampling keyboard system. Any decision to purchase should be made carefully and be based upon your current financial situation, projections for future business and how this purchase will affect your current business. Will the purchase produce new revenue, opportunity, and expand your capabilities?

Assuming you have made the decision to purchase, you now have to figure out what to purchase. This is where things get sticky. You're about to make a major investment in you business and your life, and you should approach this move with a somber attitude.

While we are on the subject of *what* to purchase, I would like to familiarize you with one other term—"planned obsolescence." Equipment manufacturers are in the business of doing two things (and in this order); making money and making equipment.

There is a very complex series of events that take place before a piece of equipment hits the shelves. As an example: research must be done to establish if there is a need for this product, and if the product is unique enough (in function or cost) to set it apart from the mass quantities of other products on the market.

The target market that the product will appeal to must be established carefully, as well as how long this product will be useful within that market. Manufacturing and advertising costs must also be established. All of this "research & product development" costs money that the manufacturer eventually recoups as the product sells.

With that in mind, the manufacturer must then establish how long this product will retain its usefulness and "flash appeal" before another product will take the spotlight. In essence, the product has a marketing shelf life—one that is usually designed into the product—where XYZ Company can expect to get its money back and hope to see profit for building and selling this gadget. However, there is one problem scenario. The manufacturer has been a product for the last two years and selling the product at full retail, but now there are other very similar products on the market and it is starting to affect sales to the point that it is no longer cost effective to build or market that product.

As a result, the company may decide to either pull it from distribution completely or to supersede (and in some cases, obsolete) it with a newer model that *may or may not be* compatible with the old model. Even worse, it's not uncommon for a manufacturer to make a last push to sell the product before introducing a new version in an attempt to clear inventory. Furthermore, the manufacturer may or may not inform the public that there is a new version that will be available soon.

The bottom line is that you could end up buying a device that will lack the manufacturer's advertising coverage hence also lack appeal to your clients, be difficult to maintain and service, or lose over half of its value the minute you take it out of the box, accelerating its depreciation value. There are some measures you can take to guard against "white elephant" purchases. First you should ask the dealer and/or manufacturer if there are plans for future expansion and upgrade of the product you are considering. Also, ask yourself if this product does exactly what you need and/or are there any other improvements that you might want later to enhance its capabilities. If so, ask the manufacturer/dealer if this enhancement could be added cost effectively at any time.

However, the most important recommendation I can make is to place software upgradeability as a priority when considering the purchase of a piece of new equipment. There are many manufacturers putting out products that only require a "chip change" to upgrade it from the old Ford to the new Ferrari. Last of all, if you do buy something and find out that three months down the road the manufacturer has discontinued it, contact the manufacturer to see what options are available. The first thing to remember is patience. Never be coerced or pushed into making a decision because of statements such as, "This is our showroom sample," or, "If you become our Beta test site we'll sell it to you at cost."

Digital dilemmas

One equipment issue, digital recorders, deserves separate mention. My impression, from monitoring the technological trends, is that we are at a fork in the road. Off to one side we have the digital tape recorder (in its various non-compatible formats), and to the other side we have the emergence of random access recording devices (including sampling devices).

Digital tape systems are in use—tried, proven and accepted. Any recording, be it analog, digital or random access, must eventually be mixed to tape (digital or analog) for mastering and eventual pressing. If it's going digital, it must be transferred to digital tape (especially for CD applications, where it must eventually be processed through Sony's 1630 digital processor/recorder).

The manufacturers of tape-based equipment have spent incredible sums of money in research and development. In order to realize a return on investment, they must continue to sell this equipment for a long time. Now, enter into the market random access recording technology with sampling rates equal to or higher than what the digital tape formats use.

The higher the sampling rate the finer the resolution, and therefore the better the sound. Further, no error correction is needed in random access recording as there is with recording tape. Random access editing is quick, and non-destructive (once you cut a piece of tape...that's it, unless you have a back-up copy). And random access hard disk recording technology is now nearly as affordable as multitrack digital tape technology.

But for now, with random access you are limited in recording time and tracks. And again, for any type of mastering and pressing you must still go to tape.

Compatibility is still a problem that plagues all the various digital tape systems. By the time the digital tape manufacturers arrive at a standard format, it will be a moot point because the industry will have accepted and adopted random access recording technology.

A by-product of all digital recording technology is clean sound (no noise, no wow & flutter). But lower noise floors can create other problems for the studio owner purchasing digital equipment. Any defects in room acoustics, noisy signal paths or noise auxiliary equipment will become more apparent in the recording.

You may have heard CD recordings that were created digitally in facilities that were once multitrack analog facilities. You're sure to hear the less-than-perfect room acoustics (flutter, echoes, bass resonances) as well as quite a few electronic "pops" and "clicks." What this could mean is that after purchasing a $200,000 digital recording system, you may then have to sink another $50,000 to $100,000 to rebuild your room.

The point of all this is that whatever you choose, think about where the system will reside. You're going to have to integrate your investment in your present facility, and if you have to spend additional money to upgrade, you need to factor that in before you make a final decision.

Research your purchase

Gather as much information as possible about the product you are considering and get a feel for "technological trends." There is quite a lot of redundant product available. The more you know about what's available, you'll be in the best position to make the right decision.

And last but not least, factor in after-sales service and support. Although this topic requires a separate article (which will appear in a future issue), the type of service you get after you've written that check makes as much difference as the service you get when you're shopping.

Many facilities are holding back on the purchase of any type "leading edge" technology. If anything, they are sticking with their analog recorders while incorporating the various types of sampling keyboard technology available. With one of these systems and an analog 8-track, you can produce quite a lush and complex product that sounds nearly as good as if it were done in one of the world's hottest facilities.

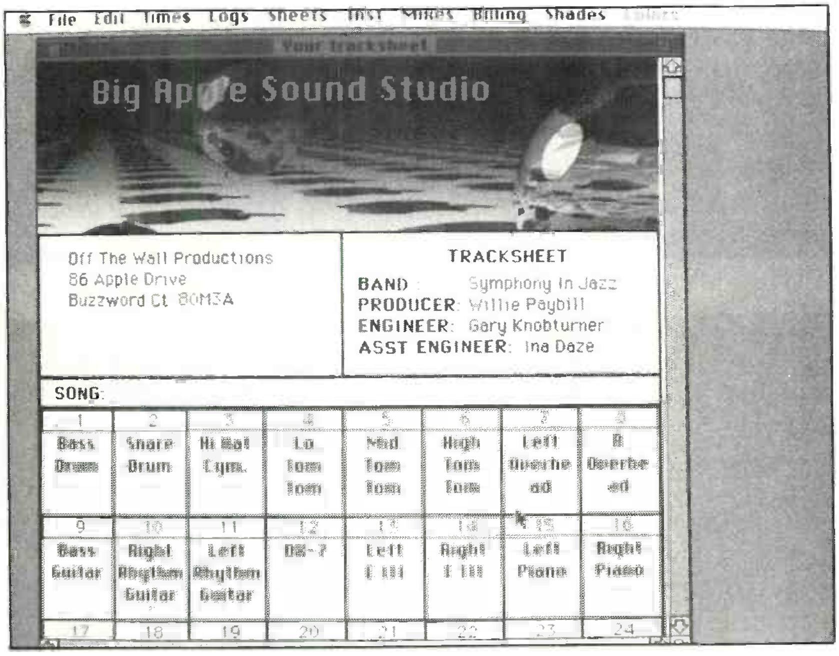
No matter what you decide to buy there are two important points to remember: identify your needs and the new products required to meet those needs, and research the options available. Find out what is being used on your favorite records, past and present. Read the trade publications to find out where a film or commercial was done, how it was made and what equipment was used in the process.

Good luck, and make good sound.

RE/P

Studio Master Plus

By Dave Barton



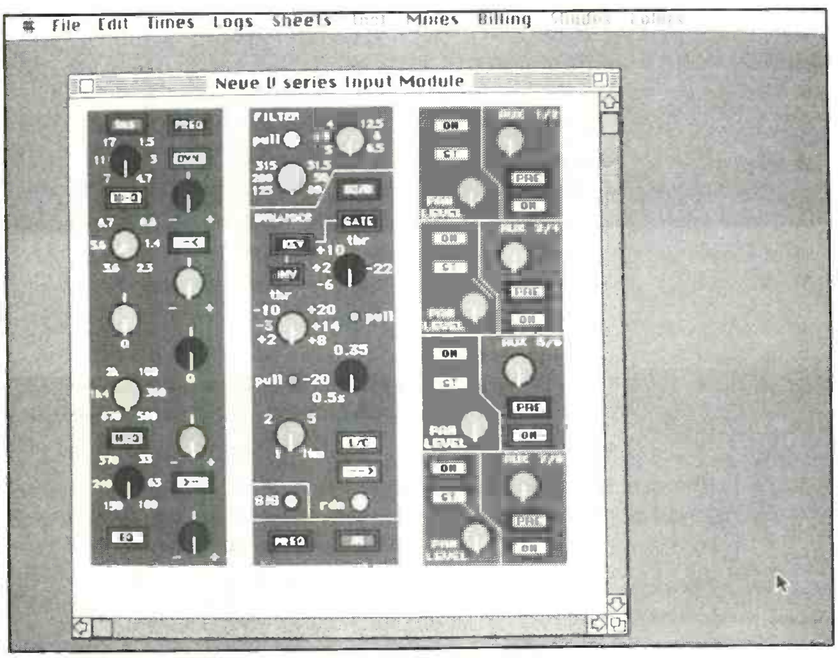
Sample track sheet from the Track Master application program. A standard form is included, or a custom form can be created.

Studio Master Plus is a system designed primarily to aid in logging and resetting a console to a particular mix. It uses both software and hardware to integrate a Macintosh computer with an already existing console using the console's audio inputs and outputs.

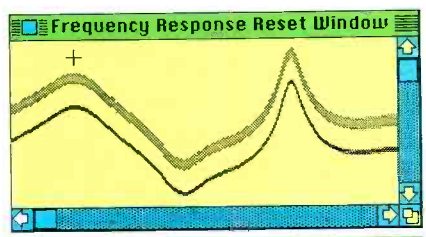
Programs

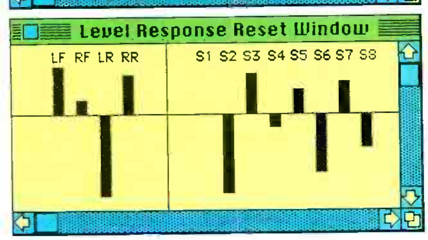
But Studio Master Plus is not just for logging consoles. Included are programs for automatic billing, creating track sheets and creating graphic representations of outboard gear (again, to facilitate logging or resetting the console).

At our studio, we have been using the system for about a year and a half, primarily for logging mixes. A processor unit interfaces between the Macintosh and the console. The processor sends a sweeping tone, from a low frequency to a high frequency, at a constant output level, to an audio input in the console, preferably an oscillator input that is assigned to every channel on the console. Also connected



Sample of a Neve V series input module created by the Outboard Master application program.





Screen dump of the frequency response reset window. The gray line represents the frequency response that was previously recorded for the channel. The black line represents the current frequency. Adjusting the EQ until the black line overlays the gray line resets the EQ.

Dave Barton is chief engineer at New River Studios, Fort Lauderdale, FL.

to the processor is the console's main buses (LF, RF, LB, RB) and the sends.

To log the console, mute all channels and start the program. The Macintosh will prompt you to mute or unmute the channels you want to store. This process takes about 15 to 20 minutes on a 56-input console.

Logging the console

When logging the console, the user does not have to "work" the computer, such as typing in commands for it to perform. The operator merely mutes and unmutes channels as the computer screen tells him to. Resetting the console, though, takes more interaction between the user and the computer. The screen needs to be fairly close to the user in order to see it clearly and to have access to the mouse.

Resetting the console is much like logging it, except that when you turn on a particular channel, it will show you two graphs. One is a frequency graph of the way the channel is supposed to be, shown as a wide gray band. The other, a thin black line, is a frequency graph of the way the channel is set at the moment.

The black line is updated by the computer so that as you make changes to the equalizer section, the line changes accordingly. The EQ section is set once the side gray line and black line have the same curve. Use the mouse to advance to the next page, which will adjust send and master bus levels.

This screen will look like bar graphs coming from a horizontal line through the middle of the screen. A bar going up from the line indicates the level is too loud and must come down to where the center line is. Likewise, a bar below the line indicates the level must be increased. When all the levels are at the rest values, there will be just a horizontal line across the middle of the screen, and the computer will automatically advance to the next channel.

One problem with plotting the equalizer this way is that there is an overlap in the frequencies of a parametric EQ, especially when shelving and filters are used. This can lead to resetting the channel's EQ to the correct frequency response but using different knobs positions to achieve the same results. Most engineers, when listening to their reset mix, will complain that certain elements don't sound right because the EQ doesn't look like the way the engineer set it.

Knob position

This is a viable complaint because they may of had a reason to have the knob in that particular position. To correct this, we decided to use the outboard gear drawing program to make up a graph of the console's EQ. Then, after printing it out, we could make a quick sketch of the knob positions as we were logging the console.

Using these sheets would help us set the knobs exactly as the engineer had put them. It also speeded up the process of resetting the console by quickly allowing the user to roughly set the EQ knobs to where they are supposed to be, then using Studio Master to fine-tune the knob positions.

I understand that newer versions of Studio Master will have a feature similar to this. First, you will be able to design a diagram of your console EQ and insert it in the logging and resetting process. Then, as you log each channel, you can use the Macintosh to draw the knob settings on the diagram. When resetting the console, the diagram for each channel comes up on the screen with the positions for the EQ for that channel.

