RECORDING ENGINEER / PRODUCER

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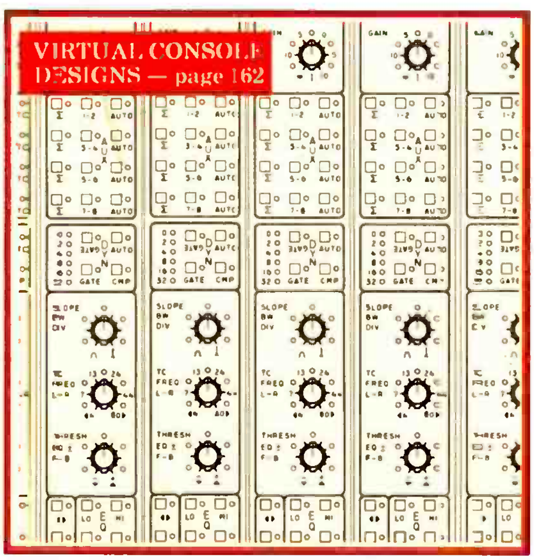
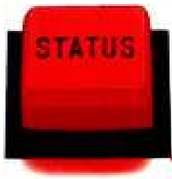
October 1985
Volume 16 — Number 5



PRODUCING AUDIO FOR • TAPE • RECORDS • FILM • LIVE PERFORMANCE • VIDEO & BROADCAST



The TS24 is the first in-line console from Soundcraft.

And it represents a major breakthrough in in-line technology, because it now makes the console far easier to understand and operate.

Believe us, this is no hollow promise. Our argument is built around two rock solid foundations. Firstly, a new concept in console layout so logical, engineers used to split or in-line consoles can start work from day one. And secondly, a set of master conditions so advanced they'll amaze you.

STATUS.

One touch of the status button will configure the whole console for each particular stage of recording, mixing, broadcasting and video post production without sacrificing any flexibility whatsoever. In other words, one touch and you're off and running.

NEW DESIGN.

Conventional in-line consoles suffer from the limitations of one long travel fader and one equaliser being shared by two signal paths. With the engineer fader reversing and moving the equaliser back and forth throughout the recording, overdubbing and mixing process to optimise the situation.

The TS24 eliminates these shortcomings, thanks to its logical design. The long travel fader is in the section called MIX, which is the signal path for both monitoring and mixing. The equaliser moves between the MIX and CHANNEL signal paths automatically by use of the master status switches. 'Soft' switches may locally move EQ and AUX sends between the two signal paths but are also automatically reset.

When mixing, the Channel sections become available as additional inputs or effects sends without the limitations imposed by more conventional designs.

DROP-IN. BOUNCE.

Drop-ins are made easy by the use of the TAPE and GROUP button (T & G). Tape and Group enables you and the musician to monitor the original track and the overdub simultaneously.

The Bounce button facility enables you to take any combination of channels with their fader and pan settings directly to the routing matrix giving you instant bounce down.

SOUND AND VISION.

To create perfect sound, you also need perfect vision. With the TS24, that's exactly what you get. Separate scribble strips are provided instead of the usual confusing double one, and the Mix and Channel controls are in clearly defined areas for easier use.

AUTOMATION.

Soundcraft have developed a unique interface to the disc based MASTER MIX automation system, which enhances its operational flexibility by totally integrating the full extent of the console muting.

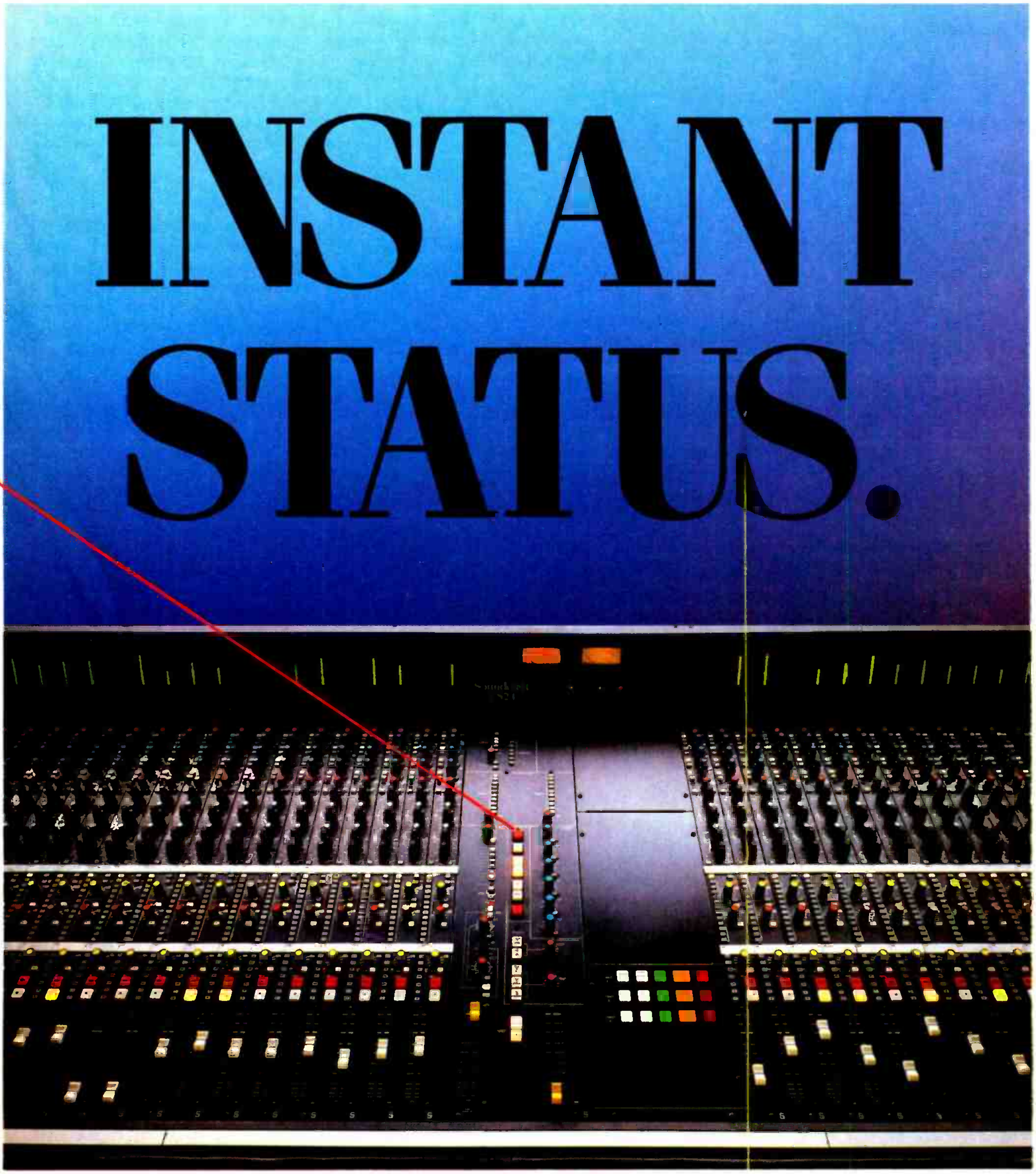
One feature of this system enables you to by-pass the Channel VCAs, thereby optimising the original recording quality.

Surprisingly enough, all this practical technology, combined with sleek good looks doesn't carry a huge price tag. So our doors are open to practically everybody.

Which only leaves us with one thing to say: if you want to keep your finger on the button in the most up-to-date mixing console design available, contact us.

