August 1986

Recording ENGINEER/PRODUCER

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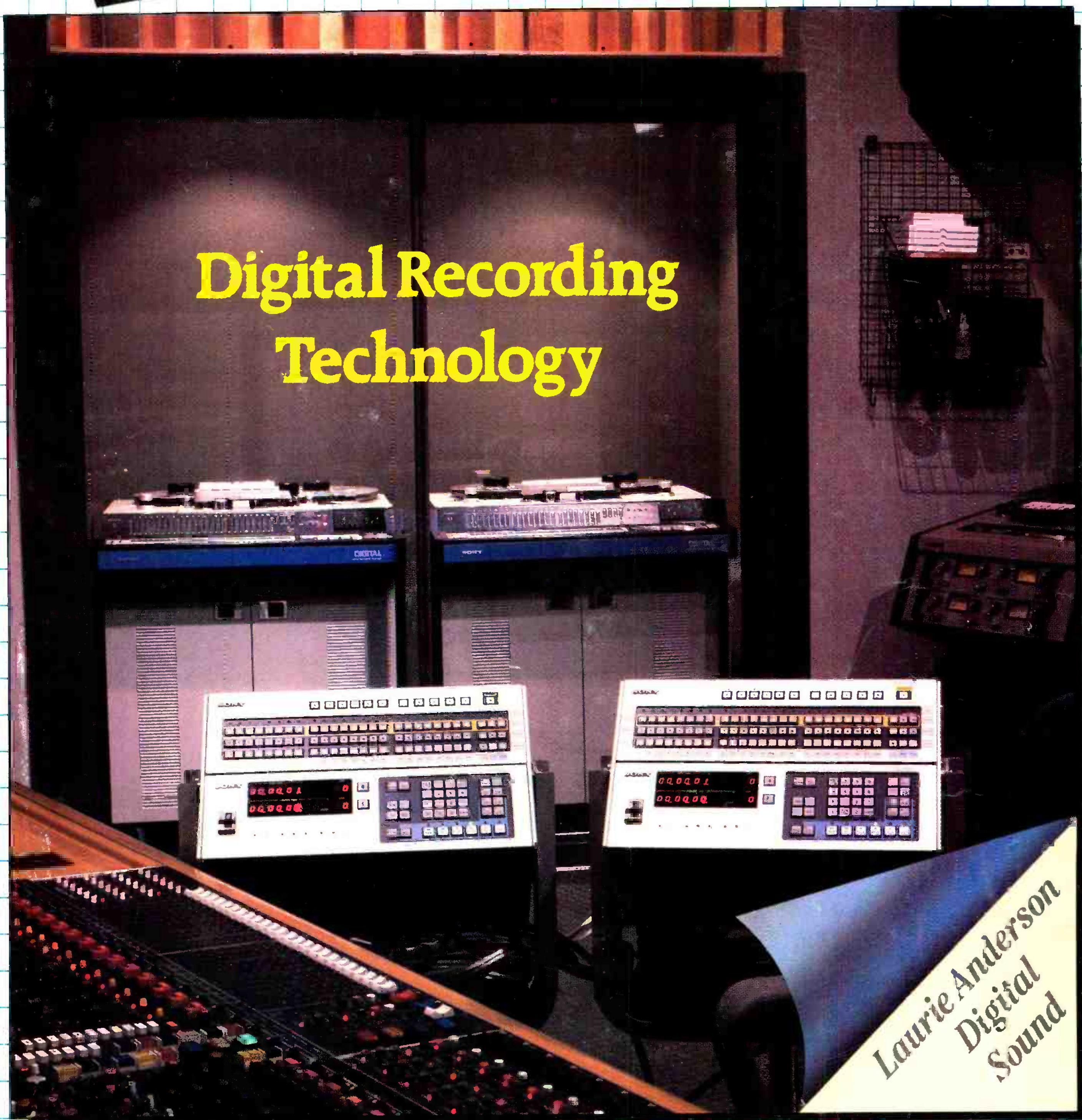

Behind Every Synclavier® There's a Success Story

Profile: **Murray Allen**

Accomplished musician, composer, engineer and businessman, Murray Allen owns one of the largest audio facilities in the world. Universal Recording offers every service in the field of audio from 24-track recording studios, film mixing theaters, and video sweetening rooms to remote sound crews and even a cassette duplication factory. Murray knows that every piece of equipment he purchases must not only provide the highest sound quality and operational flexibility available, but also be able to prove itself financially with a solid return on investment. He comments on Universal's recent completion of a dedicated Synclavier studio:

"We at Universal feel a studio cannot honestly call itself world-class nor can it claim to be state-of-the-art if it does not own and utilize a hard disk-based digital audio manipulation system. Now having said that, we have researched every such system on the market and some on the drawing board and have found only one system sophisticated enough, fast enough, with the kind of sheer computer power we need to serve our clients. That system is the Synclavier."

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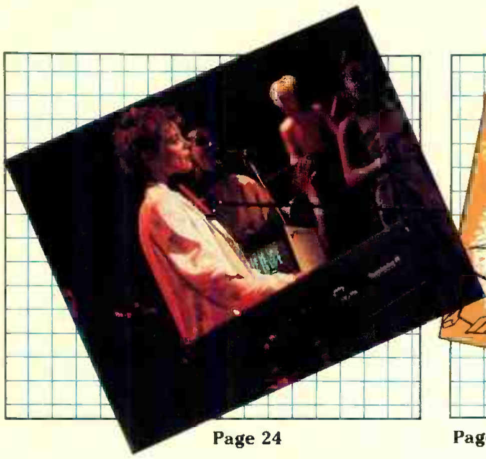

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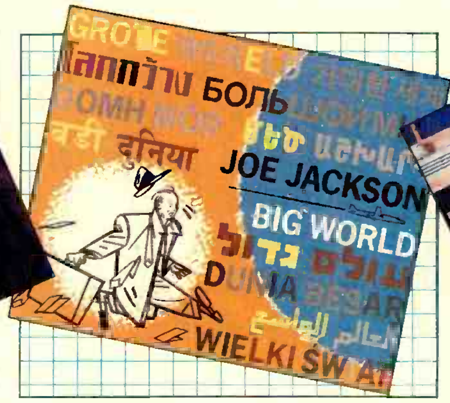

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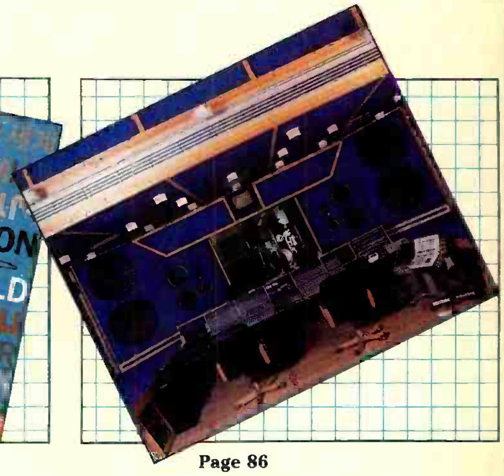

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On the cover

A pair of DASH-format digital multitracks at Master Sound Astoria Recording Studio, New York. (Photo by David Bevan.)

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News

APRS secures U.K. work permit rules for producers

After more than a year of correspondence with various immigration departments in the United Kingdom, the Association of Professional Recording Studios has managed to clarify the work permit regulations, which for a long time have prevented American producers from working in Britain's studios.

Work permits are *not* required for American producers and their staffs, provided that they intend to visit the United Kingdom solely and specifically to record in a studio and that they are being paid by an overseas record company. APRS received the confirmation from the Home Office, Britain's government department for immigration matters.

In the past, engineers and producers wishing to record in the United Kingdom have faced delays in being issued relevent work permits, often having to wait more than two months.

However, work permits would still be needed if they performed or made promotional appearances during their visits, according to the Home Office.

The October **RE/P** will have more details on this subject. For additional information, contact the APRS secretary, 23 Chestnut Ave., Chorleywood, Hertfordshire WD3 4HA; 01144-923-772907.

Software directory

RE/P is compiling a directory of business application software, which will include programs specifically written for studio management. These include software for monitoring bookings, calculating session fees, following studio traffic, maintaining inventory and specialized programs for use in the day-to-day business operations of a production facility.

Consideration will be given to packages that are prewritten (off-the-shelf), customized (originated in-house) or user-modified (from existing programs, such as Lotus 1-2-3 or Ashton Tate Dbase III).

Please limit written submissions to software that operate on a Commodore C64 and Amiga, Apple IIe, Atari ST520/1040, Apple Macintosh, and IBM PC and AT, including compatibles.

For further details, contact the **RE/P** editorial office at Suite 220, 1850 N. Whitley Ave., Hollywood, CA 90028; 213-467-1111.

Midwest music exhibition

Scheduled to be held Sept. 13-14, at the Holiday Inn in Skokie, IL, Musitech '86 will comprise a series of seminars, demonstrations and workshops covering such topics as digital sampling and sound reinforcement.

More than 40 manufacturers will be exhibiting a range of products, including 360 Systems, Akai, J.L. Cooper Electronics, E-mu Systems, Fairlight, Fender, Ibanez, JBL, Lexicon, Oberheim, Peavey, Roland, Soundtracs and SynthAxe.

For further details contact the organizers, Gand Music & Sound, 780 Frontage Road, Northfield, IL 60093; 312-446-4263.

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Recording
ENGINEER/PRODUCER

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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Editorial

Digital Directions

Although digital transports and processors have been available for several years, I detect a degree of reluctance from certain studios to invest in digital technology. As far as I can tell from conversations with several leading facility owners, the two factors at issue are format compatibility and an uncertainty with the future of digital recording.

Obviously, format compatibility is a key consideration when tapes have to be moved from one facility to another, and will only become a non-issue when there are more machines in the field. (And, with the inclusion of AES/EBU digital interfaces on DASH- and PD-format transports, digital-to-digital copies would be possible, should you elect to start a project in one digital domain and complete it in another.)

The question of future uncertainty centers on the very nature of the digital recording process. Will tape remain the primary medium for audio recording and production? Many industry observers have questioned the long-term cost effectiveness of using complex multitrack machines to access up to 32 tracks of 16-bit audio. If only because of our familiarity with the reel-to-reel analog transports from which they are derived, all current-generation digital machines use a similar design and operation.

But, it can be argued, magnetic tape is an inherently inappropriate medium for the short-term storage of digital audio. During tracking and overdub dates, we are pushing digital recording and error-correction technology to its practical limits as we punch in on previous tracks and combine tracks to free up space on the multitrack. Not to mention the physical abuse tape suffers during cut-and-splice editing. Also, the speed of finding a specific tape location is limited by the machine's mechanical transport.

A far better approach would be to relegate tape to long-term, archive applications—where its data packing density and cost effectiveness cannot be equalled—and utilize random-access memory for real-time recording and editing. With the accelerating fall in the cost of RAM and controller micros—not to mention the newer, high-capacity memory technologies currently being developed—it doesn't take a crystal ball to predict that multitrack systems with adequate recording capacity are just around the technology corner.

The choice of recording medium goes beyond the flexibility offered by non-tape based systems. What many of us overlook, I would suggest, is the historical reason for separating the recording device from that other essential item of production hardware—the console—stems from the fact that these two items have been based on widely divergent technologies.

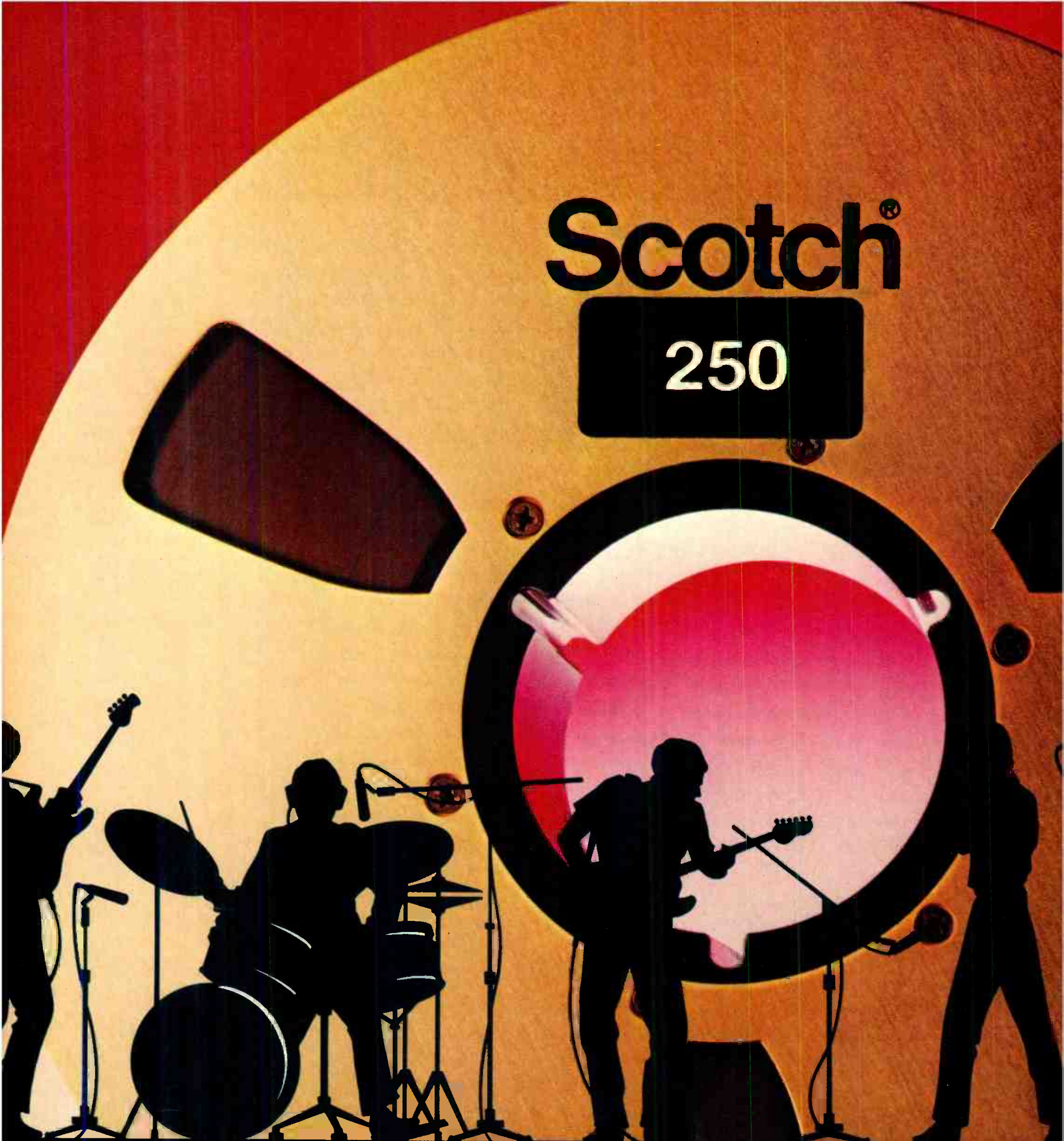
But as console topographies start to incorporate greater amounts of digital control, reaching up towards all-digital designs, the distinction made between the console and storage device will become more blurred. With so much processing power available in future-generation recording and production consoles, why not combine these two functions—along with random-access editing capabilities—into one microprocessor controlled system?

Of course, there will still remain situations where our sole interest is the real-time control of audio signals—on-air broadcast and live-performance sound being but two examples. For these applications, the amount of on-board RAM for recording could be reduced to a minimum, although the amount of digital control elements could equal that of consoles designed for conventional recording duties.

As has been stated in my previous editorials, radical advances in digital recording and console design are closely linked with the degree of "technology overspill" from the computer manufacturing industry. Something tells me that as powerful, high-speed micros and minis become the *de facto* standard in the mainstream business and scientific computer markets of the late Eighties, we will be able to put such systems to good use in the studio for digitizing, recording and manipulating audio.

Mel Lambert

Mel Lambert
Editor



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TO THOSE WHO PUT A RANGE OF MUSIC ON A ROLL OF TAPE, ONE TAPE IS TRUE.

Letters

Sampling keyboards

From: Phil Peters, Delta Stage Lighting, Mobile, AL.

As an update to our keyboard system, we recently acquired two Ensoniq Mirage Digital Multi-Samplers in addition to a Yamaha KX-88 control keyboard, Yamaha TX-816 rack and QX-1 sequencer. Terry Fryer's comment in his recent review [February 1986 issue] that the Mirage is a "rock and roll" sampling unit very well describes the unit as a whole. I am generally pleased with the DMS units, except for a few things.

Using the KX-88 to control the TX-816, a DX-7 and the Mirage units via MIDI, I need to have the units respond to all basic MIDI commands. However, the Mirage does not respond to volume commands.

Another disappointment is the Mirage's keyboard range. Limited in its design to five octaves, when played from the KX-88 most sounds—such as the piano sample—repeat octaves on both ends of the keyboard. However, on the factory string sample, the full 88 notes are represented.

This seems a contradiction, because the unit is supposed to be limited to a 5-octave range. So why does the string sample continue out to a full 88 notes, and the keyboard sample, which is really why we bought the units, stop at the last octave?

Reply from: John O. Senior, software engineer, Ensoniq Corporation

Phil Peters is correct in observing that Mirage Multi-Samplers do not respond to MIDI volume change commands. Although admittedly desirable in some circumstances, this feature is impossible to realize in the current design. The volume control on all Mirage instruments is an analog potentiometer controlling a VCA in the audio output section and is not under any digital control. This hardware configuration precludes the possibility of implementing MIDI control of overall system output level on the Mirage.

The second observation he makes, concerning the 5-octave effective keyboard range to which the Mirage will respond, also is correct. This design limitation was implemented to avoid the severe degradation of audio quality that occurs when samples are played back with extreme transpositions. Aliasing becomes significantly more problematic above the 5-octave limit and, although transpositions in the lower octaves may be acceptable in some cases, software design considerations made the decision to restrict the range on both ends the better alternative.

The factory-produced string sample on Sound Disk 3 is limited to the same 5-octave range as all other sounds, as this is a function of the operating system and not the data on the sound disk.

We hope that the Mirage units, within the limits of their design, will continue to prove musically useful to Mr. Peters and his associates, and we appreciate the opportunity to answer his questions on these topics.

On the right track

From: Greg Chastain, Wax Master Productions, Laguna Beach, CA

In January 1986, I opened a 16-track facility in Laguna Beach, CA. Because my bank was willing to loan me only $50,000, I had to start the studio on a limited budget. **RE/P** became my bible for picking the equipment that would best suit my needs within my financial constraints.

Especially important was the 2-part directory on consoles published in the April 1985 and June 1985 issues, which led me to purchase a Trident Series 65. One of the new product previews that appear in the back of **RE/P** prompted me to buy an Otari MX-70 1-inch 16-track. Your technical reviews, along with pricing, have been my most important sources of information. It seems that whenever I have a question, the answer appears somewhere in the magazine.

Since opening Wax Master, I have grossed more than $20,000. This has allowed me to purchase more goodies in the never-ending battle to get "The Sound." The goodies include a Lexicon PCM-70, two Roland SRV-2000s and a Yamaha REV-7. I have also filled out my dbx 900 rack, which now includes two 903 compressors, two 904 noise gates, two 902 de-essers and two 905 parametric EQ modules.

After owning the Otari MX-70 16-track for two months or so, I began to contemplate moving up to an Otari MTR-90 24-track, which would require an additional investment of more than $20,000. Instead, I opted for a far less expensive alternative: I went out and bought a Commodore C128, the new Sonus SuperSequencer 128 and an interface card.

With this setup, I can store all my keyboard and drum tracks on the SuperSequencer, freeing up all of the tracks on my Otari for vocals and other acoustical tracks. To do this I write one track of SMPTE time code using an E-mu Systems SP-12 drum machine. The SP-12 then reads the SMPTE code and sends out MIDI commands to SuperSequencer, allowing me to sync all keyboard and drum tracks to tape.

Now my only limitations are the 40 inputs on the Trident console.

This solution cost me less than $1,000 total, and also gives me all the SuperSequencer's special MIDI-editing features.

Using this equipment and MIDI techniques, I recently completed a project for a new artist, Jane Child, resulting in a major label contract with Geffen Records. Jane is a keyboard player and vocalist. In preproduction, we sequenced all keyboard, bass and drum tracks on the SuperSequencer, and then cut seven vocal and six guitar tracks on the MX-70. Tracks 14 and 15 were mono work mixes of all the sequenced music, track 16 being reserved for time code.

At final mixdown, all sequenced music was synchronized through time code to the multitrack performances and sent directly to the digital 2-track. With the technique, I get a very clean product, and am not limited by having only 16

Project engineer Paul Beliveau

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Letters

tracks of analog tape. Jane Child appeared in May on a *Good Morning America* special dealing with getting signed to a recording contract. (David Hartman: If you only knew how it's really done!)

I have completed more than 30 separate projects since opening Wax Master Productions, including audio sweetening of commercials for the Los Angeles Dodgers, Dimension Cable TV and Zorprin Aspirin. In March, I did a project that resulted in a top 10 single currently being played on several Southern California stations.

The whole process of putting the studio together, and making Wax Master a success, has been exciting. I know that without **RE/P**, I might have succeeded, but not without spending more money and more time, and not without running into a brick wall or two.

MIDI in the studio

From: Rob Fuston, general manager, 39th Street Music Productions, New York.

I recently saw an article in **RE/P** about MIDI compatibility in recording studios, and thought that you might be interested in what we have here in the way of MIDI interface. We have thoroughly integrated the personal computer into our recording studio as a production tool. We are using both the IBM PC and Apple Macintosh in an effort to extend our services out of the studio and into the field with preproduction services. This, I believe, is a new concept in recording studio marketing, and is being brought about by the advent of computers in music production: specifically, the computer sequencer.

Preproduction is not a new concept, of course, but with the help of computers it has now become part of the recording process. The use of sequences allows complete performances to be brought into the studio and simply transferred to tape. Recognizing that sequencer-equipped home studios now have the potential to cut into the recording studio business, we have taken the position that, rather than try to ignore this trend, we should meet it head-on. Therefore, we designed our new control room to facilitate easy transfer of preproduced sequencer tracks from any source into our system.

Our new SSL console uses time code for all of its computer functions. We use that same time code to drive our Garfield Master Beat synchronizer, which in turn drives an unlimited assortment of sequencers and drum machines [via MIDI].

A major consideration in outboard purchases was MIDI compatibility. Thus, we went with the Publision Infernal Machine 90, Lexicon PCM-70 and other outboard gear with MIDI implementation that allows program changes to be stored to computer. Our system is networked so that sequencing can be done in one location and then be transferred to another computer for filing purposes, or into the CR computer for final production.

In addition, engineers affiliated with our studio are providing preproduction services using personal computers in the field, thus giving clients the opportunity to take full advantage of the technology. Looking toward the future, computer sequencing done anywhere else in the world can be transmitted to the studio via modem.

To complement our computerized facility, we have assembled the best MIDI keyboards and drum machines available into a permanent setup controlled via the J.L. Cooper MIDI Switching Matrix. Preset MIDI routings can be called up and allow quick, no-hassle setups for multipart sequencing, MIDI-capable keyboard and outboard effects. Any sequencer or keyboard that is brought in can be easily patched into the system and immediately interfaced with the computer and other keyboards. Our Emulator II is interfaced with a Macintosh computer that also can be used for sequencing software.

All in all, it is a complete, integrated system that saves the trouble of carting in equipment and setting it up, but is flexible enough to be compatible with any equipment that a producer or engineer may choose to bring in.

We also now have the ability to store synthesizer outboard and MIDI-routing setup to the computer using Octave Plateau's Patchmaster software. Although this software has not yet been released commercially, we have been working directly with Bruce Frazier of Octave Plateau in the development stages. An entire setup of 14 synthesizers, along with MIDI outboard programs, can be loaded from the computer in about two minutes, giving any producer a virtual permanent synthesizer setup to work with whenever they come in. Synthesizer patch files can be saved in independent directories restricted for use by individual clients, or clients may call up patches and setups from our collective library.

In addition to our installation of new equipment and computers, we engaged Russel Berger, of The Joiner-Rose Group, Dallas, to acoustically redesign our control room. We were the first studio to be done by Russ' company in New York, a city where he has noted a lack of attention to control room acoustic design. The result is a room that is already gaining a reputation for having a great sound.

We have a vision of the future that sees the recording studio as the focus of a more decentralized approach to music production. We have made all of these improvements in order to remain competitive and profitable as a recording studio. Many studios boast of having "MIDI Rooms" and computers, but we have integrated this technology into our basic studio design. And, whereas much of this new technology is offered as an extra feature, we consider it part of our basic service, we have gone to great lengths to make sure that the system is as reliable as our Studer tape machines.

R·E/P

WE'VE GONE TO GREAT LENGTHS TO MAKE SERIES 600 THE ULTIMATE 16 TRACK CONSOLE.

The new Series 600 has been designed as the universal 16 track console. From ½" personal recording facilities to the professional 2" studio standard.

In analogue recording, the tape is the weakest link. Therefore, it's vital that whatever you put onto the tape is of the highest quality – whichever tape format you use.

At Soundcraft we've used all our latest technology in circuit design to ensure that the performance of Series 600 outshines any other 16 track console.

We've included internal line matching links and switches which enable the user to re-set the console from the professional +4dBu standard to –10dBV for ½" and 1" multitrack.

In addition to 16 equalised monitor channels which are separate from the input modules, Series 600 includes 16 LED bargraph meters, switchable peak/VU, plus two conventional VU meters on the mix buses.

Four auxiliary sends on each channel can be routed to six auxiliary buses. And the master module contains extensive monitoring facilities, including access to three two track sources for playback.

All line inputs and outputs (except input channel direct out) are balanced, utilising Neutrik XLR connectors. An external 19" rack mounting power supply allows for clean and stable DC voltages to the console.

Series 600 also incorporates many more features than you'd expect from a console so reasonably priced.

So, whichever tape format you use, you should go to great lengths to ensure your console is a Soundcraft Series 600.

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Managing MIDI

By Paul D. Lehrman

Anyone who has designed an automated studio knows that it involves getting various pieces of essentially non-compatible equipment to "talk" to each other. At the heart of things is some kind of digital bitstream, like SMPTE time code, that keeps things together.

But time code has limitations—it only tells things where they are in time, not *what* to do when they get there. For that, each piece of gear needs its own internal processor and memory. If a recording or mix is going to take more than one session to complete, all appropriate instructions have to be stored somewhere.

But wouldn't it be wonderful if all of those instructions—when to do something *and* what to do—could be stored on one medium, like a single disk drive in a single computer? Multiple takes, each one with its own subtle distinctions, could also be stored easily, without taking up extra memory or tape tracks, and even be merged into each other if the situation warrants.

It sounds expensive. If it means using an all-digital console, with built-in equalizers and processors, it will be. But there's another way to do it, using discrete equipment and a single data language common to a wide variety of gear: MIDI.

There are 16 MIDI channels, and each one has the capability of sending 128 different "controllers," each of which can have 128 discrete values. Some of these controllers are normally assigned to specific functions, like pitch bend or sustain pedal, but the majority are not.

With only 100 controllers, a 16-input console could have all of its faders, as well as four EQ controls and one effects send per channel, individually addressed by MIDI—and that's just using one MIDI channel. Minor modifications in the MIDI spec, already being implemented by some hardware makers, can increase the number of controllers and their resolution by as much as a *power* (not just a factor) of two.

Level processing

There is part of the MIDI spec called MIDI volume that can be used for some level-setting chores. Unfortunately, not every MIDI synth responds to this information, and among those that do, some do not mute completely, or worse, they mute the signal without removing the residual noise.

Several companies that are making a stab at MIDI-controlled level processing. Several mixing consoles and discrete devices that will mute one or more audio channels when given the appropriate MIDI commands, either a program changes or note-on/note-off data. Other companies are making consoles that use variations on the "snapshot" approach used by some time code-based or event-triggered automation system.

Several companies are making a stab at MIDI-controlled level processing.

Sophisticated as these MIDI-equipped mixers are, most of them still fail to take full advantage of what MIDI has to offer for studio automation. Snapshot mixing cannot possibly reflect the subtle changes an experienced engineer always puts into mixes. Any engineer will tell you that few fader movements are perfectly linear, and 99 moves in one song, for example, is pretty limiting when you're dealing with 32 channels of audio and multiple effects units. Instead, whatever system is recording the movement of the console controls has to be able to respond as "humanly" as possible, and that means *physical* controls, whose every nuance can be tracked.

One solution might be to use VCAs that can be individually programmed to respond in real time to a MIDI breath controller, volume controller, foot pedal, keyboard aftertouch pressure or key velocity. In this way a single MIDI channel coming from a sequencer can easily contain enough data to control several channels of audio in real time. And there's no reason why multiple units cannot be ganged.

The use of velocity values to control VCAs is a way to keep the MIDI data stream from clogging (which we'll discuss in a moment), but there is no way to translate the movement of a slider or knob into MIDI velocity. In order to implement true MIDI studio automation, several devices have to appear on the market before this can become a reality.

One, obviously, is a mixer—a "black box" most likely—that would accept data from multiple controllers and apply it to audio circuits. Perhaps some manufacturers are still sensitive to the stigma that attached itself to VCAs in the early days of studio automation, but that should really not be an issue anymore.

Another device would be a dedicated control unit, consisting of nothing but assignable MIDI controllers, so that engineers could set up the console any way they want, and be able to store the configuration, preferably as a MIDI data dump.

Third, there is software; of the three possibilities, this is closest to being ready. Many, if not most, MIDI sequencers faithfully record and play back MIDI controller changes. A few offer controller editing, mostly in the form of text processing, in which each MIDI event is displayed and can be altered; or in the form of "table operations," in which specified controller information within a certain time span can be arithmetically altered, much like a spreadsheet accounting program.

What's missing, however, is the human touch: *graphic* controller editing, which can take the motion of a controller, display it as a line on a screen and then let the user alter it by moving individual segments of the line, much like the envelope editors on many synthesizer patch-editing programs.

With MIDI controllers doing the work of operating a mixing console, the software will have to be able to scan backwards to all of the last controller settings and also output the data before playing. Considering the huge number of possible MIDI controllers, the software will have to be very selective about scanning them, lest it take forever to start the music.

Even if all of these advances were to come to pass, there would still be two serious problems to be overcome before MIDI could be relied upon to provide full-blown studio automation. One is that using many MIDI controllers means a lot of data have to be sent down the MIDI line. With MIDI's bandwidth of 31.25kHz, things can clog up pretty fast. When that happens, events get out of sync.

The other problem is that in the eyes of many high-end equipment manufacturers, MIDI still belongs in the world of performance or, when it comes to recording, "semi-pro." The products available so far, interesting as they are, do little to change that image. However, at least one high-end console manufacturer has said that it is definitely doing "something" about developing a MIDI interface for its equipment. If that's true, then you can bet others will follow, and the next step in the maturation of MIDI will have been achieved. R-E/P

Lehrman is a free lance writer, electronic musician, synthesist, producer and a regular RE/P contributor.