This allows you to visually set the EQ quickly, then tap the mouse button to see the frequency response of that channel to fine-tune the position. This is a major ad-

An Additional Viewpoint

By Stefan Ingmann

For two years, we have been using Studio Master Plus with a Macintosh Plus. From the beginning, we were delighted by the practicality of the working surface. All of the console's outputs are connected via multicore cable parallel to the computer interface. Therefore, we do not need to patch the test signal and the patch cords at the output buses.

After a short period, we were able to make a "phantom" mix with the mix computer. This was then started by a time code generator instead of using the multitrack's time code. By stopping the Studio Master's scan time for a single channel, we were able to program the channel's mutings in advance. As the mix computer muted the channels, the Studio Master stores the adjustments, leaving us time to relax. We don't have to mute the channels or turn them on one at a time anymore.

Also, the Studio Master test signal is paralleled on the patchbays, allowing us to record the EQs and the levels of the peripheral instruments. Already, engineers who have regularly been working with the Studio Master are able to reset a channel in two to three minutes. Many have realized through the graphic representation of the EQ settings and the manual matching of the old settings what the EQ actually does.

One of the strong points of the Studio Master is that the application of Mac software is possible. Therefore, every owner can design his own input/output channels, even after a few hours work. Being able to use the Mac software was a great advantage for us. With the help of a Mac utility program, we were able to translate the entire text dialogue into German.

At this point, I'd like to mention how astonished we were by good service—backup through the manufacturer, especially considering the large distance. The company is open for all improvement suggestions. At the moment, we are discussing the possibility of making the Studio Master system work locked to the time code.

It would then be possible to load time code values of cue points in the track sheet or to load other events as time code values into the system, making it unnecessary to type endless numbers into the computer.

It would be nice to be able to jump from channel to channel when recalling the settings. It is only possible to run down a whole channel at one time without the opportunity of resetting a single knob on a different channel. This means you are stuck with one channel until it is set, wasting a lot of time.

For example, if you want to set one parameter of a setting, you have to go through the whole recall menu and cannot just look at the pan pots positions from channel 1 to 56. The number of channels to be scanned is almost without boundaries because it is dependent on the software.

Either in the morning or in the evening, I let the system print out the reports and directly have a complete overview of time worked, material used up and service reports. Within seconds, the happy client can hold a personal, freshly printed invoice. Another advantage for us engineers is that we can use the same computer for sound design and sequencer programming and of course, for the recall system. Instead, we don't have another computer in the control room, only a little program on our hard disk that has left every client gasping for breath.

Stefan Ingmann is an engineer with Dusseldorf Film and Video, Dusseldorf, West Germany.

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AMS Reverberation Was Added At:   3:30 PM   Mon, Sept 23, 1986
Started 2 Track To Cassette At:   4:52 PM   Mon, Sept 23, 1986
4  Audio Cassettes Were Added At:   5:31 PM   Mon, Sept 23, 1986
The Session  Stopped At:   5:32 PM   Mon, Sept 23, 1986
Carlson Twins   A Bill Was Saved At: 5:35   Mon, Sept 23, 1986
The Session Was Closed At: 5:37 PM   Mon, Sept 23, 1986

The Session Was Opened At:   5:38 PM   Mon, Sept 23, 1986
Started 48 Track Recording At:   6:03 PM   Mon, Sept 23, 1986
2   2 Inch Tape 2500 Ft. Were Added At:   6:03 PM   Mon, Sept 23, 1986
Reel # 1   A Tape Log Was Saved At:   7:26 PM   Mon, Sept 23, 1986
Started 24 Track Recording At:   9:26 PM   Mon, Sept 23, 1986
Reel # 1   A Tape Log Was Saved At:   10:45 PM   Mon, Sept 23, 1986
2   2 Inch X 10.5 Empty Reels Were Added At:   10:48 PM   Mon, Sept 23, 1986
1  Long Distance Call   An Extra Was Added:   10:53 PM   Mon, Sept 23, 1986
The Session  Stopped At:   11:07 PM   Mon, Sept 23, 1986
Bad Boys   A Bill Was Saved At: 11:13   Mon, Sept 23, 1986
The Session Was Closed At: 11:32 PM   Mon, Sept 23, 1986

The Session Was Opened At:   10:15 AM   Tue, Sept 24, 1986
Started 24 Track Transfer At:   10:26 AM   Tue, Sept 24, 1986
Down Time Strarted At:   10:41 AM   Tue, Sept 24, 1986
Started 24 Track Transfer At:   10:52 AM   Tue, Sept 24, 1986
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Screen dump of the captain's log, showing various events during a session.

Specifications

System requirements: *Apple Macintosh or Macintosh Plus computer; Studio Master Plus software; Studio Master Plus mixing console interface.*
Power input: *110V; 220V on special order.*
Connectors: *30-pin Tuchel—Amphenol DIN connector. All connectors included.*
Mixing console input channels: *Almost unlimited; dependent on available disk storage space.*
Output: *Test signal nominal level, +4dBm; output impedance 600 balanced; attaches to consoles OSC/Slate bus or can be patched into each line input one at a time.*
Inputs: *16 standard; maximum level +28dBV; input impedance, 20k balanced; attaches to consoles mixing outputs, echo sends, effects sends, etc.*
Relays: *8 standard; contact current rating, 0.5A.*
Level response: *Logarithmic.*
Level resolution: *0.1dB.*
Frequency response: *Logarithmic or linear.*
Maximum number of frequency readings: *500 logarithmic and 500 linear.*
Number of measurements taken per channel: *1,016.*
Computer interface: *RS-422; SCSI on special order.*
Dimensions: *14x16x7.*
Applications programs: *Studio Master: billing system, maintenance log, captain's log, session log. Track Master: track sheets, tape labels, word processing. Outboard Master: logs outboard gear settings.*

vantage in both speed and accuracy in resetting the console.

Other studio uses

Along with the main function of storing console settings, Studio Master also handles some of the paperwork and business needs of studios. There is a track sheet section that allows you to quickly create track sheets from the Macintosh. As you are recording, click on a track with the mouse, then call up from a selection of instruments or type in your own to denote that particular instrument on that track.

At the end of the session, the track sheet can be printed out and be put with the multitrack tape. You can even use MacPaint to create a studio logo or design that would be put on the track sheet.

On the business side, Studio Master will do automatic billing for the studio. Pull down "24 TRK" under the TIMES menu and Studio Master will automatically start billing for 24 track at that time, at rates designated by the studio owners or manager. At the end of the session, a billing sheet or work order can be printed out with the times and costs of using the studio.

At New River Studios, we primarily use Studio Master to log console settings after mixes. Producers are much happier knowing they can get back at a later date the same mix that they are leaving with. With very little effort, a producer can make a song sound the way he wants it without spending another 12 hours to mix from scratch. From a studio standpoint, it takes much less effort and is more accurate to log and reset the console using Studio Master than to write down all the settings by hand.

We do not usually use the track sheets for the automatic billing because we are already set in our ways in billing clients and we already have plenty of track sheets printed. These are big advantages, however, for new studios or older studios wanting to upgrade their method of operations. For any studio, Studio Master can be a real asset. RE/P

Engineer Interview: Steve Davis

By Rick Shaw

An engineer at one of the premier audio post facilities talks about the many facets of putting sound to picture.

Like many of us, Steve Davis is an engineer who has worked his way through a number of audio positions. His present position at Crawford Post Production in Atlanta puts him in a world-class facility geared toward the audio for video post-production market.

Crawford Post is one of the more advanced video post houses in the country. A quick look at the rate card outlines some of the most coveted video production and digital animation equipment available anywhere. Spurred by multi-million dollar accounts like Coca Cola, Crawford's sophistication has grown rapidly during the past three years.

Because of this rapid growth, new opportunities for audio production and development are always underway. This is one engineer's point of view on how to handle the problems and creative decisions that come with the art of putting sound to picture.

RE/P: Tell me how you started out, and why you came to Crawford.

SD: From a hardware and facilities standpoint, Crawford sprung up almost instantly as an up-and-running facility. After the first year, Crawford decided it wanted to put a real emphasis on the audio department. They hired Tom Race and myself. Tom is a premier music engineer in Atlanta; I'm known as a production specialist.

RE/P: What type of work would you say is your mainstay?

SD: Right now, I would have to split it down the middle between video post-production work and industrial music recording, jingles and film scores, big industrial scores.

RE/P: Do you use the Synclavier for that or do you have someone who specializes in that area?

SD: We have a couple of people on staff who are really savvy with the Synclavier. We've only had it about a half a year. It's a very deep system, and as time goes by, everybody is getting familiar with it. We've let a couple of people, Tom Race and Gary Crawford, specialize on it so they can get up and running faster.

The Synclavier is like a little career all itself. It's a closed system, it's not like anyone else's system; the keystroke commands and the terminal commands don't really have much to do with anybody else's system as far as programming goes. It seems like almost every music session that we do, the Synclavier is involved. In at least half of our sessions, the Synclavier is used in the basic programming of the rhythm tracks.

RE/P: So do you use it to lay down most of the tracks or do you beef it up with live instrumentalists on your sessions?

SD: What tends to happen is that the writer, because he or she has the structure in mind, will write out a chart that contains the basic rhythm structure of the piece in terms of chords and wherever there are kicks, repeats and so forth. They will fill it out and put in their own creative input and add the embellishments.

RE/P: Do you tend to use the digital drums on the Synclavier?

SD: Yes, constantly. These days, the Synclavier tends to take the place of the rhythm chart. You can program it, so you have access to the feel of a piece of music instead of just working with it on an intellectual level. If you see things that are going wrong or things that need to be changed, you can make those alterations because it's all software.

RE/P: I know that you have expanded your video equipment, and when I saw the addition of the Synclavier, I wondered if you were planning on adding the Direct to Disk.

SD: You have to take a long look at that right now. Because of the format incompatibilities between the PD and the DASH system, we were very squeamish about committing to one or the other and dumping $150,000 to $200,000 into a digital audio tape recorder. We had a wait-and-see attitude, and now that has turned into an "Oh my gosh, here comes tapeless digital multitrack."

RE/P: Have you ever had a compatibility problem getting digital material in from the other studios that you need to synchronize to video?

SD: We've done digital sessions here and our console, a Neve 8128, is harnessed for it. In fact, we can be harnessed to a 24-track analog and a 32-track digital at the same time.

RE/P: What is the input configuration on it?

SD: It's 40 inputs, with 48 motorized NECAM faders on it. The extra eight are return modules that can be anything you want; you can patch reverb returns, tape machine returns or most anything else.

RE/P: Are you using NECAM II or 96?

SD: We recently installed NECAM 96 and we are just thrilled. It's not just the NECAM 96 that I'm so proud of, it's a great system, but what we've managed to do is a lot of very deep engineering and digging into some of the more subtle aspects of Adams Smith synchronizers. Now we have a system where we have NECAM 96 talking to our Adams Smith synchronizer rack, which is controlled by a Boss computer editor that Alpha Audio makes.

RE/P: What do you think of Adams Smith's new visual editing system?

SD: The Boss system has that and originated the idea, but Adams Smith has elaborated on that concept. About two years ago, I was looking into more elaborate editing systems and that is what helped me make the decision to go with the Boss.

We have two Studio A80 24-track recorders and a Neve 8128, and you can do some serious music recording with a rig like that. At the same time, we're in a video post-production house and you

Rick Shaw is owner of Music & FX in Marietta, GA and a free-lance writer.

want to be able to turn around and do audio for video post sessions in the same room.

When you configure for those two different things, the reality in the past meant pulling a bunch of cables, plugging in other cables and loading new constants in the synchronizers. There was so much room for error in that process that when you try and go back and forth several times a day and configure for different things using various machines as slaves, it becomes a tremendous amount of cable pulling. It was very bad from a reliability standpoint. It was just awkward as far as session startup and changeover. It was creating a lot of downtime, and reliability was bad.

RE/P: And the synchronizer has to learn all of the machines again.

SD: Yes, and the synchronizers don't like being changed a lot. If you'll set them up and leave them alone and use them one way, they're pretty reliable. If you start throwing them a lot of curves and changing what you want them to do, they get confused and they don't behave very well.

We've been in a two-year development program of coming up with a system where you don't have to change any wires at all and you don't have to cripple one system or another to get the other ones to work. That's where the Boss comes in, and was why I was looking at these elaborate computer editors a couple of years ago.

One thing that they have that's wonderful is a database of various constant setups for synchronizers. One configuration might be a 1-inch (video) master with two 24-tracks following it and a center-track machine. Another might be a 24-track machine as a master, with just another 24-track machine chasing it and a center-track machine so you have a synched mixdown rig without picture.

RE/P: In that database, would it also know what type of machine, model, etc.?

SD: The database is multi-tiered. One tier of it is a giant list of machines and the correct constants for those machines. That's completely editable, it's not a ROM memory. You store it on floppy disk and it can be loaded into RAM memory anytime and be sent serially to the synchronizers. The next tier is the configuration of the machines you are using. You can have a group of configurations that are different machine setups that you want to use, and from that it looks deeper into its database and calls up the correct constants for those machines, sends those out to the various serial channels that you're using and puts the constants into the synchronizers. It's really neat. That's a wonderful setup, even if you're just doing audio for video post.

RE/P: It must be really easy to change masters, then.

SD: The combination of the Boss to hold all the various data, the modularity of the Adams Smith synchronizers—the ability to build a system that will do almost anything you want it to—and a very good engineer that we have here, Jim Wyle, enable us to really make the system sing.

RE/P: Does the NECAM 96 just need SMPTE coming into it in order to function?