Soundcraft TS24

Soundcraft Electronics Ltd., 5-8 Great Sutton St. London EC1V 0BX. Tel: 01-253 6988. Telex 21198 SCRAFT G.
Soundcraft Electronics USA, 1517 20th. St. Santa Monica, California 90404. Tel: (213) 453 4591. Telex: 664923.
Soundcraft Canada Inc. 1444 Hymus Blvd., Dorval, Quebec, Canada H9P 1J6. Tel: (514) 685 1610. Telex: 05 822582.



RECORDING ENGINEER/PRODUCER

— the magazine to exclusively serve the **RECORDING STUDIO and CONCERT SOUND** industries . . . those whose work involves the **engineering** and **production** of commercially marketable product for:

- **— Records and Tape**
- **— Film**
- **— Live Performance**
- **— Video and Broadcast**

— the magazine produced to relate recording **ART** . . . to recording **SCIENCE** . . . to recording **EQUIPMENT.**

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— October 1985 Contents —

The art of engineering is serious business.

SCORPION 26/16/2
w/16 monitor

You made the decision to engineer audio because you care for the art of music and sound. The realities are that you also need to operate as a business. The audio console that you choose for your creative fulfillment is the most expensive piece of capital equipment in your facility. Serious business.

We also consider the art of engineering a serious business. We have devoted 12 years to engineering and building truly the finest audio consoles. Year after year, we have reinvested our profit back into our design, manufacturing and distribution facilities. The result is that TAC and AMEK together are among the world's largest professional audio console companies. Serious business.

The SCORPION is one of the highlights of the TAC line. With 9 different modules, 2 mainframe configurations, 5 meter packages and 8 or 16 buses, the SCORPION can be configured to suit any professional application. Of course, each SCORPION comes standard with the EQ sound and chassis design that has made TAC/AMEK world renowned. Affordable value is serious business.

We ask only that you look deeper than just an ad, a brochure, or a sales presentation before you make your next major capital investment. At TAC, we treat the art of engineering as a serious business.

TAC

AMEK CONSOLES INC., 10815 Burbank Blvd., North Hollywood, CA 91601 Tel 818-508-9788 Telex 662526 AMEK USA
TOTAL AUDIO CONCEPTS LTD., Islington Mill, James Street, Salford M3 5HW England Tel 061-834-6747 Telex: 668127 AMEK G

For additional information circle #102

See the first console specifically built for 64 track digital recording at the New York AES

Designed for the world's largest and most sophisticated recording studios, the SUPERSTAR is a 20-bit analog console with the performance, specifications, and functions necessary for digital recording. The SUPERSTAR is totally modular and totally expandable, and features 64 mixing busses for recording to two 32-track tape recorders.

DESIGNED FOR DIGITAL

Through critical analysis of design, and testing and re-testing of components, the signal path and sound quality of this console is optimized for digital recording. Quad Eight, as a part of the Mitsubishi Pro Audio Group, developed this console as the perfect companion to digital multitracks such as Mitsubishi's new X-850 32-channel recorder.

64 MIXING BUSSES

The SUPERSTAR has 64 mixing busses controlled from a central assign panel and readout. The 72 by 64 output matrix uses logic-controlled summing bus switching, providing 64 instantly selectable output busses. Using its own memory for five complete presets, it also allows automation control via a serial communication port.

COMPUMIX IV AUTOMATION

A 32-bit master processing computer records data on an 80 megabyte Winchester hard disk in real time for unprecedented accuracy in an automation system. This fourth-generation design stores four instantly accessible real time mixes plus eight compressed mixes on the hard disk simultaneously, and transfers compressed mixes to and from floppy disk. A distributed multiprocessing system, Compumix IV has individual computers handling dedicated functions at different levels of the system architecture.

INTELLIGENT DIGITAL FADER

With its own microprocessor, the IDF can operate standing alone or coupled to the automation system. Using a monolithic direct digital 8-bit encoder/fader and a membrane touch panel inputing the 10-bit internal processor, exact dB values are calculated using 14-bit arithmetic, displayed, and converted to DC using a 12-bit D/A. All functions are at 10 times scanning rate for 1/10 frame mute accuracy, and fader smoothing algorithm. There are 16 nested groups, and any module can be assigned master without changing its individual function.

The VCA circuitry is on a separate PC card that plugs onto the main module PC board. Different VCAs may be easily substituted.

PLUG-IN EQUALIZER

Finally, there's a choice! The SUPERSTAR equalizer plugs in on each input module. Normally delivered with a four-band parametric equalizer with variable frequency, bandwidth, and peak/dip level; others are available. Each module also has a variable concentric high pass, low pass filter with individual in/out buttons.

AUTOMATED EQUALIZER

Each channel module has been designed to accept an automated equalizer, making the SUPERSTAR the most advanced console available.

PLUG-IN PREAMPLIFIER

Each module's microphone preamplifier is also of top panel plug-in design. Transformers—or transformerless differential, the choice is yours. And new technology can be instantly added to your console.

Mitsubishi X-850
32-Channel Digital Audio Recorders

AES Booths
717-724

SMPTE, Los Angeles
Booth 1320

FOR WORLD-CLASS STUDIOS

The SuperStar

by quad eight

MODULE FEATURES

Each module is a dual in-line design with separate channels for recording and monitor/mixdown. Main fader (or VCA), equalizer, filter, auxiliary sends, and line trim can be switched to either channel. Each input module has eight auxiliary sends configured as four monaural and two stereo sends, with panning. They are switchable as pairs to either recording channel or monitor. Monitor/mixdown channel is selectable to two stereo outputs for simultaneously making two different mixes. All output busses are differential balanced with optional transformers. For added overall control, each module has a switch (AGM) which allows it to become an audio sub-master for a group of input modules. A signal presence/peak dual LED circuit on each module indicates peak overload at microphone preamplifier out, or equalizer out, or fader out. Unique circuitry allows all to be connected to the indicator with only the peak signal shown, without addition from the other samples.

BAR GRAPH METER

Above each module is a 60-segment LED vertical bar level meter. The metering system is switchable to VU or peak ballistics with changeable electroluminescent scales for each, VCA level indication, or two sets of spectrum analyzers in ⅓ octave increments.

TOTALLY MODULAR FRAME

The SUPERSTAR console is constructed of individual housing sections of eight modules each. The console is not limited to just a few standard frame sizes, but may be ordered with any number of inputs. Interwiring of console sections and input/output connections is all with shielded plug-in ribbon cable. High quality bantam jacks are on PC boards, arranged module by module, and plug into the mother boards by shielded ribbon cable. This feature, along with the modular frame, makes this the only truly field-expandable console.

OPTIONAL OVERBRIDGE

An overbridge is available for mounting above the primary meter bridge to house additional accessories.

LIMITER/COMPRESSOR/GATE

This is a plug-in option for the meter overbridge. It is wired directly in-line with each channel, or as a peripheral patchable processor. More than just an accessory to the module, it is a full-function studio-quality leveling amplifier.

AFFORDABLE DIGITAL

The SUPERSTAR costs *less* than other world-class consoles. And a digital package with a Mitsubishi multitrack can save you even more.

NEVE. SSL. SUPERSTAR.
See them all before you decide.

MITSUBISHI PRO AUDIO GROUP

DIGITAL ENTERTAINMENT CORPORATION

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United Kingdom: 1 Fairway Drive, Greenford, MIDDX UB6 8PW • Phone (01) 578-0957 • Telex 923003

For additional information circle #103

News Letters Views

SONY APR-5002 REVIEW

from: **Takeshi Yazawa and**
Hiro Konno
Professional Audio Division.
Sony Corporation of America

As indicated in the review article by Peter Butt [published in the August issue], there are some discrepancies between our published performance data and his data as measured.