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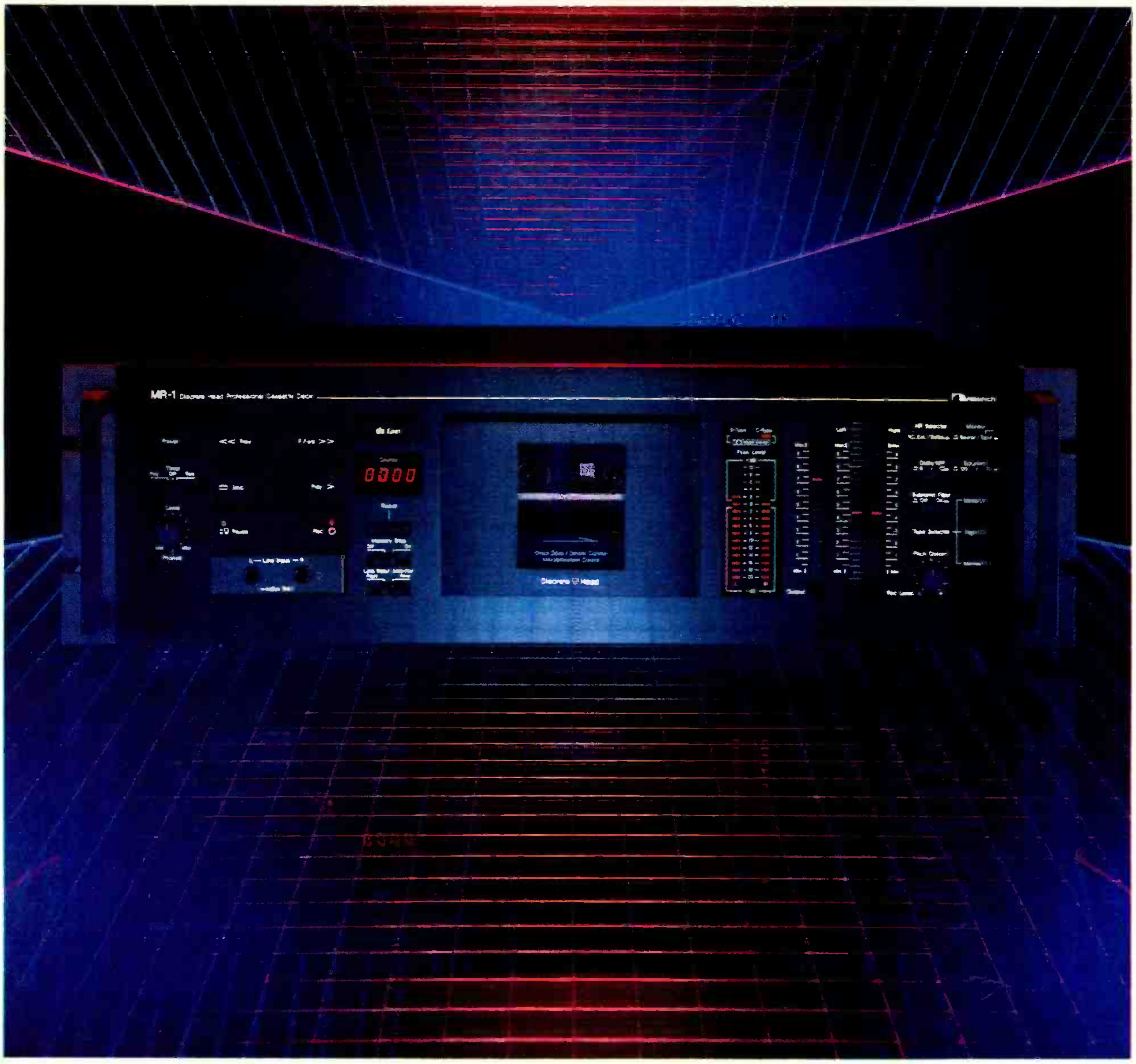

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Circle (10) on Rapid Facts Card

Film sound today

By Larry Blake

Any discussion of the technical benefits to be offered by the next generation of film sound reproduction formats—how the sound is recorded on the print—must go hand in hand with an understanding of the mindset of theater owners.

Because of the availability of Dolby Stereo prints in both 35mm optical and 70mm magnetic formats, thousands of stereo systems have been installed in the past 10 years. Asking exhibitors to make a change at this point would be foolhardy, unless the new system would offer *substantial and unequivocal* aural improvements. Equally important, the next format must be relatively maintenance-free compared with today's systems.

To this end, allow me to present a wish-list of features, in rough order of importance:

- It should have eight channels.

Both of today's analog film recording mediums—35mm fullcoat in the studio, and 70mm magnetic prints in theaters—offer a maximum of six tracks. Looking to the future, there is no reason to pay lip service to six when eight is a nice round number in relation to both the stereo dubbing consoles built today (which have between 24 and 32 output buses), and multitrack digital recorders, which are available in 24- and 32-track formats.

There would be no problem in accommodating the 24 tracks needed to record the final stereo stems (eight each of dialogue, music and sound effects). The final mix would then be "print mastered" on the 8-track digital recorders, which presumably will be here one day. (For those studios that don't invest in expensive digital multitracks, I wonder how soon someone will "invent' 8-track 35mm mag and couple it with Dolby Spectral Recording.)

Although all of this indicates that available technology will easily support an 8-track system in the studio, the *reason* for choosing eight tracks lies in the experience that would be afforded moviegoers. The format that I am thinking of would use five full-frequency channels behind the screen, a dedicated subwoofer channel and two additional tracks to be discussed later. Consider the five behind-the-screen channels.

For the past nine years, almost all 70mm 6-track films have used the Dolby "baby boom" format, which employs three speakers behind the screen for full-range, left-center-right information (recorded on tracks 1, 3 and 5), one track for the surround speakers and two for bass-extension information (tracks 2 and 4). For the 21 years before 1977, 70mm prints used speakers 2 and 4 to reproduce full-range, left-center and right-center material for a total of five channels behind the screen.

A primary reason for the change was to lower effectively the frequency response of the Altec-Lansing Voice of the Theater speaker by doubling up bass information below 200Hz. (This matter is discussed in greater detail in "The Evolution and Utilization of 70mm Six-Track Film Sound," published in the April 1983 issue of **RE/P.**)

The reason for choosing eight tracks lies in the experience that would be afforded moviegoers.

In the past two years, however, the popularity of direct-radiator speakers, such as the JBL model 4675A, with their smoother low-frequency response, has eliminated whatever need there was for channels 2 and 4 to be used for bass extension. Furthermore, most of today's 70mm mixes hardly use the boom channels (tracks 2 and 4 have been known to be left *blank*). I've never figured out what mixers can do with the boom channels if no one on the screen is caught in a gun battle or a thunderstorm.

Contrast this situation to the obvious advantages of having all five speakers on-line: pans across the screen are smoother, with no jump between left and center, and center and right. The top end of the dynamic range is increased because five full-range speaker channels can play much louder than three.

Perhaps the most important use for speakers 2 and 4 would be to function as a narrow stereo pair, with the wide pair (1 and 5) perhaps brought in only for effect, as for big music stings. The resolution provided by a 5-channel recording of a large orchestra is much greater than that of a standard 3-track. The difference is *dramatic* and can be heard in the forms of the dozens of films mixed in the "discrete" 6-track at Todd-AO, MGM and Twentieth Century-Fox.

The dedicated subwoofer track of the proposed system would finally standardize the chaotic "boom" situation today. Some theaters are only equipped with speakers 2 and 4, while others also have a stack of subwoofers. (Dolby Cinema Processors have a separate subwoofer output.) The proposed system would give the film industry a chance to simplify and codify the "boom" usage: one track for material between 30Hz and 80Hz (?), to be played through subwoofers at X level in the middle of the theater. Thus, when someone on the screen is in danger, the boom reinforcement will be consistently thundering.

Whither stereo surrounds?

In today's standard formats, one channel is fed to all surround speakers in the theater; with split surrounds, each side wall would be given its own track. Although split surrounds are possible with 70mm 6-track "baby boom" Dolby Stereo, their use precludes having five discrete channels behind the screen.

However, in this proposed 8-track format, stereo surrounds could be had without sacrificing discrete 5-channel information. Although there are other possibilities for the two remaining tracks—they could be assigned as one surround channel and one vertical channel situated at the top of screen center—most people would first think of stereo surrounds. I, for one, am not sure that they are worth the effort.

One reason I look askance at the use of split surrounds is because of what I consider to be a Catch-22: There are very few opportunities in most films to use two channels in the auditorium, and when they *do* arise, the chances are that the stereo effect is distracting and head-turning. I think sound editors and re-recording mixers have to guard against doing neat tricks in the surrounds (mono or stereo) just for the sake of doing something neat.

Again, a Catch-22: If the stereo surround field is subtle, as heard by attentive ears in the quiet of the dubbing stage, the chances are that it won't be noticeable in a noisy theater. Stereo surround movement that can cut through the "popcorn noise" will probably be distracting.

Having seen several films in Dolby 70mm split-surround format (including *Apocalypse Now, Pink Floyd The Wall,*

Explorers and *Top Gun*—fine mixes all) I'd be willing to bet that one could A/B *imperceptibly* between mono and stereo surrounds for 75% of the running time of all the above-named films.

- It should be digital and maintenance-free. Also, theaters that install equipment must be carefully policed.

The value of good presentation in the public's collective eye should not be taken lightly. Because of the hype that has preceded both the exhibition of allegedly "all-digital" and normal 70mm Dolby 6-track films, the playback of said 8-track format must be the equal to the care that will be placed in the creation of the track.

Also, just as I think that the primary benefits of digital sound recording will be filmmaker-specific—movies will be more fun and less of a pain to make when picture and sound is edited with random-access hardware—the benefits of digital sound in theaters will not totally be sound quality.

If quality were the only issue, then a format using Dolby SR would be a serious contender to anything that digital would have to offer. But SR is still analog, and film sound people long for the peace of mind that azimuth, EQ, striping quality, dropouts, and so one, are no longer agents of Murphy's law.

Another level of peace of mind could be added if there is some degree of assurance that the theaters are being looked after. Today, some films are released with more than 200 70mm prints, with the Los Angeles and New York metropolitan areas each receiving more than 10. There is less attention paid to each installation, and I've often seen bad presentations given to big films on opening weekend in Westwood, the film-going capital of Los Angeles and the United States. There is no excuse for this.

I think that both filmmakers and film-goers would benefit if the number of 70mm prints—and those in the new, improved format of tomorrow—were purposely limited to a few theaters in each large city. And when the new format is introduced, only theaters that meet strict acoustical and electrical standards would be allowed to install the equipment.

I am aware that there are myriad problems involved with policing theaters (who sets the standards, how often are theaters checked and by whom are only a few potential headaches). Nevertheless, the film community should seize this rare opportunity to improve film sound reproduction in the ears, minds and wallets of moviegoers.

- The format should be the same in both 70mm and 35mm.

The main reason 70mm is used today is because of its six magnetic tracks. For films that are originally photographed in 35mm, there are—no matter what you have been lead to believe—no important visual perks. Nor is the image "wider." In fact, because of the lack of use of boom channels in many supposedly 6-track mixes, a 35mm 4-track magnetic print would sound almost indistinguishable from a 70mm print, at about half the cost.

If the next format were offered on 35mm prints, then distributors could equip theaters with the money saved by not having to make 70mm prints, which each cost $10,000 more than 35mm optical prints. If they don't just install the systems for free, then they could lease them to exhibitors at bargain prices.

You might have noticed that I have not cited this dream format as a replacement for 35mm Dolby Stereo optical. There is no denying that the matrix places level and panning constraints on re-recording mixers. Although many clamor for a replacement, I think the film industry must respect the fact that thousands of theaters can play 35mm Dolby Stereo *today*. And there's no reason why it can't coexist with the whiz-bang format of tomorrow, thereby continuing the flow of stereo films to the Spearfish, South Dakotas, of the world that would otherwise have to be content with Academy mono.

It should be clear that I'm only commenting on the design of the format, not how the information will get on the print in the first place. (I am intentionally ruling out interlock presentation.) Despite much talk from companies proposing new formats, nothing has been heard.

What about today, while waiting for the next system? I think that a Dolby-encoded, 70mm discrete 6-track would offer superb reproduction that is largely indistinguishable from the dream format described above. (The cost and peace of mind problems of 70mm mag-striped prints still remain, however.) The discrete format would be more accepted by re-recording mixers and directors, if they only tried it and heard it for themselves. R·E/P

Blake is RE/P's film sound consulting editor.

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Circle (13) on Rapid Facts Card

Technology

By Stephen St. Croix

Now that 90dB audio is finally in the hands of the consumer, I guess we had better get it into the hands of those of us who create the product.

Whatever you may think of the way CDs sound, all music eventually will be presented to the public primarily in that or a related digital form. The marketing advantages far outweigh the remaining audio problems and even the current production complications. Advancement of the art promises to overcome the current audio shortcomings. After all, we already enjoy a 90dB+ dynamic range with ruler-flat frequency response, which was unheard of just a few years ago.

So on what devices or systems will we be recording all our wonderful future audio? Digital multitracks can do it now, for those with the budget. But is this the answer? Stationary-head, longitudinal linear PCM recording is interesting, not because it works, but because it was never really *designed*; it just sort of evolved. It operates in emulation of analog, often on transports actually designed for analog recording.

It seems that Dolby's new Spectral Recording (SR) process just may be slick enough to dent the rush to the digital multitrack, but digital tape recording is still where it will eventually end up. . . or is it?

Analog multitracks have one significant advantage right now: we all have them. Actually, there are other advantages: we all know how to use them, we know how to edit and punch on them and many of them are actually paid for.

When the concept of digital multitrack recorders first became the buzz, many of us felt that it could not be done at all. Others felt that it could not be done right. Some said it could never be done for a realistic price. Others went ahead and built them.

The fact is that these machines were built in the image of the old-standard analogs for a couple of reasons. First, it *seemed* the logical way, because everyone felt at home around analog transports. Second, limitations in the technology/dollar equation demanded that the machine be ready to store data immediately after A/D conversion. Because large RAM buffers were not realistic, something had to be waiting to take the data right away. The choice of tape was an obvious one. But it is being strained even now. A considerable portion of the data stream is committed to error management, and any trend to raise the sample rate would only aggravate things further.

Is tape the only answer?

Much of the potential of PCM recording is wasted with tape. Just because we are used to the limitation of serial-data storage for audio today, should we again impose this limitation on ourselves? If you were designing a new and better recording system would you be afraid to make it a little different than the present one, if those differences allowed vastly superior performance? Would you be satisfied if your computer only had serial data storage?

Some say that RAM-based recorders (RAMRECs?) are impossible. Nothing is impossible.

Those of us old enough to remember micros that only offered cassette storage had very little trouble learning the new disk-based storage system, once we saw the advantages.

Although a true, high-performance disk-based multitrack recorder seems difficult to build now, remember how short the time was from when you first heard of digital delay to the amazing processors available today. As memory prices drop and packaging density increases, the buffers for such a system become reasonable. As better cylindrical storage is developed, both on- and off-line capabilities become realistic for audio.

By why stop there? Isn't this merely the next step in evolution after tape? What about the real jump: massive online RAM storage. Some say that RAM-based recorders (RAMRECs?) are impossible. Nothing is impossible. (It used to be that some stuff was impossible, but now its only "not yet ready.") Some people say that RAM-based systems are silly ("who needs that kind of speed and power?"), impractical or too expensive.

Who needs speed and power? Come on, guys. We all do twice as many MIPS (million instructions per second) for half the price today as we did two years ago. We like it. We need it. With such processing power we do more work of higher quality per day every year.

RAM prices continue to fall as radical improvements in density continue. Single electron memory, interference (holographic) cube and even biological memory are just a few of the technologies being actively pursued today for the memory of tomorrow. Very soon, multiple processor systems will allow us to use this massive high-speed memory in new ways for audio.

Combined with on-board digital signal processing, these RAMRECs could be entire studios offering user-definable "everything." Want 400 takes (tracks) of that vocal to choose from? How about trimming that 40-track, 67-second spot to 60 seconds or less, without pitch shift or any side effects at all other than tempo shift? Or listening to a 24-track tune in all possible keys as fast as you can ask for it, without having to first mix down to 2-track?

With an interface concept perhaps modeled after a word processor, with features such as block move, block copy, search and change, find next, go to and undo, the state of music, film and video editing would be improved beyond the wildest dreams of today's engineers and producers. (Imagine a sampler based on such a system.)

At this writing, at least nine companies are working steadily towards making it impossible for you to resist playing Tron with your audio. They are designing RAM-based editors, computers that allow you to digitize, get into the data and stay in RAM until it's over, no matter what you want to try.

With a little planning in the area of standardization you could, of course, go through the entire recording chain without ever recording on analog. But it could also be done without *any* intermediate conversions. Music would see one stage of A/D, but all subsequent experimentation, signal processing, editing, mixdowns and generations would be internal or direct D/D, all the way to the final product. The very first D/A conversion would take place in the consumer's home.

You know what access to first-generation digital product that hasn't been through multiple format conversions will get us? Six months after this is achieved, every kid in America will expect it in every CD. R·E/P

St. Croix, **RE/P**'s technology developments consulting editor, is president of Lightning Studios and Marshall Electronic, Baltimore.

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Circle (14) on Rapid Facts Card

SPARS On-line

By Gary Helmers

The primary mission of SPARS, the Society of Professional Audio Recording Studios, is to encourage and facilitate communication between end users and manufacturers of studio equipment; among the owners of production studios; between the studio employer and the educational institutions preparing future employees; with other trade organizations representing the industry; and with local, state and federal governmental agencies that affect the production studio industry.

During the past few years, our communication with state agencies has concentrated on state sales tax issues as they apply to the professional production studio. Facilities in many states are facing difficulties in the interpretation and application of sales tax regulations. The fundamental difficulty in each state is defining what a professional studio does.

Do we engage in light manufacturing? Do we rent space or equipment? Do we sell time? Do we provide a service?

The answer is that the studio, first and foremost, provides a *service*. Like doctors and lawyers, the studio should not be required to charge sales tax on those services. Sales tax should apply only to those products a studio sells—raw tape stock.

The clarification of sales tax regulations is a task we pursue for our clients, and for the health of the industry. The studio has no financial interest in this matter; we collect sales tax from our clients and remit those funds to the tax authorities. Our primary policy must be: When in doubt, collect sales tax.

If, upon audit, it is discovered that a studio has been remiss in collecting sales tax, the studio is liable for that uncollected amount.

It is unlikely that any client would respond positively to a request for payment of sales tax that was not on the original bill. On the other hand, it is possible to receive a refund for sales tax erroneously collected and paid to the state. Not easy—just possible. Of course, that refund is then returned to the client.

The studio has a legal responsibility to be the tax collector, and is not responsible for judging the validity of resale certificates or other certificates of tax exemption issued by the state. However, we should be able to advise our clients as to the proper use of such certificates.

SPARS exists to help in these situations, and we are working hard to provide assistance in every state where there is a problem:

The clarification of sales tax regulations is a task we pursue for the health of the industry.

• *California*—The California Entertainment Organization (CEO) was instrumental in changing the state sales tax law. For more than two years following the law's passage, the CEO and SPARS worked to see that the regulation was written to interpret that law. Before the April 9, 1985, adoption of Sales and Use Tax Regulation 1527, auditors were interpreting at will. SPARS continues to work for clarification of items within that regulation. In most situations, sales tax should be charged on the sale of raw stock only. We are working with the State Board of Equalization to produce a "Tax Tips" pamphlet for production studios.

• *Florida*—At this time, studios are only charging sales tax on the sale of raw stock. Additionally, studios are not required to pay sales tax on equipment purchases. This is the result of a temporary regulation that may not continue, and may not apply to all situations.

• *Tennessee*—In August 1985, the Tennessee Department of Revenue proposed to extend the 5.5% state sales tax to include production studios. A coalition of music industry trade organizations (including SPARS) was able to quickly demonstrate the disastrous effect such a move would have on the Tennessee entertainment industry. The proposal was withdrawn in less than a week.

• *New York*—Possibly the most confusing regulations we have had to deal with exist in New York. For those studios within New York City the confusion is doubled. In meetings with state and local taxing authorities, it has been agreed that SPARS and the taxing authority will work together to produce a booklet (more like a novel) that will clarify the issues. We have requested that all New York studios send tax questions to SPARS for use in the compilation of this booklet.

Studios in many other states are facing some level of sales tax difficulty. If you are confused, proceed as follows:

1.) When in doubt, collect.
2.) Rely only on written opinions addressed to you or the industry at large. Verbal opinions, or written opinions addressed to another studio, are unsatisfactory.
3.) Contact your trade organization. We can help.
4.) When in doubt, collect.

We need your input and your support if we are to continue addressing the sales tax issue and other concerns of the professional studio industry. Even if you are not a member, we will help.

Contact the SPARS national office at Box 11333, Beverly Hills, CA 90213; 213-466-1244. R·E/P

Helmers is the executive director of SPARS.

SPARS
SOCIETY OF PROFESSIONAL AUDIO RECORDING STUDIOS

Home of the Brave cannot be described as a standard concert film, in that it doesn't document a live performance unfolding in real time before an audience and a battery of cameras. Instead, only three 35mm cameras were used, with setups carefully planned to incorporate Anderson's usual arsenal of slides, film and props. The crew would shoot many takes, sometimes without the audience present.

By the end of the 10-day shoot in July 1985, at the Park Theater, Union City, NJ, the show had been performed approximately three times. Although this might seem like a long schedule for a concert film, it is extremely short for a 90-minute feature.

Because of the many takes involved and Anderson's use of "conceptual click tracks" during concerts, the recording process started by prerecording, onto a Sony PCM-3324 digital multitrack, a set of basic tracks with the band that had accompanied Anderson on tour the previous year. The prerecorded tracks were referred to as the A-reels in the soundteam's flow chart.

The only change in the band lineup was the substitution of Adrian Belew for guitarist Sid McGuiness. (Belew had played on the *Mister Heartbreak* album.) The rest of the band was comprised of Joy Askew on synthesizers and Moog bass, David Van Tieghem on drums, Richard Landry on brass and woodwinds and Janice Pendarvis and Dolette McDonald on backup vocals.

Overseeing the New England Digital Synclavier II synthesizer during the shoot was Bob Bielecki, who had recorded the *United States Live* album. In his role as Anderson's "electronic effects designer," Bielecki is responsible for many of her best-known devices, including the "tape bow violin," the "drum suit," the "talking book" and the neon mic stands.

Sessions in New York lasted for two weeks during February 1985, first at Sync Sound and later at Blue Rock Studios.

Although some of the material was to be used as a playback track—to ensure that different takes would match in tempo and could be intercut—a conventional click track was not prepared. For example, on the song "Kokoku" the band plays to a sample of the word "shake" cycling around in an Eventide Harmonizer set to random mode. During these preproduction sessions, Anderson laid down scratch vocals not only as a guide to see how the rehearsal was sounding, but also to help her find her place during pickup shots.

Prior to the shoot, a few songs were edited by bouncing from one digital multitrack to a second, with the resulting edited 24-track tapes labeled A-1 (see Figure 1). The A and A-1 tapes were mixed down to a 4-track format on a Sony PCM-F1-encoded BVU-800 ¾-inch U-Matic VCR, using the two analog audio tracks in addition to two digital channels. SMPTE time code from the digital multitrack tape was regenerated onto the videocassette's address track.

Ungar says that "anything that we thought we might want to hear in the monitors during the actual performance was put on the digital tracks of the cassette. On the [linear] audio tracks of lower quality, we put the scratch vocals and place-keeping material."

Live performance shoot

Ungar had her hands full during the concert shoot. As well as recording the live tracks on a Sony PCM-3324 (the B-reels), she was also mixing house sound and simultaneously making a mono scratch mix for use during film editing. While this arrangement obviously precluded the need for a separate mobile truck, it did require the setting up of a full recording studio, centered on a Soundcraft 32-input TS-24 console located in the rear of the Park Theater. David Hewitt of Digital Services, Houston, assisted Baran in obtaining and setting up the equipment.

Baran and Ungar felt comfortable with this setup not only because Ungar was used to that point of view—having mixed house sound on the *Mister Heartbreak* tour—but also because it was important for Baran to be "in the thick of things."

Baran says that "it was a strange hybrid between a PA setup and a real recording studio." She was pleased that the Soundcraft, which was brand new and on its way to its owner, proved to be a "nice and clean-sounding console," and says that had the project not been all digital, the choice of consoles would not have been so critical.

Headphone monitoring was necessary during a take, after which playbacks could be conducted with either UREI 813B Time Align or Yamaha NS-10M speakers. Baran says that the audiences were "very patient during the shooting and playbacks. They understood that they were not just there to see a concert; they were part of a larger process."

The TS-24's main outputs fed the Nagra/headphone mix, the auxiliary sends being used for the PA mix, a submix of LinnDrums and Simmons drum tracks, and a submix of processed live vocal microphones that Ungar fed to monitor mixer Danny Caccavo. Onstage, Anderson had two Eventide Harmonizers (models H910 and H949); the monitor mixer received a post-fader mix

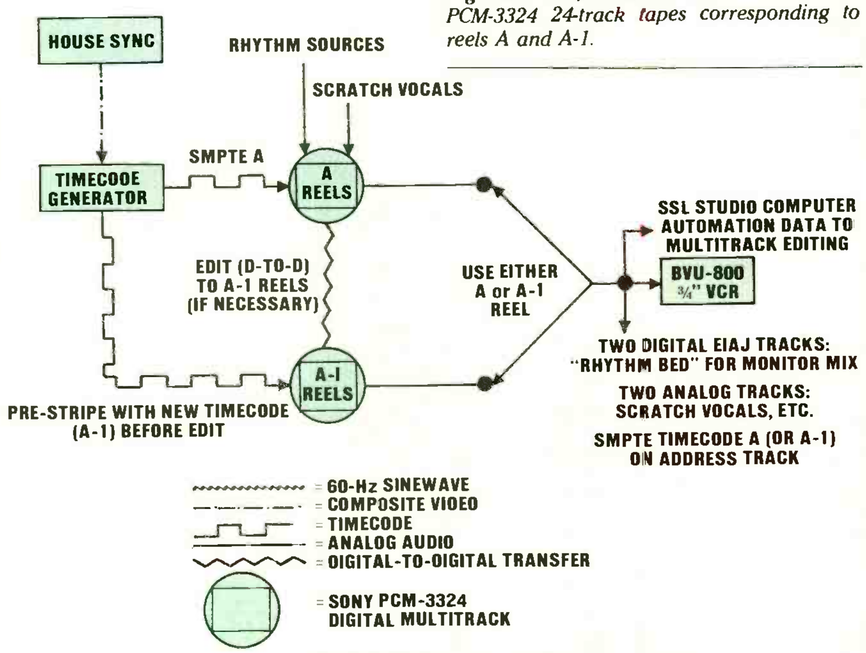

Figure 1. *Preproduction flow chart of edited PCM-3324 24-track tapes corresponding to reels A and A-1.*

Photo by Sandra Ray

Consultant Bill Marino (left) of Sync Sound and soundtrack co-producer Roma Baran at a pre-production meeting.

from the house board to ensure that the processing didn't appear in the monitors before it was in the house PA. Caccavo, in turn, sent Ungar the four playback tracks so that she could add them to the Nagra scratch mix and onto one track of the 3324.

In the best music video fashion, smoke was used for atmosphere during some scenes. Because particulate matter in the water-based smoke presented a potential problem for the digital multitrack, the prop department built a protective Plexiglas dust cover, lining the edges with foam. During reel changes, the doors directly behind the recording equipment were opened to blow smoke away from the machine. The open doors also helped cool the equipment and crew, as the temperature in the brightly lit theater reached 100 degrees.

Synchronization

A single sync source guaranteed that all audio and video transports—the digital multitrack, the playback U-matic VCR and the Nagra IV-S recording the dailies scratch mix—exactly followed the same 60Hz reference sync. The complex synchronization scheme was devised by consultants Marino of Sync Sound and David Smith of Editel; during filming the system was monitored by Bob Bielecki (see Figure 2). Gus Skinnis of Sony provided additional digital consulting.

Smith says that, with this project, "10 pounds of prevention was used, as opposed to an ounce. We didn't want to hear about *anything* going wrong, and we broke our butts."

Baran adds that the "10 pounds" were there "if only to prevent the 90-person crew from waiting for sound for even as much as 10 minutes."

An example of the kind of care being paid to the synchronization process, Marino checked many 60Hz crystals for accuracy relative to Sync Sound's master generator. The winner of the bake-off was a humble Nagra III external crystal. The output from the crystal was buffered and fed to a distribution amplifier, which in turn fed it to three destinations: the projector (see below), the sync input of the dailies Nagra and a Grass Valley 900 series sync generator that generated the 60Hz composite video sync. In turn, the video sync was distributed to the composite sync inputs of the playback BVU-800 VCR and the PCM-3324.

The 60Hz video was also fed to an Adams-Smith LG2600 time code generator that sent 60Hz/30-frame non-drop code to track 2 of the Nagra IV-S, the 3324's time code channel and two large time code display clocks for camera slating. (Because the large readouts resembled basketball scoreboards, the crew taped "Home" over one and "Visitors" over the other.)

Standard synchronization procedures for concert films entail each camera shooting time code (usually character-generated on a small CRT) until it is up to speed and then panning over to the performance. The process must be repeated if the operator stops the camera for any reason.

For *Home of the Brave*, tail slates were used, with the cameras shooting the clock *after* the take but *before* they stopped rolling. The reason for this approach was primarily logistics, because many camera setups were carefully framed and focused in regard to the stage and the process screen. Having to shoot the time code display and then pan over and refocus before the shot was considered too much trouble.

Also, starting each take required rolling the cued playback BVU-800 and recording with the 3324 and the stereo Nagra, not to mention sometimes start-

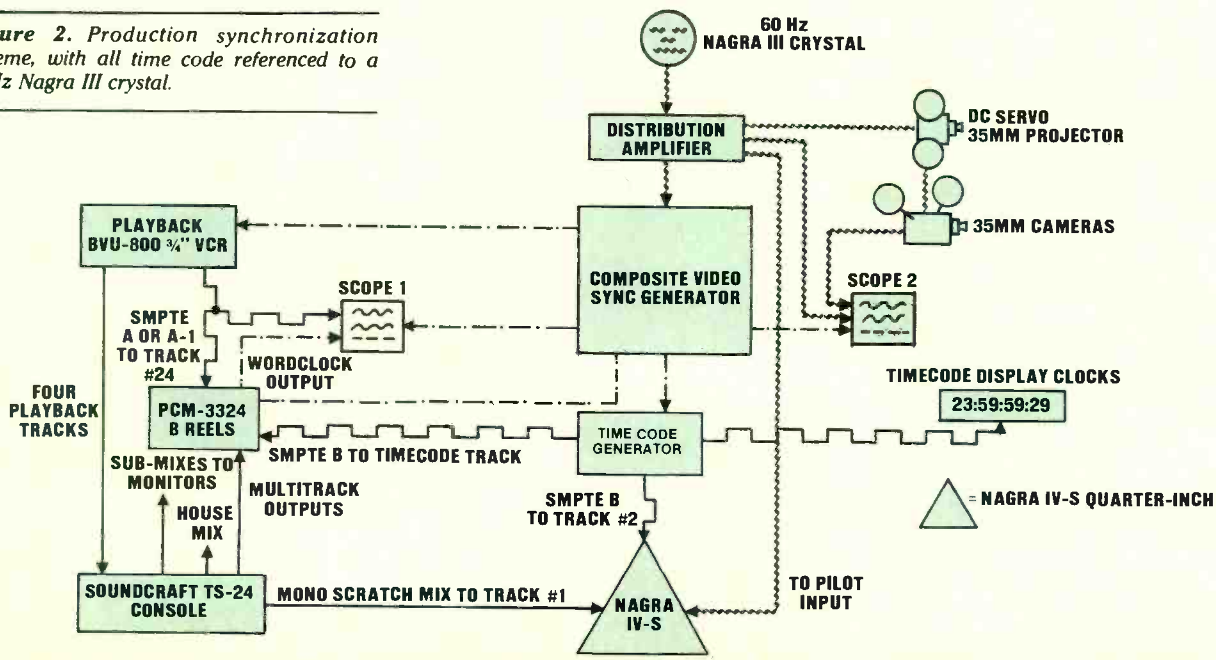

Figure 2. *Production synchronization scheme, with all time code referenced to a 60Hz Nagra III crystal.*

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ing 35mm film and slide projectors. The use of tail slates meant there was less chance of the whole operation grinding to a halt before its time.