SD: No, and therein lies one of the difficulties. NECAM 96 also does machine control, and anytime you have a system that does this, you get into this fighting between the automation system and the synchronizers.

RE/P: So does the NECAM 96 talk to the Boss and the Boss controls the machines?

SD: Actually, it's sort of a coexistence. NECAM 96 has an Adams Smith synchronizer of its own that it just uses to give transport commands—mostly like a serial interface device. Because the NECAM 96 is a computer and it wants to speak serial language, and tape recorders obviously want to speak parallel, it's an interface. So we have a single physical hookup to the machines.

RE/P: I know that some post houses have gone as far as to buy a dedicated CMX editor just for the audio position.

SD: Well, you cannot beat a computer editor if you're dealing with time code and synchronization of SMPTE coded tape to video. That's why we have three Boss systems. One is in Studio A, another in Studio B and one ready for installation in our new Studio C, which is under construction. We have a Trident series 70 console in our B studio, and we're thinking of taking it off-line as a Synclavier board and buying an automated console for our B studio.

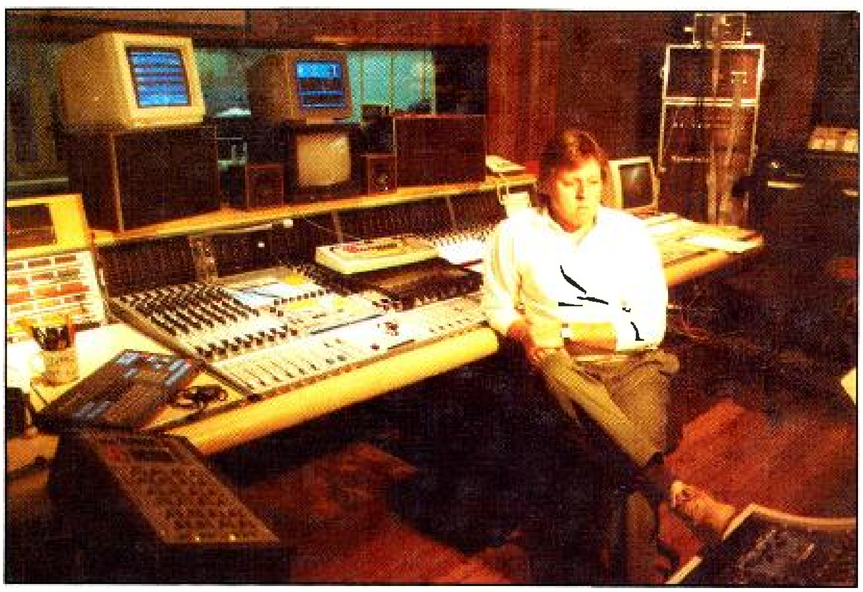
RE/P: Have you thought about MIDI automation for Studio B?

SD: Well, because we're a video house, I'm a believer in SMPTE code automation systems and not going through the MIDI translation. I've got to confess that I don't have very much direct experience with MIDI automation systems, so I can't in all fairness criticize them. I do have a certain distrust of them because I know they're going through an additional translation—from time code to MIDI clock, to MIDI data—and my experience with musical instruments is the time lag you run into.

RE/P: The only reason I would bring up this point is for the guys who may not have the budget to do what you're doing, and especially for a B studio where you want to save that client money but you want to provide some kind of automation system.

SD: Here's more of my point of view. I do want to save the client money, and a B studio is definitely a consideration. But there are many used consoles out on the market that from a sonic standpoint and a signal routing standpoint are very fully realized systems. They are just a wonderful buy, and you're then offering your client a higher level of sonic performance.

At a facility like Crawford, even in a B studio, there is a certain level of expecta-



Steve Davis sitting at the Studio A board at Crawford Post Production, Atlanta.

tion that the clients have. If you're a producer, you can choose the tools you want to work with and tailor your work to fit those tools, but when you're a facility, you've always got that phenomenon of people walking in and asking for things and you don't ever want to say, "Golly, we just can't do that."

Sometimes, that will happen, but as much as possible you want to be prepared for anything people can walk in and want to do. It's the kiss of death for a facility to say, "We just don't do that—so and so down the road does that."

Rather than going to a low-end, brand-new system using technology that's not fully compatible with my existing technology, my preference is to stick with high-level, strictly professional formats and buy good values in used gear. In our audio C room, we bought a late-model MCI-Sony 600 series console, which from a signal routing standpoint, is a terrific console, and it's got a tape-based automation system, full VCAs, 26 inputs, and it was cost-effective.

RE/P: There are a lot of them working out there.

SD: They're sort of an industry standard. This way, I can keep the quality up but still be competitive in price. But I absolutely agree for the guys who are very creative people and in the past didn't have access to any kind of professional tools. Now, because of MIDI technology, these guys can take a handful of sequencers, synchronizers and a small-format tape recorder, and they can produce terrific sounding stuff that's really fully realized, and then can do it on a budget that they can afford. It's creating a real revolution in the industry. It's something that facilities like this have to take a long, hard look at—that is, how to address that kind of thing.

RE/P: Do you have sessions where a guy brings in his MIDI system and does a score for a commercial?

SD: Absolutely. The whole gamut of that sort of thing, from guys bringing in their MIDI rig and taking it right to tape, or like other writers here in town who will go home and program what they want to do on a Macintosh, bring that in and send MIDI into to the Synclavier and use Synclavier sounds.

They'll use their own analog or DX-7 sounds at home to preview what they're writing as far as musical concept, but then come in here and substitute the digitally recorded acoustic sounds of the Synclavier for those voices.

RE/P: How do they know what sounds you have?

SD: We're very good about working together with people as far as talking to them to have a good free flow of information, because it can only help everybody. As with any sampled sounds, there is a certain technique that goes along with it. When you write for a real player, it's a given that the guy can do a pizzicato just the way you want it to be, or you have him play more legato—just a couple of verbal commands and you've got just what you want.

With a sampled sound, just because you've got the timbre of a violin doesn't mean you necessarily have the technique aspects of it. You've got to bear that sort of thing in mind, and if you're thinking the technique, you need to make sure those sampled techniques are available and that you're satisfied with how they work.

Another thing to watch out for is MIDI velocity. Something like a DX-7 that has MIDI velocity sensitivity, you can work out a sequence to your satisfaction. When you have a sampled keyboard timbre, a lot of times you'll have one set of patches that are for the instrument played *piano* but when you go to a *fortissimo* sound, you actually switch to a different sample that has that instrument being struck harder or whatever is appropriate for the type of instrument you are using. The way that's dealt with is through MIDI keyboard pressure that creates a changeover signal to the other sampled sound, and that may or may not be where you stuck your keys when you were writing your sequence.

RE/P: Of course, you can go into the Synclavier and change the values on the sensitivity of the aftertouch and so forth.

SD: Well, what we've been talking about is taking a MIDI sequencer and driving the voices on the Synclavier, so in this case you would need to go into your MIDI sequencer and make that edit so that the MIDI signals set to the Synclavier is what the Synclavier needs to see.

RE/P: I know you have done some heavy productions for Coke. Do you have any specific projects that have been challenging and stretched your imagination of what you wanted to do?

SD: That's a nice thing about being in a large post facility because you get things thrown at you that people wouldn't go into a normal recording studio and ask for. My favorite one in terms of audio is one that I did for KTLA channel 5 in Los Angeles. They did a major gonzo sales presentation to keep their local advertising clients excited about their programming. It was a big budget for a non-broadcast thing.

This particular presentation was done by Mike Cooper, a producer from Tribune Broadcasting, which owns KTLA. He's a great writer and editor and is not afraid of new techniques. He edited this thing on the Montage editing off-line system. You take all of your video source reels and you dump the video with time code onto a bunch of Beta cassettes and then there is a pretty slick computer that knows where all that stuff is. It enables you to instantly go to any one and play around with the edits. Only when you decide to keep an edit does it put it on the edit decision list. But the beauty of it is you can do non-sequential editing because you're not actually committing things to tape. You can

Some of the machines for Studio A.

do the end of the program first and the beginning of it last, and you can go back and change things all you want, at any time.

RE/P: It's like a word processor editing system.

SD: Exactly. That's probably the best analogy. It was a stereo presentation, and Mike wanted to load the thing up with effects, music, narration and bites from various programming that they wanted to use. The editing style of the thing was a very fast fluid kind of flow, where things were coming at you—it was very exciting.

In stereo, to deal with that in the video format, you've got a big problem because a 1-inch type C video machine only has two tracks. And if you're doing something with a very elaborate audio design, with A/B roll sound effects and bites, whatever, is to bounce from machine to machine. There is no possible way to do that and come out with a good mix, not to mention good fidelity.

What we did was take stereo center track (center stripe) ¼-inch tape and built up our various source reels. We took those and by locking them to house synch, we were able to input those directly into the Montage so that the Montage machine could deal with those as source reels. Mike had any one of those audio source reels he wanted to deal with at any time available in the Montage.

RE/P: So you have all your reels, and you tell the Montage this is reel number such and such, and it knows which deck to start recording automatically as you're dubbing it over onto the Beta decks?

SD: Yes. It totally manages its 17 Beta decks and its reel position on any of those decks. The Montage operator doesn't deal with tape transport control, it strictly talks in terms of the editing system.

It says, "This input is going in bin so and so and it's clip so and so," and it allows you to give it a description. It keeps track of where the time code beginning and end of that clip are. It's in film editing terms, bins and clips, just the way a film editor would literally have a stack of film bins with hooks on them and clips hanging from those hooks—that's the nomenclature that the Montage uses but it's doing it electronically.

RE/P: Have you found the DVR-1000 [Sony digital video/audio recorder] as useful for mastering your audio on certain projects?

SD: That deck is in such demand for video mastering purposes that it hasn't been practical for us.

RE/P: I'm referring to a final commercial master. Say you're doing a piece of animation with a snazzy audio track. How would you put it all on the DVR to give the client a digital master to dub from in the future?

SD: My experience has been that it has been used as a rendering medium when you're going right from a Quantel Harry or a Wavefront or a Mirage.

RE/P: Some type of digital video system?

SD: Right, when you're coming out of a digital device, you go to the DVR-1000 and it becomes the component master. But the final master still ends up being put together with supers, slates so that it's final release form still ends up on 1-inch C format.

RE/P: So you are really using it to create the main source reel before you put switched effects and other things you might want to do.

SD: Exactly. So at least 90% of it was done in digital, rather than final release materials being in the digital format.

RE/P

Facility Spotlight:

Spectrum Studios

By David Mathew

The process of constructing a four-studio-plus recording complex at Portland's Spectrum Studios.

Just about every studio begins with the distant dream of greatness. Owners of the smallest facilities have thought about which wall that first platinum record might hang on . . . someday.

Neves, SSLs, Studers and digital devices of all types are installed in many more daydreams than control rooms. Legendary recording stars grace the lounges and compliment the acoustics and accoutrements as they make plans to return again for hit after hit . . .

But success is measured in many ways, and producing platinum records is only one scale. Fourteen years ago, when Spectrum Studios incorporated in Portland, OR, founders Michael Carter and Lindsey McGill decided they would measure their operation against the high standards of quality, and support their dreams on a foundation of solid business practices.

David Mathew is an engineer and free-lance writer in Portland, OR.

Carter McGill targeted the advertising dollar. It was a slice of the pie they knew well—a wedge Portland could support. Spectrum's first studios were built with this in mind. They designed and constructed the rooms themselves with a total budget of $250,000.

Spectrum Studios

In 1973, Tom Hidley's rooms were on the magazine covers, not Michael Carter's. But Spectrum Studios served its market well. In the right hands, great things could be coaxed out of the interesting little consoles from a soon-defunct manufacturer. The trusty Scullys spooled millions of miles of tape while the slots on the splicing blocks grew wide and ragged. A third room was added to the first two, new consoles were built, tape recorders, reverbs, limiters and even engineers were added. And in a decade, Spectrum broke the $1-million-per-year mark.