As Mr. Butt's measurement method could not have possibly been the same as the one we did, we felt it was appropriate to describe the measurement method that we use in all of our tape-recorder products. It is broken down into two categories, of which detailed descriptions are as follows.

A: Signal to Noise Measurements

Using an AC voltmeter w/dB scaling for appropriate sensitivity and range, the machine is measured for the above. We use a reference fluxivity of 510 nWb/m, with a shorted Audio input, and bulk-erased audio tape for Signal-to-Noise measurements of Record Input to Playback Output. This produces a measurement of both the Record and Playback Signal processing systems. The resultant of this measurement can be expressed in weighted, unweighted, and dB(A) scales.

B: Frequency Response Measurements

Assuming that the machine has been properly aligned for flat-frequency versus amplitude response using an approved reproducer alignment tape, and has been calibrated for proper bias, Record level and equalization for the tape being used, the machine can be measured for overall frequency response — Record/Play. A precision sinewave oscillator is connected to the line input or calibration input connector and an AC voltmeter w/dB scaling is connected to the line or calibration output connector. At constant amplitude output, the oscillator is swept from approximately 10 Hz to the upper frequency limit of the device. These upper and lower frequency limits are defined as the 3-dB down points in measured outputs. Throughout this sweeping upward, the operator denotes the amplitude response variations (if any). Upon completion, the results are mapped out on a frequency versus amplitude scale and the appropriate standard deviation is noted ±. This can then be verified against a reference specification. The resultant is called the frequency response. □□□

Peter Butt replies:

Sony/MCI Product Management is correct on points A and B.

A. The (S+N)/N figures given in the tabulation at the head of the article were measured using biased tape that had not been bulk erased, and then passed over the heads in Record mode at the speed of interest, inputs shorted. It was then replayed by the reproduce head at the speed of interest. The reproducer had previously been calibrated and equalized for a reference fluxivity of 260 nW/m at 1 kHz for each of the relevant speeds. Because 20×LOG (510/260 = +8.81 dB, this figure was added to the +4 dBv meter calibration to yield a reference correction factor of +12.81 dB. The indicating instrument is a Hewlett-Packard 334A Distortion analyzer, Option H05. The average-responding 334A meter 30-kHz LPF noise reading was corrected to RMS by adding the standard 1.11 dB. The 30-kHz meter bandwidth was corrected by subtracting 20×LOG (30/20) = 3.52 dB from the reading.

The reported (Signal + Noise)/Noise data reported was derived as follows:
S = (8.81 - Em + 1.11 - 3.52) Decibels: 20-kHz LPF unweighted.

B. Frequency response tolerance data was determined by first aligning, biasing, and equalizing the machine for the tape used at each speed of interest for the reproducer characteristics required. Flux reference levels were set to 260 nW/m at 1 kHz for each case. All equalizations were set using CW [continuous wave] sinewave signals at: 1 kHz, 10 kHz, and 50 Hz. Deviations were checked with a sine sweep from less than 20 Hz to a frequency greater than the high-frequency 3 dB break at the high-end of each track/speed response. The 7.5 ips response data was equalized similarly at a signal level 10 dB below the flux reference.

These preparations having been completed, the record/reproduce/sync frequency response of each was taken by recording a squarewave having a frequency of 1.963125 Hz for the frequency band below 200 Hz, and 195.3125 Hz for the band above 200 Hz. These signals were then reproduced in the appropriate modes and digitized in the time domain. The time domain data was then operated upon to produce the Discrete Fourier Transform, and then deconvoluted by the digitized, DFTd time-domain data of the generator squarewave output waveform. This operation yields the transfer function of the transmission device responsible for the observed changes in the time-domain output response, as compared with the input-signal time-domain response.

The resulting data is then plotted in such a way as to show the two bands of

... *continued on page 12* —

Introducing the Tape

Multi-track Overdubbing with Synclavier's New Direct-to-Disk™ Recording System

During the AES Convention from October 13-16 in New York, New England Digital will debut the latest Synclavier enhancement, the Direct-to-Disk multi-track recording system.

Used in conjunction with the state-of-the-art Synclavier, the combined system will offer a stand-alone "tapeless" digital recording and post-production environment. In other words, the capability of a $1,000,000 studio at a fraction of the cost, *and it's all digital!*

The system to be premiered will feature four tracks of recording to disk at 50kHz, with storage capacity up to 500 megabytes. The production model will support up to sixteen tracks and offer the capability of 100kHz sampling or stereo sampling at 50kHz.

The Direct-to-Disk system will be controlled from the Synclavier's 32-track digital recorder. Complete vocal and instrumental tracks, recorded continuously, can now be part of any Synclavier recording. For example, a user could incorporate a recording of an acoustic instrument or vocal track(s) into their existing Synclavier polyphonic sampled, synthesized, or MIDI tracks.

Be sure not to miss this revolutionary advancement!

The Midi System that Works!

Along with all the amazing capabilities of the Synclavier system, you can add the flexibility of MIDI. Now it is possible to incorporate the dynamics and timbres of your favorite keyboard with the Synclavier memory recorder. For example, record your keyboard into the Synclavier's 32-track digital memory recorder or trigger the timbres of the Synclavier from one of your favorite performance instruments.

The standard Synclavier MIDI system features one channel in/four out. The standard system can be expanded to 8 inputs/32 separate MIDI outputs. Plus, the new Synclavier software gives the ability to slide tracks forward or backward in time, eliminating the MIDI related delays which have plagued most MIDI systems.

New England Digital Authorized Distributors **Atlanta** Songbird Studios 404/35 5955 **Boston** Syntone Inc. 617/267-4137 **Dallas** Lightning Music & Sound Inc. 214/387-1198 **Nashville** S

ess Recording Studio

Other Fantastic Synclavier Features

Besides these great new additions to the Synclavier system, the system also offers:

Polyphonic Sampling
(16-Bit/50-100 kHz) Up to 32 voices and 32 megabytes of R.A.M.

Multi-Channel Outputs
SMPTE
Velocity/Pressure Keyboard
Automated Music Printing
Guitar Interface

Instructional Video Cassettes

If you're interested in relaxing at home and learning the basics of the Synclavier system, you can now purchase three video cassettes which guide you through its basic features and operations. Send your check for $175 per set (not sold separately) plus postage and handling. Complete printed documentation is also available for $200 per set.