In order to keep the shutters of the three 35mm Panavision cameras in phaselock with the shutters on the process projector during the shoot, director of photography John Lindley, used a Panavision shutter control device. This box makes the camera shutter follow the projector and is used during process photography with rear or front projection.

To help the crew keep an eye on sync, two dual-trace oscilloscopes were mounted on top of the Soundcraft console. The 60Hz sync was sent from the projector to camera 1, and then daisy-chained to cameras 2 and 3. The sync output from camera 3, in turn, was sent to a scope. If any camera went down or went off speed, the crew was aware of this immediately. (Sometimes only one camera was connected to the projector.)

On the other trace of the same scope was displayed the 60Hz output from the master crystal, allowing the crew to monitor the beginning and end of the process. Whenever there was no film projection, the cameras were freed from their sync cabling and allowed to run "wild" on their own accurate crystals.

One trace of the second scope was used to display time code from the BVU-800 playback deck, with the 3324's 44.1kHz square wave word clock output on the other. The trigger input of both scopes was the 60Hz video signal from the sync generator, and *not* 60Hz sine wave from the crystal.

According to Smith, "the video triggers a lot better than the [60Hz] sinusoid because of the fact that it has a fast rising edge to it."

The 3324 performed flawlessly during the shoot, and Smith notes that "the only thing you had to be really careful about is the lockup, that the machine was running synchronously [on the external time base.] The machine has a very slow time constant; in other words, if its video reference should jitter, you don't want the machine to suddenly change speed or it will mute [because the sampling frequency, which tracks the video reference, will fall outside its nominal tolerance window for reliable D-to-A output].

"So you have to be careful about the stability of the video time base and constantly check that the lock between the 60 cycles and the video is *extremely* well maintained.

"Remember that video [sync] is divided down from a 31.5kHz square wave that's the time base for all video, not including the color information. Because you're multiplying the 60Hz crystal by 525, any jitter in the 60 would be immense in video, and would cause the 3324 to lose lock. You *have* to make sure that in the event it does lose lock, you don't get a mute or a major speed variation."

Picture editing

In light of the existence of the Nagra IV-S TC tape machine, which records time code on a center track, an observant reader might wonder why this model was not used, because it would allow a *stereo* scratch mix to be recorded. The answer is simple: By using a standard stereo Nagra IV-S, ¼-inch to 35mm mag transfers could be done at any transfer house. Center-track time code is still not commonplace. Additional reasons were that only a mono mix was needed and that this IV-S was the machine owned by Jim Anderson, who oversaw the Nagra and the dailies transfers.

However, while the ¼-inch playback portion of their dailies transfers was standard, the 35mm mag portion was not. In order to have time code recorded on the mag film without going to the considerable expense of making 35mm fullcoat transfers, the crew used 35mm mag heads that allowed time code to be recorded on the usually blank balance stripe of 35mm single-stripe.

The mono guide mix recorded on track 1 of the Nagra was placed on the standard 200-mil track. (Single-stripe 35mm mag is also much easier to handle and splice than fullcoat.)

All picture editing was done on 35mm film by Lisa Day, who previously had cut the concert films *Let's Spend the Night Together* and *Stop Making Sense.* Day says that although film was chosen over videotape, primarily because of budgetary considerations, she missed the speed inherent to the off-line video editing process. (Editing on tape would have presented the filmmakers with another technical problem: relating the in/out time code numbers of the edit decision list to the workprint.)

When picture editing was finished, the next step was to assembly edit the A (playback) and B (live) digital multitrack

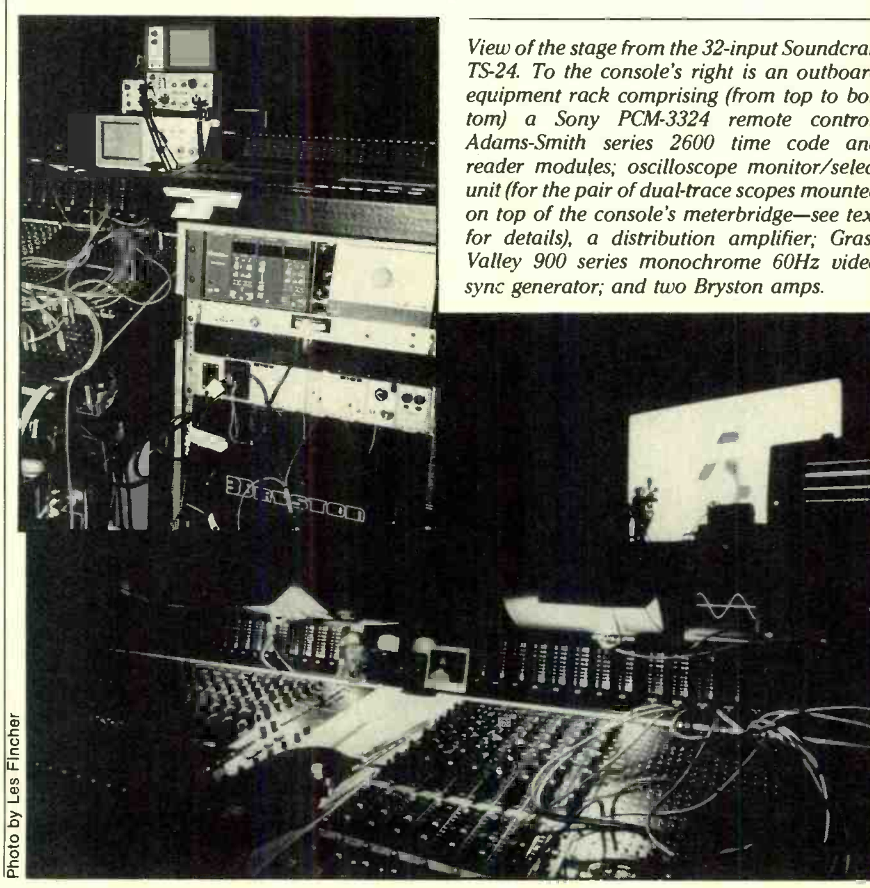

Photo by Les Fincher

View of the stage from the 32-input Soundcraft TS-24. To the console's right is an outboard equipment rack comprising (from top to bottom) a Sony PCM-3324 remote control; Adams-Smith series 2600 time code and reader modules; oscilloscope monitor/select unit (for the pair of dual-trace scopes mounted on top of the console's meterbridge—see text for details), a distribution amplifier; Grass Valley 900 series monochrome 60Hz video sync generator; and two Bryston amps.

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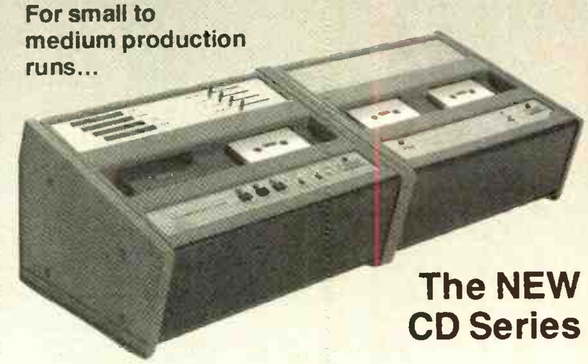

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Roma Baran (left) and Leanne Ungar behind the Soundcraft TS-24 console during the live concert shoot at Park Theater, Union City, NJ. In the rear is Jim Anderson, checking the Nagra IV-S recorder.

tapes to create an edited 24-track C-reel conforming to the 10-minute film-edit reels. Day's editing assistant, Lori Mazilo, took apart her audio "worktrack" (the roll of 35mm mag stripe with edits) at every splice, attached yellow leader to the ends and ran each edit through a time code reader. The start/stop points of each edit were written up as an audio EDL for use during the multitrack conforming stage.

In preparation for multitrack editing and mixing at Sync Sound in New York, the film workprint and audio worktrack were transferred to 1-inch videotape. Three windows were burned into the black areas created by the hard-matting of the film's 1.85:1 aspect ratio onto the standard 1.33:1 TV picture.

One window displayed 35mm feet/frames, with another time code (non-drop 59.94Hz/29.97 fps) starting at 01:00:00 for reel 1, 02:00:00 for reel 2, etc. These two windows began at the standard 12-foot (8-second) picture start frame.

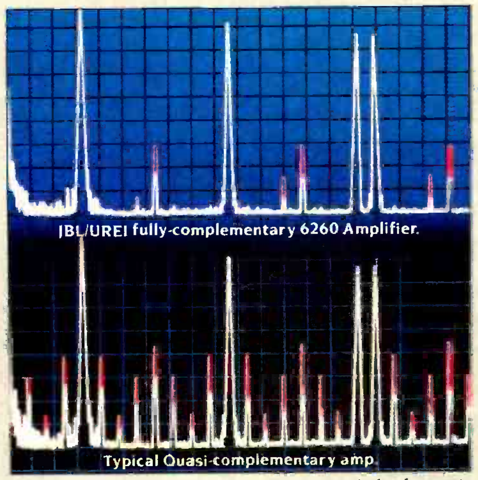
The third window displayed the edited time code from the 35mm balance stripe. The film editors placed a diagonal line across the picture frame corresponding to each sound edit, to assist the sound crew in locating the edits that were not "straight across"—i.e., at the same place on picture and sound.

The guide track was recorded on track 1 of the 1-inch videotape, with the new time code for the reel recorded on track 2. This tape was used later at Sync Sound to "jam sync" and pre-stripe the time code and guide track on the multitrack C-reels. Also, two U-matic dubs were made for use in post-production interlock. R·E/P

Part two of this article, to be published in the October issue, will cover the multitrack music editing and mixing stages of *Home of the Brave*, as well as providing details of the Dolby Stereo print mastering and 2-track soundtrack album mix.

Blake, **RE/P**'s film sound consultant editor, is a regular contributor.

WHAT TO LOOK FOR WHEN YOU LISTEN TO A POWER AMPLIFIER.



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Digital Audio for Videocassette Duplication

By Everett M. Carroll III

What video-based and stationary-head transports are available, and what system is best suited to videocassette duplicators?

Videotape recorders traditionally have included audio recording capabilities only as a necessary evil. Then came stereo VCRs with Dolby noise reduction, and people started plugging them into decent audio systems. But, before these Dolby-equipped stereo machines could become really popular in the marketplace, Beta and VHS Hi-Fi machines were introduced, boasting performance nearly as good as a digital recorder.

As a result, professional videocassette duplicators had to take a good, hard look at how to make audio for our programs that would be worthy of the better home systems now available.

This article will describe some of the ways in which audio reaches the finished videocassette, and some of the criteria used to select a digital audio processor for videocassette duplication.

First, a look at how most programs are now mastered for videocassette duplication. Most professional duplicators use either 1-inch C-format VTRs or ¾-inch U-matic format VCRs. One-inch machines have reasonably good audio performance, and the industry seems to be moving rapidly toward a Dolby-A encoding as standard practice. A properly aligned 1-inch VTR will offer a flat frequency response out to about 22kHz. But even with noise reduction, these machines have as much as 15dB less dynamic range than a domestic VHS Hi-Fi VCR and produce significant amounts of wow and flutter.

U-matic VCRs are less able to produce good quality audio. They run at much slower tape speeds and, therefore, have more trouble with noise. They also pose limitations on usable bandwidth, have more wow and flutter, and exhibit phase instabilities between the stereo pair of tracks.

With digital audio, dropouts can be corrected up to a certain size (the same holds true for digital video), and a good error-correction scheme can take care of problems caused by normal aging or even some cases of more severe damage. But analog techniques as used in both the video and FM audio systems of the "Super" ½-inch Beta and VHS machines are sensitive to even small dropouts, and therefore the tapes will show the effects of aging.

In general, the narrower the tape, the more problems can be expected with dropouts and physical damage, and the sooner any particular tape will show these problems because of normal wear and/or abuse.

Crest National Video provides duplication services for both VHS and Beta videocassettes. With an operator handling only loading and unloading of cassettes, multiple machines can be chained together for high-volume turnaround.

Some other formats used in mastering include 2-inch Quadraplex (rapidly losing ground because of high tape cost and high maintenance requirements); 1-inch B-format (not widely used in the United States); and ½-inch Beta, VHS and the M-II format.

The idea of using a well-kept ½-inch VHS Hi-Fi VCR as a mastering machine is interesting, but even if the audio performance is good the video after two generations of ½-inch videotape could be nothing short of terrible. I suppose you also should consider the new "Super" Beta and HQ VHS machines for this purpose, but the quality of the video would be noticeably lower than the larger-format machines and the tapes would age more quickly.

In addition to the sources listed above, several duplicators are using digital audio masters locked to video with a time code track, a technique that can produce truly spectacular results with a VHS Hi-Fi VCR. The added dynamic range of a good digital audio master is perfect for augmenting the effect of a dramatic thriller or keeping audiophiles content with musically oriented soundtracks. It seems likely that this method, or another yielding similar quality, eventually will become the standard for professional videocassette duplicators.

Purchasing considerations

All of which brings up this question: Which digital audio format makes sense for duplicators? Some of the considera-

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DYNAMIC RANGE is the spread between maximum output level (MOL) and noise (tape hiss). It is a major criterion of tape quality because it shows the true capacity for music. Tapes with high output and high levels of tape hiss are really no better than low output tapes with low noise. It's the difference between output and noise that matters.

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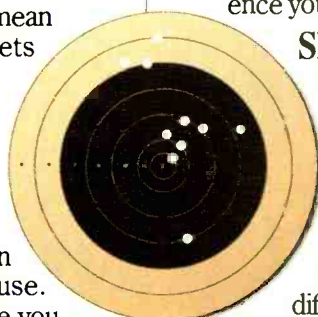

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Chrome on the range.

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tions a person must keep in mind when looking for a digital audio processor suitable for professional videocassette duplication are:

- It is often necessary to have more than two hours of recording time available, sometimes as much as three full hours for a feature film.
- High reliability is a must, because the master will need to be run quite a few passes—perhaps more than 100. (Small duplicators have more trouble in this area because more passes are necessary to fill a large order with their smaller inventory of duplicating slaves.)
- The ability to reliably interchange tapes between different playback locations. For a large videocassette duplicating facility, which has several facilities located in different parts of the country, it is necessary to be able to send a master to any location or to receive a master from any outside suppliers without encountering interchangeability problems such as phantom dropouts or glitches.
- The ability to perform basic frame-accurate editing (not just assembly).
- Cost-effectiveness.

Reliability criteria

When I was first approached about the possibility of using a professional 16-bit system with ½-inch VHS tape (which was considered necesssary to provide the 2- to 3-hour replay time for film soundtracks), I was concerned about potential reliability problems because of dropouts, aging or tape damage. As a result, I set up a thorough torture test.

Starting with a cheap VHS-format tape, a short segment of program was recorded. The tape then was put into an endless loop on a JVC BR-7000-UR Hi-Fi duplicator, and a counter attached to accurately record the number of passes. After 450 passes no additional indication of cyclic redundancy check errors was apparent. Just to make things more interesting, the tape was deliberately damaged.

In one section of tape I created a longitudinal as well as a transverse crease, actually causing a small area of the tape to completely lose its oxide. At this point the tape was played back, and although an error severe enough to cause interpolation occurred (interpolation is a form of error correction that replaces data that have been completely lost with calculated data), the problem was virtually inaudible. Perhaps in direct comparison with a clean copy of the same material the difference would be obvious, but I could not hear it at all.

The tape then was placed in an endless loop again and checked after every 200 passes. After 2,176 passes I stopped the test. The amplitude of the RF envelope on the tape was only about 50% of its original level (indicating that the tape was severely beaten by now), but the program played flawlessly, including the portion of tape that was without oxide.

There are many possibilities for setting up a digital audio system for use in video duplication. Some of these are listed below, along with comments concerning their suitability for use in professional duplication.

Professional video-based systems

• *Sony PCM-1610/1630.* These digital processors provide a 2-hour capability only when used with a 1-inch VTR. (It is not possible to record two hours or more on ¾-inch U-matic format and the process is not sufficiently reliable when used with a ½-inch VCR.) The system is somewhat costly; a typical 1-inch VTR/TBC combination costs at least $50,000. Add a second VTR to ease dubbing of masters, and you have a substantial investment.

Furthermore, mainly due to minor format differences between machines offered by various manufacturers, the 1-inch C-format and PCM-1610 combination is not always reliable. I'm not saying this is a bad system—far from it. However, PCM-1610 tapes mastered out-of-house on other types of 1-inch C-format VTRs have had dropout problems more than just a few times.

The PCM-1610 does include a full list of professional features, including balanced inputs and outputs, with or without transformers. The system also offers simple editing capability with any editing VTR and comprehensive editing with Sony's DAE-1100 editor. The processor is being used now by several large duplicators.

Besides the 1610, Sony now is offering an upgraded version of this format, the PCM-1630. It offers oversampling, with newly designed analog and digital filtering—features that increase audio performance while maintaining compatibility with tapes produced on a 1610.

Other new features, RAW and RAR (read after write and read after read), also improve performance and confidence monitoring while recording. RAR can be achieved only in conjunction with Sony's new DMR-4000 U-matic recorder, which is equipped with an optional DABK-1630 interface board and has redundant data reading to improve playback reliability. RAW can be accomplished by using any recorder with video-confidence monitoring capability.

• *JCV VP90/900.* Also a professional system, the VP900 has features nearly identical to the Sony PCM-1610. The unit allows simple editing with any video editing machine and crossfade editing with JVC's AE-900V professional editor.

It also offers several other capabilities that make it almost ideal for videocassette duplicators. The system provides reliable operation using ½-inch videotape; the capability to digitally transfer directly from PCM-1610 format (using an optional FC-900 interface); and direct conversion from 14-bit, EIAJ-format masters using another optional interface available from JVC.

Because a videocassette duplicator is by definition likely to have many professional ½-inch VCRs available, the production of multiple dub masters for distribution is painless and cost-effective.

A central control module provides start, stop and rewind commands for dozens of slaved VCRs.

Within the facility's video control room, two interlocked Ampex VTRs provide picture playback, and the PCM audio.

After working with this system for more than a year, I have seen it have only one problem with a master. This was caused by a factory-defective tape that had a burr on an interior guide, causing a continuous longitudinal scratch. Even with this problem, the VP900 still made it through about 40 passes before occasional dropouts were able to get by its powerful error-correction algorithms.

• *dbx model 700 CPDM processor.* This is an interesting machine. It can be used with ½-inch videotape and is capable of frame-accurate editing with an editing VCR.

It is reasonably priced for a professional system. Something to keep in mind is at present a dbx 700 master cannot be transferred to any PCM system without going to analog somewhere in the process. It seems that this processor, although exhibiting great sonic performance, would not be the best choice for our application.

Semi-professional video-based systems

• *Sony PCM-F1.* The ubiquitous F-1 processor and its counterparts, the PCM-701, PCM-501 and the Nakamichi DMP-100, certainly have proven themselves to be star performers in the price/performance arena of professional digital recording. Never intended as a pro audio product, these EIAJ-standard machines boast more powerful error correction than the Sony PCM-1610 (presumably because the processors were intended for use with consumer VCRs). Other features include portability, 14/16-bit operation and compatibility with ½-inch VCRs, for extended recording time and reduced tape costs. These processors require modification to allow external video-sync operation, but this is a relatively easy task.

Even though these units (especially the new PCM-501) have fairly good error-correction capability, care is required in selecting tape stock. High-quality tape is necessary, especially in the 16-bit mode, which has reduced error-correction capability. (The 16-bit mode "borrows" data recording space within the standard 14-bit EIAJ format that normally would be used to extend error correction.)

Editing capability is another problem with these processors, although systems that will ease frame-accurate editing ostensibly are now available from at least two vendors. I don't know of a duplicator using any of these processors, but I am certain they could be integrated into a workable and cost-effective system.

• *JVC VP-100/101.* These processors are JVC's implementation of the 14-bit EIAJ standard. For the most part, the above description of the PCM F-1, *et al.*, also holds true for the VP-100/101.

These two processors are nearly identical, with the VP-101 being the "professional" version; dc coupling throughout is said to provide a clarity and impact that almost makes one forget that it is only a 14-bit recording. The noise floor is audible at high listening levels but is still better than a VHS Hi-Fi VCR or 1-inch C-format videotape. The VP-101 has a rather unusual capability in that it provides external sync capability. (What JVC means by this is a signal that can be supplied to the playback VCR to synchronize its servo system to the processor's word clock.) I'm not quite sure what this feature was intended for, but in any case modifying the VP-101 to lock to external video sync would not be too difficult. By the way, the VP-100 doesn't need this modification, because it has the ability to lock its internal clock to video being replayed from VCR.

• *Sansui X-1 Tricode.* Again, this is a 14-bit, EIAJ-standard machine and is subject to most of the same description as the above examples. The X-1 has been around for some time, and although it has an excellent error-correction system and good audio performance, I have not encountered many of them.

Digital stationary-head recorders

• *Sony PCM-3102/3202.* These two DASH-format, reel-to-reel machines—the 3102 runs at 7.5ips, and the 3202 Twin-DASH at 15ips—promise some nifty features for editing and post-production. They will be supplied with a built-in chase-lock synchronizer capable of locking to external time code, a feature that will save the expense of a separate synchronizer costing at least $10,000. Additional savings could be achieved if the cost of a DASH machine is less than the total cost of a "professional" video-based processor and a companion video machine.

But, for a duplicator, the PCM-3202 Twin-DASH system would restrict recording time to 90 minutes. Possibly as much as 130 minutes could be obtained with 14-inch reels if the 3202 is capable of handling them, but this would still be a limitation for some film soundtracks.

The DASH-S PCM-3102, running at half the speed of the 3202, is capable of the

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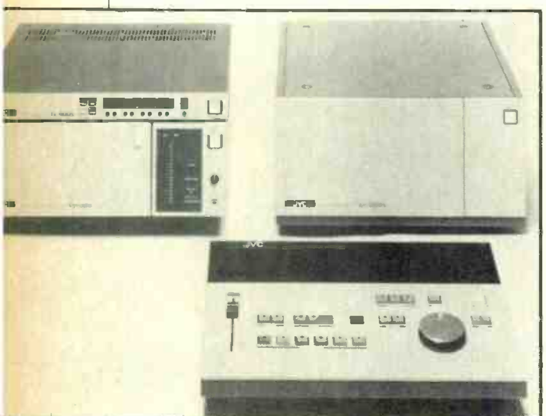
Sony PCM-1630 digital audio processor.

JVC digital audio system comprises a VP900 processor (top left), TC900V time code unit and AE900V editor.



required extended recording time, but its use raises the question of finding compatible masters. Unfortunately, the 3102 and 3202 use two entirely different recording formats; tapes produced on one machine will not be usable on the other. To my knowledge no dual-speed machine is planned by any of the DASH-aligned manufacturers.

All of which would make finding the appropriate DASH master confusing and possibly difficult. For a large company that has a need to distribute dub masters to different locations, at least two DASH transports would be required.

In contrast, a video-based digital audio system could make multiple, simultaneous dubs using VTRs and VCRs that form a normal part of the facility, and thus would not require purchasing additional equipment.

- *Studer D-820X-2.* Again a Twin-DASH machine, the D-820X-2 is limited to a recording time of approximately 130 minutes using 14-inch reels. Studer representatives have informed me that the company presently does not intend to offer a 7.5ips, DASH-S-format machine. However, the 820 also will be available with an optional chase-lock synchronizer and numerous features that would be dynamite in post-production editing.

The potential objections to using this machine for duplication are the same as those cited for the Sony DASH machine.

- *Mitsubishi X-80/86.* Mitsubishi's original stationary-head, 2-track machine, the X-80, has been in day-to-day use for quite a while now. The new PD format X-86 is incompatible with the X-80, although an option will be made available for X-86 users to replay X-80 tapes.

Both recorders have features similar to DASH machines, including razor blade editing and provisions for locking to various types of external sync (composite video, for example).

For duplication, the X-80 and X-86 pose the same kind of time limitation as DASH-format machines; with 10.5-inch reels you have only one hour of available recording time at 15ips.

Other formats of interest

- *Sony BVH-2800/2830.* This new 1-inch C-format VTR is based on the highly successful BVH-2000/2180 transport with the addition of PCM audio recording capability. The BVH-2800 offers two hours of recording time, and the 2830 offers three hours.

PCM data are recorded on the videotape using an area normally reserved for the vertical interval of an incoming video signal. The vertical-interval portion of a video signal generally is not needed by 1-inch C-format machines, because they are normally used with a digital time base corrector that replaces the vertical interval upon playback.

Although the two machines may be considered too expensive for many duplicators, the operational advantages of having both video and high-quality audio recorded on a single piece of tape, to be played back on a single machine, could offset the extra cost.

- *DVR-1000/DVPC-1000.* Yet another new product from Sony, the DVR-1000/DVPC-1000 is a component digital video/digital audio cassette recording system. It provides four channels of 16-bit linear PCM encoding at a sampling frequency of 48kHz and uses ¾-inch tape in a large cassette enclosure different from conventional U-matic housings.

This seems to be a high-quality system that would be well-suited for post-production editing and other applications involving multiple generations of tape between the original and end product. For duplicators (and probably quite a few post houses as well) the $140,000 cost of the system may be prohibitive. Its 76-minute time limitation also means that longer movies would require the coordination of more than one transport.

- *Panasonic AU-650.* This version of Panasonic's M-II format is not actually a digital machine, but I thought I'd sneak it in anyway. The AU-650 is a component video recorder that also features two tracks of FM audio. The FM audio system used in this machine, much like the VHS Hi-Fi available in consumer VCRs, offers more than 80dB of dynamic range and flat frequency response out past 20kHz. It performs well enough to produce good-quality dubs for the home video market and at less than $40,000 is a less expensive alternative than some others mentioned.

The AU-650 is capable of recording only about 90 minutes of audio and video, but considering the machine's cost, it may be worthwhile to use two units for longer movies. The quality of video recording also is very good with this system, and tape cost is lower than with 1-inch C-format machines.

All things considered, this M-II format machine opens up some interesting possibilities.

In closing, I would like to say that a workable audio/video lockup system could be set up with nearly any of the above-mentioned products. A 14-bit EIAJ processor could make a cost-effective system, while the stationary-head machines perhaps would offer compatibility with more audio studios or post-production houses that undoubtedly will find this format attractive.

The M-II format could be used as a cost-effective high-quality audio/video source, while the new digital video recorders and 1-inch C-format VTRs with digital audio probably will fill the high end of the market.

With care and proper integration of the equipment into your working area—including clean storage of tapes and recorders, quality tape stock, coordination of systems with outside suppliers of masters and so on—achieving a reliable, high-quality dubbing operation should not be a problem. R·E/P

Carroll is president of EMC Engineering Services. Previously, he was senior project engineer at VCA/Technicolor, a company that describes itself as possibly the world's largest duplicator of home and industrial videocassettes.

Crest Video photos by Elizabeth Annas.

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Ian Eales: Ian has served as technical audio consultant and as engineer for numerous rooms on the west coast. Ian's work includes David Foster's "Best Of Me" LP and Shiena Easton's "Kept Secret". Ian has worked extensively with Al Jarreau, including his latest album "High Crime".

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Circle (23) on Rapid Facts Card

Recording Joe Jackson Live to Digital

By Lauren Block

Producer David Kershenbaum details the recording of *Big World* at New York's Roundabout Theatre.

Many would consider that, in terms of breadth of recording and production experience, David Kershenbaum knows his field inside and out. During his many years in artist production, Kershenbaum has produced gold and platinum records with artists such as Joan Baez, Air Supply, Cat Stevens, Graham Parker, Eddie Money and Duran Duran.

During his six years at A&M Records, he spent three of them as vice president of the A&R department, where he was responsible for discovering and signing Joe Jackson. He produced the Jackson albums *Look Sharp!*, *I'm the Man*, the Grammy-nominated *Night and Day*, and *Body and Soul*.

Now an independent producer, Kershenbaum continues to work with Jackson. Early this year, the team recorded a direct-to-digital session at New York's Roundabout Theatre. Kershenbaum describes the resulting double album, *Big World*, as a "real achievement" for himself and engineer Michael Frondelli.

"Joe and I wanted to record direct-to-digital 2-track for a number of years, but we were looking for the right vehicle. We felt that this time the material had the kind of form where we could actually pull it off."

What made Kershenbaum choose to record Jackson direct-to-digital, when most sessions these days are firmly based on multitrack techniques?

"There were a couple of reasons for us-

ing digital 2-track for *Big World,*" he says. "One was there was a great risk involved; Joe and I wanted simply to prove at this point that we could do it. We were both sure that we could complete the project successfully—all we had to do was get all the pieces together.

"We also wanted to catch the spontaneity of the band playing with all that 'rawness' and 'energy.' We thought we would control the feel, without it sounding too contrived. We wanted the album to sound unique, and not like just another studio album; we wanted the live album to have its *own* sound."

Session planning

The chain of events leading up to this direct-to-digital session started in late 1984, at a Masonic Lodge in New York, where Jackson and Kershenbaum first wanted to record a live, digital 2-track album. But, because of problems with sound leakage, the team decided only to remix the multitrack tapes to digital 2-track, optimistically labeling this experimental session as a "learning experience."

In the fall of 1985, Jackson and Kershenbaum selected the band for the 5-show digital session. Drummer Gary Burke and guitarist Vinnie Zummo were already lined up, but a bass player was needed. After a series of auditions, Rick Ford got the job. To allow the new quartet to run through Jackson's newer material, it rehearsed for about six weeks.

Just before Christmas, the band played unannounced live dates in a series of East Coast venues, including New York's CBGB's. After that, Kershenbaum had to find a suitable hall to record the digital 2-track sessions for *Big World.*

"The room had to be live to give us the ambience we needed, but not so live that it would be uncontrolled when we got there," Kershenbaum says. "We wanted it to be an album recorded live, not a live album that sounded like someone had held up a cassette recorder in the back of the room."

They finally settled on the Roundabout Theatre in New York. Having sorted out power, access and permit requirements, the band was put on a serious production schedule in preparation for the session, Jan. 23-25.