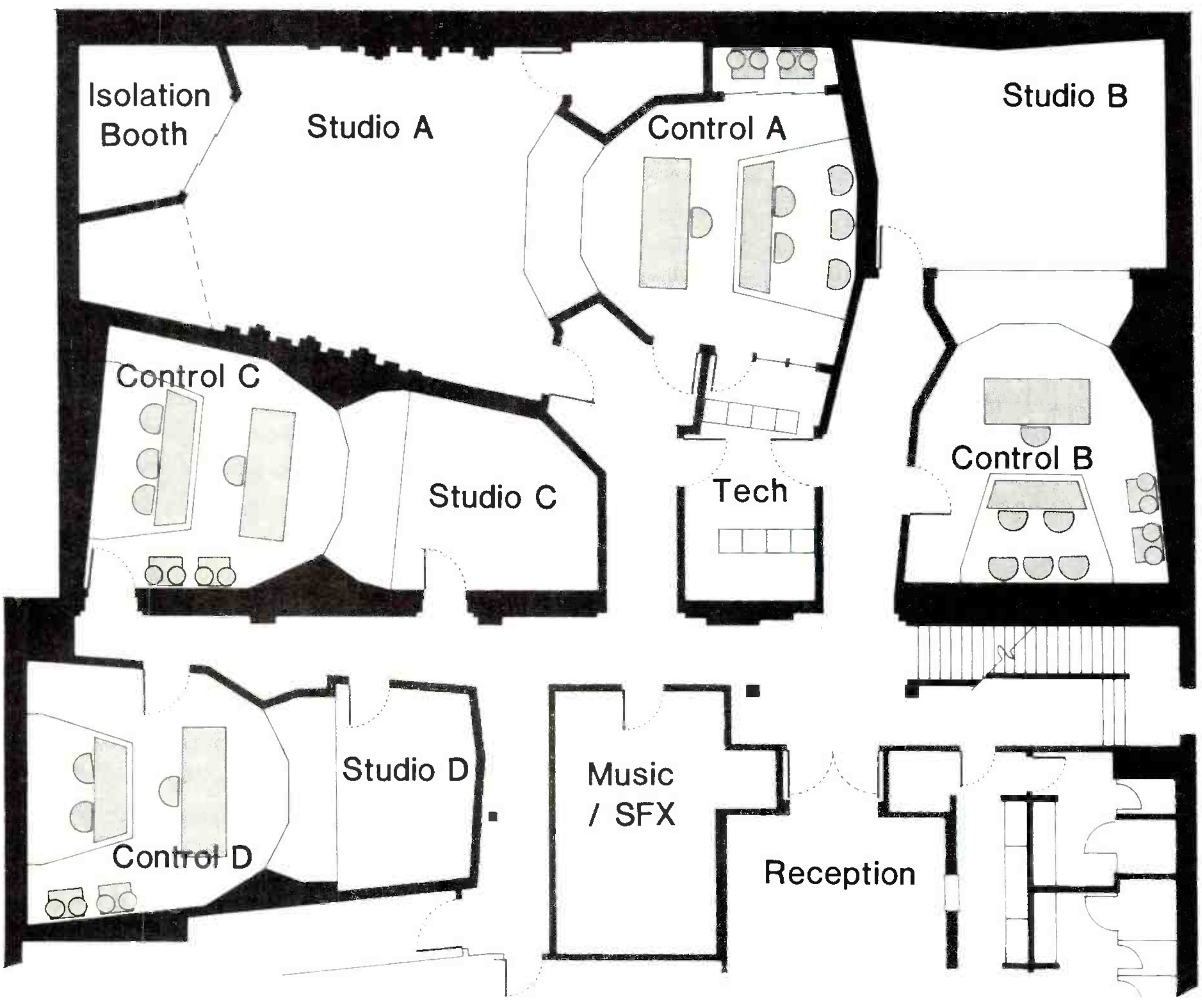
In many ways Spectrum was, and is, an unusual facility. Although many albums, singles, scores and jingles were recorded there, music recording was always the softest part of the business. Spectrum was known for top-quality media work: radio spots, supported by a large duplication system; sound tracks for television spots; film sound, both on location and 16mm dub-down; and audio visual and multi-image tracks.

Supporting all this was one of the largest music and effects libraries on the West Coast, a free voice casting service, and an excellent technical design and maintenance staff. Awards covered the walls and ran down the halls.

McGill had helped enough agencies hook up their playback systems to realize there was a market in this. Friendly advice grew into a large systems design division—creating electronic board rooms throughout the West.

Carter and McGill had planned to move

Floor plan of Spectrum Studios.



to a new facility within five years of incorporation. Five years became 10, then 11, then 12. Spectrum found itself in the position of turning away business on a regular basis; there was simply not enough studio time available.

The decision to expand

The type of work Spectrum was doing was changing too. Video sweetening was replacing slide shows and 16mm sound dubbing. Portland artists were gaining record contracts, and Spectrum's rooms were losing music clients to better studios in Portland, Seattle or Los Angeles.

The need to move intensified. The staff searched sectors of the city for suitable space. They contacted and compared studio designers. They ordered console brochures and evaluated monitors. Michael Carter talked to bankers, developers and brokers. The slow, painful, expensive and exhilarating process had begun.

"We needed more space, and a long lease," Carter says. "We needed to have more rooms to book, and we had to look at finally getting a proper music studio."

A staff engineer researched studio designers and presented a report. By evaluating their published design, experience and philosophies, Carter and McGill narrowed the field down to three designers. The first two were well-known names, with offices in Southern California. The third was someone they had never heard of, Russ Berger of the Joiner-Rose Group in Dallas.

"Lindsey and I got on an airplane," Carter says, "and met and talked to each of those three guys, and looked at their different rooms.

"There was just no question; there was one clear choice for us to make, and that was Russ Berger. The reason was his designs. We saw rooms that we thought worked right, and he fit Spectrum. He would work with us in the way we wanted."

"They were looking for a space," Berger says, "that would put them in a position to compete with anyone along the West Coast, not something that was as good as anyone else but something that was superior—that would stand them apart—because they were not in a major market. The nature of their work dictated certain special requirements such as the tremendous volume of production work.

"Now, there are really three things you have to balance whenever you're doing a facility: user requirements, acoustic science and budget," Berger says.

With their broad palette of services, Spectrum's "user requirements" indicated a large, flexible facility. Of course, they wanted to both increase their capabilities and improve the technical quality of their work. And Carter and McGill decided it was essential that they remain in Portland's downtown area near the city's advertising agencies.

Design specifications

"Spectrum required extremely low noise floors," Berger explained. "They mandated that this facility be appropriate for their needs for the next 10 years, and be ready for whatever forms digital might take during that period of time.

"Then there were certain functional requirements that every studio has: where the engineer likes to sit, which side the equipment is on, where the clients sit, vision requirements, access and degress, privacy, studio tours, and on through a whole multitude of requirements, some of which we prompted them for and some they came up with themselves."

When it came to the acoustics, neither Berger, Carter or McGill would allow science to be shortchanged.

The design at Spectrum is in line with the trends of the last few years: spacious control rooms; dead in the front and live and diffuse in the rear; large producer/synthesizer desks behind the consoles; a separate tech room for the video recorders and much of the electronic equipment; a glass-doored machine room in Control Room A to keep the listening area quiet; and well-isolated relatively live but diffuse, studio acoustics.

"The budget," Berger continued, "is the third part of the equation. You can't have all three—user requirements, ultimate acoustic science and low budget.

"However, there are many ways of applying common building materials to get the best performance. How you do this depends on your situation. If your labor is expensive, then sometimes you'll want to take a more materials-intensive approach. If materials are extremely expensive, then the converse is true.

Berger added that the contractor in Portland did a particularly good job. He was interested and paid attention to detail. Because of this, common building materials could be used in their optimum configuration.

"Really, the thing that goes when you're trying to hold the budget down is exotic finishes. I think Ingrim Associates [the architecture and design company] did a really nice job with the limited budget they had for finishes. Although the finishes are rather Spartan, it's an attractive space."

Site selection

Russ Berger provided Spectrum with a list of site criteria specifying minimum square footage, clear ceiling heights, structural considerations and proximity to sources of environmental noise and vibration such as freeways, airports, fire stations and potholes.

Carter and McGill finally chose an existing building in downtown Portland. Although Berger had recommended against this site on the basis of its limited space,

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Studio dimensions and consoles

Control Room A: *25'x23'*
SSL 6056E 56x32 computerized console
Studio A: *38'x30' with isolation booth*

Control Room B: *21'x22'*
MCI JH-536 36x32 automated console
Studio B: *27'x19'*

Control Room C: *20.5'x22'*
Custom 16x16 console
Studio C: *17'x16'*

Control Room D: *20.5'x22'*
Custom 16x16 console
Studio D: *11.5'x17'*

Scoring Suite: *20.5'x22'*
Custom 16x16 console

the location and the terms made available by the seller proved impossible to ignore. The process had been continuing for much too long, and now the owners of the existing studio building were seriously talking about eviction and demolition.

"We spent about two years trying to select a site," Berger says. "We went through three schematic designs of different sites before the downtown location was finally selected. We brought our equipment up and did original site surveys, measuring noise and vibration to find out what the environmental impact was going to be. It was fairly heavy truck traffic, and street irregularities caused vibration problems in the building. There was an adjacent uphill climbing lane for trucks that caused direct impact on the roof and exposed walls."

A quieter site with a bit more room would have been less expensive, given the same performance goals. But for Spectrum, the downtown location was critical.

"There is no optimum," continued Berger, "there are always limitations, and because of the location, extreme isolation measures were necessary due to the limited space in which the five studio functions had to fit. It would have been nice to have some more space, but I don't feel the performance of the room was compromised. It did have excellent clear ceiling heights though, which helped immensely."

Construction begins

Demolition of the interior began in June 1986. However, not long into the process, the project sustained a major setback.

A workman allowed a beam he was removing to fall to the concrete floor below. The floor, which had been approved by the structural engineer, cracked open.

"We were planning on building floated floors on top of the existing concrete floor structure," Berger says. "When the beam dropped, it became apparent there had been an underground spring years ago that had washed out all the infill underneath this slab, which made a hollow cavern under there."

The failure of the floor cast doubt upon the walls as well. It eventually took six months, 35 14-inch diameter concrete piers sunk 30 feet into the earth, a thick new concrete subfloor, reinforcement of the walls and a complete steel girder infrasturcture to get back to zero.

The structural fix alone cost several times more than the entire start-up capital for the original studios 14 years ago. There was no doubt that all the non-essential items had to be dropped out of the design.

The vaulted reception area with the cupola? Gone. The beautiful office cabinetry? Later. Cut by cut the list was trimmed to correspond with the new reality.

"The beauty of it," Berger added, "is that it gave us an exceptional base slab to build on, which would have been much too expensive to consider unless the structural requirements had come into play."

Michael Carter by now found himself de facto project manager and out of studio production for the duration, coordinating the implementation of Russ Berger's designs.

Construction management

"Russ indicated from the start that we would not need studio experienced carpenters," Carter says. "Any competent general contractor and good workmen can build a studio. If they read the plans and do what the plans say for them to do, it'll come out fine.

"In our case we flew our project superintendent and our project foreman to Dallas. They saw facilities that were both under construction and completed, and met with Russ Berger and Richard Schrag, so they both had a much better idea of what they were going to be building. That helped a hell of a lot. Through the whole first layout and everything—things that were critical—they understood. They had seen what we were trying to do."

The new concrete base slab is eight to 10 inches thick. The floated studio and control room floors are each separate slabs of concrete, poured on the base slab on a sandwich of sand, plywood and plastic and isolated from the walls and from each other by 1-inch compressed fiberglass.

The 20-foot high studio walls, formed of grout-filled concrete block and reinforced with steel bar, sit free-standing on the base floor. The ceilings are acoustically isolated from the walls.

Between the control room and the studio, the walls stand 4½ feet apart. Mounted in each wall are the ¾-inch laminated glass studio windows. The control room monitors are located in this wall (or space, depending on your point of view).

An important aspect of Berger's studio design is proper monitor positioning and mounting. "There is an optimal range of elevation to place a monitor speaker," Berger says, "somewhere between 12° and 15° up above the median plane of the head. Then you find a compromise between that and a standing position, because many people like to work standing or on a stool. The lower you can get the speakers, as opposed to the antiquated stick-'em-on-the-ceiling, the less problem you have with reflections off the console. The console top really wreaks havoc with the image.

"Of course, if you throw console-top monitors up there you lose much of what you've gained, but at least those are removable. People use console-top monitors for a variety of reasons, but usually it's to get around room problems. If you have a good set of monitor speakers and a properly designed room, there's no reason why you can't use the mains. It certainly fills the room better, gives you a much broader bandwidth signal to work with and usually provides better listening conditions for everyone else in the room also."

Berger added, "Over the last six or seven years our experiments have confirmed that you need to mechanically isolate the speakers from the surrounding control room shell surfaces and the structure. We've also found that by mechanically tying the monitor speaker cabinetry into a massive structure rather than resiliently mounting it, low-frequency reproduction quality is dramatically improved.

"In fact, one of the biggest problems we find in remodeling jobs is that loose and flimsy panels (two or three layers of sheetrock can be considered loose at certain frequencies, depending on how they're mounted and supported) are usually the major problem with low frequency reproduction in the room. When a speaker is coupled directly to these flimsy surfaces, the whole low end lights up. You wind up with lots of bass in the room but very poor imaging and extremely irregular frequency response."

Berger added, "We've found we need to isolate the monitors from the structure, but still couple them to something massive to greatly improve the low-frequency response. We accomplish this by pedestal mounting directly on a massive structure, or by coupling them through a rigid, conductive support to a massive structure."

Spectrum Studios primary studio equipment

Otari MTR-90 II 24-track (2)
Ampex MM100 16-track Mitsubishi
X-86a 2-track digital
Otari MTR-12 CTTC 2-track
Ampex ATR-102 CTTC 2-track
Ampex ATR-104 4-track
Scully 280B 4-track (10)
Scully 280B 2-track (7)
Scully 280B 1-track
Nagra IV-STC CTTC 2-track
Nagra 4.2L 1-track
Dolby M24H (2)
Dolby 361 (4)
Dolby SR (2)
dbx 180 (7)
dbx 187 (3)
Audio Kinetics 3.10 Synchronizer
Sony BVH1100a Type C 1-inch VTR (w/Dolby Cards)
Sony VO-5600 3/4-inch VCR
Lexicon 224XL
EMT 140st
Lexicon PCM-70 (2)
Yamaha REV-7 (4)
Yamaha SPX-90 (2)
UREI 813C (2 pair)
UREI 811C (2 pair)
UREI 811 (1 pair)
UREI 6500A (2)
Yamaha PC2002 (6)
Yamaha PC1002 (4)

Control room and studio communication

One concern at Spectrum centered around the visual communication between the control room and the studio. Russ Berger's designs place the control room monitors low and connect the speakers very solidly below, often on a large base of concrete. In the examples Spectrum studied, this left a control room window much too narrow for comfortable viewing.