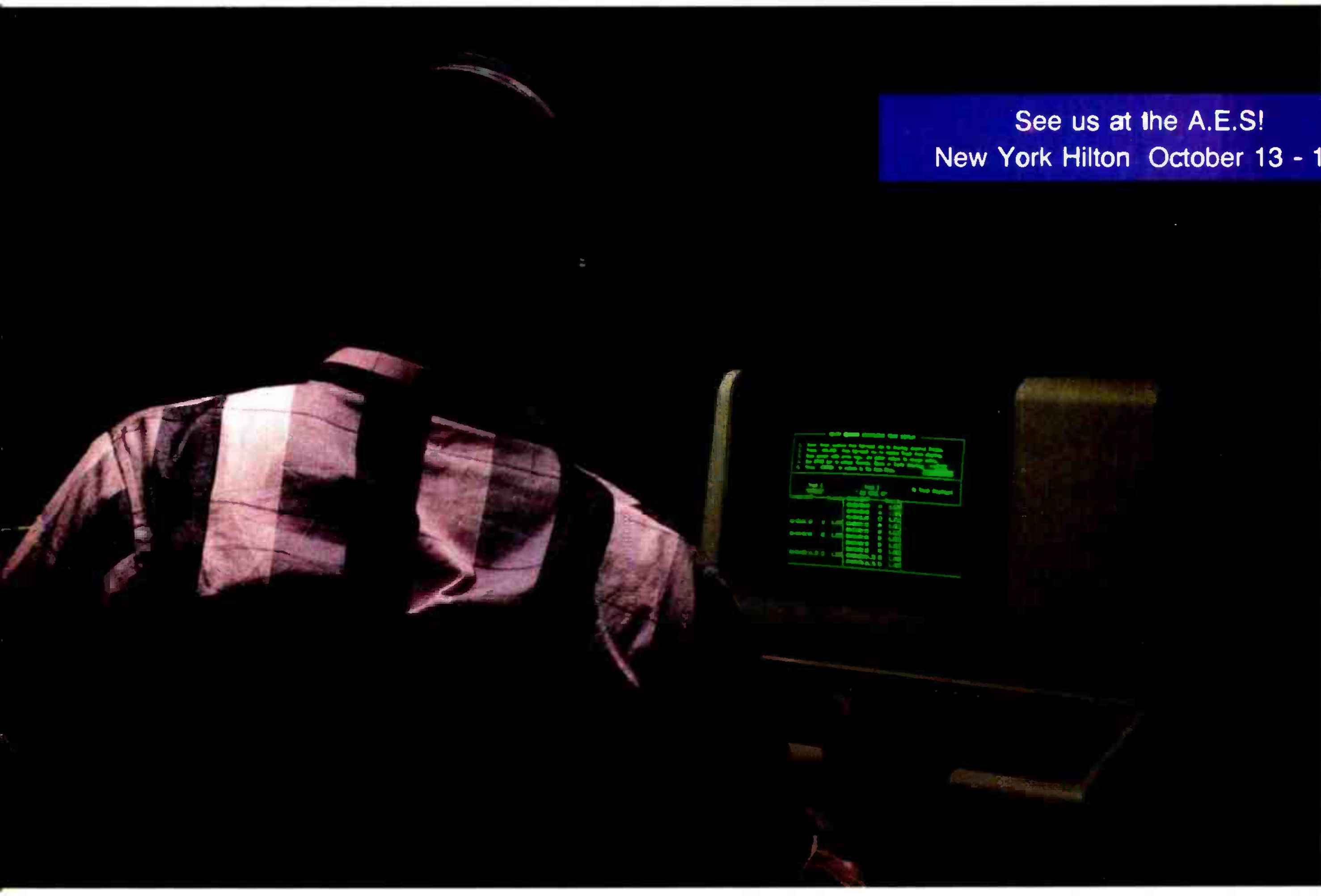
For more information or a personal demonstration, please call New England Digital or one of our authorized distributors:

New England Digital
White River Jct, VT 802/295-5800

Los Angeles
New England Digital 213/651-4016

New York
Digital Sound Inc. 212/977-4510

Synclavier®
DIGITAL MUSIC SYSTEM



See us at the A.E.S!
New York Hilton October 13 - 16

Synclavier operator captures continuous live vocal overdub

For additional information circle #106

. Digital 615/327-4343 **Israel** Syntron Ltd. 51 Ben-Zvi Blvd. Ramat-Gan **London** Turnkey 202-4366 **Montreal** Digital Music Systems 514/284-0609 **Toronto** Gerr Electro Acoustics Ltd. 416/368-0523

LETTERS

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data as a continuous transfer function. The frequency response data are then reported from inspection of the plotted magnitude data to the tolerances indicated by the manufacturer (±2 dB).

The variance in magnitude response indicated for the test results is due to the energy of the squarewave signal occurring at odd multiples of the fundamental frequency. For convenience, I choose to call data obtained by the above method the Dense Spectrum response, as it contains many more frequency components than does the single- or swept-frequency sinusoidal magnitude response. High-frequency response is shown more pessimistically this way, as the squarewave harmonics act to contribute to their own recording bias signal in a way proportional to their respective magnitude and frequency. I feel that this approach better represents the system response for the case of linearized analog magnetic record/reproduce systems than does the pure sinewave method, as sinewaves rarely constitute modulation signals commonly encountered. I grant that this is a more severe representation of the record/play magnitude frequency response. I do think it is more representative of common application.

I have begun to suspect that reality rarely has the charm of fantasy. ■■■

EXPOSING AUDIO MYTHOLOGY

Laying to Rest Some of the Pro-Audio Industry's More Obvious "Old Wives' Tales"

by John H. Roberts

In this month's column only I'd like to address some of the subtleties to balancing signals for transmission across the room, and across the country.

A popular misconception is that transformers are the *only* way to truly balance a line. Not only is this untrue, but many transformer-coupled lines are imbalanced by incorrect termination, or floated from ground intentionally.

Before we get into a strict definition of what is and isn't balanced, let's back up a minute and look at *why* one would want to balance a line. The basic goal is to transmit a signal from point A to point B with maximum fidelity.

There are many things that can happen between here and there to degrade your signal, such as: signal losses in the line; crosstalk between the channels; interference from outside noise sources; and ground-potential differences.

Signals within a given piece of equipment are usually routed around single-ended configuration, based upon the assumption that ground potential at all points within that box will be virtually identical. But for signals sent over any distance (like the inside of a large recording console), we must consider differences in ground potential. The most popular, and cheapest, way to correct for ground-potential errors is to use a differential summing amplifier (Figure 1).

$$V1 = Vs + Vg \times [R2/(R1+R2)] \times [1+(R4/R3)] + Vg(-R4/R3)$$

For R1=R2=R3=R4,

$$V1 = Vs + Vg(0.5\times 2) + (-1\times Vg)$$

$$V_{load} = V_{source}$$

The ability of the circuit shown in Figure 1 to reject this common mode of ground potential is directly related to the matching of R1 to R3, and R2 to R4. Typically, 1% resistors will be used in such circuits with critical applications being trimmed. The ratio of R1, 3 to R2, 4, need not be unity, allowing this topology to deliver boost or cut and still provide substantial attenuation of common mode signals. The ability to reject common-mode signals is called Common Mode Rejection Ratio (CMRR) and, as you can probably guess, is measured in decibels.

A second, somewhat more expensive, way of isolating ground-potential differences is to use transformer coupling (Figure 2). Since the signal is transferred from the transformer's primary to secondary winding as a magnetic flux, the secondary voltage is floating, and can be tied to any reference. This characteristic makes the transformers useful in isolating large potoential differences, and even allows some simple