About Dec. 27, Frondelli and Kershenbaum began attending Jackson's club dates. Everything had to be charted out; nothing could be left to guesswork. When all details had been well-

Producer David Kershenbaum (left) and engineer Michael Frondelli in Guy Charbonneau's Le Mobile truck.

Closeup of Joe Jackson's Yamaha piano and accompanying pair of AKG 414s. An acoustic blanket and Sonex acoustic foam were used to help prevent sound leakage.

Stage layout for Big World*—a live, digital 2-track session at the Roundabout Theatre.*

rehearsed, the next step was to find an engineer who could provide front-of-house sound without producing sonic problems for the recording crew. Al Tucker was the producer's choice, and he joined the project at the beginning of the year.

Frondelli and Kershenbaum now turned their attention to the selection of the mobile recording truck. Having settled on Guy Charbonneau's Le Mobile, the duo spent about a week going through the Neve-equipped truck and adding outboard gear, including Sony PCM-1630 and PCM-1600 digital processors. Normally housed in the truck are two Studer A-800 24-tracks and two A-810 2-track machines; however, the analog tape decks remained dormant all three nights. Backup multitracks weren't even run, just in case a fix would be needed. *Big World* would be *live* 2-track, with no repairs or overdubs.

The actual recording of the warm-up dates started on Jan. 6, with Frondelli and Kershenbaum spending several days in Le Mobile as they mixed live multitrack tapes in real-time to practice their stereo balances. During these run-throughs they also had time to develop the basic levels needed on every song. Finally, reverb and room balances were worked out, and an overlay panel was designed for the Neve 8058 console's monitor section to mark preset levels.

Input designations were also developed and the corresponding tape strips hung on a wall of the truck. Assistant engineer Eddie Ciletti, who was hired to look after the PCM-1630 and 1600's companion Sony BVU-800 and 820 U-matic videocassette recorders, had the task of passing the next strip to Kershenbaum and Frondelli as each song ended.

The two video decks received identical digital and analog signals. On the analog tracks, channel 1 received a mono mix for cuing, while channel 2 was stripped with SMPTE time code. In addition, both machines were run in tandem, with a couple of minutes of lead time on one machine to ensure total coverage of the entire performance.

"Hands would fly during the song crossovers," Kershenbaum says. "Everyone had to move their faders up to the right starting levels. All the echo sends and returns were marked with little colored dots, which corresponded with what you push and what you don't push. It took only 27 seconds to set up for the next song, is a major accomplishment.

"During rehearsals, we thought we had it all in the bag—until, that is, we got to the Roundabout Theatre. Then everything changed. Every console setting was different because the amount of room sound was so unlike that in the previous clubs we had been playing. We had two days to get it together before the show, during which time we rebalanced everything and had to keep doing sound checks.

"For the most part, what you hear on the record is exactly as it went down."

"In addition, we would take digital mixes across town to Masterdisk, where Bob Ludwig would do trial cuts of rehearsals. The sound quality had to be accurate, because there was no going back and remixing."

The crew only had three nights and five shows in which to get the takes for the album. To ensure clean outros to each song, the audience was asked not to respond until the music ended. The few times that the audience did applaud early, the crew retook the song.

"But for the most part," Kershenbaum says, "what you hear on the record is *exactly* as it went down."

Sound leakage

To overcome sound leakage problems among on-stage microphones, Kershenbaum and Frondelli had to come up with some novel solutions. As Kershenbaum explains, the acoustics of the Roundabout Theatre didn't help.

"There was a big dome over the stage that acted like a parabolic reflector. Anything that hit the stage would bounce off into the dome, and just disperse all over the place. It was very hard to control."

To cut down sound reflections, the stage area was covered in thick carpet, and heavy drapes were hung around the rear walls. Also, absorbent baffles isolated the drums, bass amps and the quartet of backup singers.

The vocal microphones used were AKG C535s. Jackson and the background vocalists worked close to their mics to further reduce sound spillage.

"The 535s were excellent for both live sound and recording," says Kershenbaum. "Which was important for us, because there had to be a PA system in the room for the audience."

Kick drum and toms were both miced with Sennheiser MD421s, and top of the snare with a Shure SM-81. Overheads were a pair of AKG C460s, and the hi-hat was covered with an AKG C452.

Because of additional on-stage leakage, Frondelli built a chicken wire cage around the bass drum and covered it with an acoustic blanket. This arrangement had the two-fold effect of allowing the MD421 to be mounted away from the kick drum, without having the microphone pick up too much leakage from nearby instruments.

Another leakage problem that had to be solved was the grand piano. Kershenbaum recalls that Jackson used a Yamaha CP80 electric grand piano on the warm-up sessions because the stages used couldn't accommodate the size of a conventional grand. On this digital session, however, Jackson was adamant that the real grand be used. But how to provide sufficient sound isolation?

"If microphones had been placed very close to the strings and sounding board," Kershenbaum points out, "the sound would have been very choked, and you don't get the overtones resonating the way you'd like."

To overcome these problems, Frondelli removed the piano lid and raised it about four feet. A U-shaped strip of ¼-inch plywood was then built around the piano harp. The piano enclosure was wrapped in Sonex acoustic foam and an acoustic blanket to prevent leakage from the nearby drums and on-stage monitors.

Piano micing consisted of a pair of AKG C414s mounted near the top and bottom strings, plus a Crown PZM and two Countryman Isomax microphones.

In an attempt to reduce sound leakage in the background vocal mics, the four singers were moved to the side of the stage in front of the curtains. Baffles were then placed between them.

Because they were also being taken direct, Kershenbaum recalls that the micing of the guitar and bass posed few problems.

Outboard effects

Processing equipment included GML model 8500 microphone pre-amps and model 8200 parametric equalizers. Also added to Le Mobile's outboard racks were a collection of API model 560 and Pultec equalizers, two Roland SDE300 digital processors, a Publison model 90 Infernal Machine for pitch shifting, Fair-

Photo by John Bellissimo

David Kershenbaum and Joe Jackson at Masterdisk, New York, checking a DMM copper master.

child and dbx compressors and limiters, an AMS RMX-16 digital reverb, an EMT 251 reverb, and various de-essers and noise gates to further reduce sound spillage between on-stage mics.

Kershenbaum says that he used the room acoustics of the Roundabout Theatre to create the necessary ambience on the tracks.

"To put some space around close-miced instruments and vocals, we used the EMT 251, AMS RMX-16 and Lexicon 224X," he says.

Digital advantages

Although the *Big World* session represents Kershenbaum's first experience with a direct-to-digital 2-track recording, he is a definite convert.

"Digital is identical to the original source, with the exception of a slight loss at the very high top end. Also, you get compression on analog tape that you don't get on digital. So, if you like that compressed sound, you have to duplicate it electronically, because digital tape isn't going to help you.

"Sometimes when you're recording drums, it's an advantage to let the tape compress a little bit, or oversaturate. But you can't do that with digital; it's a very clean reproduction. Also, small amounts of guitar amp noise, for example, are there for you to hear off-tape—in contrast to being somewhat obscured by tape hiss from analog.

"With digital recording you have to be so much more aware of any extraneous clicks, pops, noises and hisses. You also have to remember that you're not going to have any tape correction at all; the tracks are going to sound *exactly* as you heard them.

"When I record analog, I usually add a little more top end than it needs—a couple of dBs at 15 or 16kHz—because I know that after a week's overdubbing, the tape is going to start losing it. That's

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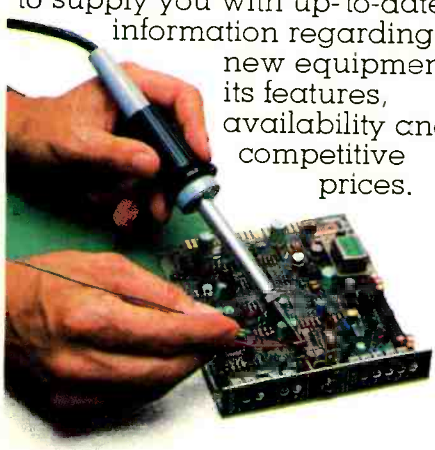

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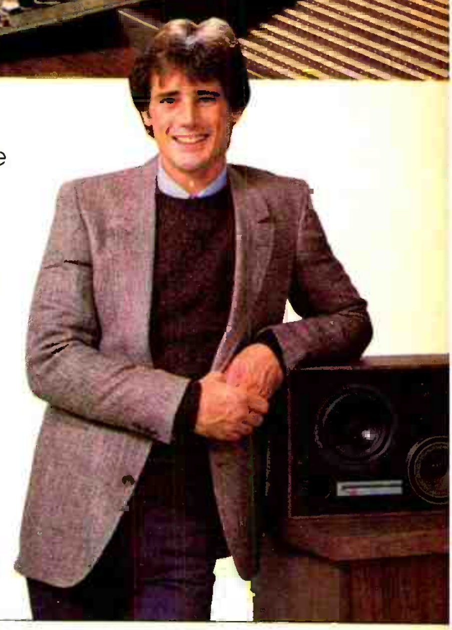

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Joe Jackson's latest album is the result of three nights of live direct-to-digital sessions.

2-album, 15-song release from PCM-1630 masters. (Only three sides were cut with program material; the fourth was left blank.)

Briefly stated, DMM is said to improve transient performances, reduce distortion and pre-echos, and offer a flatter frequency response, compared with conventional vinyl mastering techniques.

"When the normal masters are sent to the pressing plant, mothers are made, from which they make the pressing stampers," Kershenbaum says. "But, at this point, the record starts to lose top end. With metal mastering, however, you can grow a stamper directly from the original DMM master. By eliminating the intermediate mother in the process, you can retain a lot more high end." [The DMM process is detailed in "Direct Metal Mastering," in the October 1985 RE/P—*Editor*.]

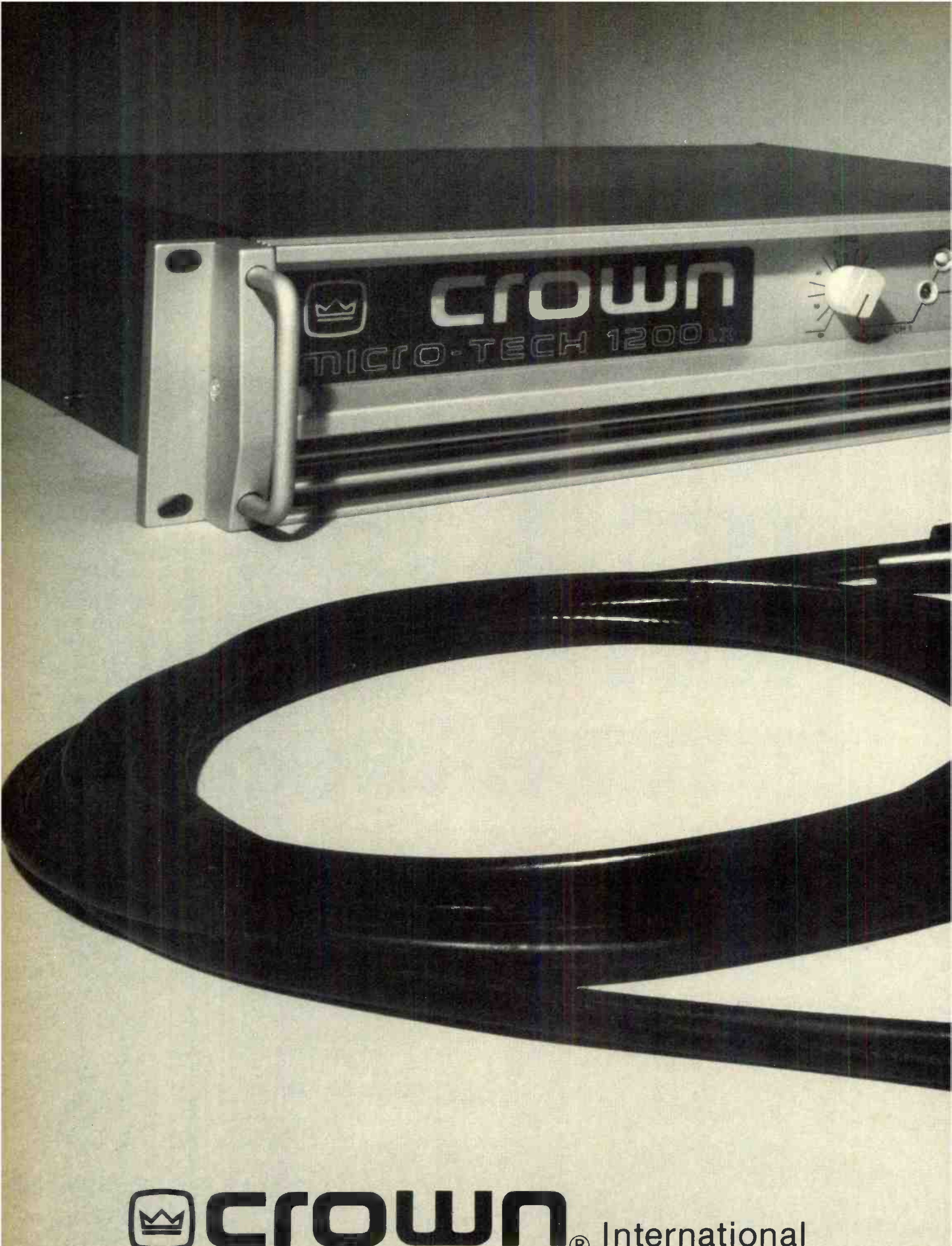
"First," says Masterdisk's Scott Hull, "we cut a lacquer reference disk, just to add any last-minute equalization or compression, because it is virtually impossible to play back a metal DMM master on a conventional turntable. Although you *can* play it back, you'll be deceived in terms of sound quality.

"The combination of DMM and digital technology produced a clean sound. But DMM has its own inherent sound quality. It's the same argument used with analog vs. digital sound: some people like it, and some people don't. In my opinion, DMM more accurately translates program material on a master tape to a vinyl album, and that's what David and Joe wanted with this digital session."

If his previous track record is anything to go by, David Kershenbaum will be among the first to employ techniques as they become available. Risk and personal challenge are what motivate this busy producer. R·E/P

Block is a free-lance recording engineer and writer.

Photos by Stephanie Chernikowski except where indicated.



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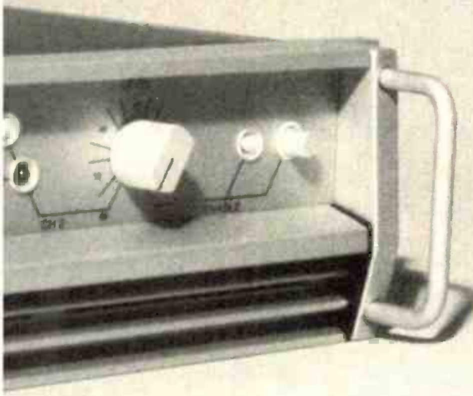

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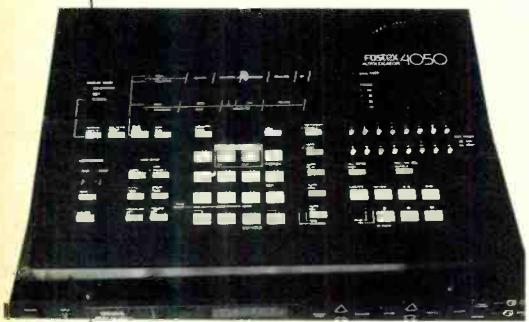


The model 4050 synchronizer/autolocator will operate with all Fostex reel-to-reel machines.

Continued from page 50

mable functions are based on reading time code locations. The 4050 will read and generate 24fps, 25fps and 30fps drop/non-drop time code, but will not regenerate code. The reader automatically determines which type of code it is reading, and displays this information on the front panel. If the code being read is different from the type that was preset via the back-panel DIP switches, the preset display will be lit and the input code display will blink. This is a well thought out feature, and one that can save the user hours of troubleshooting. To supplement this feature, an LED shows whether the preset code is even present on the tape.

The MIDI control section is musician-oriented, and a very musical way of looking at recording control. Although it is perfectly reasonable for a user to enter cue points as time code locations, musicians tend to feel more comfortable with entering such information in terms of bars and beats. Time signature and tempo may be entered and changed wherever you desire within the song.

Based on the time signature and tempo changes, a song's *total time* is calculated by the 4050, and may be displayed. This feature is handy for making minor tempo changes that will help you fit a tight cue or musical section into the proper time allocation. Once time signature and tempo have been programmed, if you change the total time of a song, the 4050 recalculates and automatically changes the tempo to fit your cue restrictions.

Programming all song information is quick and easy, and there is even a copy mode that acts similarly to that featured on a LinnDrum. A start time can be entered that will trigger MIDI song pointer information for an external sequencer, and act as a reference for positioning throughout the tune.

Despite the fact that the song pointer worked reasonably well, one problem exists with the 4050's MIDI capabilities. The unit sends a constant stream of MIDI note-on commands, which could potentially play havoc with your sequencer. We found that if the 4050 MIDI cable was left plugged in for a couple of minutes, the Opcode MidiMac sequencer software running on the Apple Macintosh would rack up hundreds of measures. Then, when the play command was given, the sequencer would try to play catch up. I'm sure Fostex will take a look at this as more people find their sequencers running away.

A 10-location memory bank allows the storage of all the required information for a tune. Each memory bank location will store: 10 cue points (0-9), locate, preroll, auto return, zone limit, relative reset, start time, measure set and tempo set—pretty much everything you would need to go back into a tune to do some overdubs or re-recording. All of this stored information may be saved to tape and reloaded at any time. In order to avoid losing any important information, the 4050 will verify all stages of the save and load process.

All four time base operations—real time, relative time, bar and measure—are displayed on a legible, alphanumeric LED readout. Real time is presented in hours, minutes, seconds, frames and subframes (tenths only). Relative time, which reads tach pulses from a tape machine and may be used when code is not present, is displayed in the same format with a plus or minus label to indicate relative location. Bar displays tempo, bar, beat and sub-beat (24 divisions) for some precise tempo change programming. Measure displays both bar and measure information. The display provides extremely precise control and is complemented by a clearly labeled panel that uses a number of LED indicators to keep you informed of what's going on.

Operationally, the 4050 was an impressive piece of equipment. The combination of MIDI- and time code-based controls allowed me to run the entire operation from a central point. Once a MIDI sequence had been set up on the Mac, and the 4050 was programmed, everything could be performed automatically; we just sat back and watched the hardware go through its paces.

The 4050 also generates "phantom" code, which allowed us to trigger the MIDI sequence without rolling tape. In this way, we were able to rehearse parts and punches without constantly running tape over the heads and decreasing the high end. Autopunch points can be rehearsed and a beep will sound once the cue point has been reached on the track. It should be noted, however, that the B-16D did not respond with the same speed as the input-monitor switching and rehearsal beeps indicated. As a result, to correct for the delay, we had to adjust our punch-in and -out times by five frames.

There are only a couple of things that Fostex should take another look at. A damping control allows individual compensation for different tape transports. We found that the review B-16D would fail to locate every 10 or 12 attempts, and also was sometimes slow in finally reaching the correct locate position, not knowing quite where to stop. This problem could have been caused by the review sample, but I did not find a solution. [Details of the new E-16, which now supersedes the B16-D and features an improved transport design, can be found in an accompanying sidebar—*Editor*.]

Other owners of this same Fostex combination have expressed having similar difficulties. Given that the 4050 is totally software based, a solution should pose few problems for Fostex, which in the past has been very good with software development and is constantly working on updates and new features.

It also should be mentioned that a serial interface port is provided for adventurous users wanting to control the system from a personal computer. Communications parameters are set via rear-panel DIP switches.

The model 4050 is certainly no dumb machine, and it will tell you when you're making a mistake. If you try entering invalid data, press programming buttons in the wrong order or make a protocol error, a loud beep will sound and a coded error message flashes in the display window. Because no reference to these codes is provided in the user's manual, I never knew what I was doing wrong. Fostex says that a list of error codes will be out soon.

The written manual, although not bad, could be clearer. There is a guy at Fostex that spends half of his day on the phone answering questions from confused owners. He told me that software updates have been coming in so fast that the company hasn't yet had time to put together a proper manual, and to "Please be patient, it's forthcoming." Fostex has

Any of the slaves may be placed in lock to allow individual control, if so desired. The 4035 will synchronize in one of three modes: frame, sync or auto lock. Frame lock is the tightest sync, and locks frame by frame. In this mode the slave will follow the master's wow and flutter. In sync lock mode, the slave will not follow the master's wow and flutter. Fostex recommends using auto lock, which allows the 4035 to select and use the most accurate lock mode.

The 4035's display shows time code locations of the master, slaves or offset between the two. Offset can be programmed with 0.1-frame accuracy and then, as the tape is rolling, fine-tuned with the offset trim button. (I understand that an update from Fostex will allow the user to view the offset down to 0.01 frame while setting trim.)

Model B-16D 16-track

The B-16 ½-inch 16-track is available in three basic versions: a standard model that features a belt-drive capstan motor,

Stop Press: Fostex Introduces E-16

At the end of July, Fostex ceased production of the B-16 machine and replaced it with the new E-16 16-track machine. Designed to offer all the B-16's functions, the E-16 has these added features:

–built-in 2-position autolocator;
–autoreturn and autoplay functions for loop or repeat capabilities;
–rewind and fast-forward tape shuttling at 15ips;
–direct-drive PLL capstan motor;
–improved transport control;
–timed bias circuitry for "gapless" punch-outs; and
–footswitch control of cue No. 1.

Pro user price for the E-16 is $6,995.

Circle (194) on Rapid Facts Card

New E-16 ½-inch 16-track machine.

The 24-input CMC-24 console incorporates the CARS microprocessor-controlled routing and mute system.

2-head (erase and record/play) transport layout, 15ips speed, integral Dolby B-type noise reduction and −10dBV input and output levels; the B-16D reviewed here, which is identical to the standard model except for the addition of a direct-drive, phase-locked-loop capstan motor; and the B-16DM, which features a third head for tape reproduce and 16 channels of independent Dolby encode and decode for off-tape monitoring.

So how did all of this recording and synchronization hardware work together? Well, not bad, but not great either. The model E-2 ¼-inch 2-track slave locked perfectly with the JVC ½-inch VCR master, and with the proper offset programmed into the 4035 synchronizer controller the soundtrack transferred in perfect sync. Having relaid all the musical cues, which were triggered from the Mac by MIDI song pointer to the time code-striped B-16D, we found they were in their proper places. The master time code track transferred perfectly to the 2-track during mixdown, and needed to be set only once.

The biggest problem we encountered with the B-16D transport occurred in search mode. The machine took a long time to locate; sometimes it hung up and rocked back and forth over a cue point. I think that all the problems we experienced could be traced back to the fact that the 4030's servo adjustment LEDs did not function. Because these indicators were necessary to set up the system for the dynamics and tape handling of different machines, we recorded A/440Hz on the appropriate deck and made the adjustments by listening for octaves.

Although this procedure should have taken care of the adjustment, it didn't quite get the job done. Fostex told me that other people have experienced this same LED failure, so I guess it's time for the company to take a look at the problem. I should mention that the model E-2 2-track did not experience these difficulties after setup.

The lock light on the 4030 and 4035 only shows frame lock, not subframe mode. If you need true subframe accuracy, it is best to allow another six seconds to ensure correct lockup. (Although we couldn't use the system to lock together two multitracks—simply because we only had one machine—it would have been a good test to see how well cymbals held sync when recorded, using overhead stereo mics, onto separate machines.)

Fostex is not standing still with these synchronizer products. It is now working on a unit that will read vertical interval time code (VITC), as well as handling house sync, jamsync and more. These features will allow the user to really nail down cue points with VITC, by taking advantage of video "thrill" wheels to rock and roll the tape while still reading time code. Although this was a capability I missed when working with the 4035/4030, one must keep in perspective how much Fostex has accomplished for the money.

The 47-minute video manual for the 4030/4035 is recommended because it covers some areas omitted by the printed manual. The 4030/4035 combination was fairly straightforward to operate, and familiarity with the 4050 gave me a head start when learning the 4035.

Model CMC-24 recording console

We recorded the music tracks onto the B-16D using the AHB CMC-24, a compact studio recording console equipped with CARS (computer-aided routing system).

An in-line configuration with 16 full-feature channels and eight tape/line return channels, the CMC-24 provides microprocessor-controlled routing to 16 output buses. (The board is also available with eight and 24 full-feature input channels, as the CMC-16 and CMC-32, respec-

tively.) Each full-feature input is equipped with a 3-band, sweep-frequency EQ section and four echo or cue sends that may be routed in turn to six send outs. Balanced mic and unbalanced line inputs, as well as an unbalanced send-receive jack, are provided.

Input and monitor channel muting can be controlled by the microprocessor-based CARS system. In fact, the way the routing is set up, all monitors may be used during mixdown to provide 48 possible inputs (24 channels + 16 monitors + 8 effects).

The console operates at −10dBV levels and should be interfaced with compatible equipment. Although well-matched with the Fostex tape machines, interfacing the CMC series with +4dBm units might cause some problems.

Features that make this console stand out from its competition are the possibilities provided by CARS. Channel bus assignments can be programmed and memorized, as well as channel and monitor mutes. These assignments are stored in 32 internal storage locations.

I used the CARS mute control during mixdown to simply step through a series of mute setups that would normally have demanded all my attention. The system helped to keep tape hiss and channel noise to an absolute minimum, along with providing me more options in channel combining and solo juggling.

The one thing I didn't like about the CARS system was the bus assignments. During overdubs, the assignment of a number of synthesizers to their prospective tracks became a slow and attention-consuming task. It was not nearly as fast to set up and use as standard bus routing buttons.

By connecting a Commodore C64 computer via a back-panel socket and optional CMI-64 interface unit, the console's storage possibilities increase significantly. Also, the information is now presented on the C64's video monitor in a much more pleasing visual display. Channel inputs (with instrument names or signal sources entered from the C64 keyboard), tape track assignments, bus routing patches and mute patterns may be stored to floppy disk and displayed graphically, along with song titles and tape-counter settings.

A 2,048-event sequencer is also provided for setting combinations of routing and mute patterns and can be run manually, triggered by an off-tape clock signal or by an external drum machine. (For external control or tape-sync, a CMS-64 interface also is required.)

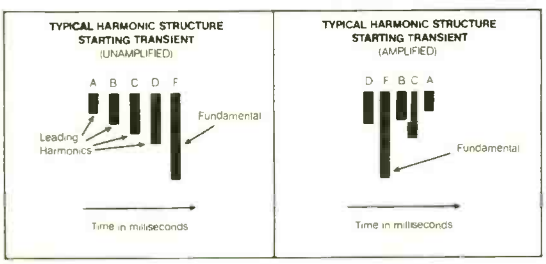
The CMC-24 was fairly easy to use, and the manual covered all operations clearly. I would like to see a high-pass filter added to the EQ section, which can be especially useful when the low-frequency boost uses a shelving function, and even more so with narrow-gauge multitracks. I also think that a polarity-reversal switch on the mic inputs would be handy for multi-mic situations.

Other than these minor points, I know of no other board in this price range that offers the degree of control sophistication provided by the CMC series.

"I like the way BBE™ brings out the 'live' dimension in sound."

Bernie Grundman

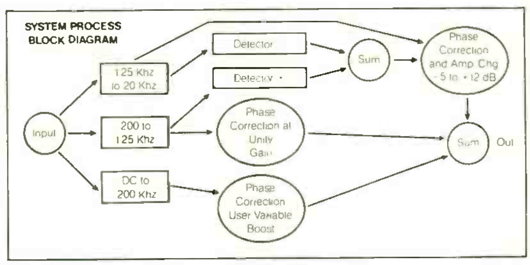
When it comes to cutting discs, Bernie Grundman is the master. Ever since the late 60's, the music industry's top artists have been entrusting Grundman with the delicate task of preparing their record masters. The list of satisfied clients includes Stevie Wonder, Prince, Michael Jackson, Olivia Newton-John and Lionel Ritchie, to name just a few. So when a critical listener like Bernie Grundman lists himself as one of BBE's satisfied customers, it's quite a compliment.



"Unlike an equalizer, BBE is a dynamic process. It gives me the high end I want without the sibilance and without creating energy levels too intense for my cutting lathe to handle. I can foresee BBE becoming a very valuable tool in the mastering process."

The BBE 202R is a vital link in the chain of precision components used at Bernie Grundman Mastering in Los Angeles. Even the finest components, you see, can be undermined by that all-important interface between amplifier and speaker. That's where phase and "overhang" distortion develop, due to voice coil characteristics, reflected impedance from the environment, crossover impedance anomalies and the mechanical properties of dynamic speakers. The relationships among the fundamental frequencies, their leading harmonics and between the leading harmonics themselves become distorted in both amplitude and time. The result? Muddiness, poor imaging and pinched, colorized sound that lacks the presence and punch of the real thing.

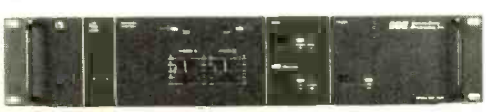
The BBE 202R puts the clarity and sparkle back into amplified and recorded sound. We like to think of it as the "unprocessor." Rather than artificially altering the original source, BBE restores the natural harmonic balances that were present in the live performance. How? First it divides the audio spectrum into three bandwidths. Then it applies phase correction across the full spectrum and dynamic high frequency amplitude compensation as required. BBE's continual sampling of the mid/high frequency relationship allows this correction to take place



automatically. Convenient front-panel controls let you boost low frequencies and regulate the amount of high-frequency amplitude correction to suit your needs. There's no encoding or decoding involved, so BBE can be used anywhere in the recording chain—from individual tracks on a multitrack tape to a mastering lab monitoring system.

For disc mastering perfectionists like Bernie Grundman, BBE makes an audible difference. Shouldn't BBE be a part of your next masterpiece?

To find out what the BBE 202R can do for your sound, call us toll-free at 1-800-233-8346. (In California, call 1-800-558-3963.) Or write to us at Barcus-Berry Electronics, 5500 Bolsa Avenue, Suite 245, Huntington Beach, CA 92649.



BBE™ All the sound you've never heard.™
Barcus·Berry Electronics, Inc.

Circle (35) on Rapid Facts Card

Model E-2 2-track machine with center-track time code.

[In addition to the CMI-64 interface for the Commodore C64, which allows external control of the CARS routing and mute assignments, and the saving of console configurations to floppy disk for later recall, Allen & Heath Brenell recently unveiled at the summer NAMM show a self-contained controller for the CMC series. The CMR-MIDI provides 100 MIDI memories for external program changes of console status; 100 routing plus 100 muting memories; and a 10-song sequencer with MIDI synchronization and song-pointer implementation. Console configurations and MIDI channel data can be stored in a removable RAM cartridge. Also unveiled was the CMC-24 MkII, which now features a redesigned master section with enhanced controls; and a redesigned return input section that allows any one or more channels routed to tape-track outputs 1 thru 8 to be subgrouped on returns 17 thru 24, thereby providing a submaster control for respective channel outputs and rerouting to a selected pair of channel outputs or the stereo mix—*Editor.*]

Model BR-8600U VHS editing VCR

The JVC BR-8600U ½-inch VCR served as the video master on our scoring project and was a great machine to work with. The editing wheel allowed me to spot my cue points on a frame-by-frame basis, and the VCR did a nice job of recording stereo audio, with or without Dolby B-type noise reduction. The machine has extensive video editing capabilities, especially when connected to another JVC deck using the company's editing controller and remote.