"We wrestled with it on the airplane," Carter explained. "The architect, Dan Ingrim, and I made drawings, and when we got to Dallas we told Russ that we had to have vision underneath those speakers. We worked on a give-and-take level with Russ and the result is something I think is contributory to the recording studios: the monitors are supported by a 3-inch steel pole, grout-filled. We had it chromed so it looks good. This pole is bolted to the slab—it's not connected to the floating floors. On top of that is a ¾-inch steel plate welded on to this rod—that floats in the middle of the speaker soffit. On that are three little chamfered rods on which the speaker sits."

"The speaker is completely free-standing," Berger added. "It's not bridged or rigidly connected to any of the surrounding walls or ceiling, yet it's tied firmly to the ground. The support itself is mounted on an isolated section of base slab, below the floated floors. The monitors are then sealed into place with a fabricated gusset, using a sound-rated barium-loaded vinyl flap.

"Another important advantage of this mounting is the flexibility it offers. The monitor speakers can be changed out in about 20 minutes, flushed into the wall with no construction required at all. If you have a client who comes in and requires a special type of speaker, you can supply him with that. And in two or three years when the current monitor is now an outcast, you can replace them."

Once the thick windows and massive walls are in place, you might think the isolation would be pretty good. But until the door holes are plugged, there is no isolation. The isolation finally achieved is limited to the weakest link—the doors.

Berger usually specifies sound-rated doors. A sound-rated door is built with recessed seals, which are pulled tight when cams in the hinges drop the 400lbs door slightly as it meets the jamb. But because they cost more than $3,000 each, many studio owners decide to build their own doors.

"Spectrum's studios are among the quietest in the world. We spent about three and a half days up there testing, and the noise floors we found were typically below our threshold of measurement; well below NC 15."

Russ Berger

"People build up these ridiculous configurations," Berger says. "You know, double solid-core doors with lead sheeting and machine rubber and all this other stuff all screwed and glued together, and they pull out their beads and rattles and then they stick some scab-on door seals on there.

"They're limited by the seal, an STC [isolation rating] at a maximum of around 36, typically around 32, and that's as high as you can get. The doors that we used in Spectrum are up more than STC 50, and they give predictable results.

Spectrum asked their engineering staff for ideas throughout the entire design process. The question of video monitor mounting for the sweetening rooms was tossed back and forth.

"We looked at all the magazines," Carter says, "every studio photograph that we've ever seen, and we never saw anything that fit our criteria. We wanted a video monitor that was closer to us, not way back on the wall. When you're doing video sweetening, you want to be able to study that image. A projector provides a large picture, but the screen is between you and the talent and if you're doing ADR that's a problem."

Spectrum's staff designer Mark Gottwig provided the solution. He suggested mounting a video monitor on a steel shaft and hanging the monitor from the ceiling just past the meter bridge of the console.

"Mark came up with a 10- or 12-foot pole with the wiring inside it connected to a yoke, which holds the monitor," Carter says. "There is a remote-control electric winch, which raises and lowers the monitor."

The mixing rooms all use RPG Quadratic Residue Diffusers across the back wall to provide an even field of diffuse sound, a common feature in current control rooms. Berger's design for Studio A breaks new ground, though, in his use of large, low-frequency QRDs in the main music room.

"Spectrum requested a studio with a lively but neutral-sounding environment," he explained. "They wanted it free from coloration with an even, broadband decay. During the design of the studio and the diffusers we contacted Peter D'Antonio of RPG to consult with us.

"We implemented a step 7 series quadratic residue diffuser, a low-frequency diffuser. These things are approximately 12 feet high and about 20 feet long. They are field-fabricated and they run along the non-parallel opposite sides of the studio's longest walls.

"These diffusers, along with special shaping and acoustical finishes, provide an environment I think is really nice to listen in, play music in and record in. We've used large QRDs three or four times before in performing arts facilities and rehearsal rooms, but this is the first time that we've installed them in a studio."

"It was easy to adjust to the new studios," engineer Mike Moore says. "The first night we got the console and the power amps up, we spent the evening listening to CDs. That's all it took. The imaging is incredible. I've never heard a control room

Construction at Spectrum Studios included double walls of reinforced grout filled concrete block completely decoupled from the other wall.

Construction phase of Control Room B.

Installation of isolation material along low-frequency quadratic residue diffuser in Studio A.

that felt the same way.

"The part I enjoy is wandering around the control room while I mix. The sound is that even. There is no sweet spot; it's all sweet."

Jim Rogers, chief engineer, designed and installed the wiring interface. Six months before any solder was melted or connectors crimped, he began the documentation. Imaginary wires were numbered and assigned to imaginary conduit.

"Fleetwood Mac dropped in for a couple of days just before Christmas, overdubbing for an upcoming Westwood One program. It was the first time we've had artists as particular as they are record here, and it worked out great.

"John and Christine McVie say it was a nice change to work in a clean new facility, not just an older room fitted with new equipment. They really liked the feel of the whole place."

Mike Moore

When the new base slab was poured, he was there first with the PVC pipes and the stout UFER ground, a long copper bar imbedded in concrete sunk deep under the tech room.

We laid PVC highways from each control room to the tech room," Rogers says. "We have spare capacity in A and B to prewire an entire new console system, if need be, without taking the existing system down.

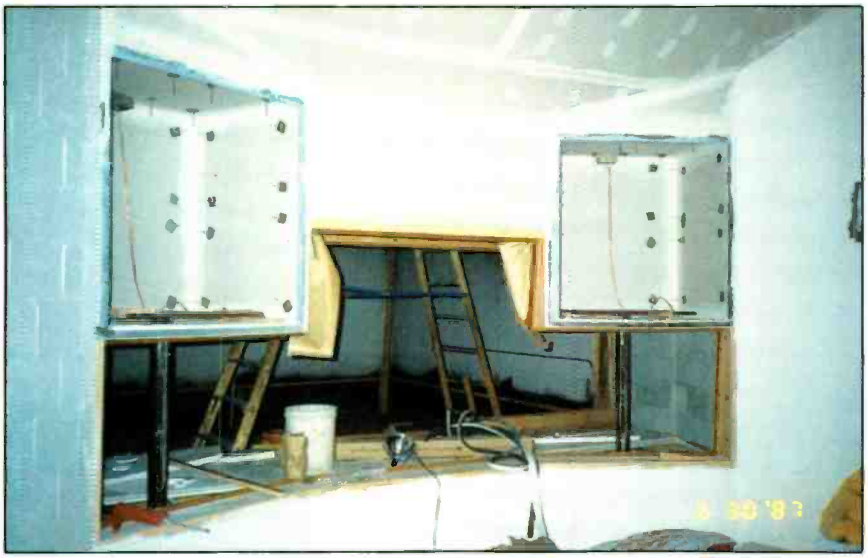
"There are video and audio patchbays in the tech room, so you can tie any control room or studio to one another. For example, we miced the piano in A, ran the lines clear back to D, and then ran the cue sends from the D console up to A, all while a mix was being done on the A console."

Grounding

All the grounding returns to the tech room. Hospital-grade isolated-ground outlets are used in the control rooms, with a star-grounding system providing separate ground returns for each outlet. A massive copper bus bar connects the grounds to the UFER shaft.

"When I hooked up all the shields and made the final ground connections from each of the control rooms," Rogers says, "all the mysterious buzzes went away. The noise floor dropped. It was like magic."

Rogers chose Gepco wire for all of the audio runs in the studio. The studios are all wired balanced, with pin 3 of the XLR-type connectors high, he explained. "Most of the gear we have here is pin 3 high. If something comes in wired pin 2 high, I modify it internally. We have maintained absolute polarity all through the chain, so



Front of Control Room B showing monitor pedestals under construction.

Control Room A features a SSL 6056E 56x32 computerized console.

Studio A features a Yamaha C3 grand piano.

a positive pressure impulse in the studio results in a positive pressure impulse from the monitors."

The mains power comes to the studios through large isolation transformers in the basement. Technical power, which was designed by Richard Schrag of the Joiner-Rose Group, is all on the same electrical phase.

Moore appreciates the foresight in the wiring. "We're wired for the future. There are six MIDI ties mounted at the producer's desk in A and B. They can be interconnected via the tech room.

"Frankly, I have yet to find a valid use for them," Moore says. "Most of the guys that come in have a self-contained system. But we're trying to stay ready for it. Just having the extra conduit and ties in the tech room will be handy if and when a new digital interface system is developed."

Studio equipment

Much of the equipment for the new studios came up the street from the old facility. The three custom consoles, the automated MCI JH-536, the 18 Scullys, the Ampex ATRs, the video system and the EMT plate were all carefully reinstalled in their new digs, along with racks of outboard gear and shelves of microphones. But there was still a lot of room to fill and new markets to address in these quiet new studios. The engineers knew they needed a top-flight console in Studio A. The great acoustics alone wouldn't sell the room.

"The income from the new capacity we have is in balance with the increased cost. In the first two months our recording work has gone up 34%. We're making a reasonable profit and have a good cash flow, right off the bat."

Michael Carter

"We narrowed it down to Calrec and Solid State Logic," Carter says. "We talked at length with both representatives and brought in sample channels from each, and set them up side by side so everybody on our staff could listen to them, push buttons and see which one they felt good about. We went nose-to-nose in our own place, just as thick as we could get it, to see what surfaced as our decision. We weighed the features and the sound and the reputation.

"All things were considered, not the least of which was the overall opinion of our staff of 10. It was all put together, and we ordered the SSL."

The final cost of construction, excluding new equipment, was $2,215,000. Was it worth it? Check back in five years for the full story. But the initial reports are good.

RE/P

Automation: The Purchase Decision

By Erika Lopez

Is your facility considering the purchase of an automation system? Here are some aspects to consider before buying.

The decision to automate your studio is not an easy one. It can be a large investment both financially and technically. Can your client base afford the increased rates? Do they really require the added flexibility that automation provides?

We all remember the early days of automation. A box attached to the console faders via dc control lines—delay problems from processing codes on tape and the inherent noise that went along with it. Automation systems have come a long way since then.

The advent of SMPTE-based and disk-stored computer automation systems essentially eliminated delays from track bouncing and greatly decreased errors, while also adding increased flexibility and saving a lot of wasted time.

Still, some studios are reluctant to cross over into automation. High costs are one of the major prohibiting factors in automating your studio. The MCA (MIDI Controlled Attenuator) automation systems can offer cost-effective solutions to automation. Systems such as the Twister, MidiMation and Iota MIDI Fader will remember a sequence of setups by taking "snapshots" and then recalling these setups with a manual command. These automation systems allow the user to decide what and where to automate (i.e., auxiliaries, group outputs or levels).

Automation systems

These MIDI-based automation systems are ideal in sound reinforcement applications. In a theater application, for example, scene changes can be memorized and level changes and mutes can be brought up on consecutive nights, thereby saving the operator the manual reset of levels required at every new song or set change.

The next step up from these basic automation systems is to integrate a computer. Manufacturers have brought user-friendly automation as close as your Apple Macintosh, IBM PC or Atari ST. Easy-to-understand video graphics and simplified controls further the ease of these automation systems when computerized. Based on MIDI time code, these systems are slower than SMPTE-based systems. However, they offer a relatively flexible automation system for under $20,000. Some, in fact, can be purchased in blocks of eight inputs for under $4,000 and upgraded as needed.

More complex automation systems include VCA (Voltage Controlled Amplifier) systems. I have found that many engineers and producers don't like working with VCA faders because they feel they degrade the sound. But manufacturers listened, and now VCAs can be bypassed—either with bypass circuits built into the console or with retrofit automation systems. This permits minimal use of VCAs, but allows the mutes to be controlled by the automation system.

When you begin working with moving fader automation systems, you no longer need to contend with VCA faders, as the faders are controlled by servo motors directly interfaced to the computer.

Additional benefits of moving fader systems include the ability to read changes instantly, and the visual indication of your overall mix levels when all the faders are moving.

To get a feel for the purchasing decision of a major automation system, we talked to Jay Antista, co-owner and chief maintenance engineer at Lion Share Recording Studio in Los Angeles. Jay takes us through a brief history of automation at Lion Share, and talks about the two new George Massenberg Labs moving fader automation systems they installed in November 1987.

"As more and more tracks are used for today's album productions, we're seeing two 24-track machines, a 24-track and a 32-track, or sometimes even two 32-track machines locked together. Two, even three people can't make all the moves needed to mix down that many tracks. It's just too complicated to make all the fader moves and do all the fine tuning needed," Antista said.

"Mixing with computer automation allows an engineer to work the mix in sections and fine tune each section to perfection. The computer can make the moves for you and memorize them for later work as needed.

"Another problem that arises when you mix from a 48-track machine is noise. Even with digital machines, you have A/D and D/A noise problems when you bring tracks back into the console. Before automation, it might take an engineer hours to erase breathing, coughing, foot shuffling, or whatever from the unused portions of tracks. But with computer automation you can mute tracks during the mix—saving a lot of time and affording a quieter mix," he said.

"Once you've gotten your tracks clean, you can go back after a session and polish up the mix, put the 'musicality' into it. Computer automation allows the old adage 'fix it in the mix' to actually become a reality."

Lion Share has always strived to be a mixing studio, so becoming an automated studio was an obvious choice.

"We bought a 56-input Neve with Necam 1 for Studio A in 1981. The Necam 1 was a good automation system, but it only provided two moves, faders and mutes. A lot of merges were necessary. You couldn't stop to mute a track and then carry on, you had to keep going and then go back for all these merges. We updated Studio A with Necam 2 in early 1987.

"In 1984 we put a 48-input Neve into Studio B with Necam 2 automation. This was a better system in that it allowed you to stop the machine, back up, put it into a suspend mode, make your changes, and then the computer would catch up when you carried on. It allowed for more freedom for the engineer to make mistakes, he could go back and correct it without having to merge at the end of the mixdown," he explained.