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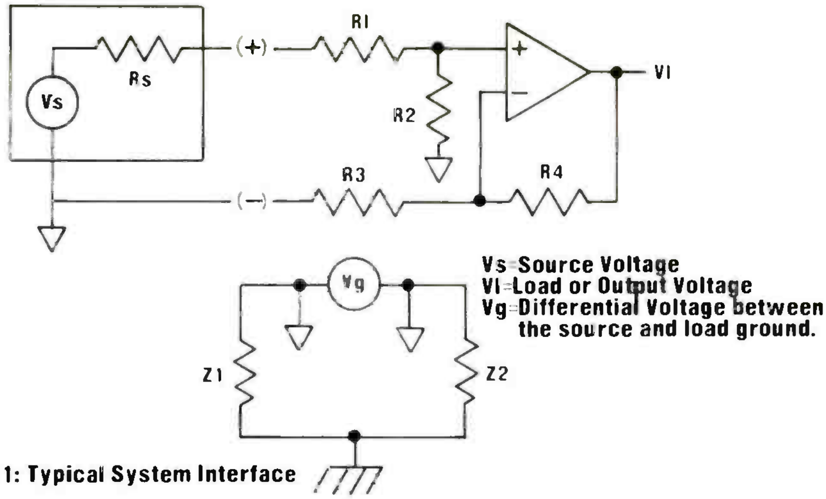
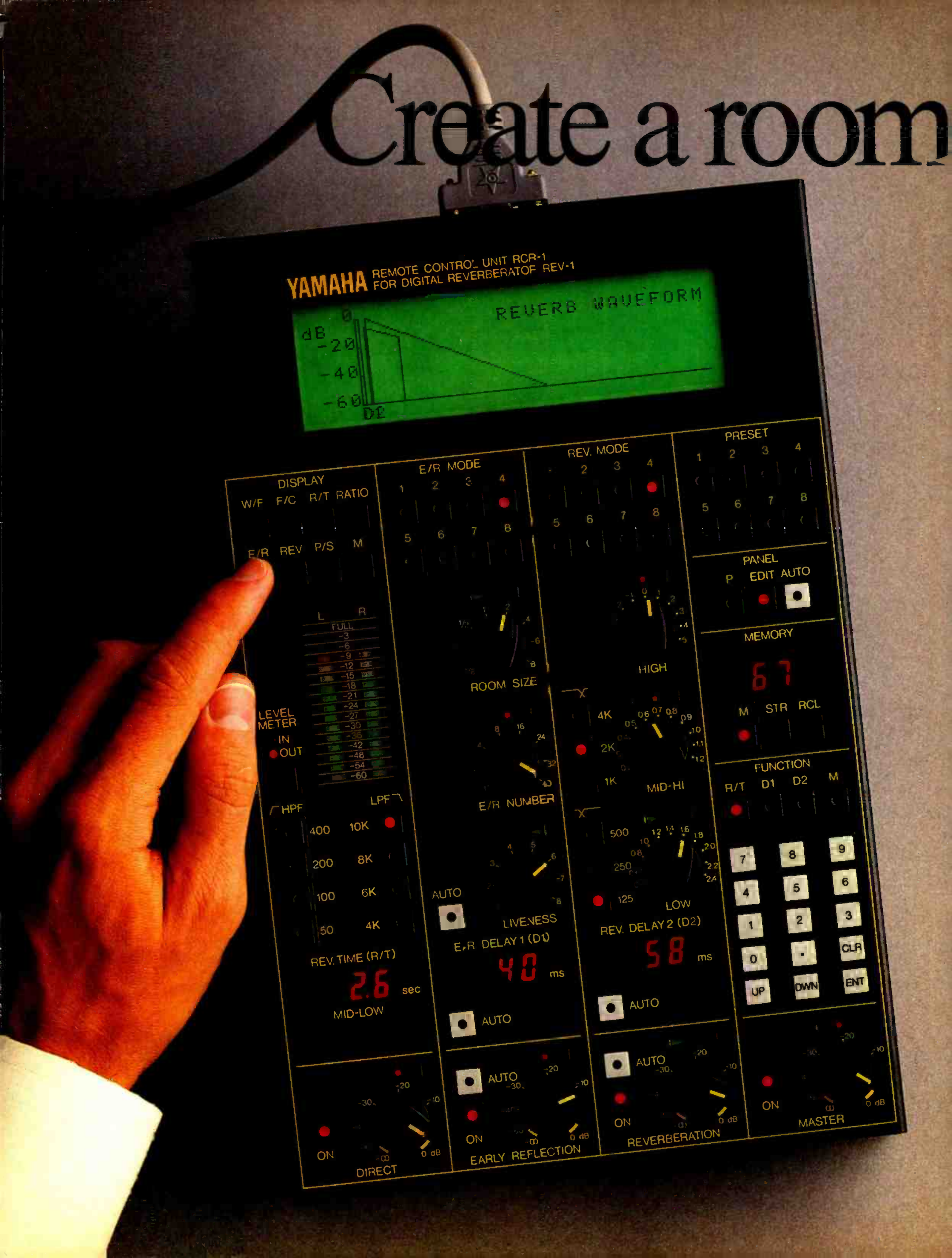


Figure 1: Typical System Interface


Create a room
YAMAHA
REMOTE CONTROL UNIT RCR-1
FOR DIGITAL REVERBERATOR REV-1
REVERB WAVEFORM
DISPLAY
W/F F/C R/T RATIO
E/R REV P/S M
E/R MODE
REV. MODE
PRESET
PANEL
P EDIT AUTO
MEMORY
M STR RCL
FUNCTION
R/T D1 D2 M
LEVEL METER
IN
OUT
ROOM SIZE
HIGH
E/R NUMBER
MID-HI
HPF
LPF
LIVENESS
LOW
AUTO
E/R DELAY 1 (D1)
40 ms
REV. DELAY 2 (D2)
58 ms
REV. TIME (R/T)
2.6 sec
MID-LOW
ON
DIRECT
EARLY REFLECTION
REVERBERATION
MASTER
CLR
UP
DWN
ENT

with a view.

We'd like to open your eyes to the incredible REV-1 digital reverb. Because it gives you unheard-of control over virtually all reverb parameters. And something that has never been seen in any type of reverb: the capability to "look" at the sound as well as hear it.

The remote unit that controls the nineteen-inch rack-mountable unit has a lighted high-resolution LCD display that graphically depicts the results of the adjustments you make.

So getting just the right reverb sound is no longer a question of trial and error.

"EARLY REFLECTION" display mode showing room size and relative level and time of discrete reflections.

"REVERB DENSITY" display mode showing level and relative time of subsequent reverberation.

"REVERB TIME" display mode showing difference in reverb time in each of four frequency bands.

"MEMORY TITLE" display showing the titles of internal ROM memories.

The logical grouping of the parameter controls on the remote also makes it easy to create any effect you like. Then store it in any of 60 memories for instant recall.

The remote also contains 9 additional RAMs so you can store programs and carry them with you to use anywhere there's an REV-1.

And there are 30 additional ROMs with factory preset sounds. Many of which can be completely edited (as can the user-programmed sounds) by using the LEDs to tell you the set value or indicate in which direction to move the control so you can easily and precisely match the value of the originally programmed sound.

And the sound itself is far superior to any other digital reverb. The REV-1 uses specially developed Yamaha LSIs to create up to 40 early reflections and up to 99.9 seconds of subsequent reverberation. So the effect can be as natural (or unnatural) as you want it to be.

We could go on about the REV-1. Tell you about its 44.1 kHz sampling rate that provides a full 18 kHz bandwidth to prevent the natural frequency content of the input signal from being degraded.

How it has a dynamic range of more than 90 dB for the delay circuitry and more than 85 dB for the reverb circuitry.

But why not take a closer look at the REV-1 at your authorized Yamaha Professional Audio Products dealer. Or for a complete brochure, write: Yamaha International Corporation, Professional Products Division, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Ave., Scarborough, Ont. M1S 3R1.

YAMAHA

For additional information circle #109

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addition or subtraction of signals. For example, connecting two secondaries in series with the same polarity (dot indicates positive polarity) will sum the two signals, while connecting the windings in series with opposing polarity will difference the two signals.

While both of these circuits are adequate at suppressing ground-potential errors, they are limited in their ability to eliminated noise signals induced into the interconnecting wires. Unless the signal source has an infinate impedence to earth ground (not likely), the two lines (positive and negative) will have different impedences to ground. These different impedences will cause slightly different noise voltages to be induced in each line, and thus not provide complete noise cancellation. An impedence can be added in series with the negative lead at the sending unit, equal to the single-ended source impedence, balancing (there's that word) the impedence to ground (Figure 3). This addition will improve rejection of common-mode noise and crosstalk, at the expense of doubling the effective source impedence.