The question here is whether you need this degree of sophistication for scoring sound cues. Unless you are doing audio and video layback, you're paying quite a bit extra for some excellent video-editing features. When Fostex provides interfaces for other VHS machines with remote connectors, much less expensive decks will be able to perform a similar task. As far as I know, the JVC BR-8600U is the only such transport for which Fostex has developed a suitable control interface.

A minor drawback with the BR-8600U is that it doesn't provide tach-pulse output. If you want the Fostex synchronizer to chase tach in video fast-forward or rewind, a simple internal modification must be performed to route tach pulses to the remote-control socket. A friend of mine has performed this mod on his

How to keep your mixing board from holding you back.

Today's technological revolution is a battle fought on two fronts: quality and cost. Recent advances in recording equipment let you do more than you ever could before – And for less money. But what about your mixing console? Can it keep pace with recent giant steps such as the significant advances in the quality of analog and digital recording equipment? Without sacrificing your budget?

Program Mix Control

If you're mixing on a RAMSA WR-T800 series console, the answer is yes. RAMSA's 8-buss consoles are a smart choice for 8 or 16-track recording. They offer the *quality* you need to bring your recording chain up to today's demanding specs.

RAMSA's T-Series consoles save you time, too. Each input channel simultaneously accepts Mic, Line and Tape signals. A time-saving feature, the Program Mix control and stereo Solo switches let you choose the signal source you want without re-patching. So you get from basic tracks to final mix in record time.

Because with the RAMSA T-Series, you get a mixing board that does much more for less without holding anything back.

Pushbutton Input Selection

RAMSA

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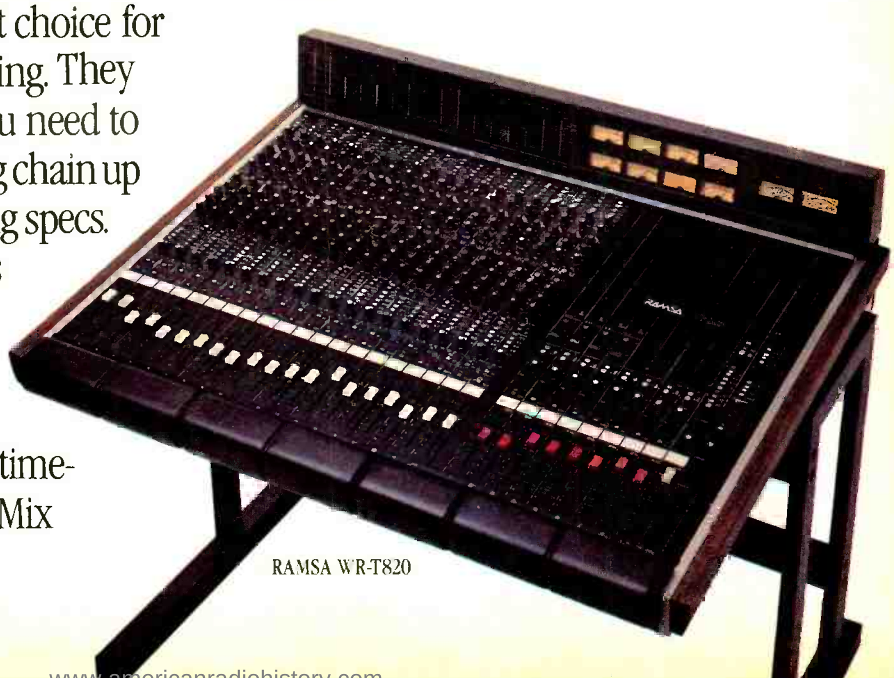

RAMSA WR-T820

Circle (36) on Rapid Facts Card

Using Opcode Systems MidiMac Sequencer

By Denis Hannigan

For the past couple of years, musicians who use MIDI sequencers have been waiting for a dependable program to run on conventional personal computers—something they could use and trust on the job. It's beginning to happen.

For this project, we interfaced an Apple Macintosh 512 running Opcode Systems MidiMac sequencer software (version 2.0) with a Fostex audio post system controlled by the model 4050 autolocator.

My tapeless MIDI studio consists of a Yamaha DX-7, two TX-7 modules, an Oberheim Xpander, a Yamaha SPX-90 effects processor and a LinnDrum that was recently MIDI-retrofitted by J.L. Cooper Electronics. I route this equipment through a Roland MPU-104 MIDI input selector and an MPU-105 output selector, using Southworth Music Systems' MIDI interface box for the Mac. Using a Roland MSQ-700 sequencer as a second MIDI data recorder, I transferred some music files from another Macintosh sequencer into MidiMac.

Before I began to write the score, I took timing notes from a ½-inch VHS videotape copy of the film that had a time code window burned into the picture, thereby providing reference times for the length of each cue.

I program synthesizer parts in one of two ways: either composing the entire cue into one sequence or building the cue by chaining sequences together. The latter mode allows me the flexibility of making changes to different sections of the cue, without having to reprogram the entire piece.

When programming percussion, I either first record all the drum parts into the LinnDrum's internal memory, and then transfer the completed tracks into MidiMac, or record them directly into the MidiMac sequencer, using the DX-7 as a controller.

Either way, I have control from one source; centralized control is one of the advantages of this technology.

By connecting the DX-7 to the SPX-90 effects processor via MIDI, I was able to select reverb and effects programs from the synth's control panel. By pushing one button, I recalled preselected synthesizer sounds and effects, a feature that helps save studio time if you adjust reverb and effects settings during preproduction.

After the music was composed and programmed, Bob Hodas entered all the timing points into the model 4050. When we were ready to record the sequence onto the Fostex B-16D, I connected the 4050's MIDI Out to the Macintosh MIDI In.

I set the MidiMac sequencer into external trigger mode, and hit the play button. The sequence then waited for MIDI song pointer and start commands from the 4050.

It worked perfectly. I was able to walk away from the Mac and let the 4050 perform the start, stop and chase functions.

MidiMac records up to 26 sequences containing 16 MIDI data tracks each; any sequence can then be recorded onto another to create subsequences. The software was more than adequate for my sequencing applications, which are generally in the range of 10,000 to 20,000 MIDI events per composition.

There are three windows on the Macintosh screen at all times. The file window keeps track of active and inactive sequences, which are also selected and opened from this window. The sequence window displays synthesizer type, patch number and/or name, MIDI channel selection, tempo, cue length, record/mute assignments, looping and quantization. The editor window displays a counter, allows access to record/play, fast forward/rewind, and has punch-in and -out functions. Step-record and editing modes can also be entered from this window.

Step recording is a breeze with MidiMac. The counter gives you a bar, beat and sub-beat readout that is easy to follow. A complete palette of note values to choose from, including triplets, is extremely helpful.

MidiMac lets you filter out unwanted MIDI data from whole tracks or just notes that fall within a certain range. You can even change note velocities and transpose notes that are in a chosen range.

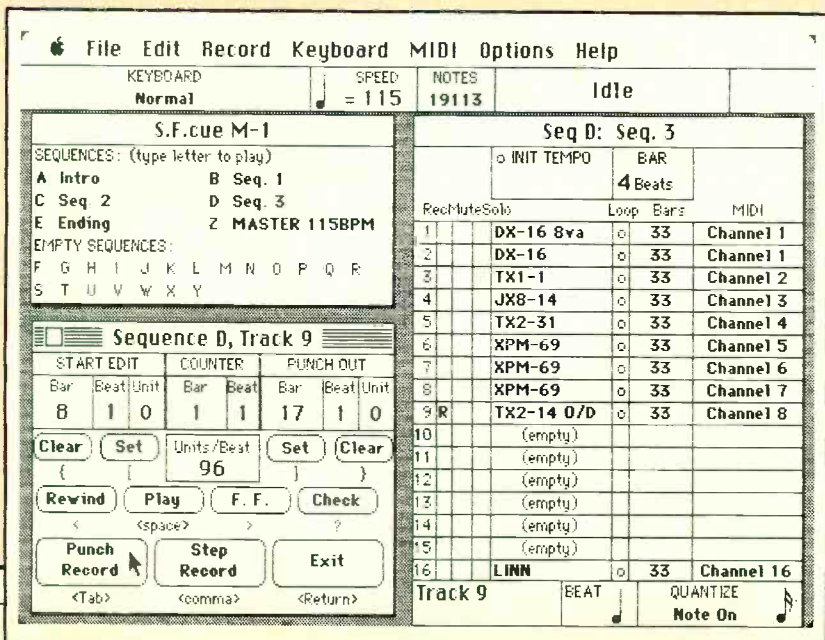
Merging and unmerging MIDI data also is possible. I took a composition from another Macintosh sequencer program, recorded it onto one track of a Roland MSQ-700, and then into MidiMac. After selecting the unmerge-to-track function, the composition was unmerged to its separate tracks/MIDI channels.

The program also lets you transpose and trigger sequences from the keyboard; splitting the keyboard also is possible. It has an arpeggiator that plays in forward, reverse, alternate and random patterns.

Opcode Systems MidiMac is a powerful, fast and easy-to-use sequencing tool. More importantly, it's unobtrusive.

While MidiMac quietly keeps track of all pertinent details, I can get on with creating music.

Screen dump of Opcode Systems MidiMac sequencer software for the Apple Macintosh.



Hannigan is a composer/synthesist, and a contributing writer to **RE/P**.

The Thin Man becomes a Fat Boy

. . . or how a Mirage Digital Multi-Sampler can make your DX-7* Deluxe

The DX-7 is a marvelous machine, but quite a few of you think it could use a little fattening up. DX sounds are punchy and crisp, but a tad on the thin side. Not to worry. With a Mirage Digital Multi-Sampler and a MIDI cable, you can change all that.

While the DX uses operators, algorithms and sine waves to create its sonic personality, the Mirage uses multi-sampled waveforms of actual acoustic instruments for sounds with acoustic richness and character. Just connect the MIDI Out of the DX-7 to the MIDI In of the Mirage, power up your system, and turn yourself on to the hottest performance set-up going.

Partners in Crime

If killer sounds will help you steal the show, the DX and the Mirage are perfect partners in crime. There are over 100,000 sound combinations among the available DX and Mirage sounds. Rather than list them all, here are a few favorites.

Dualing pianos. DX and Mirage keyboard sounds complement each other perfectly. The electric piano sounds in particular combine the synth punch of the DX with the realistic timbre and dynamics of the Mirage. In fact, any synthesized sound takes on a new dimension when combined with the sampled acoustic counterpart. Strings, brass, mallets and fretted sounds take on a new personality when doubled on the Mirage.

Mirage sounds + DX sounds . . . over 100,000 combinations

The bells are stringing. The impressive bell sounds of the DX come alive with orchestral richness when combined with the Mirage string sounds from Ensoniq Sound Library Diskette 3. Depending on how you balance the levels, the effect can be either subtle or startling. While you've got the strings loaded into the Mirage, try some of the DX synth bass sounds (especially percussive variations) and hear how well the strings support the bass.

Digital Doo-wop. The "Tah" and "Doo" vocal samples from diskette 17 add new life to many of the old standby DX sounds. Try the "Tah" and DX brass sounds together. Use the "Doo" with the caliope. The human voices add a jazzy sophistication to even clichéd patches.

The Special Effects Department

With pitch and mod wheels, velocity sensitivity, after touch, breath controller and pedals, the DX is among the most expressive synths. If the truth be known, the new Mirage operating system (version 3.0 and up) was developed specifically to take advantage of these DX features.

Through the magic of MIDI, the DX and the Mirage can be configured many different ways. For instance, you can modulate the Mirage LFO from either the DX mod wheel, breath controller, foot pedal controller, volume pedal, after touch or even the data entry slider. And all independently of how you are controlling your DX.

So you can use after touch to modulate a DX string sound while using the DX mod wheel to control vibrato of the Mirage sampled strings.

The Mirage has the ability to vary the mix between the two oscillators of each voice. The solo rock guitar sound on diskette 6, for instance, has a heavy guitar sound on one oscillator and a harmonic feedback sound on the other. You can vary this mix with any of the DX control functions. A favorite of Mirage/DX players is to use the DX after touch to control the mix. Playing the keyboard normally gives you that "wide-open-through-a-couple-of-stacks" sound, and pressing extra hard will bring in the feedback. A little practice with the pitch and mod wheels will earn you a convincing guitar technique.

Remote Territory

Changing sounds and programs on the Mirage is simply a matter of pressing a few buttons, but if you want to rack mount your Mirage you can just as easily change sounds and programs right from your DX

Just 11 pounds of Mirage can make your lean DX a fat boy

over MIDI. Pressing one button on the DX can change your entire set-up from a sweet string background to a sizzling solo sound on both the Mirage and DX.

A Marriage made in Malvern

The Mirage/DX partnership is a natural. Although the instruments are designed and built on opposite sides of the globe, they go together like hot dogs and mustard (or sushi and soy sauce). If you own a DX-7, bring it down to your authorized Ensoniq dealer and let it spend some time getting friendly with a Mirage Digital Multi-Sampler.

ENSONIQ Corp.: 263 Great Valley Parkway, Malvern, PA 19355 ■ Canada: 6969 Trans Canada Hwy., Suite 123, St. Laurent, Que. H4T 1V8 ■ ENSONIQ Europe: 65 Ave de Stalingrad, 1000 Brussels ■ Japan: Sakata Shokai, Ltd., Minami Morimachi, Chu-O Building • 6-2 Higashi-Tenma, 2-Chome • Kita-ku Osaka, 530

ensoniq®

DX-7 is a trademark of Yamaha International Corp. Mirage is a trademark of Ensoniq Corp. You're welcome to trademark any nicknames you have for either. The Mirage Digital Multi-Sampler retails for $1395.

Circle (37) on Rapid Facts Card

machine and reports that it works fine.

Not having access to the machine's schematics, I couldn't perform the necessary modification. Instead, once the video master was set at the specified cue, the slave machine would be parked manually in the general area of the master, and all machines started.

Model SPX-90 digital reverb

I used the Yamaha SPX-90 digital effects processor on this project both for recording wet tracks and during mixdown. With 30 factory presets and storage for 60 user presets, the MIDI capable unit offers a great deal for its price. However, I did find the 12kHz bandwidth to be restrictive, and used the unit only as a side-chain effect.

Any straight-through program, such as the compressor, parametric EQ or ADR/noise gate, sounded as if the top end was chopped off compared with the regular signal. However, I did like several effects programs and used the reverb and gate, stereo flange and various reverb programs with success.

Session conclusions

I will have to admit I was surprised that the B-16D sounded as good as it did. We got some nice tracks down with no special problems. Crosstalk was minimal, and there was no perceptible wow or flutter. Published frequency response is 40Hz to 18kHz, but this did not seem to be noticeable on playback. There did seem to be a slight rise in the playback low end, even though the calibration tape looked all right. If the faders were pushed way up and we monitored just the B-16D output, a lot of low "bubbling" was heard where I would normally expect to hear only hiss.

The final judgement must come on mixdown, however, and the 15ips master we got from the model E-2 translated well no matter where we played it. The E-2 2-track did its job well, was easy to edit on and could even memorize a couple of cue points for auto returns and repeats. The center-track time code capability makes it a desirable machine.

As a whole, the session went smoothly and our desired goals for the film soundtrack were accomplished. Except for the problems noted, we thought that this system would provide a solid proposition for someone who needs to turn out soundtracks on a low- to mid-priced budget. It is interesting to note just how sophisticated narrow-gauge equipment is becoming, and it will be even more interesting to watch the direction Fostex takes as it expands its features into an area that has always been reserved for the pro. R·E/P

Hodas is **RE/P**'s evaluations and practices consulting editor.

Photos by Elizabeth Annas.

Product Descriptions

Fostex B-16D

Description: 16-track, ½-inch tape machine.
Frequency response: 40Hz to 18kHz, ±3dB.
S/N ratio: greater than 80dB, weighted.
Distortion: THD 1% at 1kHz.
Pro user price: $5,900.

Circle (185) on Rapid Facts Card

Model E-2

Description: 2-track, ¼-inch tape machine, with center-track time code.
Frequency response: 30Hz to 26kHz, ±3dB.
S/N ratio: greater than 70dB weighted.
Distortion: THD 1% at 1kHz.
Pro user price: $3,600.

Circle (186) on Rapid Facts Card

Model 4050

Description: MIDI synchronizer/autolocator, 10-position; 10-bank memory used for storing a "snapshot" of front-panel settings, totaling 100 cue memories; loop/repeat functions; up to 99 seconds of selectable pre-roll; serial data port.
Pro user price: $1,300.

Circle (187) on Rapid Facts Card

Model 4030

Description: synchronizer module; Time code reader; 0.01-frame resolution (drop/non-drop frame); "daisy-chaining" capabilities; serial data port for computer interface.
Pro user price: $1,500.

Circle (188) on Rapid Facts Card

Model 4035

Description: synchronizer controller; 10-position autolocator; up to 99-second selectable pre-roll; programmable punch-in and -out functions with rehearse capabilities; individual transport control of up to one master and three slaves.
Pro user price: $500.

Circle (189) on Rapid Facts Card

Model 8710

Description: Interface; when used in conjunction with a model 4035 remote controller, the unit designates each model 4030 synchronizer as slave 1, 2 or 3; supports up to three slaves.
Pro user price: $500.

Circle (190) on Rapid Facts Card

Yamaha SPX-90

Description: MIDI digital processor.
Format: mono-in/stereo-out; 30 factory and 60 user presets.
Dynamic range: reverb, greater than 75dB; delay, greater than 81dB.
Range times: program-dependent, zero to 99 seconds; 16-bit D/A convertors; 31.25kHz sampling frequency.
Pro user price: $745 without remote.

Circle (191) on Rapid Facts Card

Opcode Systems MidiMac

Description: sequencer; records up to 26 sequences, each containing 16 MIDI data tracks; stores up to 25,000 notes and 50,000 MIDI events; stores pitch bend, velocity, patch-change and after-touch information; runs on Apple Macintosh.
Pro user price: $200.

Circle (192) on Rapid Facts Card

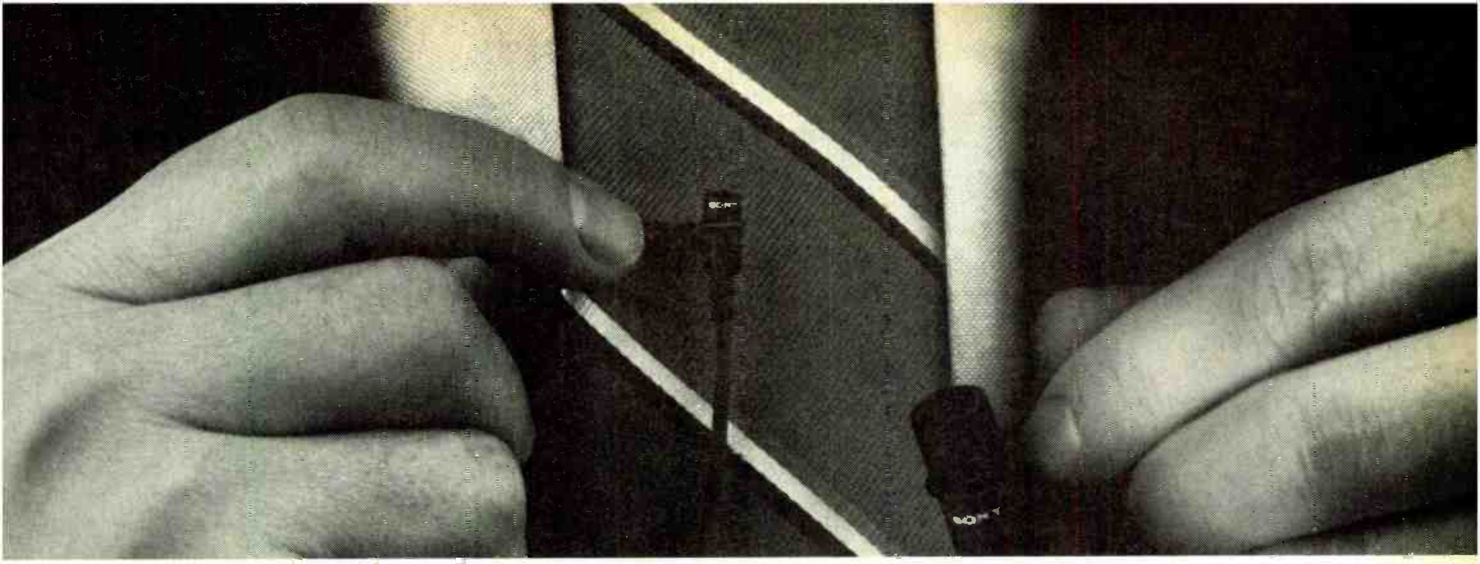
AHB CMC-24

Description: 24-input console.
Format: 24-inputs; 16 outputs; 16 tape returns; six auxiliary outputs.
Frequency response: 20Hz to 30kHz, +0/−1dB.
Distortion: between 100Hz and 10kHz, THD is 0.04%.
Equalizer section: ±14dB boost/cut; HF range 5kHz to 15kHz; mid 300Hz to 3kHz; LF 40 to 400Hz.

Unveiled at the summer NAMM show in Chicago, the new CMR-MIDI is a self-contained controller that includes 100 MIDI memories for external program changes of console status; 100 routing memories; 100 muting memories; and a 10-song sequencer with MIDI synchronization and song-pointer implementation. All data are recorded in a removable RAM cartridge. The controller also supports step- and real-time programming, with full editing facilities.

Pro user prices: CMC-24 console, $7,200; CMI-64 interface, $420; CMS-64 synchronizer, $210; SMPTE Software, $1,200 (tentative); CMR-MIDI, $980.

Circle (193) on Rapid Facts Card



THE ONLY COMPANY THAT DIDN'T HAVE TO IMPROVE ITS LAVALIER MICS JUST DID.

As the world's leader in lavalier microphones, we've just taken some very important steps to ensure that we retain that distinction.

For one, we've taken our mics in a new direction with the addition of the Sony ECM-66 unidirectional. Its wide-angle cardioid pattern provides better off-axis frequency response than the classic pattern—while also providing an unprecedented level of isolation from ambient noise.

We've also made the least of a good thing: The new ECM-77 is the smallest microphone in Sony history. We've even made our new cable housing smaller—and more durable.

We've made more of some good things, too. The new ECM-55, for one: the latest refinement of our successful ECM-50 series.

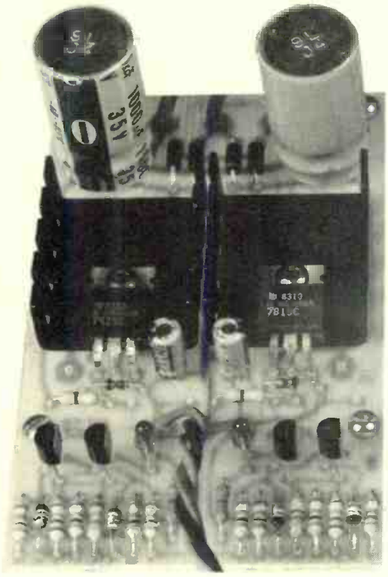
And we've expanded our line of accessories—with new color windscreens; pencil-type, safety-pin and necklace-type clips; and a power supply holder that clips to your belt.

Sony lavalier microphones operate on either a single AA battery or phantom-power. You also have a choice between black or satin-nickel finishes; and XLR, pigtail or Sony wireless-compatible output configurations.

So to see (barely) and hear (very clearly) the results of Sony's refusal to rest upon its laurels, call your Sony representative: Eastern Region, (201) 368-5185; Southern Region, (615) 883-8140; Central Region, (312) 773-6000; Western Region, (213) 639-5370. Or write to Sony Professional Audio Products, Sony Drive, Park Ridge, NJ 07656.

SONY
Professional Audio

Circle (38) on Rapid Facts Card



Building a Dual Bipolar Power Supply

By Jon Gaines

At the heart of every piece of audio equipment is a power supply—a simple yet essential circuit that converts 117Vac to one or more dc voltages which, in turn, power the active components of an audio circuit.

Op-amps, transistors, LEDs and other active devices require a steady direct current (dc) power supply to function.

In most contemporary signal processing equipment, the power supply is a bipolar, regulated type, which means that it supplies two output voltages simultaneously, and usually these are positive 15V and negative 15V. Voltage regulation ensures that the output remains constant, regardless of varying load conditions.

This article describes the design and construction of such a typical bipolar supply. Also described is an optional output voltage monitor, which is a section that measures the output voltages of the supply and, under normal conditions, lights two LEDs.

The project will be useful to anyone who builds their own studio equipment, or would like to try. In fact, the power supply makes a good first-time project that then becomes the basis for other construction projects. It may be used with virtually any audio circuit that requires ±15V supplies or as a good spare supply to temporarily, repair out-of-service equipment, and should prove to be particularly useful as a bench supply in the design of new equipment.

When working on home-built audio projects, it is easy to accidentally short or disrupt the power supply lines, leading to peculiar symptoms that may have you investigating the wrong part of the circuit. In the event that the supply voltages become abnormal, having a visual monitor may save time and confusion.

The parts required for the project are commonly available and require no special handling precautions. Additional parts references are listed separately.

The finished circuit board is compact—2.5 by 3.5 inches—and mounts easily inside existing equipment, or in readily available enclosures.

Circuit description

Figure 1 shows the PSU's schematic. The circuit can be viewed in three basic sections: transformer/rectifier, capacitor-input filter/regulator and output voltage monitor.

The power transformer primary must be fuse protected, and a power on-off switch used if desired. With a primary voltage of 117Vac, the transformer secondary voltage should fall within the range of 26Vac to 50Vac (center-tapped), with a nominal value of 40V at 0.5A being ideal. Below 26Vac, regulators will be unable to maintain proper regulation; above 50Vac, regulators may be damaged. Choosing a secondary voltage somewhere in the middle of the range gives a supply that is tolerant of varying load conditions and fluctuating ac line voltages.

The transformer secondary wires are connected to a full-wave bridge rectifier consisting of diodes 1 through 4, with the secondary's center-tap wire connecting to circuit ground.

Capacitors C1 and C2 smooth the rectified dc voltage at the inputs of the voltage regulators; a value of 1,000μF is typical for this application. Points *A* and *B* are unregulated dc outputs that will not be used in most applications, but find occasional use.

Regulators IC1 and IC2 are 3-terminal integrated circuits. This type of regulator is extremely popular, because of its ease of application, minimum external parts required in practical use and low cost. Output-voltage accuracy and ripple rejection also are excellent. The LM Series of regulators is available with various output voltage ratings for both positive and negative polarities, and can be specified to handle up to 1.5A, although the type recommended for this project is limited to 0.5A output.

Although the regulators have a nominal 15V output, they can be modified to produce a higher output voltage by changing a single resistor value, as noted by the asterisks on the schematic. Table 1 gives appropriate resistor values for some typical output voltages.

Additional features of the IC regulators are internal thermal overload protection and short-circuit current limiting, making them essentially immune to overload and accidental shorts.

It is important to remember that, in order for the regulator to do its job, the unregulated dc voltage at its input must be at least 3V higher than the desired dc output; that is, at least 18Vdc for a 15Vdc output. However, a dc input higher than 35V may damage or destroy the regulator. An ideal 40V transformer (center-tapped) will supply each regulator with roughly 28Vdc, which allows room for variations in ac line and load conditions without degrading regulation.

Capacitors C3 and C4 enhance the stability of the regulators, while clamp diodes D4 and D5 prevent reverse-voltage conditions from causing latch-up or damaging the regulators.

The output-voltage monitor circuit is an optional feature that neither improves nor degrades the basic performance of

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the power supply, but simply provides a handy visual indication of correct power-supply operation.

Using four zener diodes as references, a voltage reference *window* is created. If the output of the power supply is correct, give or take a couple of volts, the LED will light. If the output voltage exceeds 17V or falls below 13V, the LED goes out.

A simpler circuit would only connect an LED between each output to ground, lighting whenever a voltage was present. However, a regulator could be broken and the LED would still light, negating its usefulness. Regulator failures are unpredictable, and the output voltage could be anywhere from 0V to 35V in the failure mode.

The circuit presented here covers all the bases at minimum extra cost. Note that the resistor values given are for a nominal 15V supply. For higher output voltages, the voltage-divider resistors should be scaled down in value.

The LEDs may be of any shape, color and size, with brightness controlled by resistors R11 and R12. LEDs do not need to be mounted directly on the circuit board and may be moved to another location in a chasis or front panel.

Safety

Every effort has been made to ensure safe and successful assembly of this power supply. Everything in the circuit after the transformer primary is low-voltage at low current, and presents minimal hazard. No high-voltage ac is exposed on the circuit board. However, the transformer primary and fuse do attach to the 117Vac line, so you must exercise caution and good sense in the assembly and use of this project.

To minimize the exposure to high-voltage ac, when wiring up the fuse, power switch, power cord and power transformer, use wire nuts, shrink tubing, electrical tape, and/or insulated connectors as appropriate. Use a fuse as indicated in the schematic, but do not substitute a higher current fuse should the correct one blow. Use a 3-wire, grounded-type line cord, and ground it to the power-supply enclosure if one is used for the project.

Before plugging in the power cord, look around and make sure you know what's in contact with what. If you don't feel confident that you know what you're doing, stop and seek the help of a qualified technician.

Dual PSU assembly

For those who wish to etch and drill their own circuit board, actual size PC artwork is shown in Figure 2. You can also hand-wire the board on perforated board, although use of the circuit reduces the possibility of wiring errors and improves reliability. Physical layout is not critical, though you should keep the layout compact.

Consult the assembly guide, Figure 3, and the parts list to assemble the circuit board. Pay particular attention to the polarity of all diodes, capacitors, transistors and LEDs. Use a medium-heat soldering iron with resin-core solder.

To avoid overheating the components, prime the tip of the iron with a drop of solder before applying it to the component lead, then add a bit more to make a good connection. Avoid using excess solder, and watch out for bridges between copper pads.

It is preferable to mount the IC regulators and heat sinks to the circuit board before soldering, which reduces the stress on the solder joints that might occur if you soldered the legs first.

Note that if you want the output voltages set at ±15V, R1 and R3 are omitted, and R2 and R4 replaced by wire jumpers.

When all components are mounted to the circuit board, clip off excess lead lengths and double check your work.