Erika Lopez is owner of Audient Marketing Services in Mission Hills, CA, specializing in the pro-audio industry, and a free-lance writer.

Upgrading automation systems

In 1987 the studio decided to upgrade again and added a George Massenburg system.

"We tried the Necam 96 system and liked it. It worked well and did all the things it was supposed to do. However, we felt the George Massenburg Labs system offered more options and room for future expansion. We talked to every engineer, artist and producer who came in and found a lot of acceptance for the GML system.

"We know that our newly installed systems will increase our business. We've already had calls from people who have worked at Lion Share in the past and gone elsewhere for GML automation, and are now ready to come back."

Eddie Ashworth, chief engineer at Total Access Recording in Redondo Beach, CA, also feels automation is an absolute necessity for their 24-track studio.

"Primarily due to today's complicated mixes, we feel automation is essential. It's a time-saving device, as well as a creative tool. We chose Audio Kinetics MasterMix Automation for our 52-input Amek G2520 console. We didn't feel the need for moving faders. The MasterMix system is very user-friendly and lives up to all expectations.

"The control offered to artist, engineer and producer during mixdown is great. Projects can be done faster and sound the way a client wants—because of the flexibility the automation affords—and with a lot less trial and error. We firmly believe it gives our studio a cutting edge to be automated. We can't imagine staying competitive without it," Ashworth said.

But is it right for everyone?

Not everyone agrees that automation is expedient for the job required. There are many engineers who feel that automation is a hindrance to the mixing process.

The importance of automation depends upon the type of projects your studio attracts. Most album-oriented facilities find automation to be a good investment for their studio. If your business caters more to a commercial-production clientele, perhaps automation isn't as important.

A good example of this would be 24-track mixing for broadcast ID packages. Here the engineer may have as many as 50 or 60 different mixes to complete in one day. Out of this 50 or 60, there will probably be about 10 different pieces of music, with each requiring several alternate mixes. It is not possible to accomplish this kind of mixing volume if the automation system has to provide new mutes and fader moves every few minutes throughout the entire day. Granted, these don't always come out as master album mixes, but an experienced commercial production mixer can come close without automation.

The flexibility automation provides is unquestionable. It lets the operator create a string of extremely complex events—play it back, work on it in sections, and then cut and paste sections of the mix as he likes. Experimentation with the mix becomes much easier. As a creative tool, automation is unparalleled, not unlike a word processor of sound.

The future is looking toward systems that can be purchased with a small number of automated faders and mutes, and can offer the expansion flexibility that many smaller studios need.

There is one point that everyone agrees upon: Automation is going to be with us from here on out, and the easiest systems to operate will generate the greatest demand. RE/P

Contents

Attention Studio Owners
Studio Update has been expanded, making it the industry's most complete source of studio news. But to make it work, we need your input:

—Talkback: Do you have a tip on how to run your business better? Send it to Talkback. If we print it, we'll pay you $50 and credit you and your studio.
—Studio News: the latest in equipment updates and personnel additions.

Mail to: Studio Update, RE/P, P.O. Box 12901, Overland Park, KS 66212.

Talkback

Using a Cellular Phone for Remote Communication

During the recent New York Marathon, ABC Television encountered a complex communication problem. A roving motorcycle cameraman and the network operations center needed to communicate. Shooting live, the cameraman needed interruptible fold back (IFB) to receive camera directions from the broadcast director.

A/T Scharff Rentals, New York, modified a cellular telephone, allowing the communication. Using a Mitsubishi Mesa 55, the camera operator called into ABC's dial-up IFB system. Supplied in an over-the-shoulder configuration, the phone could be carried on the motorcycle and operated hands-free.

A Telex earpiece allowed the operator to hear the camera directions through one ear, while concentrating on shooting. Two 10-hour battery packs and an antenna system designed to mount on the motorcycle were also supplied. According to ABC, the phone worked so well that the external antenna was not used.

Have you encountered a problem or unusual request during a recent job that required a unique solution? We would like to share it with the industry.

Send it to "Talkback"; if we use it, we'll pay you $50. "Talkback" is a new forum for sharing your solutions to difficult production situations other engineers may encounter. In a continuing effort to educate, we feel this type of information is helpful and will display your professional abilities. This is not a tech tips column; rather, the focus is on solutions to problems—technical or non-technical.

Each month RE/P will select one or more "Talkback" pieces for publication.

To submit: In 1-2 pages, describe the job, what the problem was and what you did to solve the problem. Include any supporting documentation, such as diagrams or photos, that would help explain the situation. If we publish your entry, you and your company will be fully credited.

Send material or inquiries to Michael Fay, Editor, RE/P, 8885 Rio San Diego Drive, #107, San Diego, CA 92108.

Spotlight

Planet Dallas

By Dan Torchia

Recently upgraded, Planet Dallas is continuing in its original goal of serving music production clients.

In a town where studios are known mainly for jingle production, Planet Dallas, which caters soley to local music production clients, is an exception. Although that decision occasionally causes financial difficulties—after expanding to 24 tracks, the studio endured a particularly slow period—the future as a full-fledged facility looks good.

Located in a 100-year-old house a few minutes from downtown Dallas, Planet Dallas was formed as a music production studio for local musicians and also features a music publishing division. Most projects are completed by Rick Rooney, studio manager, engineer and one of the studio's owners, and Patrick Keel, in-house producer. This enables the studio to provide a total production/publishing package for local musicians.

"I personally have always worked on music productions," Rooney says. "We love it, even though we have had some hard times."

Dan Torchia is staff editor of *RE/P*.

Studio background

Planet Dallas first opened in 1984 as an 8-track studio, with a goal of providing a professional, high-quality product at low cost. After three years as an 8-track, the studio decided to upgrade to 24-track. Renovation started in January 1987; by May, the studio reopened.

"The client base that we serve is in dire need of a facility that is affordable," Rooney says. "The larger studios have a price schedule that is pretty high, and we decided to offer a top-notch studio with rates that the local musician can afford."

Planet Dallas' advantage, he says, is that clients can work at a studio that is dedicated to music production. They can work when they want to, rather then having to wait until after midnight at larger studios. There's a sonic payoff as well as a financial one, Rooney says.

"We get a lot of comments from bands that say, 'Our products always sound squashed and tinny.' People come in wanting stuff remixed or totally recut because they worked with an engineer who works

At a glance

Studio dimensions: *40x45.*
Control room dimensions: *20x15.*
Owners: *Planet Dallas Inc. (James K. Devlin, CEO; Rick Rooney, president; Patrick Keel, vice president).*
Studio manager, engineer: *Rick Rooney.*
Staff producer: *Patrick Keel.*
Studio design: *Carl Yancher, Lakeside Associates, Los Angeles.*

Studio equipment
MCI-528B 28x24 automated console.
MCI JH-24 24-track tape machine.
Sony JH-110 2-track machines.
dbx noise reduction system.
Monitors: Lakeside Associates custom monitors with TAD components; Yamaha NS-10s, Auratones, Quadraflex.
Monitor amps: Yamaha PC-2002.
Lexicon Prime Time.
Delta Lab DL-4.
Yamaha REV-7s.
Lexicon PCM-60s.
Yamaha SPX-90s.
Master Room Lx-305 reverb.
Dynamite noise gates.
Brookes Siren compresser and de-esser.
UREI compressor limiters.

Available instruments

Yamaha Recording series drums.
Korg DDD1 drum machine.
Simmons SDS7 electronic drums.
Ensoniq ESQ and Mirage keyboards.
Casio CZ-101.
Other equipment/instruments available on the client's request.

Address: Box 215029, Dallas, TX 75221; 214-521-2216.

all day with jingles. I'm not degrading those guys, because that's an art in itself."

Along with Rooney's engineering experience, clients also are able to work with an experienced music producer in Patrick Keel. A former drummer, Keel began producing in Austin, TX, winning a variety of local music awards for his synth programming and production work.

The ability to offer additional services besides recording gives the studio an advantage, Keel says.

"We've started to develop a reputation as a great place to go to in the Southwest for talent," he says. "We're someone that the labels can regularly work with, which makes it easier for them."

Economic slowdown

As tough as it is to survive soley as a music production studio, the studio's situation was aggravated soon after the renovation was complete. Part of the client base went to other studios to do their projects, something that was expected. But the local studio community was also experiencing an extremely slow period, a result of the decline in oil prices that has affected the entire Texas economy.

"I got so concerned that I called every major studio in town and said, 'What is going on?' A couple of studios went out of business, and everybody said that business was the worst it's been in six to eight years."

The studio cut their rates from $50 and hour to $35, regular clients returned, and the studio is now "seeing light at the end of the tunnel."

Studio layout

Physically, the studio is divided into two rooms, each totaling about 400 square feet, which are acoustically isolated from each other but visually accessible by windows. The control room measures 20x15. Although the size may be off-putting to some potential clients wanting the "big room" with the "big sound," they usually are convinced that the smaller size is not a problem after hearing samples of previous productions, Rooney says.

"It would be nice to have the big reverberant room, but we deal with it, and it works," he says.

Carl Yancher of Lakeside Associates in Los Angeles designed the studio.

"When looking for designers, I called everyone I could think of," Rooney says. "Carl and his crew were the only people who called us back. There's not a room like it, and it sounds really nice."

Because of limited budgets, the studio had to be particularly careful when selecting equipment, particularly the console and tape machines. The studio looked for high-quality used equipment that did not have a lot of hours on it and would not deplete their budget. Most of the outboard equipment was carried over from the 8-track days, although they did purchase some new equipment.

The studio's console is an MCI-528B 28x24 console that already had automation. The console was 14 years old, and was upgraded with help of Bob Spalding from Studio Supply in Nashville, changing all the capacitors, the relays and the VCAs.

"The MCI is a world-class console in that it's got all the parameters on it that we would need," he says. "It's an in-line desk, so it's not gigantic. You don't have two different sections."

Rooney credits the studio's success to James K. Devlin, the studio's majority owner. Formerly a music publisher in Austin, Devlin was solidly behind the concept of serving the local musician and try-

ing to break them out, Rooney says. That sort of support is not available when you're financed by a bank.

"He obviously went far and beyond the call of duty," Rooney says. "That's why we were able to break out of the 8-track. There was a large need in the community, yet nobody was willing to take the chance.

"When you're a local musician, you cannot pay $100 an hour for studio time. Where are you going to go? You've got to go down to the guy who's got his stuff in his bedroom or you've got to do it live, and it's still going to sound slipshod. That's why we did what we did." RE/P

Studio News

Northeast

Chung King House of Metal (New York) has completed Studio B with the addition of a Neve V series console with 60 inputs and NECAM 96 automation. Other equipment adds include Studer A80 24- and 2-track recorders, Tannoy FSM monitors and Parraux amplifiers. *241 Centre St., New York, NY 10013; 212-219-8485.*

Shelton Leigh Palmer & Co. (New York) has named Carl Rosenberg as executive producer of special projects. He most recently was the musical producer for the 1987 National Junior Champion of Fox Television's "Star Search." *19 W. 36th St., New York, NY 10018; 212-714-1710.*

Power Play Studios (Long Island City, NY) has moved to a new location with acoustic and architectural design by Benchmark/Downtown Design. The facility features a 48-track SSL 4000 console, Sony multitrack and mastering recorders, and UREI monitors. Two additional recording suites and a mastering lab are planned within the next 18 months. *3812 30th St., Long Island City, NY 11101; 718-729-1780.*

Metropolis Music (New York) has installed a D&R Score Automation System in the facility's 8000 series console. *1650 Broadway, New York, NY 10019; 212-246-8420.*

Person to Person Productions (Litchfield, CT) has added Aimee Chiariello to its staff. She formerly was production manager at Masterdisk in New York. *Box 546, Cobble Court, Litchfield, CT 06759; 203-567-9012.*

Platinum Island Recording Studios (New York) has installed a Solid State Logic 4000 series console with 44 inputs in a 56-input frame and the new G series EQ. *676 Broadway, New York, NY 10012; 212-473-9497.*

Sigma Sound Studios (Philadelphia) has named Mary Ann Campo as director of marketing of media services. She previously was with Graco Manufacturing. *212 N. 12th St., Philadelphia, PA 19107; 215-561-3660.*

Great Immediately Recording (New York) has installed a WaveFrame AudioFrame digital workstation, said to be the first in the New York area. The workstation features 16 voices with 16 megabytes of RAM, a Compaq 385 computer interface and an internal sampling rate of 23MHz. *423 W. 22nd St., New York, NY 10011; 212-206-8561.*

Power Play Studios.

Chalet Sound (Manasquan, NJ) is a new facility featuring an Amek Angela console with Mastermix automation, Otari MTR-90II 24-track and MTR-12 2-track recorders, Meyer 833 monitors, and AMS and Lexicon signal processing. Also included is a MIDI production center and a Dolby XP24SR Spectral Recording system, said to be one of the first installations on the East Coast. *2323 Highway 34, Manasquan, NJ 08736; 201-223-0836.*

Gnome Productions (New York) has recently completed an audio-post-for-video upgrade and has installed the following equipment: a JVC 6650 ¾-inch VCR, Sony video monitor, and Fostex 4030 SMPTE synchronizer with remote control and 4010 SMPTE reader/generator. Gnome also has added the latest operating software revision for the NED Synclavier. *251 W. 30th St., New York, NY 10001; 212-594-7554.*

Sunset Productions (New York) is a new studio offering music, film, TV and advertising services. Equipment includes a Harrison 4832 automated console, Telefunken/Studer tape machines, more than 30 synthesizers, Sycologic MIDI Matrix, an IBM PC and Apple Macintosh II. Ben Tao is the owner; Mirisa Armstrong is the studio manager. *226 E. 54th St., New York, NY 10022; 212-832-8020.*

Milbrook Sound Studios (Milbrook, NY) has added an Otari MX-80 32-track tape machine with optional 24-track headstack.