However, the resultant circuit will still suffer from an imbalance in the impedence to ground as seen by a normal (sometimes referred to as "metallic") signal. This imbalance will cause signal currents to flow in the various equipment ground paths. The voltages generated by these ground currents should be common-mode, and thus reducible by the differential amp. It is good engineering practice, however, to avoid these currents in the first place, and not tempt Murphy to visit.

The easiest way to satisfy all of these requirements, and get some extra benefit

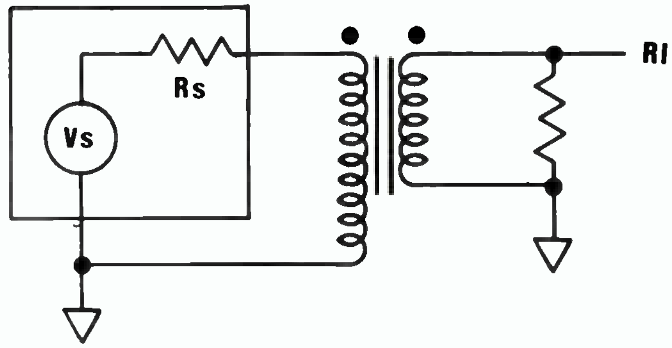

Figure 2: Unbalanced Transformer-Coupled Interface.

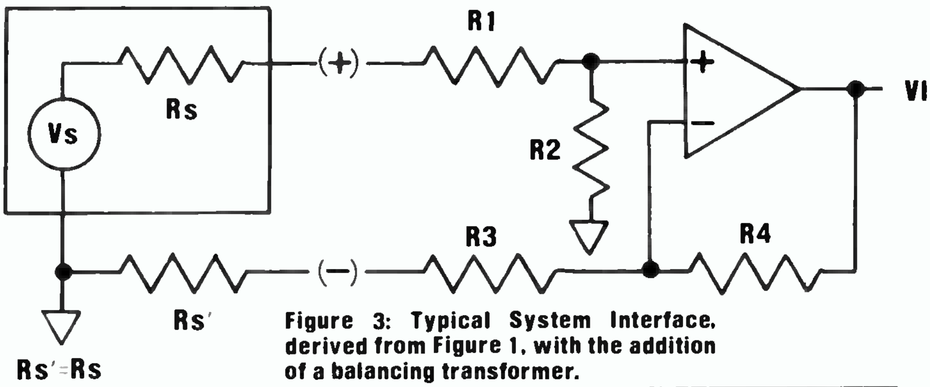

Figure 3: Typical System Interface, derived from Figure 1, with the addition of a balancing transformer.

THE PEAVEY CELEBRITY™ SERIES

Designed to have everything... except competition.

At Peavey Electronics we're dedicated to our commitment to design and manufacture high performance products at realistic prices. We've underlined that philosophy with our Celebrity Series line of microphones.

The Celebrity Series feature large diameter diaphragm/voice coil structures for increased sensitivity with the ability to handle high sound pressure levels. These higher output levels allow for significantly less mixer gain and are a tremendous aid in maintaining good signal-to-noise ratios.

Perhaps the most important characteristic of any performing microphone is reliability. The design of our cartridge/shock mount system increases ruggedness as well as isolation capability to insure long-term performance under severe field conditions.

Our microphone screen utilizes extremely heavy gauge wire that has been "junction locked". Once the screen is formed, we do not stop there. The heavy wire screen is "fired" in an oven after forming, thus causing the plated wire to "fuse" at all interconnecting points. The result is an unbelievably durable "brazed" wire windscreen that will hold together under the most severe abuse. After the ball windscreen is formed, brazed and coated, a precision urethane foam pop filter is fitted to minimize the undesirable proximity effects. This special acoustically transparent foam protects the entire sound system by breaking up explosive high SPL pressure waves created by close vocals or close miking percussion instruments. For those applications requiring even more acoustic screen from wind noise, etc., Peavey offers special external colored wind noise filters that slip over the screen and internal pop filter.

While outwardly, the appearance of the Celebrity Series is somewhat conventional, the aspect of "feel" has been given heavy emphasis since our experience has shown that performers prefer a unit that not only sounds right and looks right, but must also have a comfortable balance, weight, and overall tactile characteristics.

Special "humbucking" coils (models CD-30™ & HD-40™) have been designed into the microphone element that effectively counter-balance any hum that might be picked up from external sources. Performers who play clubs where hum from light dimmer switches or other sources are a problem can appreciate this unique feature.

We invite comparison of our Celebrity Series with other cardioid microphones. You'll see why we feel that in terms of performance, features, and price, there is no competition.

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For additional information circle #111

to boot, is to break the signal up into two opposing polarity versions of the original. The positive polarity version is sent at one half its original amplitude down the positive line, and the opposite polarity version sent at one half the original amplitude down the negative line. When these to signals are recombined in a differential summing input, such as one of thse shown in Figure 4, the original full-level signal is recovered, while common-mode noise is suppresed and ground potential errors rejected.

In addition, crosstalk will be dramatically reduced, for two reasons: first, you are now only sending one half the original peak level down the line; and secondly, there will be some self-cancellation by the two closely-coupled but opposite polarity signals. Note: you can get all the benefits of balanced outputs/lines with the simple differential input of Figure 1B, except for the freedom from ground-signal currents.

Now for the strict definition of balancing: A "balanced" output will provide two equal (but opposite polarity) signals, symmetrically situated about ground or some appropriate reference potential. The impedence to ground from both positive and negative outputs will be equal for common-mode signals and for normal signals. Note: the common-mode impedence may be different from the normal impedence, as long as it is always the same at both outputs. A "balanced" line and input will have identical impedence restraints, with the "balanced" input having very closely matched but opposite polarity gain at its input ports.

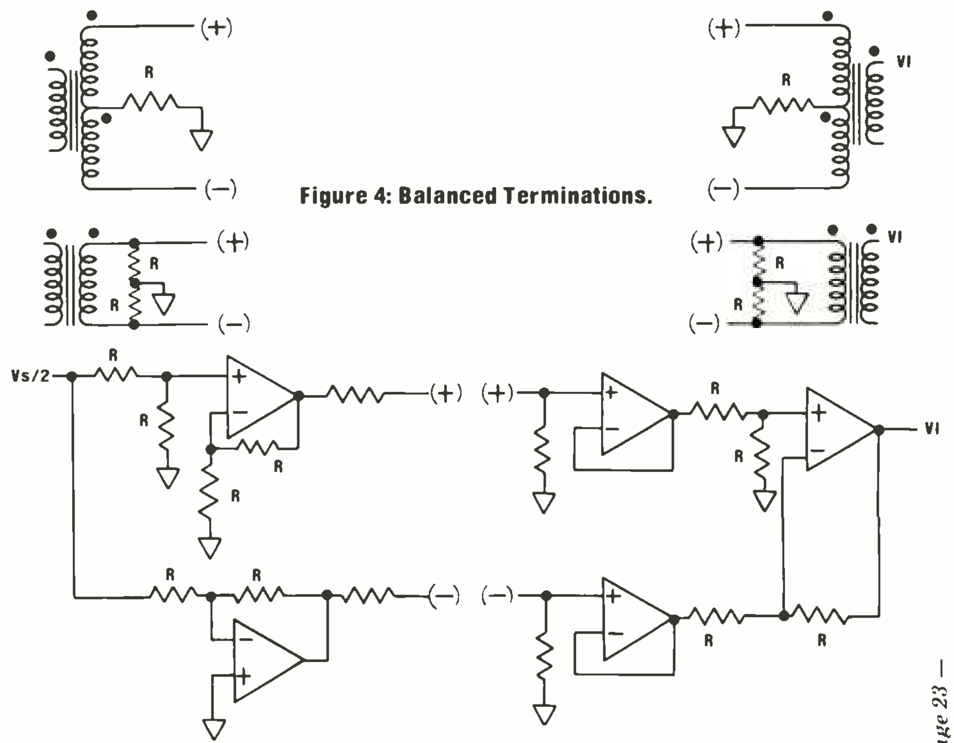


Figure 4: Balanced Terminations.