External wiring

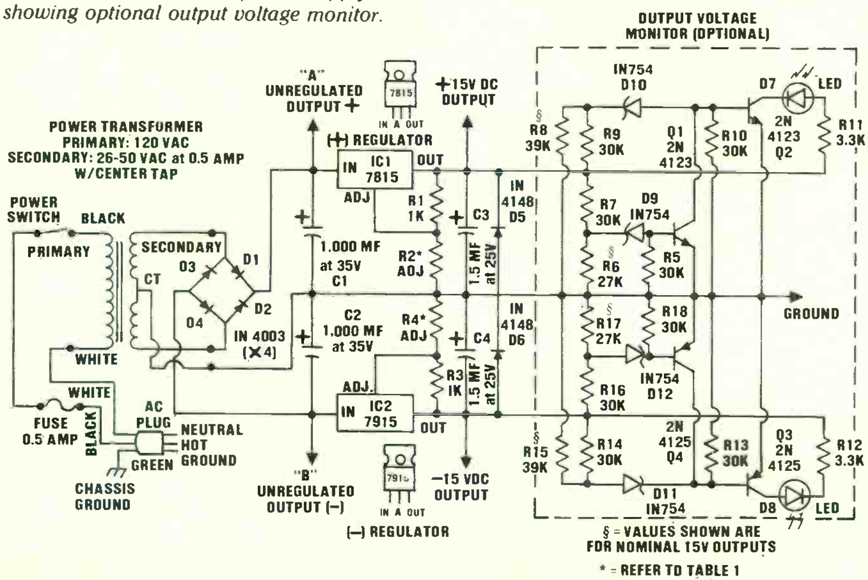
Figure 4 shows the power supply mounted in a typical enclosure, and can be used in conjunction with the schematic diagram to show proper wiring. For purposes of illustration, the terminals in the photograph were left uninsulated although, in practice, they should not be.

The power cord is a grounded-type, with the green ground wire attaching directly to the chassis wall as shown in the upper left-hand corner of the photograph. The cord itself is secured with a plastic strain relief. The white (neutral) wire from the power cord attaches directly to the transformer primary, which in this case is also a white wire. The connection was made with an insulated butt splice connector. The black (hot) wire from the power cord connects to one terminal of the fuseholder, then proceeds from the fuesholder's other terminal to one terminal on the power switch in the upper right-hand corner of the photograph. The second terminal on the power switch then connects to the remaining transformer wire, which is black on this transformer.

On the secondary side of the transformer (left-hand side in this photograph), the center-tap wire (striped) connects to the top of the circuit board at the point marked *CT*. The two remaining secondary wires then connect to the points on the circuit board marked *secondary*.

The twisted group of wires exiting from the middle of the PC board near the bottom of the photograph is the output of the power supply. These wires can connect directly to another circuit board, or attached to a suitable connector. Banana

Figure 1. Schematic of power supply unit showing optional output voltage monitor.



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Parts List

Resistors

All resistors are 0.25W; 5% tolerance; carbon film.
—R1, 3 (see text): 1kΩ (brown-black-red).
—R11, 12: 3.3kΩ (org.-org.-red).
—R6, 17: 27kΩ (red-violet-org.).
—R7, 9, 10, 13, 14, 16: 30kΩ (org.-blk.-org.).
—R8, 15: 39kΩ (org.-white-org.).
—R2, 4: see Table 1.

Capacitors

—C1, 2: 1,000μF at 35WVdc.
—C3, 4: 1.5μF at 25WVdc.

Diodes, transistors and ICs

—D1, 2, 3, 4: 1N4003 silicon rectifier diodes.
—D5, 6: 1N4148 silicon diodes (sub: 1N914).
—D7, 8: LEDs.
—D9, 10, 11, 12: 1N754A zener diodes.
—Q1, 2: 2N4123 transistor (sub: 2N3904).
—Q3, 4: 2N4125 transistor (sub: 2N3906).
—IC1: LM7815C positive 3-terminal regulator.
—IC2: LM7915C negative 3-terminal regulator.

Other parts

—Printed circuit board.
—Power transformer.
—Two heat sinks for TO-220-style case.
—Fuseholder with 0.5A 3AG-style fuse.
—Power cord with strain relief.
—Power switch, if needed.
—Wire nut, shrink tubing, standoffs and screws.

Parts are available from Gaines Audio, Box 17888, Rochester, NY 14617, 716-266-0780, as follows:
—Complete kit as described, including circuit board, transformer, all components except chassis: $42.95, plus $2 shipping.
—Complete kit fully assembled and ready for shipping: $57.95, plus $2.
—Circuit board only: $9.95, postage paid.

For V_{out}	R_{ADJ}
15V	Jumper
16V	47Ω
17V	91Ω
18V	130Ω
20V	240Ω
22V	330Ω
24V	430Ω

Table 1. R2 and R4 values for various output voltages. Note: If Jumper is used for R_{ADJ} resistors, R1 and R3 may be omitted.

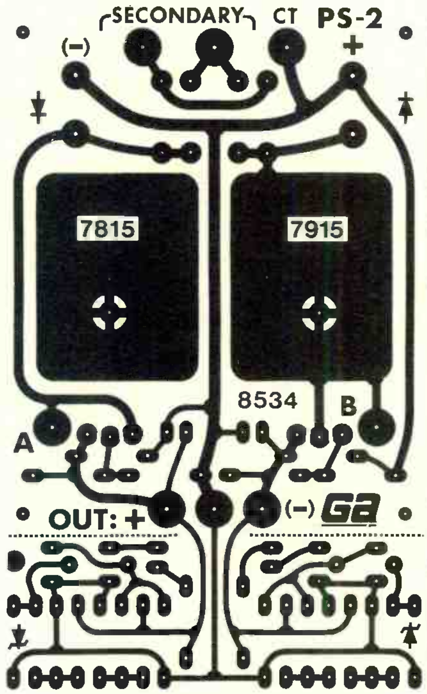


Figure 2. *Acutal-size PC artwork.*

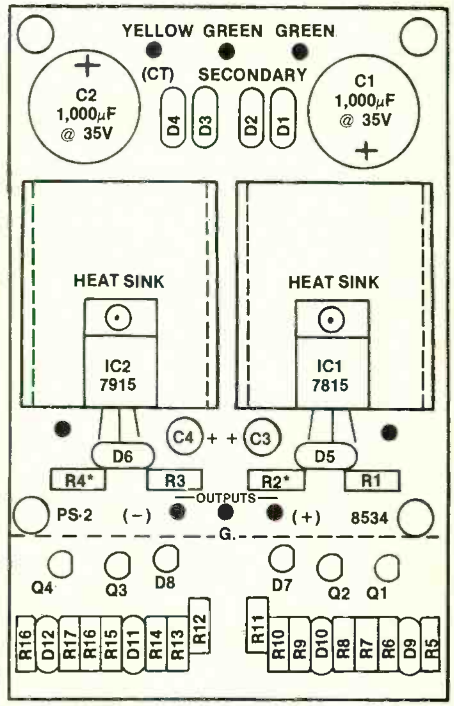


Figure 3. *Assembly guide for PSU.*

jacks, XLR connectors, and ¼-inch TRS phone jacks are popular choices for power supply connectors.

Testing the power supply

Assuming that you've put everything in the right place and soldered carefully, the supply should come right on and work, with the two LEDs beaming their approval. If they don't, immediately unplug the supply and feel around the circuit board to see if any components are hot. If not, it may be that the power supply is working, but the monitor circuit is not. You'll need to check with a voltmeter or scope to decide that.

If the fuse blows constantly upon plugging in the power cord, look for errors in the first section of the circuit: improper ac wiring, transformer wiring, or diodes 1 through 4 installed incorrectly. If the dc voltage at the input and output of the regulator is identical, check R1 through R4, or replace the regulator. If the input to the regulator is correct but the output is zero, look for a short circuit in the output or monitor circuit wiring, check clamp diodes D5 and 6 or suspect the regulator.

If the project seems to be working correctly, and the LEDs are glowing, you can check the functioning of the output monitor circuit by deliberately short circuiting each output momentarily to ground. (The regulators are internally protected against short circuits and will not be harmed.) The appropriate LED should immediately go out, and then reilluminate when the short is removed.

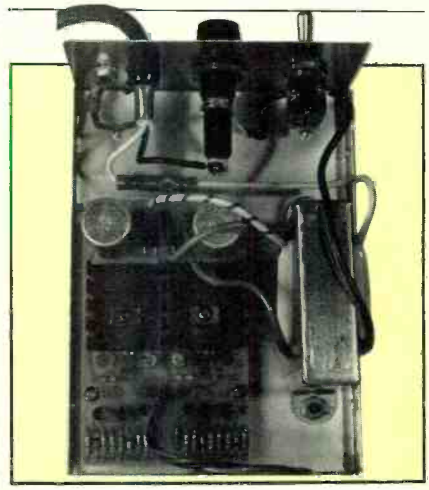
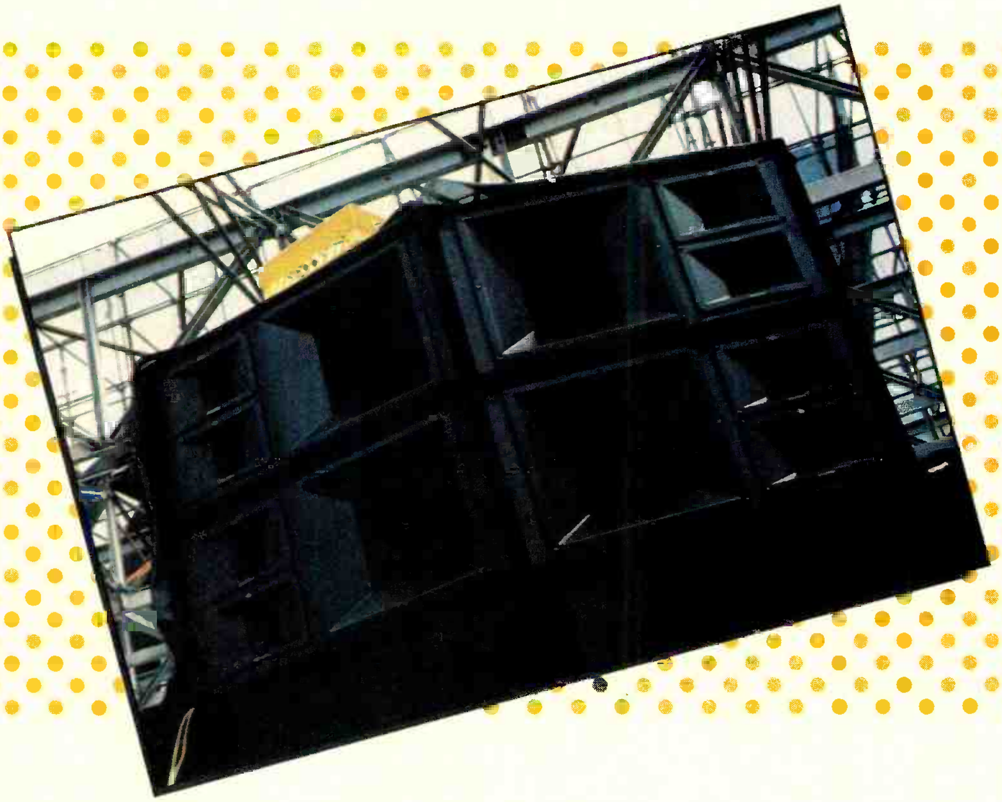


Figure 4. *Completed power supply.*

In use, the LEDs should glow fairly brightly and the regulators warm to the touch, but not so hot that you can't keep you finger on them. Capacitors, diodes and resistors in the circuit should not get hot. R·E/P

Gaines is director of the engineering staff at Ashly Audio, Rochester, NY.



Field Testing New Concert Sound Technology

By David Scheirman

One of the latest full-bandwidth systems features an integral signal processor in a easy-to-rig package.

The prebuilt, full-bandwidth loudspeaker system packages developed and brought to market by several manufacturers within the past few years have given those in need of sound reinforcement systems some interesting options. Considerable time can be saved when a purchaser chooses a prebuilt system. In such instances, the end-user relies on the manufacturer for research and development.

When an integrated electronics package is also available, the end-user has even fewer decisions to make regarding drive components. The inclusion of power amplifier racks and cabling, along with optional hanging hardware points, makes it easier than ever before to purchase a complete, ready-to-use sound system.

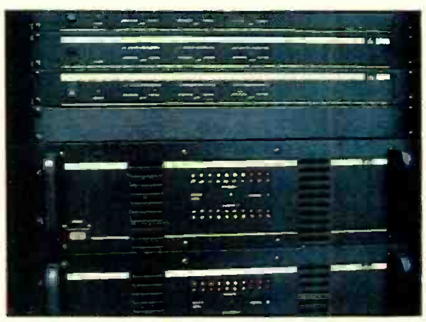


Figure 1. Smart System processors combine active electronic crossovers with phase compensation circuitry and excursion/thermal overload protection. Shown are the X3-M1 processors, along with Crest power amplifiers.

In attempting to make a purchasing decision, sound system users are faced with a growing number of choices regarding which available, prebuilt sound system will best suit their particular needs. Vocal and music reproduction, live performance and high-level playback use, indoor and outdoor applications . . . rarely have sound systems offered ideal performance for widely varying situations. A sound system representing the "perfect compromise" has yet to be designed.

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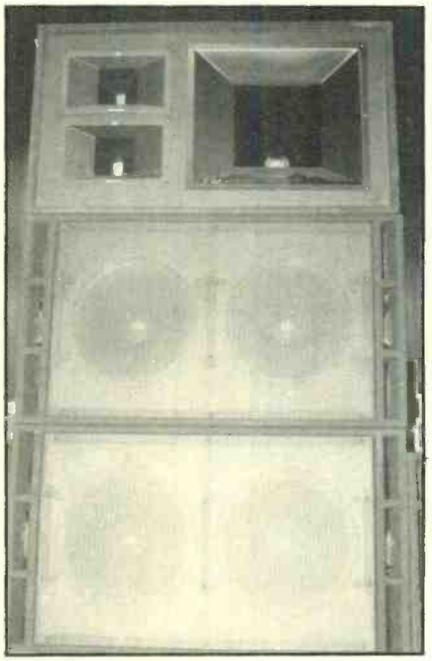
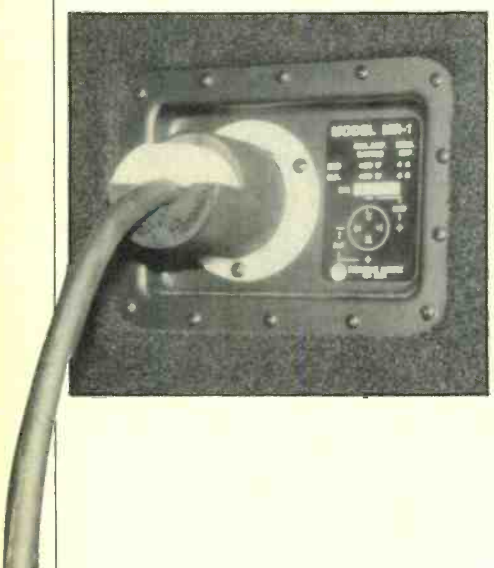


Figure 2. *The MR1 and LR2-M loudspeaker modules are identically sized. Each enclosure is 48"x27" and is wedge-shaped to enable the easy assembly of arrays.*

Figure 3. *Enclosures may be specified with heavy-duty, twist-lock road connector speaker cable panels. Enclosures are covered with all-weather carpet.*



improve sound quality and maximize acoustical output from compact enclosures is Renkus-Heinz. In 1983 the company's SWG series introduced "fail-safe" compression driver operation, with electronically enhanced and extended high-frequency performance.

In 1984, the Smart Crossover concept was expanded to include driver protection and performance enhancement for the low-frequency section of speaker systems. In 1985 the company introduced compact subwoofers and the Smart Jr. System, a single-amplifier "smart" sound system intended for use in live music clubs, discos, church and theater installations, and general sound reinforcement.

The combination of these various technologies into the company's new MR1/LR2-M modular full-range speaker system, introduced this past spring, represents an attempt to create an all-purpose, high-level system for portable and installed uses.

Smart System processing

Renkus-Heinz engineers have stated that, for ultimate reliability in sound reinforcement systems, there is a need for automatic speaker protection. [This concept is detailed in "A New Approach to Power Handling," in the March 15, 1985, issue of *Sound & Video Contractor.]*

Loudspeaker components have obvious power handling limitations. When a loudspeaker's operating limits are approached, such complications as setup mistakes, system oscillation and operator error can lead to loudspeaker failure.

Driver protection for both low- and high-frequency loudspeaker components was an initial design criterion for the current generation of Smart System processors. The Renkus-Heinz processor systems deal separately and independently with excursion and thermal overload limits of each speaker. A new circuit developed by the company is said to work with the problem of loudspeaker overexcursion at high operating levels, while allowing the use of lower crossover frequencies with compression drivers. The circuit allows compression drivers to have their lower frequency response limit to be extended down into the midrange region.

The MR1/LR2-M loudspeaker system is intended to be used with the model X3-M1 processor, a unit that combines an active 3-way electronic crossover with integrated subwoofer drive capabilities, phase compensation for time-coherent system performance and excursion/thermal overload protection for all drivers in a single package that is one standard rack-space high (Figure 1).

An 18dB per octave crossover slope is used with crossover frequencies for the MR1 system set at 300Hz and 2kHz. Smart processors are said to constantly monitor the signals applied to each loudspeaker and, if required, take corrective action. In the case of the X series processors, such corrective action includes shifting the crossover frequency point. The "floating crossover" enables the speaker systems to achieve extremely high average sound pressure levels with no damage to the speaker components.

An automatic loudness compensation circuit (rear-panel switchable) boosts the system's low-frequency output up to approximately 10dB at lesser volume levels; high-frequency equalization (the inverse of the normal, mass-induced roll-off of the compression drivers) offers flat frequency response to 17kHz uniformly within the coverage angle of the horn and is said to eliminate the need for high-frequency tweeters. The low-frequency boost is gradually removed as the signal level to the system is increased.

In summary, benefits afforded by use of the Smart processor are said to include lower crossover frequencies for compression drivers (allowing optimum performance in the operating range), extended bass frequency response by as much as one-half octave with the use of enclosure equalization and protection of the speaker components during heavy use.

Enclosure design

Due to processor calibration and programming required for optimum matching with loudspeaker components and enclosures, the Smart processors are each dedicated to particular system designs. The controllers cannot be adapted for general use with other speaker systems.

The enclosure configurations chosen for the MR1/LR2-M project include wedge-shaped boxes, facilitating the easy construction of cluster arrays. The MR1 module (mid/high-frequency) and the LR2-M subwoofer are identically sized. (Figure 2.)

Each speaker module is 48 inches wide and 27 inches high, with a depth of 24 inches. The mid/high module weighs 160 pounds, while the LR2-M subwoofer weighs 175 pounds. A stack of three cabinets (one MR1 and two LR2-Ms) has a net weight of 510 pounds.

Figure 4. The SR1 houses a single 15-inch loudspeaker and one compression driver. The LR2 subwoofer contains a pair of speakers, manufactured for Renkus-Heinz, by PAS.

Figure 5. Gemini Stage Equipment and Lighting supplied a pair of SR2 enclosures to the Fort Worth Blues Festival for use as on-stage sidefill monitors. Shown here is Tim Cain of Gemini's sound department.

of invited audio industry members. Initial listener reactions to this first outdoor testing included surprise at the high sound pressure levels available from a relatively small point source, and comments on the system's ability to convey impressive amounts of low-frequency energy to the rear of the outdoor amphitheater's seating area.

The true test of any sound reinforcement system that is intended for live performance use, however, is an actual musical performance. To more accurately gauge this speaker system's performance, I attended one of the first concerts to use the stacks, the Fort Worth (TX) Blues Festival, held this past April.

The annual event was held at the Tarrant County Convention Center, which has approximately 60,000 square feet of floor space. A crowd of 6,000 seated at tables attended the event, which featured entertainers such as the Rose Brothers, Betty Wright, Bobby "Blue" Bland and Johnny Taylor.

The listening area was an acoustically poor room for amplified musical performances; concrete floor and masonry block walls contributed to a highly reverberant enclosed space. The 40-foot high ceiling was supported by steel girders that were not acoustically treated.

Sound system services were provided by Gemini Stage Lighting and Equipment, Dallas. Started by brothers Dell, Terry and Tim Cain in 1984, this regional sound and lighting company became a Renkus-Heinz Smart System dealer in 1985. The equipment obtained for retail sales demonstrations was put to use as part of the company's rental stock.

"The sound-system rental business is a very competitive market," says Dell Cain. "We have found a need for sound systems that are well-packaged and easy to use. They have to be practically foolproof as well, because much of our work is with clients who are interested in equipment-only rentals. We can't afford to have something that gets taken out and then blown up."

Originally, Gemini purchased multiples of the Renkus-Heinz SR1 enclosures and LR2 subwoofers. (The SR1 comprises a single 15-inch loudspeaker and one compression driver on a constant beamwidth horn mounted in a compact trapezoidal box. The LR2 houses a pair of 18-inch loudspeakers in a rectangular vented enclosure, as can be seen in Figure 4.)

For the festival, Gemini brought in these enclosures, along with a pair of SR2 boxes for on-stage sidefill monitors (Figure 5). The total loudspeaker complement for each side of the stage included eight 18-inch speakers, four 15-inch speakers, a single SSD-5600 midrange compression driver and six 2-inch compression drivers (Figure 6).

This particular event provided a unique opportunity to compare the part of the stack housing the large-format compression driver, with the more traditional type of system featuring 15-inch loudspeaker cones for midrange reproduction.

A Soundcraft 800B-32 mixing console was set up so that the two different parts of each stack were fed from the console's output matrix. Levels were set with an oscillator so that power amplifier sections for each part of the stack clipped simultaneously. In this manner, the relative performance of the two types of speaker systems could be observed.

Figure 6. Each loudspeaker stack for the 6,000-person audience contained eight 18-inch PAS speakers, four 15-inch EV-15L speakers, a single Renkus-Heinz SSD-5600 midrange compression driver and six 2-inch compression drivers.

Comparison results

The part of each speaker stack that included the large-format compression driver (Renkus-Heinz SSD-5600) apparently had an increased average sound pressure level output, both audibly and as measured with an SPL meter. The large-format compression drivers did reproduce music well, particularly during playback of prerecorded material. This type of transducer does seem to be

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DIGITAL SIGNAL PROCESSING FROM ROLAND

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suitable for applications beyond the vocal-only, large outdoor venue announcement systems, although it is becoming a common sight at stadiums and racetracks.

As well as producing slightly increased sound pressure levels in the midrange frequency region, the SSD-5600 seemed to offer better articulation on music program material in that 2-octave region centered at approximately 1.5kHz. From 750Hz to 3kHz, intelligibility throughout the listening area was improved with the MR1/LR2-M stack, as compared with the stacks containing 15-inch midrange drivers.

Live music use

During the Fort Worth Blues festival, I noticed a surprising amount of available system gain from relatively small loudspeaker stacks. While listening to live music, I noticed that there was a particularly strong system response in the 80Hz to 160Hz region. These two system characteristics were possibly due to the Smart processor; that unit's automatic loudness compensator and bass boost gave the overall sound of the musical program a very strong bass presence.

In general, the on-stage instruments that were reproduced most authentically with that particular system proved to be percussion. The show's drum sound was dramatic, and the kick drum carried well throughout the listening area. It was oc-

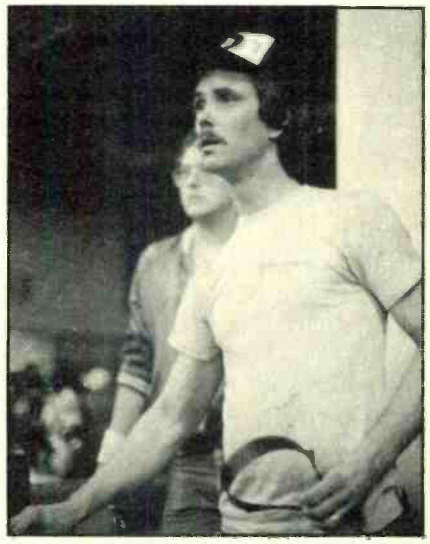


Figure 7. *Dell Cain, head of Gemini's sound department, at the house mix position for the festival.*

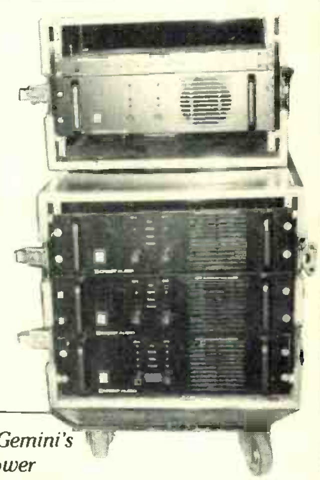


Figure 8. *Gemini's compact power amplifier racks housed Crest model 4000s.*

casionally the most prominent part of the mix, although not intentionally. Low-frequency resonances were difficult to remove with console equalization.

Signal processing equipment Gemini supplied to accompany the 800B mixing console included White third-octave series 4000 filter sets, dbx model 160X compressor-limiters and effects devices from Lexicon, Yamaha and Roland.

"When it comes to purchasing new gear, we try to get a feel for what different rental customers want by looking at the contract riders that come in for the many one-nighter shows we do in our area," says Cain, who is head of the sound division (Figure 7). "We also try to carry product lines for our retail sales division that can be used with our rental department."

Gemini's compact power amplifier racks housed Crest model 4000 amplifiers (Figure 8). The same units were also chosen for the company's stage monitor system (Figure 9).

Figure 9. *The festival on-stage monitor mix position. Crest power amplifiers and a Soundcraft console were used.*

Speaker system controls

As with other prepackaged loudspeaker systems that use integrated electronic signal processors, the Smart System processors were located at the stage, in the power amplifier racks. The house mixing position was approximately 120 feet away from these units. I found the lack of control over the various relative levels of the system's frequency bandpasses to be somewhat frustrating. Unlike other processor-controlled loudspeaker systems, even the balance of the subwoofer levels in relation to the rest of the system could not be adjusted at the mixing position. Instead, gain settings on the power amplifiers themselves were changed as needed. During the show, this required an intercom and helper.

However, this lack of control does not bother some sound system operators. I found myself thinking about creative solutions to the problem: perhaps a console-located VCA array with an additional control cable running back to the power amp racks?

Subjective comments

Having listened to the SSD-5600 large-format midrange compression driver with both prerecorded playback material and live music indoors and outdoors, I felt that this driver has some potential applications in both areas.

The driver's exceptionally efficient reproduction characteristics in the midrange frequencies tended to sound a bit strident indoors, with live musical material that included both male and female voices and full horn sections. These same characteristics, however, gave the identical system an apparent advantage over cone-midrange devices with prerecorded music playback in an outdoor setting.

For several reasons, the live performance that I attended did not offer optimum conditions for using and evaluating a new speaker system. I would welcome the chance to experiment with the system again.

The MR1/LR2-M system seems to be particularly well-suited for use as a high-level sound reinforcement system for theme parks, sporting events and anywhere that vocal and music program material must be distributed to large crowds. Like other Renkus-Heinz Smart Systems, the new MR1/LR2-M sets up quickly and easily, due to its integrated processor. It seems to offer more sound-generating capabilities, pound for pound, than some other commercially available speaker systems.

As an integrated engineering project, the system represents an attempt by one manufacturer to address a question that is decades old: How best to build and package a full-bandwidth, portable sound-reinforcement system that works well in a wide variety of applications?

By following the recent trend toward wedge-cut, arrayable cabinetry, and the use of integrated electronic signal processors, Renkus-Heinz has established a system that is competitive in the pro sound marketplace and should be listened to by those interested in ongoing evolutionary developments of prepackaged loudspeaker systems.

It seems probable that the stage of advancement in the field of sound reinforcement system design has been set for new developments in the electronic processing of audio input signals. It is certainly easier to effect changes in loudspeaker performance electronically, rather than attempting to stretch the boundaries of available materials and manufacturing technology for loudspeaker construction.

Loudspeaker systems available from several different manufacturers that incorporate electronic signal processing as an integral part of the sound system package are possibly the wave of the future. A decade from now, it may be difficult to purchase a new speaker system that does not come equipped with its own dedicated electronics package, fine-tuned for optimum performance. R·E/P

Editor's note: The mention of specific products is not to be taken as an endorsement by **RE/P** or Intertec Publishing. The system has been detailed for the purpose of satisfying reader interest and educational needs.

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

Photos by David Scheirman.

Design and Construction of Puk Studios

By David Rideau

A new residential facility is equipped with a 5-way custom monitoring system and LEDE acoustics.

To many engineers, producers and musicians who have used the facility, it is no small exaggeration to say that Puk Studios represents one of the most interesting and innovative approaches to acoustic design in recent years. Taking a rare, "spare-no-expense" attitude, its designers transformed a previously obscure studio operation into an ultra state-of-the-art recording facility.

The studio's history, as told by John "Puk" Quist, owner/chief engineer, is an interesting one. Originally a quaint farm nestled in the Danish countryside, the site was bought in 1966 and transformed into a monastery that ran a drug rehabilitation program. In 1975 the owner died and the run-down property was purchased by Birte Quist.

At the time, Birte was not living on the farm, but in a nearby town where she met John, an aspiring musician/engineer. Quist soon realized that the farm buildings would make a perfect home for his newly purchased Soundcraft console.

From these humble origins, Quist's studio grew to become a simple, but well functioning, facility that was used to produce many Danish hits.

Not content with success on the local level, Quist set out to secure funding to build his dream studio, which would raise Puk to an international standard.

Studio financing

To understand the problem of financing and maintaining a studio in Scandanavia, one should be somewhat familiar with the region's record and film industry. On one hand, you have an audience that has come to expect a high level of technical excellence, as seen and heard in U.S. and other countries' releases. But at the same time, there exists a limited buying audience, which almost eliminates the chance of making a profit. (The population of Denmark only is 6 million.)

The film industry gets around this problem with generous government grants to filmmakers. The Scandinavian record industry is not as fortunate and survives by allotting modest budgets to its recording artists. Such budget restrictions are reflected in session payment from the producer to the bottom of the food chain: the recording studio.

So how does a Danish studio owner survive? Two top rooms in the area—Easy Sound Recording, Copenhagen, Denmark, and Polar Studios, Stockholm, Sweden (ABBA's studio)—have done so by building their rooms in cities that guarantee a certain amount of international traffic. In both cases this method has worked well.

Quist, however, wanted to keep his facility where the only traffic is an occasional cow. In this case, it was a gamble that paid off. The Danish government, interested in generating revenue in this less-populated area of the country, was willing to provide a large, low-interest loan, the first payment not being due for two years. Puk was on its way.

Design concepts

The idea was simple enough: to build an extremely advanced recording facility, with no architectural restrictions. All design parameters had to conform to digital standards of dynamic range, frequency bandwidth and system distortion.

The team chosen to execute this idea included Andy Munro, of Munro Associates, London, who was mainly responsible for the control room design; and Ole Lund Christensen and Knud Rosenskjold, of the Danish company SLT, who would take care of the technical and electronic aspects. Architect Mogen Hansen, who was responsible for much of the interior design, translated construction assignments into reality for the local building crews.