Frankford Wayne Mastering (New York) has installed its fourth Sony PCM-1630/DMR-4000 digital mastering system with DTA-2000 for compact disc preparation. Other recent equipment adds include a Sony PCM-2500 R-DAT recorder and a Harmonia-Mundi digital transfer system. *1697 Broadway, New York, NY 10019; 212-582-5473.*

Southeast

Scene Three (Nashville) has promoted Nick Palladino to vice president. The studio has also added an Adams Smith 2600 audio editing system. *1813 Eighth Ave. South, Nashville, TN 37203; 615-385-2820.*

Memphis Sound Productions (Memphis, TN) has taken delivery of an Akai-Linn MPC-60 sequencer/drum sampler, using sound donated by percussionist Terry Elam. *315 Beale St., Memphis, TN 38103; 901-525-5500.*

Midwest

Brown & Brown Recording and Music Productions (Portage, MI) has added a Roland S-50 digital sampling keyboard, dbx de-essers and noise gates, a Deltalab Super Timeline and Sequential Circuits Digital Drums. *Box 224, Portage, MI 49081; 616-327-8352.*

Southern California

Track Record (Hollywood) has moved to a new facility in North Hollywood, featuring a 30'x40' tracking room, two iso rooms, a 20'x22' control room designed by Lakeside and two MIDI production studios. *5102 Vineland Ave., North Hollywood, CA 91601; 818-761-0511.*

Devonshire Studios (North Hollywood) has completed a major renovation of its studios, to provide audio-for-video post-production. Equipment upgrades include a Neve 56-input 8128 console with NECAM 96 automation; Sony and RCA 1-inch video machines; five Sony BVU-850 VCRs; five Adams-Smith 2600 synchronizers; and two Emulator II digital samplers, with Macintosh SE interfaces running Sound Designer and Q-Sheet software by Digidesign. *10729 Magnolia Blvd., North Hollywood, CA 91601.*

Record Plant Recording Studios (Los Angeles) has named Rose Mann as manager of the LA Record Plant. She is returning to the post after a year of travel and studio consulting. Dawn Roberts has been named director of Livingston Audio, Record Plant's rental company. *1032 N. Sycamore, Los Angeles, CA 90038; 213-653-0240.*

Northern California

Music Annex Studios (San Francisco) has upgraded its video synchronization facilities, allowing 3-machine lockups in both Studio One and Studio Two. Its Q-Lock syncrhonizer has been updated, and an MTM Mag recorder has been installed. *69 Greene St., San Francisco, CA 94111; 415-421-6622.*

England

Advision (London) has installed a 48-module version of the Harrison Series 10 console. *23 Gosfield St., London W1; 01-637 2758.* RE/P

Send studio news, including openings, equipment additions, renovations and personnel changes, to Studio Update, **RE/P**, Box 12901, Overland Park, KS 66212.

NEW PRODUCTS

New England Digital post-production system

PostPro is a specialized version of NED's Direct-to-Disk system for film and video post-production. The 8-track recording and editing system offers twice the recording time and fidelity of competing systems, the company claims, as well direct digital transfer to DASH, PD and AES/EBU formats.

Circle (141) on Rapid Facts Card

Electro-Voice N/DYM 1 compression driver

To be introduced at the Paris AES, the driver uses neodymium technology, which E-V first used in its N/DYM mics, and allows for lighter weight and a more efficient structure. The driver weighs just 7½lbs and has a 5½-inch diameter, compared to 28lbs and a 8½-inch diameter of non-neodymium drivers. Flux density in the gap is 2.25 tesla.

Circle (163) on Rapid Facts Card

Opcode Systems Timecode Machine

Converting SMPTE time code to MIDI time code, the Timecode Machine handles all applications that use SMPTE time code to sync audio or video tape with MIDI and sequencer tracks. Features include reading SMPTE code, generating MIDI TC, generating "direct time lock" used by Mark of the Unicorn's Performer and, with a computer, striping SMPTE code onto tape. The unit reads all varieties of SMPTE code.

Circle (143) on Rapid Facts Card

Benjamin International bulk tape eraser

Model 24-022A is an updated version of the 24-022 eraser and is designed for tape reels up to 11 inches in diameter. The updated model has 10% more erasing power, ensuring more effective and faster erasing, the company says. The on-duty cycle also has been increased to five minutes on, 15 minutes off.

Circle (140) on Rapid Facts Card

Phi Tech's "MIDI and the Modern Guitarist" guide

The 16-page guide covers MIDI guitar applications and has sections on enhancing the guitar, effects control, sequencing, and arpeggiating and tricks with MIDI. The guide also contains information on using guitar MIDI systems in the studio and in live performance.

Circle (152) on Rapid Facts Card

Soundcraft Twister automation

An 8-channel automation package (expandable to 64), Twister works with any MIDI compatible computer. Up to 99 snapshots of each mix are stored; with an external computer, dynamic control of levels, mutes and VCA subgroupings are stored. Up to 64 channels can be controlled on one MIDI channel. By using two MIDI channels, mixes may be modified until the desired mix is achieved.

Circle (149) on Rapid Facts Card

QSC MX 2000 amplifier

An addition to the MX series, the 2000 is a dual-monaural amp, allowing each channel to operate as an independent amplifier while sharing one ac cord and power switch. A new forced-air cooling system permits high-duty cycle operation with low impedance loads and is adjustable by a 2-speed front-panel control. The unit occupies three rack spaces; rated power is 625W per channel at 4Ω.

Circle (150) on Rapid Facts Card

Bruel & Kjaer 4011 microphone

The type 4011 is a prepolarized condenser mic, with a first-order cardioid directional characteristic, and combines a flat on-axis frequency response with a uniformly smooth off-axis phase and frequency response. The mic has a metalized diaphragm and a P 48 phantom-powered transformerless pre-amp. Frequency response is 40Hz to 20kHz at 30cm; SPL before clipping is 158dB.

Circle (151) on Rapid Facts Card

Tektronix Pacesetter oscilloscopes

The new series are said to offer popular automated features found on more expensive oscilloscopes. The 2245A and 2246A are the first 100MHz scopes that offer cursors, readout and automatic setup, the company says. In addition, the 2246A also can store and recall up to 20 front panel setups, allowing for repetitive testing or service applications.

Circle (118) on Rapid Facts Card

Audio Media Research MIDI Director

MIDI Director is a hand-held wired MIDI remote control, featuring multi-function MIDI control, song/sequencer start, stop and continue control, a 40-character by 2-line LED display and 16-segment control key matrix. Powered by a 9V battery or an optional external power supply, the unit also has a battery backup for program memory during power down.

Circle (119) on Rapid Facts Card

Sola SPS/R standby power source

Designed to protect sensitive electronic equipment from ac power problems, the SPS/R features a transfer time of 1ms, compared with a 8ms-to-10ms transfer of conventional systems. A portable, plug-in unit, the SPS/R has a 3:1 crest factor load. For protection against load inrush, the unit's overload capacity ranges from 1,400% for one cycle to 300% for 10 seconds.

Circle (142) on Rapid Facts Card

Aphex Type C Aural Exciter improvements

Now with the model number of 103A, the Aural Exciter has new circuitry that reduces output noise, less noise enhancements of noisy signals and continuous operation from −10dBm to +4dBm. Output noise is close to −100dBm; the sidechain is automatically gated to reduce noise during quiet passages.

Circle (126) on Rapid Facts Card

BGW Ground Touring amplifier

For live sound use, the amp delivers 900W per channel into 2Ω, or 1,800W bridged mono into 4Ω. Three rack spaces high, the unit includes dc speaker protection and can accommodate two BGW crossover cards.

Circle (127) on Rapid Facts Card

Atlas/Soundolier mic boom

The SB-10WE is a boom/stand designed for studio use. With hard rubber ball-bearing swivel casters, the unit is adjustable from 43 inches to 60 inches. The 60-inch boom comes with a 90° mic adapter for installation and directional positioning of a standard threaded microphone holder.

Circle (128) on Rapid Facts Card

NEW PRODUCTS

Intelligent Music M software

Previously available for the Macintosh, the interactive composing and performing system is now available for the Atari ST. Users specify basic music material as notes and chords and determine the ways that the basic material will be transformed. The music is performed by manipulating the screen controls, playing control keys on a MIDI keyboard or by "conducting" with the mouse.

Circle (129) on Rapid Facts Card

Furman Sound LC-6 stereo compressor/gate

The unit consists of two limiter/compressor/noise gates in a single rack-space chassis. The two channels may be used independently or linked for stereo via a push-button switch. Each channel has seven controls and a bar graph LED meter that indicates the amount of gain reduction.

Circle (130) on Rapid Facts Card

MasterVision "Touche Ross Video Tax Guide" videotape

The videotape from the Big Eight accounting firm contains an analysis of tax reform, a section on dealing with an IRS audit and tips on tax planning in this tax year and years to come. Also included are discussions of tax planning for investments, for small businesses and the self-employed, and for real estate investments.

Circle (153) on Rapid Facts Card

Kalglo Electronics DLP232-1 data line surge suppressor

The unit is designed to protect printers, plotters, networks, CPUs and other devices using a 25-pin serial interface. Preventing equipment damage and garbled data, the suppressor fits in-line to protect the 11 most used lines—1 through 8, 11, 20 and 22. The response time is less than 1ns, and the capacity is 115 joules.

Circle (144) on Rapid Facts Card

Coda Finale music printing software

Finale features a built-in transcription and notation intelligence that eliminates edit-display cycles, allowing more user flexibility. Running on 1-megabyte Macintosh and IBM platforms, the software allows music notation to be printed using information from MIDI equipment, point devices or computer keyboards. When users play a MIDI keyboard, the software uses a proprietary "time tagging" method to print the music.

Circle (145) on Rapid Facts Card

Resonate update of Listen software

Version 2.1 of Listen, ear-training software for the Macintosh, includes a level-selection menu, a rewritten manual and an improved user interface.

Circle (120) on Rapid Facts Card

NEW PRODUCTS

Research Technology International Tapechek dropout analyzer

Model D11 tests videotapes for video and RF dropouts before recording, editing or duplicating, and can be used with any videotape format. An audible alarm sounds when the dropout count exceeds preset levels, and a built-in printer provides a hard copy report of tape condition. Readouts include total dropouts, "bad" minutes of tape, dropouts per minute and elapsed evaluation time.

Circle (121) on Rapid Facts Card

Audio Kinetics LTC gate array

The DAK 010 is a SMPTE/EBU longitudinal time code processor that tracks incoming TTL level signals over the range of 1/50 to 200 times play speed, and packs the data into four time code and four user bit registers. Data are configured in these registers in the same format regardless of the direction of the incoming LTC stream.

Circle (122) on Rapid Facts Card

Roland E-660 digital parametric EQ, R-800 digital reverb

The units are said to be the first pro audio processors that incorporate the new digital audio transmission standard developed by the AES, allowing signal processing entirely in the digital domain. The E-660 is a 2-channel/4-band parametric EQ, with the center of each band being individually adjustable. The R-880 is a 4-channel reverb, each channel being independent, which allows a variety of configurations.

Circle (123) on Rapid Facts Card

Tascam MTS-30 MIDI/FSK translator

Aside from offering MIDI/FSK conversion, the unit used MIDI Song Position Pointer data to synchronize playback from any point within a composition. An automatically switched MIDI out/thru terminal is included, allowing operators to drive a drum machine while recording the sync tone from sequencer to tape, and to drive the drum machine and sequencer on playback without repatching.

Circle (124) on Rapid Facts Card

Akai DR1200 PCM digital recorder

Using a tape format developed by the company, the DR1200 uses 8mm videotapes to record up to 17 minutes of audio. Sampling resolution is 16-bit linear, and sampling rates of 44.1kHz and 48kHz are switch-selectable. A separate analog track is available for recording time code, and up to three units can be connected using the DL1200 locator.

Circle (125) on Rapid Facts Card

Ear Works percussion sampling disc on CD

"Volume 1 Percussion" contains 156 percussion sounds that were recorded digitally, including Acacia wood bowl, Chinese tom-toms, dumbeck, log drums and thundersheet. The compact disc is designed to be used by any sampler, and all sounds are presented in their full duration to provide a choice of sampling lengths.

Circle (160) on Rapid Facts Card

JBL 2450J compression driver

The driver features JBL's first use of neodymium magnet technology, and is smaller and lighter than any comparable compression driver, the company claims. The driver also has a Coherent Wave phasing plug, which provides in-phase combining of sound waves, allowing smoother response and a clearer, cleaner signal. The driver is said to be 1dB more efficient than the company's 2445J, which uses a ferrite magnet structure.

Circle (161) on Rapid Facts Card

Passport Designs MIDI Transport interface

A dual MIDI interface for the Macintosh, the MIDI Transport operates as a Mac MIDI interface or as a stand-alone MIDI/SMPTE interface. Incorporating SMPTE-to-MIDI time code conversion, it can sync to all SMPTE formats or its own variable rate FSK signal. MIDI Transport is also compatible with the J.L. Cooper PPS-1 and provides two MIDI Ins and five MIDI Outs, and audio tape in/out.