Electronic Versus Magnetic

It is possible to meet these requirements easily with op-amps as well as transformers. Since op-amps are much cheaper and smaller than even a low-grade transformer, you might wonder why anyone would ever use one. Well... there are several reasons:

First, transformers make an excellent

... continued on page 23 —

September 1985 No. 6

ECHO TIMES

EMI: AMS

At a recent studio managers' conference held at EMI Abbey Road Studios in London it was unanimously agreed that pieces of AMS outboard equipment would be made available for every control room in all EMI recording studios worldwide. The delegates represented studios from EMI's international network including Japan, Australia, New Zealand, U.S.A., Germany, Sweden, South Africa, France, Holland and the U.K.

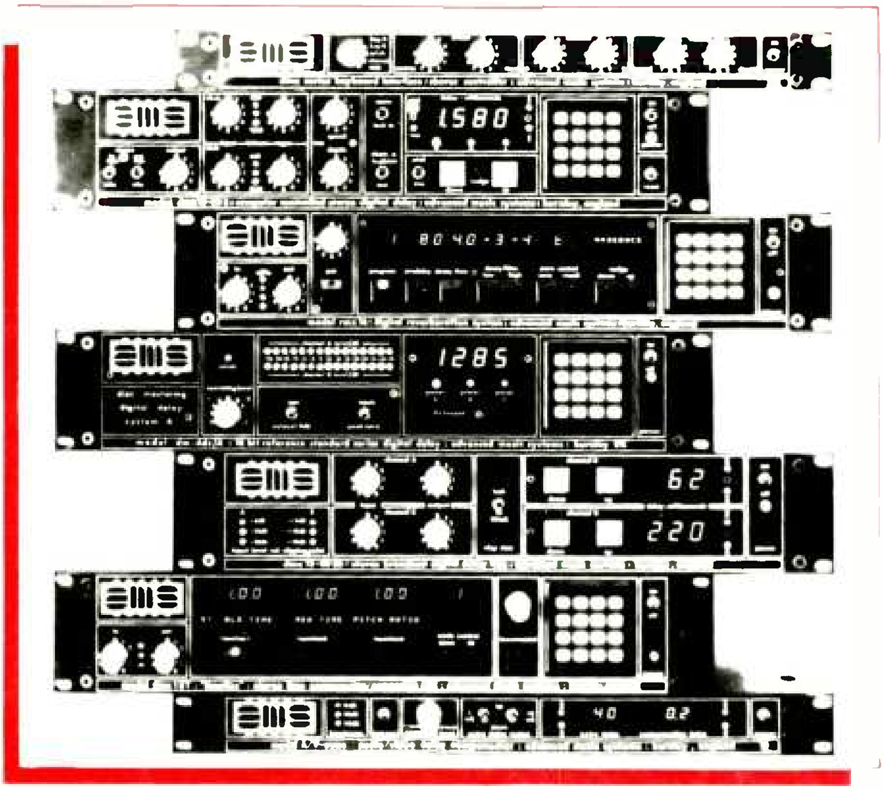
NEW·NEW DMX 15-80S Two Channel Sampling

A new update has now been introduced for the DMX 15-80S giving users the option of sampling and triggering two independent pieces of information.

Each of these samples is controllable as with the original single loop – continuously looped, manually single triggered or triggered by audio input. In the case of audio triggering, audio input sufficient to illuminate either the channel A or channel B input LEDs will result in triggering of the sample stored on that channel of the unit.

NEW·NEW RMX 16 Memory Expansion

The RMX 16 can now be supplied with memory expansion to increase the number of factory set programmes from 9 to a number capable of accommodating all AMS factory set programmes available at any one time. New programmes for the RMX 16 will still be made available on bar code allowing those owners with remote terminals and wands to immediately take advantage of new software issued.



NEW·NEW TIMEFLEX

AMS Timeflex has continued to prove its popularity with audio, video and film post production facilities by providing very high quality audio time compression. Timeflex is capable of operating in mono, dual channel or stereo modes and for this reason contains additional circuitry to that offered with standard AMS pitch changers to ensure complete phase matching of channels when used on a stereo signal.

A new interface card for AMS Timeflex is now available providing communications to external audio, video or film machines. The two standards currently offered are RS 422 and 9K6/19K2 tacho signals. These interfaces allow Timeflex to automatically correct audio pitch should the machine to which it is interfaced be vari-speeded. Alternatively, again using this interface, Timeflex can behave as master simply allowing the user to enter a new play time and accordingly Timeflex will accurately alter the machine speed and correct the audio pitch.

TIMES SIX

The popularity of "Echo Times", particularly in the U.S.A., as a medium for keeping owners, users and potential owners of A.M.S. equipment up to date with the latest developments has not gone unnoticed. We have received many requests to supply back-issues and accordingly reprints of all previous issues have been made and complete sets are now available on request.

The following people were interviewed in issues 1 to 5, all discussing their uses and applications for A.M.S. units: Martin Rushent, Kevin Peak, Air Studios, Hilton Sound Rental Company, Tom Bailey (of the Thompson Twins), Phil Collins, Humberto Gatica, Jeff Lynne of ELO, Paul McCartney and Hugh Padgham.

People in the know

"The AMS DDL is used to provide variation on the various rhythms, especially the bass drum rhythms. Effects used on 19 were setting the delay to a semi quaver's length so that instead of a steady four on the bass drum you get sixteenth notes in succession. A reverb with a long delay time could then be added to the original bass drum but omitted from the echoes for extra effect. Another effect that was used was to make the echo fall on an existing beat such that phase elimination would occur.

Also sometimes I add a bit of white noise to the snare by playing it onto a track from a synth, just to make it sound bigger. And I've found ways of using the AMS to make the sound much bigger."

Paul Hardcastle talking in an interview with Richard Walmsley in Electronic Soundmaker and Computer Music magazine.

"It is generally felt in digital circles that hard-disc editing is the way of the future, and with AudioFile, AMS has beaten many of its larger competitors. The software possibly needs a little refinement, but I for one am looking forward to the day when I can install one of these devices in Tape One."

Bill Foster of Tape One studios talking in Music Week

"One of the stars of APRS 85 was AudioFile from AMS."

Jim Evans of Music Week.

"On Mag element A there was an LCR band mix, mag B contained Sting's vocal on track one, and the girl backup vocals on tracks two and three, and on the last three-track mag element there was bass and stereo audience. AMS digital effects were summed onto selected tracks during this mixdown: "AMS mania" according to Aaron."

Brad Aaron talking to Larry Blake of Recording Engineer Producer magazine about "The Police Synchronicity Concert" film.