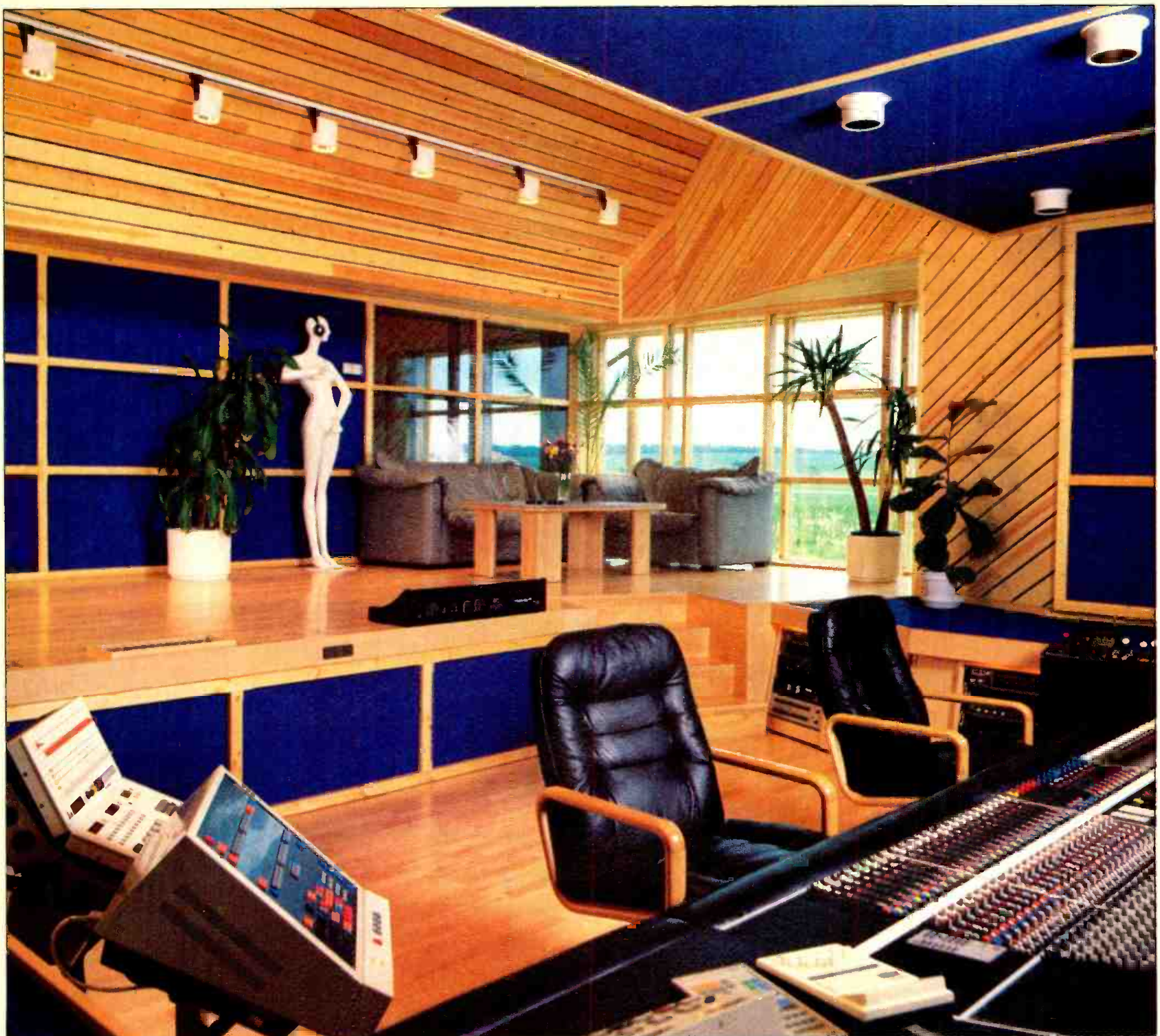
The basic approach for all aspects of the design was to establish a goal, look at it theoretically and then try to develop it in real terms. This often led to abandoning traditional ideas or trying several approaches to solve a particular problem. Sometimes the discussions between Puk owners and the designers didn't lead to an immediate agreement, but eventually a working solution was reached.

Control room acoustics

For Quist, the most important consideration was the control room design. As a working engineer, he realized that more and more recording was taking place on his side of the glass. So in addition to the traditional area in the control room for the engineer and producer, there should also be a stage for several musicians and their instruments.

Quist took this concept a step further, feeling it could be a psychological boost to have the musicians raised on a platform behind the console, so that the players could look down on the engineer and producer seated before them.

Quist also felt that it would be advantageous to incorporate a number of windows in the room design to provide



views of the scenic countryside, generating an airy atmosphere with natural lighting.

It was agreed as much as possible to base the control room acoustic design on the Live-End/Dead-End (LEDE) concept. Admitting that no design is perfect, Christensen pointed out that the well documented results of time-delay spectroscopy (TDS) demonstrate that no reflective surfaces should be placed around the monitors to reduce early reflections off the walls into the monitoring position behind the console.

Andy Munro was the first to admit that "the design of Puk's control room represented new territory for acoustic theory." The room's volume of 3,200 cubic feet, he says, "created a special problem, in that a room of this size—even if well-damped—normally would have an RT60 of 0.5 seconds. Because conventional LEDE designs would not work in such a large space, a new approach was taken to ensure that

A dedicated tape machine room houses (from left to right) a Mitsubishi X-800 digital 32-track, an Otari MTR-90 24-track, MTR-10 2-track, Mitsubishi X-80 digital 2-track and a second MTR-10 2-track.

path lengths of reflections were optimized for both the monitoring and rear performance areas."

The room was, in effect, split into two parts. The large rear space is virtually self-contained, yet adds enough diffused sound field to the front to create the impressions of a rigid rear boundary when one is sitting at the console.

"Room acoustics perceived by musicians working in the rear of the room is two-fold," Munro says. "Sound emanating from the monitors maintains good stereo imaging, without excessive bass lift, while sound created at the back of the room—vocals, for example—sound crisp with their own natural ambience. The geometry required to achieve this degree of acoustic control involved a very high ceiling and critically damped, slat diffusers.

"The problems of early reflections and diffraction around the console have been solved by the use of an acoustic walkway around the front half of the room. It can support a person's weight but, at the same time, is 99% absorbing."

Recording areas

Directly behind the new control room is what's referred to as Recording Area

A. This is an extremely live room, housing a Bosendorfer grand piano, with three of the walls made of glass to take optimal advantage of the impressive view available from this side of the studio complex. To further enhance the room's acoustic and aesthetic properties, a cathedral ceiling has been created on one side, with vertical beams and glass in between. To control standing waves, all the remaining walls and ceilings use the same method of damped-slat diffusers as found in the control room. The floor is pine.

The larger Recording Area B has been positioned more traditionally in relation to the control room. Once again, the musicians are on a slightly higher level than the engineer and producer facing them through the control room window. This room also has a cathedral ceiling and wooden floors, but is less live than area A because there is less glass. Originally part of the farm, the room retains the old beams, from which suspended curtains can be pulled across as required to modify the room's acoustics.

View from the performance area in Control Room A, which features a 56-input Audio+Design/Calrec UA-8000 console and custom 5-way monitoring system.

Recording Area C faces Puk's original control room but is intended to be available to both control rooms. One end of the room's acoustics is live with a beech platform and pine ceiling. The acoustically dead area of the room is carpeted and the walls are treated with dense acoustic tiles.

In addition to the three main recording areas, a vocal booth to the right of the new control room is described as the only truly dead acoustic environment in the facility.

The existing Studio B is somewhat modest in comparison to its newer brother. Monitoring is provided by JBL model 4333s, and the control room is equipped with the same 28-input Soundcraft series 2400 console (serial No. 0001) that Quist brought to the farm several years ago. Although not state of the art, a smart producer could use this room for preproduction and synthesizer programming, allowing the new Studio A to be used to continue the recording process.

Monitoring system

Design of a control room monitor system proved to be one of the most difficult, yet most rewarding, aspects of the project. Quist informed Christensen that he wanted the system to be able to operate at very high sound pressure levels. Although such a request is not that unusual for a monitor system designed to handle the frequency response and dynamic range of digital playback, it would be quite a design challenge.

When Christensen and Munro worked on the basic room design in 1982, both realized that Puk probably was the first digital-capable control room being designed from the ground up.

The design parameters upon which they agreed were: a dynamic range of

Instruments available to clients include New England Digital Synclavier II and Fairlight CMI Series II synthesizers.

Puk adds Second Control Room

*In early July, **RE/P** learned that Puk Studios recently completed construction of a second, identical control room. Studio B, also designed by acoustician Andy Munro, of Munro Associates, London, features a 56-channel Solid State Logic SL4064 console, three Sony PCM-3324 digital multitracks (available for use in both Studio A and B) and a Fairlight CMI Series III digital synthesizer.*

According to studio owner John Quist, "Even though we are very pleased with the sonic qualities of the Calrec UA-8000 console in Studio A, there was a demand amongst our clients for an SSL-equipped studio, which is why we chose the SL4064 with Total Recall and Primary Computer for the new facility."

In addition, the studio complex has exchanged its existing Mitsubishi X-800 digital 32-track for two new X-850s, which Quist says allows clients to make digital-to-digital safety copies of multitrack masters.

"Also, it's the ony way to utilize the cut-and-splice feature of the X-850, because you have to copy the spliced tape in order to store it properly," he says.

During subsequent evaluation tests with a Techron TEF-10 analyzer, Munro reports that overlay comparison of the time delays spectrometry plots of both rooms indicated practically identical RT60 and acoustic performance in studios A and B.

around 120dB, an extended (flat) frequency response and a negligible amount of background and monitor noise. The only way these exacting criteria could be achieved was to build a very large room, which would eliminate low-frequency standing waves. The end result was 3,200 cubic feet of space in the control room.

Monitor loudspeakers were to extend in frequency response from 17Hz to 25kHz, with appropriate rolloffs at each end. Christensen and Munro wanted a monitoring system that could relate in subjective terms to other, larger cabinets, but with a subwoofer providing another octave on the low end. Such a configuration also had to maintain the 96dB dynamic range of digital playback with transient peaks approaching much higher SPLs. Using a background noise level of 20dB as a reference—representing the projected noise floor of the control room—it became obvious that no commercially available system could accommodate the dynamic range requirements while maintaining low distortion levels.

Recording Area A features a Bosendorfer grand piano. To improve visual communications, at right is a window linking the area with Control Room A.

Finally, SLT and Munro designed their own monitor systems. Christensen's main role in the project was to carry out the tremendous amount of research required. With a degree in electroacoustics, he was qualified to put together all the personnel and information needed to get the job done.

When attempting any serious acoustical task anywhere in the world, especially in Denmark, one company somehow always comes into play: Bruel & Kjaer. This time it was Peter Ladegaard, a B&K employee using the company's TDS measuring system, who helped Christensen develop the best monitor configuration for the new control room. One of Christensen's conclusions was that to reproduce a 20Hz wave as opposed to 40Hz, the law of physics requires four times the movement of air—therefore, the LF cone should be four times the normal size.

When questioned about the usefulness of having a system that could reproduce an earthquake, Christensen was quick to point out that, until now, "analog recorders have been kind enough to filter anything below approximately 40Hz. But, with digital recordings and Compact Disc players in full swing, it was imperative for a session engineer to know *exactly* what is going on below 40Hz."

Rosenskjold was called upon to develop the necessary crossover network for the monitoring system. The result was an electronic 5-way system, with the crossover frequencies satisfying the complex requirements of each driver. Highlights of the system include an inverted JBL model 2241 acting as a large-dome tweeter, thereby eliminating the problems of phase and reflections that the large horns tried earlier had produced. The component that would meet the designer's low-frequency/high-level requirement was a Fostex model L869 30-inch woofer.

The critical choices of system components and Rosenskjold's crossover adjustments were done on-site with information gathered by Munro using a Techron TEF-10 analyzer.

"Without time-delay spectrometry to provide anechoic measuring conditions," Munro says, "it would have been far more difficult to obtain clean data at this point of the project."

The system is said to deliver up to 140dB SPL on transient peaks, giving a potential usable dynamic range of 120dB (assuming a background noise floor of 20dB).

The surface of the monitor wall is totally absorbent and the speaker mountings are completely isolated from the room to avoid transmission of sound vibrations. Because of the system's high SPL and low-frequency performance, Munro decided that the housing of each monitor should be made of dense concrete.

The 5-way amplification system driving the cabinets consists of E.A.R. amplifiers for the high-frequency drivers, with the balance of power being supplied by custom-built SLT MOSFET amplifiers operating in bridge mode. The total amp power is 4,000W per side, bringing the total monitor system cost close to $30,000.

Puk Studios Equipment Summary

Consoles: 56-channel Audio+Design/Calrec UA-8000 with Audio-Kinetics MasterMix automation (Studio A); 28-input Soundcraft series 2400 (Studio B).

Tape machines: Mitsubishi X-800 digital 32-track; Mitsubishi X-80 digital 2-track; two Otari MTR-90 24-tracks; two Otari MTR-10 2-tracks; Sony PCM-701 processor (all transports are shared between Studios A and B); Nakamichi cassette decks.

Outboard effects: EMT 251, Klark-Teknik DN780, AMS RMX-16, Lexicon 224XL, Yamaha REV-1 and Quantec QRS digital reverbs; EMT 140S stereo plate; two AMS DMX-1580 delays; two Eventide H949 Harmonizers; Eventide Instant Flanger; Lexicon model 92 delay; Aphex Studio Aural Exciter II; Aphex Compellor; two dbx model 165A compressors; dbx model 903 compressor, model 902 de-esser and model 904 noise-gate modules; Audio+Design Transdynamic and Vocal Stresser; Klark-Teknik DN30/30 graphic equalizer.

Monitor system: 5-way custom configuration, using two 30-inch Fostex subwoofers, two 15-inch Fostex units, an 8-inch JBL model 2445 and a Fostex bullet tweeter on each side (Studio A); in addition, JBL model 4350, 4333, 4312 and 4411 cabinets; Yamaha NS-10M and Auratone Sound Cubes are available as required.

Instruments: New England Digital Synclavier II; Fairlight CMI Series II; Bosendorfer grand; Oberheim OB-8 synth; Fender Rhodes; Hammond A-100; Slingerland trap set.

Additional equipment: Audio-Kinetics Q.Lock 3.10 time code synchronizer; Garfield Doctor Click; Friend Chip SRC.

Sound isolation for digital

Because of the stringent dynamic range requirements of the control room design, it was decided from the beginning to construct a completely separate machine room to reduce background noise from the recording equipment. The newly purchased Mitsubishi X-800 32-track and X-80 2-track digital machines, as well as almost all fan-cooled pieces of equipment, are housed in this location. Units that must remain in the control room are located in a special rack system, which was designed so that each individual rack could be rolled to the middle of the room. From this position an engineer can make critical adjustments to outboard gear at the optimum listening point in the sound field. Acoustic treatment of the rack ensures that it does not cause sound reflections or upset the stereo imaging.

Another source of background noise, the air conditioning system, was eliminated by Quist's spare-no-expense approach. The system was built to

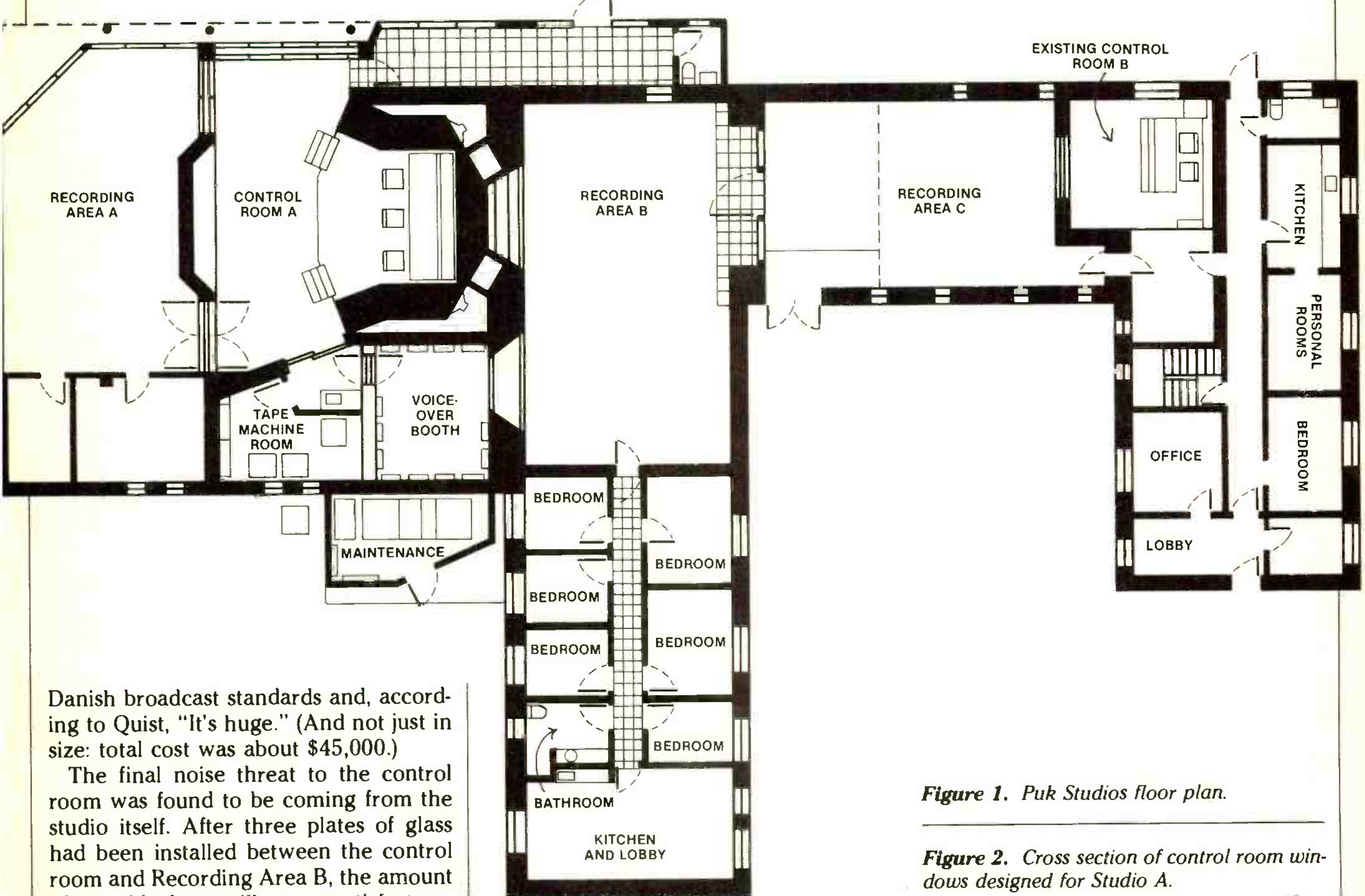


Figure 1. Puk Studios floor plan.

Danish broadcast standards and, according to Quist, "It's huge." (And not just in size: total cost was about $45,000.)

The final noise threat to the control room was found to be coming from the studio itself. After three plates of glass had been installed between the control room and Recording Area B, the amount of sound leakage still was unsatisfactory. To overcome the problem, the original glass was replaced by 0.7-inch laminated panes. There are now three being used, each consisting of a glass-plastic-glass sandwich, with the middle pane being thinner to eliminate any common sound resonances.

The construction process took about six months. Standard floating construction techniques were used, with all the rooms being completely isolated from each other through breaks in the foundation. As an added bonus, the area's soil is predominately clay, which acts as a natural floating isolation material.

Another main acoustic consideration involved the large number of windows facing the outside world, which had to be carefully designed and constructed.

Console selection

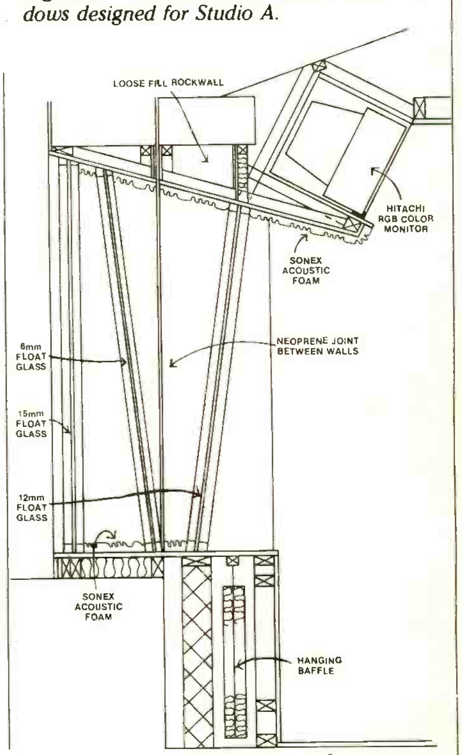
When it came time for choosing a console, Quist became interested in a new board from Audio+Design/Calrec. He had learned that Polar Studios researched the capabilities of various boards and eventually had chosen Calrec to build a custom console. Realizing that Polar has a reputation for never cutting corners, the Puk team decided to take a look and ultimately purchased a UA-8000 that includes a mixture of traditional Calrec and Polar specifications.

Puk's UA-8000 comprises a 64-channel mainframe fitted with 56 inputs. Each channel module offers two inputs, mic/line and monitor; during mixdown there are a total of 112 available channels.

Before visiting Puk, I hadn't seen or heard of the UA-8000. (I thought Calrec made only broadcast mixers and microphones.) During a month-long project at the studio, I had the opportunity to work on the console. From personal experience, there aren't very many good things I can say about the majority of consoles on the market. For example, I'm tired of manufacturers cramming in every possible "in-line feature" and almost always willing to sacrifice the integrity of the signal path.

Not so with the UA-8000. I was par-

Figure 2. Cross section of control room windows designed for Studio A.



ticularly impressed with the board's basic sound quality. Even the mic pre-amp was respectable. And with an in-line dynamics chain, zoned status control and two VCAs per channel module (channel and monitor fader), the UA-8000 handled the most complicated setup.

To augment the console's primary functions, Quist has equipped it with an Audio-Kinetics MasterMix VCA-based automation system. Even though MasterMix has several shortcomings involving its disk storage system—for example, you cannot label mixes—I did remix an entire album using the system.

The studio is well equipped with microphones, including Neumann M49s and U47s (tube-type), a Calrec Soundfield and Sanken and B&K models.

One of the finest features of Puk Studios is that it owns both a New England Digital Synclavier II with sample-to-disk and music printing options (30Mbyte Winchesters allow up to five minutes of mono sampling) and a Fairlight CMI Series II, which the facility soon hopes to update to 16-bit, Series III specifications. Both systems are included in the rental cost of the new Studio A.

Session experience

I asked Quist if he considered it profitable to build a world-class facility in an area that, in U.S. terms, would be the equivalent of the outskirts of Blue Rapids, KS. He confidently stated that "there would always be bands, even solo artists, who want a clean, quiet environment in which to develop their musical ideas." So far Quist has proved himself right and has kept the facility booked solidly during its first year of operation.

During my visit, a German band was in Studio A with producer Terry Britton, who produced Tina Turner's "What's Love Got to Do With It?" Britton and the band seemed to be very pleased with the facility and were using the performance area in the control room to record synthesizer parts.

The band members were living at the studio in a dormitory-type complex that has space to comfortably sleep at least six people. I stayed the night at Puk with my son Vince and an American engineer, Robert Macias, who was interested in seeing the facility. Even with Britton's band, a Danish band in Studio B and our unexpected visit, Birte Quist (now the facility's general administrator) kept things running smoothly. The studio staff caters to your needs, and meals are included with a session booking in Studio A.

As we chased cows with Pilot, the studio dog, picked fruit off the trees and generally enjoyed the pastoral setting, I imagined this could be the perfect relief from a tough recording schedule.

The studio complex is located on the peninsula side of Denmark in Jutland, which is just to the north of West Germany. The easiest way to reach Puk from the United States, or any major city in the world, is to fly to London or Copenhagen. From either location there are direct flights several times daily to the international airport at Aalborg. Then it is an hour's drive to the studio.

Closeup of Control Room A's right-hand monitor bank.

(Quist guarantees that his Land-Rover will be there to pick you up in even the most extreme weather conditions.)

From personal experience and conversations with visiting and staff engineers, Puk Studios seems to meet all of the design team's expectations. I found that the facility has the same light and airy feeling that Frank Lloyd Wright expressed in many of his architectural works, and one in which a fine line exists between the building and its environment.

The control room platform is about 12 yards wide and shares the same view to its left as Recording Area A. Standing there beneath the cathedral ceiling, I imagined that the environment could be an exhilarating space for musicians.

I'm not particularly fond of large monitors and find most of them to be misleading. But compared to most, the Puk 5-way system provides good imaging depth and clarity. And it could play loud! I would speculate that because of its extremely low distortion figures the system could trick you into using higher SPL levels for longer periods of time, a potentially dangerous situation for one's hearing.

When monitoring at a reasonable level, however, and moving back and forth on the rear platform, there was a close facsimile of what I heard down at the console. One exception was a reduced sound level, which decreased even more as I moved to either side of the performance area. On the plus side, the various people working in the control room can choose their own SPL in relation to that set up by the engineer.

As might be expected, Quist exceeded his projected construction budget. (I understand that it had tripled by the time Puk Studios was finished.) He wasn't even phased by the cost of updating equipment. Although it is safe to assume that he is content with his dream studio, Quist already has plans for a 4-unit apartment complex to provide even more privacy for visiting clients.

His only complaint is that "we've been so busy, I can't get time in my own studio." But isn't that the type of problem most studio owners are looking for? R·E/P

Rideau is a free-lance engineer/producer, and is a regular **RE/P** contributor.

Studio update

Northeast

Howard Schwartz Recording (New York) has reopened Studio West, which now is said to be large enough to accommodate 25 musicians. The studio specializes in commercial, film/video scoring and record production. The control room centers on a 48-input **Solid State Logic SL6000E** console equipped with **Total Recall** automation, linked to a **Sony PCM-3324** digital 24-track machine. Other tape machine acquisitions include a **Sony JH-24** 24-track and **JH-110C** 4-track, and a **Studer A820** 2-track with

Howard Schwartz: reopens Studio West.

center-track time code. For time code lockup, an **Adams Smith 2600** synchronizer/editing system also is provided. Outboard equipment includes: **Dolby XP24** noise reduction units; **EMT 140S** and **250** reverbs; **Lexicon PCM-70, -60, -42** and **Prime Time** reverbs; **Pultec** and **Orban** equalizers; **UREI 1176** limiters; and **UREI model 813** and **Yamaha NS-10M** monitors. The facility also is capable of tape duplication, high-speed and real-time film dubbing, satellite uplink/downlink and has access to six sound effects libraries. *420 Lexington Ave., Room 1934, New York, NY 10170; 212-687-4180.*

Kajem Recording (Gladwyne, PA) has added an **Otari MkII MTR-90** 24-track to its recent 48-input **Solid State Logic SL4000E** console acquisition. For 48-track lockup, the studio has purchased an **Adams Smith 2600** time code synchronizer. Outboard purchases include **Massenberg** EQs, **Drawmer** tube compressors and noise gates, and a **Roland SBX-80** synchronizer. *1400 Mill Creek Road, Gladwyne, PA 19035; 215-877-2513.*

Manhattan Recording (New York) has purchased an **Otari MkII MTR-90** 24-track. Opening 18 months ago as an 8-track studio, MRC's original design was "definitely geared toward eventual expansion to 24-track format," recals president **Nick Gutfreund.** "We felt it would make more sense to start with a high-quality, narrow-gauge format, and expand as the business grew." According to chief engineer **Billy Straus,** the facility has recorded a variety of studio dates; "everything from 2-track digital jazz sessions to full-blown rock productions. With the MTR-90, we are looking forward to more bookings than before the [Otari] purchase." *135 W. 14 St., New York, NY 10011; 212-929-5960.*

Southeast

Island Studios (Richmond, VA) has expanded its recording format from 16- to 24-track by adding a **Sony JH-24 24-track** with autolocator, and an **Allen & Heath Brenell EX-8** 8-channel expander unit for its **AHB model 2416** console. Also reported is the area's first installation of an **Octave Plateau Voyetra Eight,** an 8-voice polyphonic, MIDI-capable synthesizer with built-in sequencer, arpeggiator and pressure-sensitive keyboard. Outboard equipment purchases include a **Roland SRV-2000** digital reverb, **Yamaha RX-11** drum machine, **Eventide Harmonizer, Nakamichi 480** cassette deck and microphones from **AKG, Sennheiser** and **Audio Technica.** Joining the facility's staff as chief engineer and manager is **Victor Bensoff.** An assortment of vintage instruments made by Fender, Ludwig, Sonar, Hohner and Korg also are available to clients. *9 W. Grace St., Richmond, VA 23220; 804-643-2022.*

South Central

Dallas Sound Lab (Irving, TX) has redesigned Studio B, a voice-over and radio commercial specialization room. Because of "client demand" and "to better accommodate radio production," a 24-channel **Sony JH-636** console has been installed. In addition, a new **JBL 4430** monitoring system, and **UREI** and **Valley People** compressor/limiters also were added. **Neumann, AKG** and **Electro-Voice** microphones round out the new equipment list. *6305 N. O'Conner Blvd., Irving, TX 75039-3510; 214-869-1122.*

Appletree Sound (Dallas) is a new, electronic music studio equipped with a **Studer A80** 24-track and an **Otari MTR-12** 2-track. Outboard equipment includes **Klark-Teknik** and **Lexicon** digital reverbs, and **dbx** complimiters. According to manager **John C. Cluts,** the studio "incorporates the latest computer-based recording as well as conventional tape technologies. The studio's SMPTE implementation gives the ability to sync computer-oriented information with tape." In-house MIDI-capable instruments include an **E-mu Systems Emulator II, Ensoniq Mirage** digital samplers, three **Yamaha TX-816 racks** and one **DX-7** synthesizer (for which the facility reportedly has a 2,400 voice sound library). *2150 Royal Lane, Dallas, TX 75229; 214-243-1626.*

Midwest

Brown Institute (Minneapolis) has purchased 16 **International Tapetronics/3M Omega** mono cartridge machines. Said to be one of the nation's oldest broadcasting schools, the facility is constructing a three-story building, which is scheduled for completion some time this summer. Students will be able to work in a TV studio, video-editing facilities, five 4-track studios and one 8-track studio. *3123 E. Lake St., Minneapolis, MN 55406; 612-721-2481.*

Studio A (Dearborn Heights, MI) has expanded its **New England Digital Synclavier** digital music system by adding a velocity- and pressure-sensitive keyboard, and MIDI/time code capabilities. Along with a **Roland JX8P, Sequential Prophet 2000, Yamaha TX-7** synthesizer module and **Apple Macintosh,** the

Dallas Sound: renovated Studio B.

Studio update

Synclavier is housed in the facility's newly opened keyboard overdub room, which houses a **Sony JH-24** 24-track and **Yamaha NS-10M** reference speakers. New outboard acquisitions for the studio's main control room include a **Roland SBX-80 Sync Box, Yamaha REV-7** digital reverb, **SPX-90** MIDI-capable effects processor and **dbx model 160X** complimiter. *5619 N. Beech Daly, Dearborn Heights, MI 48127; 313-561-7489.*

Woodland West Recording (Olathe, KS) is a new 24-track facility whose primary focus is album-oriented production. The facility's president, **Jack Black**, says, "We built this complex from the ground up, basing the acoustics on Jerry Milam's LEDE design." The 2-studio complex comprises Studio A, which houses a 15'x25'x30' studio and 20'x20' control room. Available to clients are a 24-input **Baur** console, **Sony JH-114** 24-track, **Tascam model 38** 8-track and two **Otari MX-5050B** 2-tracks. Outboard equipment includes two **Yamaha REV-7s**, a **Lexicon PCM-60** and **AKG BX5** reverbs; three **DeltaLab** DDLs, **Ashley** noise gates and compressors, and two **dbx model 160** compressors. The facility's monitoring system is constructed from a custom tri-amplified **JBL Heil** system, with 3-way **Audio Logic** crossovers; reference speakers are **JBL 4313s, Electro-Voice Sentry 100s** and **Auratone 5Cs**. Control room B is an 8-track preproduction studio where, according to Black, audio-for-video clients build basic tracks and then move to Studio A for overdub sessions. Housed in this room is a 28-input **AudioArts** console, and **Otari MX-5050B** 8-track and 2-track machines. As needed, outboard and monitoring equipment are brought in from Studio A to B. Interconnected to both control rooms is a MIDI control center. Although still under construction, the room will employ tie lines, and rack-mounted, direct-inject and MIDI Thru boxes. *20260 W. 114 Terrace, Olathe, KS 66061; 913-829-9378.*

Woodland West: new 24-track studio.