Circle (146) on Rapid Facts Card

Hal Leonard Publishing's "The Sound Reinforcement Handbook"

Commissioned by Yamaha and written by Gary Davis and Ralph Jones, the book is an unbiased reference book on audio reinforcement, for nearly every level of knowledge. In addition to sound reinforcement, the book covers most major aspects of professional audio, including recording, broadcast and fixed installations.

Circle (147) on Rapid Facts Card

Community Light & Sound CS28M floor monitor

The monitor shares the same components as the company's CS25 loudspeaker and is designed for situations where space is a valuable commodity. The monitor contains a 12-inch LF driver; at 3kHz, high frequencies cross over to a PZT driver coupled to an exponential horn. Measurements are 14"x15"x22¼"; power handling is rated at 100W rms 250W program, while maximum SPL is 117dB at 1m.

Circle (148) on Rapid Facts Card

E-mu Systems Emulator Three

The Emulator Three Digital Sound Production System is designed to be a musical instrument, post-production workstation and digital effects processor in one system. Capabilities include true stereo sampling, 16 voice, 16-bit linear data format and up to eight megabytes of internal RAM for up to 135 seconds of sampling time. The system is available in a keyboard or rack-mount version.

Circle (167) on Rapid Facts Card

JBL/UREI 7110 limiter/compressor

The unit features soft-knee compression curves adjustable from 1.5:1 through infinity:1. Users have complete control over threshold, attack, release time and output level. The unit also has an automatic preset button that engages a program-dependent variable attack and relase circuit, and fixes the compression ratio and peak/average blend controls to critically accepted settings.

Circle (168) on Rapid Facts Card

White Janssen archival, stock film library

The company's film library is now available for music video production, offering a way to produce videos and reduce costs. The library contains more than 2 million feet of original 16mm film and 150,000

NEW PRODUCTS

feet of original 35mm film. The company says it will deliver a 1-inch master of the needed footage within 21 days, at a guaranteed cost.

Circle (169) on Rapid Facts Card

Neve Prism series

Derived from the V series console, the Prism series comprises a 4U rack with a 10-module capacity. Two modules available are the Format Spectrum equalizer and the mic amp/dynamics unit, which includes a compressor/limiter/gate/expander. Both modules have electronically balanced line level inputs and outputs; the entire rack may be powered from the existing console or by a 2U power supply.

Circle (170) on Rapid Facts Card

TDK R-DAT tapes

The tape is of super finavinx metal particle design, and is available in three lengths: 60-, 90- and 120-minutes.

Circle (162) on Rapid Facts Card

MIDImouse Sonicflight software

The software is an editing, programming and librarian program for the FB-01 and Atari ST. Features include full mouse support, standard GEM interface, editing and storage of two patchbanks and three configurations and graphic display of algorhythms, waves and envelopes.

Circle (164) on Rapid Facts Card

Ramsa WR-T820B console

The WR-T820B is an 8-bus console that uses high-speed op-amps at critical gain stages throughout its circuitry. Flexible routing and switching allows up to 48 inputs and eight addressable aux sends; full-function LED and VU metering are included.

Circle (166) on Rapid Facts Card

Total Audio Concepts automation interface

The interface, available as an option on TAC Scorpion and Matchless consoles, allows the consoles to be run on the automation systems made by specialist manufacturers. The interface is universal, allowing end-users to choose the most appropriate system for their needs.

Circle (154) on Rapid Facts Card

Solid Support Industries adjustable mixer stand

The AM-10 is designed for mid-sized mixing consoles and will hold mixers ranging from 27 inches to 47 inches wide. Consoles are cradled by 2-inch lips and are set at 26½ inches high. Two of the four casters are locking; the unit can support up to 250lbs.

Circle (158) on Rapid Facts Card

Full Compass Systems DAP-320 processor

The unit is a general-purpose digital signal processor with open-ended architecture, and can be configured by changing software and/or plug-in modules. Features include limiting, compressing, expanding, gating and peak-limiting simultaneously in the digital domain; graphic display; variable sampling rate, FIR filtering and DAT notch filtering.

Circle (156) on Rapid Facts Card

Hardware revision of Digital Creations automation

The company's Moving Fader console automation system has been redesigned, and features conversion accuracy to 10 bits and a new touch sensor design that has an auto-calibration mode. Servo drive amps are now mounted in the system rack, eliminating head at the faders and facilitates maintenance. All system calibration is now performed by the computer, which eliminates trim pot adjustments.

Circle (157) on Rapid Facts Card

Electro-Voice DeltaMax speaker system

DeltaMax is a processor-controlled speaker system in a 12-inch, 2-way and 15-inch, 2-way versions. The electronics provide conventional frequency division, speaker protection, time delay and equalization to the speaker. Both systems are constructed in trapezoidal wedge, forming a 30-degree wedge. The 12-inch version uses the DML-1122; the 15-inch, the DML-1152.

Circle (165) on Rapid Facts Card

New operating software for AKG TDU 8000

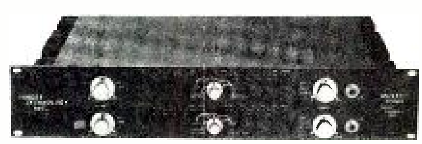
The digital delay line now offers a SAFE program, protecting currently used delay times from inadvertent readjustment. The upgrade is avalable as a field-installable EPROM from the company; future units will be shipped with the software.

Circle (155) on Rapid Facts Card

Target Technology TTQ-400 power amplifier

The new power amp is four, 40W RMS (8Ω) amplifiers using four 150W LM12 power op-amps. Each amp has a voltage controlled gain stage that permits remote control of level individually or in parallel.

Circle (190) on Rapid Facts Card



Fane Acoustics loudspeakers

The U.K. company's range of loudspeakers are now available in the United States. The range is comprised of the Crescendo M series, Medusa, Studio series, Co-axial series and horns, compression drivers, crossovers, tweeters, glass fiber horns and J series horns.

Circle (159) on Rapid Facts Card RE/P

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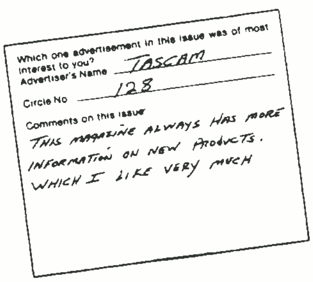

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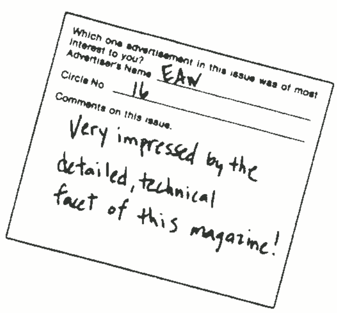
Advertiser's Name TASCAM
Circle No 128
Comments on this issue
THIS MAGAZINE ALWAYS HAS MORE INFORMATION ON NEW PRODUCTS. WHICH I LIKE VERY MUCH



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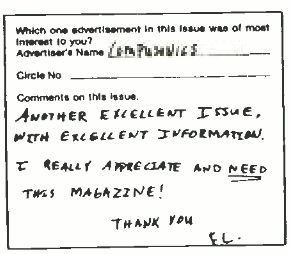
Advertiser's Name EAW
Circle No 16
Comments on this issue.
Very impressed by the detailed, technical facet of this magazine!



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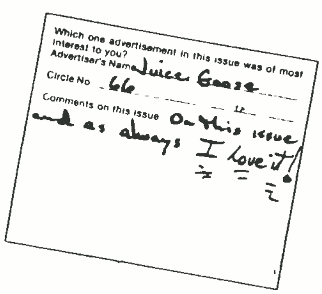
Advertiser's Name
Circle No
Comments on this issue.
ANOTHER EXCELLENT ISSUE, WITH EXCELLENT INFORMATION.
I REALLY APPRECIATE AND NEED THIS MAGAZINE!
THANK YOU
F.L.



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Advertiser's Name Juice Goose
Circle No 66
Comments on this issue On this issue and as always I love it!

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Finally: an HF driver that doesn't turn cymbals into trash can lids

It's easy to assume that the best-selling HF driver is also the best-sounding. And it's easy to conclude that the performance of the leader represents the highest achievable standards. Easy, yes. And wrong.

Because now there's something better: the Electro-Voice DH1A—a driver that **demands** comparison with any other HF transducer you have ever heard or used.

Spend a few minutes in comparative listening and prepare to revise your idea of what constitutes "reality" in the world of sound above 10 kHz. Hear, for the first time, accurate reproduction with all the presence and vibrancy of a fine home audio system. Experience clean, distortion-free sound without "ringing," the result of the diaphragm's inability to control inertia—to stop its own mass when required by fast transients.

To create the DH1A EV engineers chose a magnet with extremely high flux density, plus an optimized, balanced magnetic circuit to ensure more exact correspondence between the input signal and diaphragm movement, further improving transient response.

The 10 kHz breakup you've heard as a "sound-smearing" effect in "their" driver is avoided in the DH1A by using a 3-inch, rather than a 4-inch, diaphragm. This moves the first mode intermodulation distortion all the way out to 15 kHz, providing piston-like diaphragm motion to a point well beyond audibility.*

A field-replaceable diaphragm makes the DH1A still more useful. We also added the option of 8- and 16-ohm impedance match. And our EV-exclusive PROTEF™ feature that guards against voice coil damage.

We don't ask you to accept all this without proof. Make us prove our claims. Audition a DH1A today. Hear for yourself how easily you can prevent cymbals from sounding like trash can lids.

For more information, write Electro-Voice, Inc., 600 Cecil Street, Buchanan, MI 49107.

EV® Electro-Voice®
SOUND IN ACTION™

Circle (2) on Rapid Facts Card

*For convincing proof of the performance difference between the DH1A and today's leading driver send for actual TEF® "waterfall" curves, plus additional product information. Write to Electro-Voice, Inc., 600 Cecil St., Buchanan, MI 49107.

IN PART

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IN CONCERT

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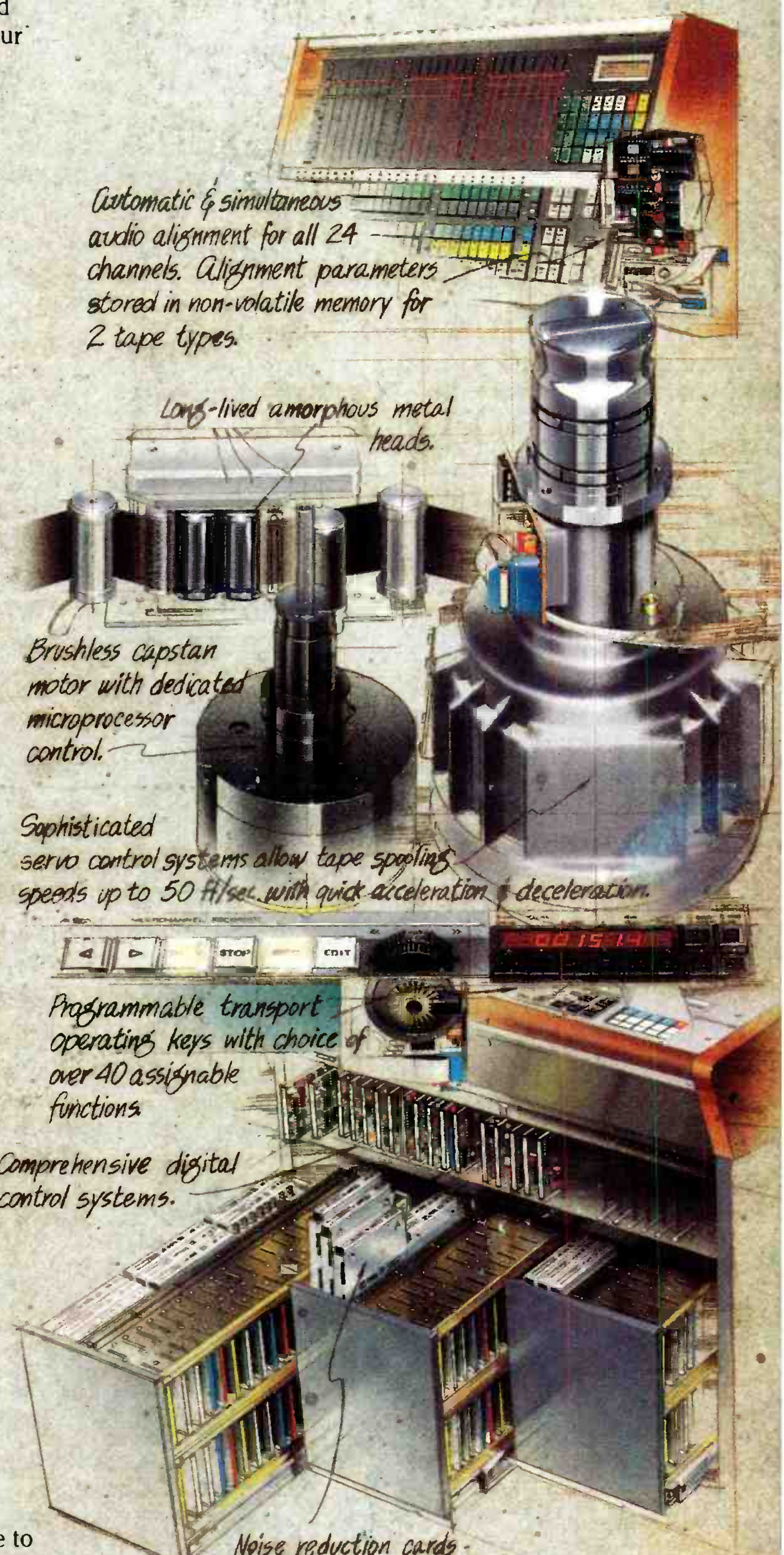
Studer A820 Multichannel Recorder



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