Southern California

Omni Recording (Los Angeles) is a new 16-track recording facility measuring 2,000 square feet. Featuring a 20-input **Hill Audio series J** console and 16-input **Panasonic Ramsa model 8816** desk, the facility has a **3M 56** 16-track with autolocator, **Otari MX-5050 MkIII** 2- and 8-track, and **Tas-**

Studio update

cam 80-8 8-track, the latter two machines equipped with dbx noise reduction. Outboard equipment includes a **Studio Technologies Echoplate II** stereo plate reverb, two **DeltaLab Effectron IIs, Symetrix** complimiter, **UREI 1178** stereo complimiters and **Roland RE501** chorus echo units. Monitoring is provided by **JBL model 4311**s and **Tannoy, Dalquist, Auratone** and **Yamaha** speakers. In-house instruments include a **Knabe** baby grand, **Roland Juno 60** synthesizer, **Fender Rhodes, Simmons SDS-5** electronic drums, **Oberheim OBX** and **DMX** drum machine. *5951 Venice Blvd., Los Angeles, CA 90034; 213-933-6664.*

Master Control (Burbank) has added a pair of **Lynx Timeline** synchronizer modules to provide 48-track capability for a recently acquired 52-input **Solid State Logic SL4000E** console. New outboard deliveries include **Lexicon PCM-70** and **-42** digital reverbs, a **Yamaha REV-7**, and an **AMS 15-80S** stereo digital delay with dual-channel sampling. *3401 Burbank Blvd., Burbank, CA 91505; 818-842-0800.*

Northern California

OTR Studios (Belmont) has added synthesizer programmer **Larry Monast** to its engineering staff. Using an **Apple Macintosh** with **Mark of the Unicorn** and **Digidesign Sound Designer** software packages, Monast specializes in MIDI programming, operating the facility's MIDI network for transfer of client's preprogrammed MIDI data into the studio's Macintosh. In addition, a **Sony PCM-701** digital processor, **Tascam 122** and **Nakamichi MR-2** cassette machines have been installed. *Belmont, CA; 415-595-8475.*

Different Fur (San Francisco) has taken delivery of a **Sony PCM-3324** digital 24-track, previously owned by **Windham Hill Recorders**. In addition, the facility has purchased a **Yamaha REV-7** digital reverb and **SPX-90** MIDI-capable digital effects processor. *3470 19th St., San Francisco, CA 94110; 415-364-1967.*

Mountain

Gravity Recording (Nogales, AZ) has purchased a 36-input **Neve V-series** console. According to owner **Miguel Crisantes**, he selected the V-series because he "wanted a company that would provide the best service and the newest technology." *Nogales, AZ; 602-281-1746.*

Telescene (Salt Lake City), a 16mm/35mm film and videotape production facility, has purchased an **Otari MTR-90** 16-track and **Audio Kinetics Eclipse** synchronizer/editor for audio sweetening. The facility recently remodeled its control room to include **UREI 811B** monitors, **Orban** complimiters and equalizers, and **Otari** cassette duplication equipment. *2185 South 3600 West, Salt Lake City, UT 84113; 801-973-3140.*

Studio update

Canada

Smooth Rock (Calgary, Alberta), said to be the first 24-track recording studio to be set up in the area, was designed six years ago by **Rudi Breur** and **Tom Hidley**. To celebrate its recent anniversary, the following equipment has been purchased: a transformerless **Sony JH-24** 24-track with **Autolocator III**; an **AMS RMX-16** digital reverb; **E-mu Systems Emulator II** digital synthesizer and **SP-12** drum machine; and a pair of **Yamaha NS-10M** close-field monitors. *No. 1-D, 624 Beaver Dam Rd. NE, Calgary, Alberta T2K 4W6; 403-275-6110.*

National Film Board of Canada has taken delivery of a 36-input **Neve 51 series** console, with six subgroups, six main outputs, 6-track monitoring capabilities, **NECAM 96** automation and VCA subgroupings. The console has film assignment (left-center-right and surround) and width controls on every channel. Other features include bar graph metering with peak hold, a user events controller (which allows mute buses and insert points to be programmed under NECAM) and a custom unit to allow remote control of all film dubbers and other peripheral machinery located in the facility's machine room.

Colombia

Instituo Macinal de Radio y Television (Bogota) has installed a 20-input **Westar** console supplied by the **Mitsubishi Pro Audio Group**. The console is equipped with two main stereo output buses, 4-band equalization on each input module, transformerless balanced input, standard VU meters and linear faders. Peripheral devices supplied with the desk include **Westrex EQ-444** 4-band EQ, two **CL-22** complimiters, two **NSD-120** noise gates and a **System 5** digital reverberation unit. The console interfaces with **Ampex MM-1200** 16-track and **ATR-800** 2-track machines.

Discos Fuetes (Colombia) has acquired a 28-input **Quad-Eight Westar** console for Studio A, to be installed by **Tom Franken** and **Jim Masure,** New York-based technical audio consultants.

England

Royal Northern College of Music has installed a **Studer 900 series** console based on series 902 circuitry, and fitted with A-series modules and reverse action faders.

Musician, producer and studio owner **Mike Oldield** has installed a **Harrison series 10** Virtual Console for music recording, video and film post-production.

The **British Broadcasting Corporation** (London) has installed a **Sony PCM-3324** digital 24-track into its custom-designed Digital Control Vehicle.

R·E/P

New products

Ramsa WR-8428 post-production console

This console features 28 inputs with switchable mic, line and tape source selection. Optional stereo input modules can be used to accommodate tape or cart machines, additional effects returns and other signal sources. Each input module is also equipped with a direct insertion jack for applying external processing devices to individual channels.

Each input module routes to a total of 10 mixing bus lines: four group, two master, two send and two echo. The

group and master buses can be monitored in stereo, allowing separate program and audition stereo monitoring modes to be set up.

The console's signal routing system terminates in an output configuration made up of four group outputs, left and right stereo masters, mono master and four matrix outputs. In this way, the unit can simultaneously provide a Dolby surround mix, a stereo mix and a mono mix.

Each channel module features a 3-band equalizer and two effect and send controls to provide pre- or post-fader and post-EQ signals for effects sends and foldback.

Circle (152) on Rapid Facts Card

Neve DTC-1 digital transfer console

Working with three mastering facilities—Sterling Sound and Masterdisk, both located in New York, and Disc Mastering, Nashville—the company has developed the DTC-1 digital transfer console for Compact Disc preparation of Sony PCM-1610/1630-encoded or AES/EBU standard master tapes.

Based on a 2-channel version built for Tape One, London, the new desk has three stereo channels (two sets of digital and one analog) to allow for real-time crossfading. Sampling frequency can be set to either 44.1kHz or 48kHz.

Also featured are a 4-band digital equalizer section, dynamic controls and a combination limiter, compressor, noise gate and expander, all of which function digitally.

An instant reset function, which comprises a "snapshot" of certain console settings, allows setups to be stored on floppy disk for later use on other DTC-1s. All electronics are housed in a separate rack for location in a separate machine room.

Availability of the first DTC-1 is expected by the fall.

Circle (161) on Rapid Facts Card

New products

DN305 noise masking processor from Klark-Teknik

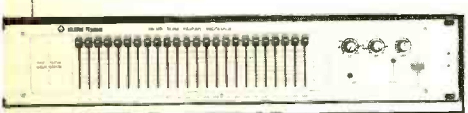
Incorporating a pseudo-random white- and pink-noise generator, shaping filters and a 24-band third-octave cut-only equalizer, the unit allows up to 15dB attenuation at third-octave centers between 63Hz and 12.5kHz.

Low and high filters provide 12dB per octave slopes with turnover frequencies variable between 30Hz to 250Hz, and 250Hz to 8kHz, respectively.

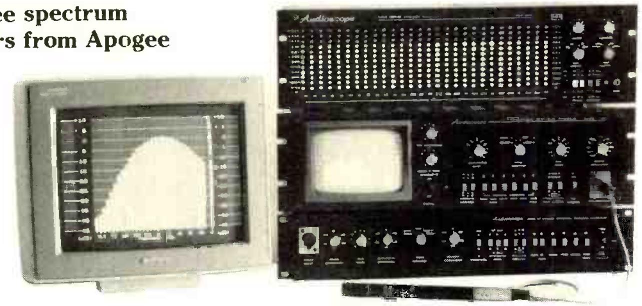
An equalizer section uses slide potentiometers in a graphic display layout and includes an LED overload indicator.

Typical specifications of the model DN305 include a minimum load impedance of 600Ω; nominal output level and EIN (for equalizer only) greater than −90dBm, 20Hz to 20kHz, unweighted.

Circle (164) on Rapid Facts Card



Three spectrum analzyers from Apogee



Unlike conventional third-octave, real-time spectrum analyzers, which typically use fewer than 16 LEDs in each frequency column, the Audioscope model 3113 uses an external color video monitor to display 0.25dB resolution over a 55dB window. Video resolution and range are said to be further enhanced by highly selective flat-topped and steep-sided 6-pole filters, which extract more valid information by minimizing unwanted inputs from neighboring frequencies.

The model 3013 is a third-octave, real-time spectrum analyzer with integral video monitor constructed from switch capacitor filters. This unit also can drive additional external large-screen monitors. Resolution is a quoted 0.25dB coupled with a 45dB display range.

The model 2813 covers the 32Hz to 16kHz range and uses a conventional LED display. Features include peak hold, color-coded LEDs, automated selection and digital interface to an external oscilloscope.

Circle (155) on Rapid Facts Card

New products

Stereo time compressor/expander from Lexicon

Incorporating a time code reader and reference output, the model 2400 is designed for both polyphonic music and voice information, and is said to employ newly developed digital signal-processing algorithms to provide mono compatibility and stereo imaging. While maintaining original audio pitch, it is designed to change the running time of video, film or audio program material.

All encoding is 16-bit linear PCM. The unit interfaces with time code synchronizers or directly to tape machines with time code-follow capabilities via a pair of XLR inputs and outputs. By generating variable-rate time code, the model 2400 is able to control play time over either master or slave transports. Two RS-422 serial ports, dc and tach outputs, and a tach input are provided for interfacing to other transports.

Machine interface is offered in two modes, master or slave. In the master mode, the model 2400 can accelerate or decelerate a transport according to original and modified running times. In the latter mode, the unit measures current transport speed, corrects pitch shift and displays the speed factor on the front.

Pro user price for the model 2400 is $19,000, without remote control.

Circle (156) on Rapid Facts Card

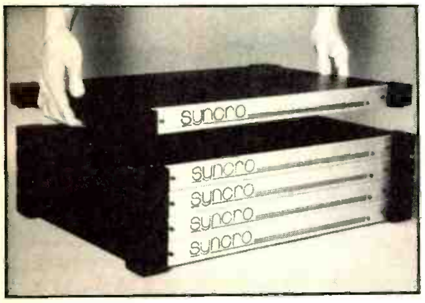
Syncro time code synchronizer from Soundmaster

Used in conjunction with the Soundmaster audio editing system program, the Syncro synchronizer/editor comprises an interface, controller box and integrated synchronizer software. The system is designed to communicate with an IBM PC controller via 48-line parallel bus, and contains internal RAM for storage of more than 400 events.

Features include variable-speed synchronization from ⅓ to 3X play speed; the capability to synchronize EBU with SMPTE time code (24 fps, 25 fps, 29.97 fps and 30 fps, drop or non-drop frame); designation of master machine via PC keyboard; and modular construction for daisy chaining up to 16 units (one Syncro per transport).

Soundmaster software enables a controller utility to support more than four transports, with simultaneous status display on an IBM PC. An edit list utility enables 2,550-event RAM storage in 10 discrete lists, a chain function for linking

New products



event lists, storage of lists on 5.25-inch floppy disks, list management functions and real-time automated auto assembly.

Suggested pro user price of the Syncro is $2,500 per box with interface; Soundmaster software sells for $5,000.

Circle (165) on Rapid Facts Card

Studiomaster series of mixing consoles

Featuring MIDI-controlled automation, the series II is available in three mainframe configurations: 16x16x2, 16x8x2 and 16x4x2. An 8-input channel expander and an 8-channel tape monitor expander are options.

Each input channel features switchable phantom power, pad, phase reversal, line and tape remix selection, continuously variable gain with detent, 4-band equalizer, EQ bypass switch, six auxiliary sends and returns, pan, channel-on switch and channel-solo switch.

Subgroups offer solo, sends to auxiliary 5 and 6, pan and monitor, and a 12-segment LED bar graph.

Under external MIDI control, the mixer's mute of input channels and auxiliary returns can be used during remix and mixdowns. Initially, MIDI-control software will be available for a Commodore C64 and IBM PC, with other versions available according to user requirements.

Pro user prices range from $500 for a 16x4x2 mainframe console to $6,750 for a 16x16x2 version.

Circle (154) on Rapid Facts Card

Bose Modeler software for sound system design

This software for the Apple Macintosh enables prediction and analysis of sound system performance in arbitrary spaces. Developed in conjunction with David L. Adams Associates, acoustical consultants, Modeler allows users to predict the time of arrival of sound at a listener's position and to create sound maps in which gray shades (rather than numbers) correspond to sound pressure levels. The program allows users to view a loudspeaker's characteristics as a 3-dimensional object.

Space is represented geometrically and acoustically on the Mac's screen as a series of adjacent planes and accompanying surface materials. The geometrical representation of a space can be rotated to show plan, elevation or a variety of oblique views. A zoom function allows the space to be enlarged or reduced to focus on a particular area. Sound sources are represented in the program by their sensitivity and dispersion characteristics.

The program features an open database, which allows it to be used with any manufacturer's products, including limited production and unique devices.

Circle (166) on Rapid Facts Card

Shure FP42 stereo portable mixer

Designed for location and remote broadcast applications, the stereo mixer features four inputs switchable for mic- or line-level, level control, center-detented stereo pan pot and cuing. Also featured is a concentric stereo master level control.

Preliminary specifications include a quoted 30Hz to 20kHz, ±2dB frequency response, EIN at less than −129dBV, and distortion at less than 0.4% THD. A pair of VU meters monitor stereo input, while a peak indicator LED illuminates at 7dB below 5% THD clipping point.

A built-in limiter enables a threshold level from +8dBm to +14dBm (adjustable), with a typical attack time of 3ms.

The unit may be operated by battery or ac power, and is priced at $720.

Circle (159) on Rapid Facts Card

New products

500 series loudspeaker system from Meyer

The new integrated unit incorporates two, full-range 500 series loudspeaker monitors, model 500 stereo amplifier, and optional model 501 subwoofers.

Features include polarized, locking connectors; internally mounted rigging blocks; removable fiberglass mesh grilles; recessed hand grips; and a flat-black textured finish.

The integral 1,200W amplifier includes a signal-processing circuit card that contains frequency response and phase equalization, protection circuitry (featuring RMS limiters and front-panel LED fault indicators), and adjustable level-matching controls to accommodate +4dBm and 10dBv systems.

The system's frequency response is quoted at 40Hz to 16kHz, ±3dB (30Hz to 16kHz, ±3dB with optional model 501 subwoofer). Maximum SPL is 130dB.

List price of the 500 series system, including two speakers and amplifier, is $4,800; the model 501 subwoofer sells for an additional $950.

Circle (172) on Rapid Facts Card

CV series loudspeakers from Community

Designed as four compact loudspeaker systems for fixed installations, the enclosures are constructed from high-density particle board with internal bracing network and formica veneer.

The models CSV25, CSV35 and CSV38M incorporate ferro-fluid cooled bass drivers, removable grilles and T-nut mounting capabilities.

To complement the monitors, the company has introduced four low-frequency bass reflex cabinets, which are said to match the CSV series in size, appearance and performance. These compact direct radiators are driven by 15- and 18-inch ferro-fluid cooled loudspeakers and reportedly are suited for 2- and 3-way systems requiring "solid low-frequency reinforcement."

Circle (157) on Rapid Facts card

New products

Titus stereo correction unit

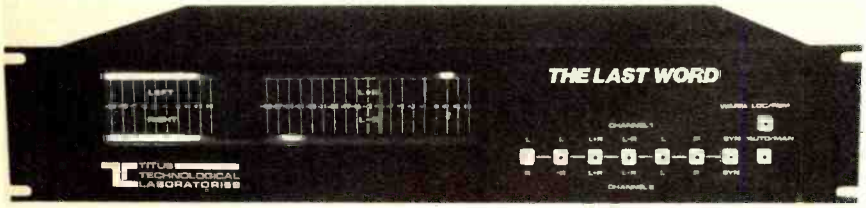
Described as the first microprocessor-based stereo audio correction device designed to correct almost any problem that a recording or broadcast facility could encounter, the Last Word monitors any stereo source and automatically or manually corrects audio problems.

Until now, the company says, an engineer had to watch oscilloscopes and listen carefully to the final product to determine if the outgoing audio was phase and polarity correct, and if both channels were present. The unit automatically corrects for loss of a channel, and for phase and polarity problems on its inputs.

Under manual control, the unit can alter the audio according to preset conditions, including reproducing stereo source, producing an out-of-phase stereo source, producing the sum of inputs, producing the difference between inputs on the outputs and bringing up the stereo synthesizer.

The unit can be used pre- or post-audio processing for detecting and correcting channel loss or polarity inversion, as well as audio time base problems, or to bring up an alternate stereo audio source with a loss of channel, or upon command.

Circle (160) on Rapid Facts Card



BC Engineering AC-400 automated gain controller

Designed for level control and crossfading during post-production, the AC-400's operational features include four pairs of XLR-type inputs and outputs (active balanced at 50kΩ); a continuous gain reduction of zero to −30dB, then a step to −70dB; and a rate of gain change between 0.25 seconds to more than 20 seconds.

Each of the four channels has two controls, one to set rate of gain change and another to set level of gain reduction. Gain reduction transition can be triggered (from pulse or relay enclosure) via a front-panel push-button or by one of two remote lines. The four channels may be used separately or coupled for stereo crossfades.

Technical specifications include a quoted frequency range of 20Hz to 20kHz; S/N ratio greater than 110dB, unweighted; and THD greater than 0.4%, unweighted.

Circle (163) on Rapid Facts Card

New products

PF series of pianos from Yamaha

These two electric pianos are MIDI keyboards featuring the same FM tone generation and internal sound systems used in the DX series synthesizers.

Both the 88-key PF80 and 77-key PF70 allow MIDI interfacing to sequencers, other synthesizers and tone generators. Enhanced MIDI control section permits both keyboards to be used as MIDI controllers, enabling MIDI split, merge, pitch bend and program change functions.

Both instruments have 10 preset voices, including acoustic and electric piano with vibes and clavinova. Control is provided for stereo tremelo and chorus, soft-pedal intensity, equalization and tuning. Also included is an internal stereo amplifier and two speakers.

Suggested retail price for the PF70 is $1,395; the PF80 is priced at $1,595.

Circle (171) on Rapid Facts Card

Garfield introduces time code card for Linn 9000

The interface card plugs into one slot inside the Linn 9000, making the unit time code-compatible when used with a Garfield Master Beat unit.

In addition to the 9000's MIDI In, Out and Thru jacks, the new card provides a second MIDI input, which reads MIDI clocks, allowing the user to merge separate MIDI clock and note sources. It also has three MIDI Thru jacks, one of which serves as a rechannelizer.

Furthermore, the Garfield/9000 offers time code-controlled erase and control over the 9000 in play or record modes. It also includes the sync-mod necessary for synchronization with tape or other clock sources.

Suggested retail price for the Garfield/9000 card, including installation, is $449.

Circle (169) on Rapid Facts Card

New products

Cetec Vega PRO1 wireless mic system

The system is offered in two versions: PRO 1-B bodypack or PRO 1-H hand-held units.

The bodypack unit consists of a model T-37 transmitter and model R-31A receiver. The latter features two LED bar graph displays for RF signal and audio levels. The power transformer supports both 115Vac and 230Vac operation, and external dc. Model T-37 accepts virtually any electret lavalier microphone via a miniature 4-pin connector.

The model T-36 hand-held transmitter uses an Electro-Voice BK-1 condenser element and black windscreen.

Both systems can operate on any crystal-controlled frequency between 150MHz to 216MHz.

Circle (167) on Rapid Facts Card

Blank Software announces update of waveform editing program

The updated V1.1 of Sound Lab will now accommodate the Apple Macintosh Plus and the rack-mount Ensoniq Mirage Digital Multi-Sampler. New waveform editing/processing features have been added, including a high-speed sampling rate conversion algorithm for "improved frequency response and increased looping flexibility," a 3-D time-domain waveform analysis display; and fine-resolution editing within memory-page boundaries.

Other new features include enhanced librarian functions, remote MIDI keyboard operation and improved wave data transfers.

Sound Lab update is available free of charge to registered owners of Sound Lab 1.0.

Circle (162) on Rapid Facts Card

Time code and MIDI interface from Southworth Music Systems

The JamBox/4 reads four independent MIDI inputs, provides a time code reader/generator function, communicates with either Apple Macintosh or Atari 1040 PCs, and sends selected data to any of the four MIDI outputs.

Supported by a special version of the company's Total Music sequencer software, synchronization is provided at 24, 48, 96, 192 and 384 clocks per beat. DIN sync is provided at 24 clocks per beat, with separate start/stop line, plus audio click sync in/out at one click per beat.

The time code reader provides dropout detection circuitry to jam sync in real time, or to provide off-line jam sync capabilities for time code recovery. A time code generator offers slow rate control. Also, LEDs are provided for each MIDI input and output, and tempo.

The company says that the unit will operate with previous releases of Macintosh MIDI software programs—to provide MIDI merging and output routing—but that it requires software modifications to support four times MIDI speed mode and direct time code synchronization.

Suggested list price for the JamBox/4 is $389.

Circle (151) on Rapid Facts Card

Ibanez SDR-100 stereo digital reverb

The unit features 16-bit independent or simultaneous linear digital processing for one or two channels. Effects modes include hall, room, gate, reverse, dual delay, auto panning and dual reverb. In all, 30 factory presets and 70 user programmable presets are available.

Programmable parameters include reverb time, pre delay, early reflection time and level, room size, gate time, feedback controls and stereo 4-band equalizer. Patches can be recalled via MIDI commands, front panel controls or the optional model IFC60 foot controller.

Circle (158) on Rapid Facts Card R·E/P

News

Continued from page 4

AMS AudioFile deliveries scheduled this summer

After two years of development, U.S. deliveries of the AudioFile random-access digital recording and editing system have begun.

The system now features a redesigned control surface with 13 software-defined function keys, two thumbwheel controls, nudge buttons, numeric and alphabetic keypads, and a set of machine transport controls.

The company reports that more than 20 AudioFile systems are currently in use at broadcast and post-production facilities around the world.

Phil Ramone premieres Dolby SR process

Working at The Power Station studio, New York, producer Phil Ramone used some of the first production Dolby SR modules in late April to prepare stereo music tracks for the film *Ruthless People*, from 32-track digital. The Dolby Stereo soundtrack features music specially written for the movie, including songs performed by Billy Joel and Mick Jagger.

Dolby Spectral Recording is said to share some important features with digital recording; resembles conventional analog recording in other ways; and adds important advantages of its own. In film post-production, for example, SR allows conventional film editing and mixing equipment to be used. Unlike digital recording, SR is said to provide a high degree of security against the accidental clipping of high-level transients, cutting down on the need for retakes.

AMEK APC-1000 price correction

The price of an AMEK APC-1000, which appeared in the June "New Products," was quoted in sterling rather than dollars. A 48-input APC-1000 with dynamics control has a pro user price of $206,100; a 64-channel version with GML Moving Fader Automation and dynamics control is $349,650.

Marshall Electronic address correction

Due to a typographical error on page 24 of the June issue, we inadvertently stated that Marshall Electronic was based in Culver City, CA, and not Baltimore, MD.

The confusion arose because of the similarity in name with Marshall Electronics, manufacturers of Mogami cables and accessories, which is located in Culver City.

AMS buys Calrec

At press time, **RE/P** learned that Advanced Music Systems had purchased Calrec Audio. AMS will handle European distribution of Calrec consoles and microphones. In the United States, Audio + Design/Calrec, Bremerton, WA, will continue distribution until the end of the year, when a new network will be established. Further details will appear in the October issue.

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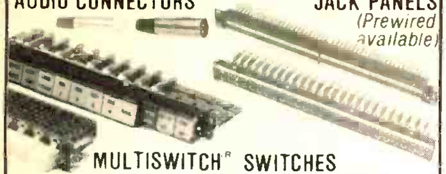

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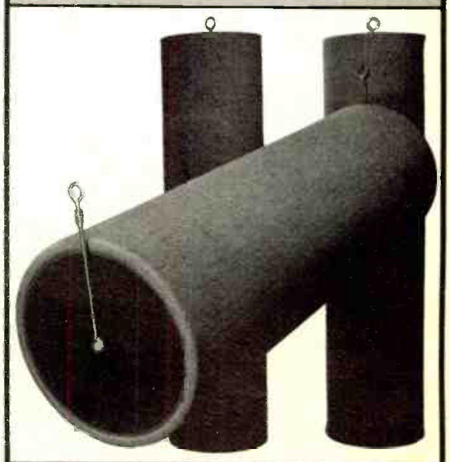

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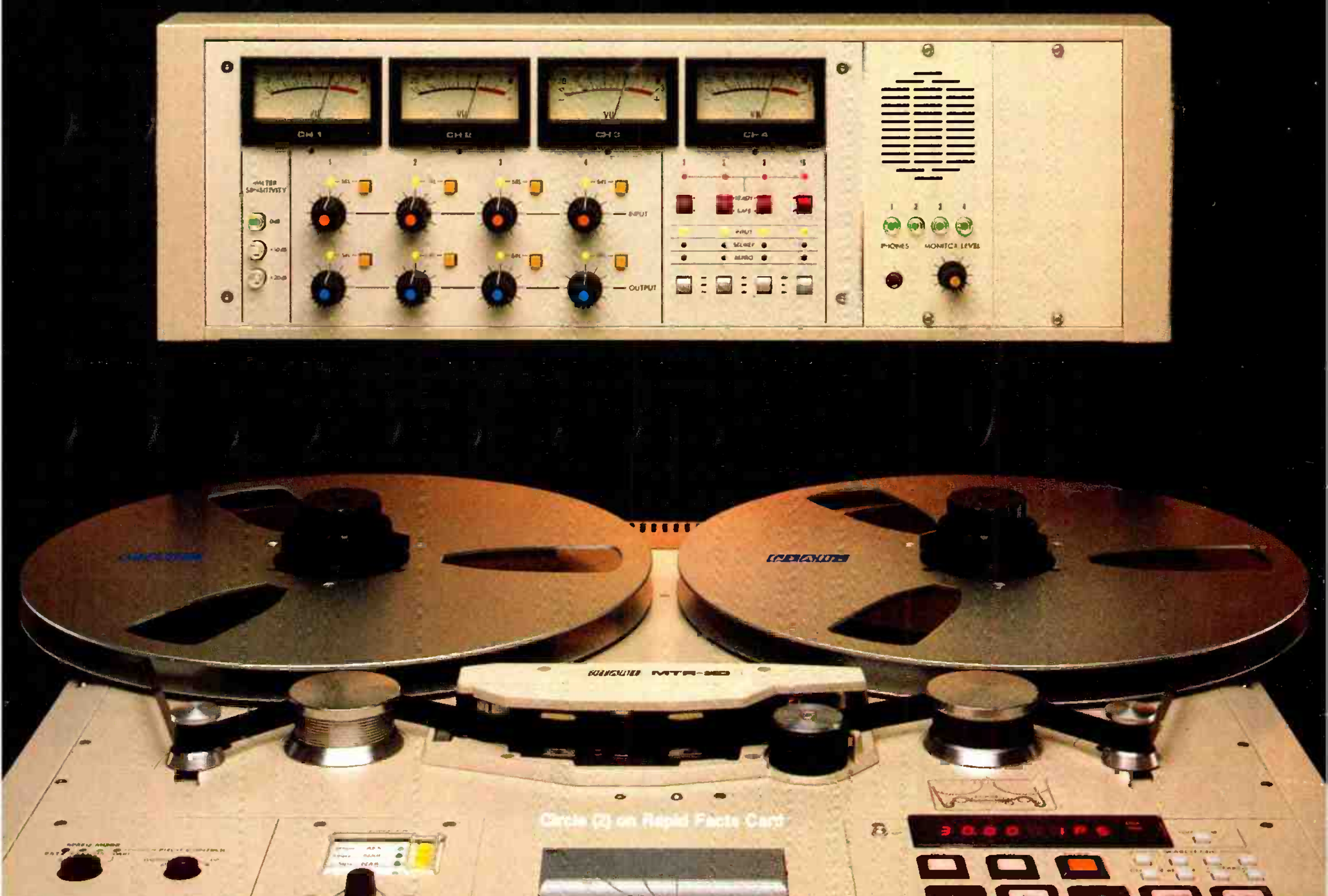

CH 1
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CH 4
INPUT
OUTPUT
PHONES
MONITOR LEVEL
MTR-20
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Studer 961/962: Small Wonder

It's a wonder how a console so small can do so much ... and sound so good!

The Swiss have a special talent for making great things small. A case in point: the new 961/962 Series mixers from Studer. In video editing suites, EFP vans, remote recording, and radio production, these compact Studers are setting higher standards for quality audio.

Sonic performance is impeccable throughout, with noise and distortion figures well under what you'd need for state-of-the-art digital recording. By refining and miniaturizing circuits developed for our 900 Series production consoles, Studer engineers have squeezed world-class performance into suitcase size.

The 961/962 Series is fully modular, so you can mix-and-match modules to meet your requirements. The 961/962 features stereo line level input modules with or without 3-band EQ, plus mono mic/line inputs and master module with compressor/limiter. Other choices include a variety of monitor, talkback, auxiliary, and communication functions. The 961 frame holds up to 14 modules, the 962 accepts up to 20.

Other new features in the 961/962 Series include improved extruded guide faders, balanced insert points, FET switching, electronic muting, Littlite® socket, and multi-frequency oscillator.

Thanks to its light weight, DC converter option, and sturdy transport cover, you can put a 961/962 mixer on the job anywhere. And, with Studer ruggedness and reliability, you can be sure the job will get done when you get there.

Packed with performance and features, 961/962 consoles will surely make a big splash in audio production circles. Small wonder. Call your nearest Studer representative for more details.

With snap-on cover, mixer is road-ready in seconds.

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