"After all, recording in 1985 is not like recording even in 1982. A little bit of the modern technology had kind of passed them by while the band was regrouping (after Lionel Richie went solo). They saw the AMS gear lined up in the outboard rack, and they couldn't believe it. We were sampling drums: we'd have a guy come in, but we wouldn't use him playing – we'd just sample his kit. Then we would have the track programmed on a Linn and we'd replace the machine bass drum or snare with sampled sounds.

It took the making of this album for the band to embrace the new technology."

Dennis Lambert talking about the making of the Commodores "Nightshift" album in an interview with Mel Lambert and Ralph Jones of Recording Engineer Producer magazine

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"If we're doing the cymbal parts separately, I'll use an AMS stereo timeprocessor with no delay using pitch changer on A channel reading 1.005 and on B channel reading 0.995 (1.000 is normal pitch). If you send the left hand cymbal track to the B channel of the AMS which returns on the right hand side, and send the right hand cymbal track to the A channel of the AMS which returns on the left hand side, this gives a nice zingy spread to the cymbals without being too splashy."

Producer Steve Brown talking to Janet Angus about his work with Wham, ABC and many others in HSR magazine

APRS '85: Stuart Nevison of AMS with Stewart Copeland of the Police, discussing AMS AudioFile.

APRS '85: Ian Noble of AMS discussing AMS AudioFile with John Paul Jones formerly of Led Zeppelin.

"Is there any outboard equipment you particularly like?"

"Well I really like our AMS reverb, it gets used on nearly everything."

Muff Murfin, studio owner talking to Paul White of HSR magazine.

"One thing we did was to take the kit out into the live foyer, record the snare onto digital, pick up a good sounding hit and dump it into the AMS digital memory. Then in the mix we triggered it from the normal snare and added it to the overall sound to give a bigger Ambiance."

Producer Chris Kimsey discussing the track Kayleigh by Marillion with Jim Betteridge in International Musician and Recording World.

"When I mix I like to have a couple of AMS delay lines, minimum, an AMS reverb, as many Pultec (valve) equalisers as there are in the world because I love to record drums through them."

Chris Tsangerides talking about studio work with such bands as Thin Lizzy with Peter Buick of Sound Engineer magazine

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"Outboard equipment is also comprehensive with AMS 1850S and RMX16 units, a Yamaha Rev 1 and the Lexicon 224. Nick also has thoughts for the future in this area, "I would dearly like to get the AMS Audiofile. It would be absolutely super – both for our audio clients and straight audio use".

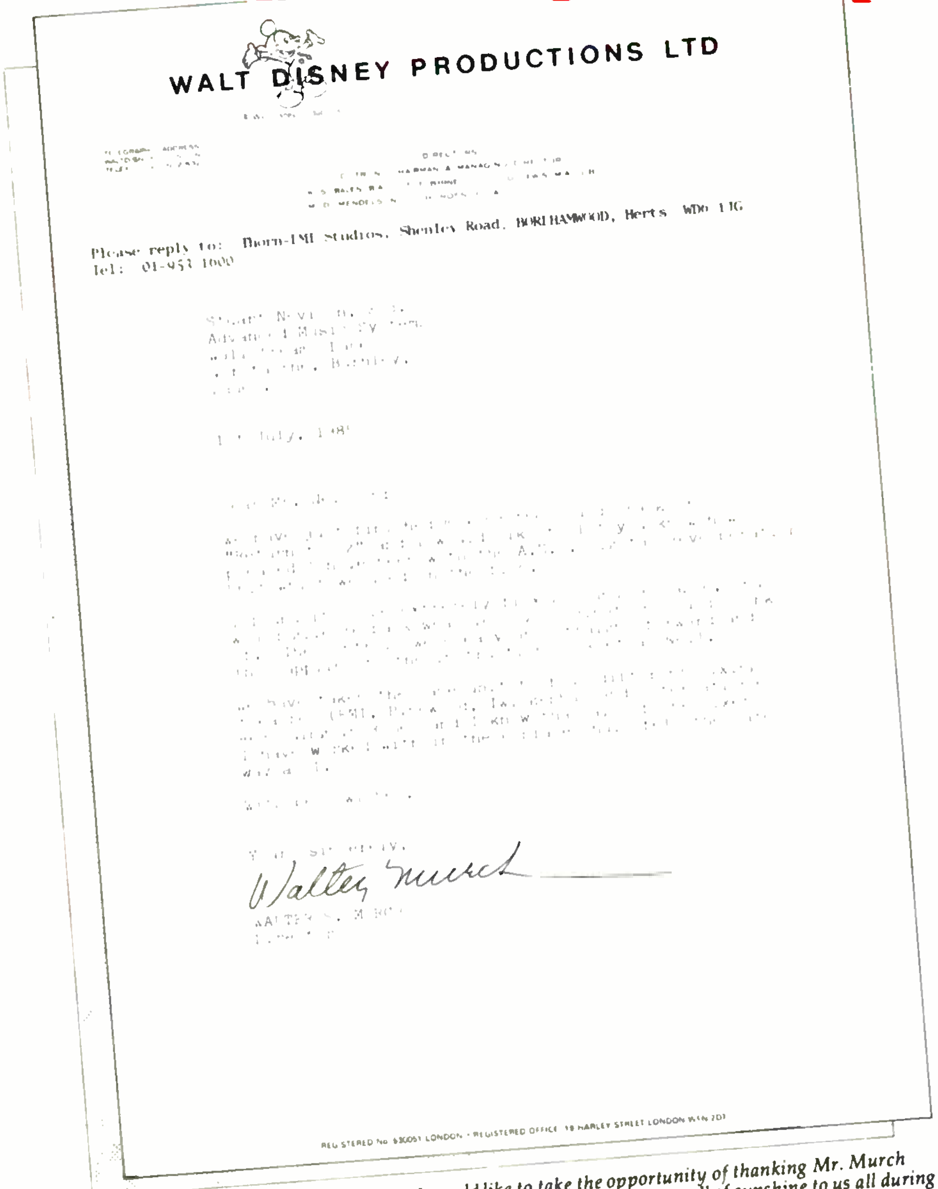
Nick Turnbull talking to sound engineer

"On the Go West album we only had the MSQ 700, which was our lifeline. Now we've got the SXB, which links up really well with the TR 909. It's great for programming, triggering the AMS and stuff like that.

"For the Radar album we used the SRC – I like to have that facility because you can change the drum patterns if new ideas come up. On some of the tracks we had to pull out whole bass lines and relocate them with the AMS. It's like painting pictures – you can just rub a bit out and move it. It might take two hours but I'll pick out a couple of things I can use somewhere else and it sounds really whacky."

Go West producer Gary Stevenson talking to Peter Buick of Sound Engineer magazine.

Disney • Disney • Disney



WALT DISNEY PRODUCTIONS LTD

Please reply to: Thorn-EMI Studios, Shenley Road, BOREHAMWOOD, Herts WD6 1JG
Tel: 01-953 1600

Walter Murch

Following receipt of the above letter, I would like to take the opportunity of thanking Mr. Murch on behalf of all the staff and workforce at AMS. His letter brought a spell of sunshine to us all during what must be one of the coldest and wettest English summers on record!

Ray Parker Jnr.

Ray Parker Jnr. is one of those people who never cease to amaze you as to how many projects they have been involved in or even how many successful songs they have written. Although not particularly big in England, Ghostbusters gave Ray three separate attacks at the British charts – firstly on the singles release, secondly on the release of the Ghostbusters film and finally it climbed the charts again as a 12" mix.

A.M.S.: *Briefly, what is your history?*

R.P.J.: The first 5 or 6 years of my career I worked as a studio musician and got involved in a series of different projects ranging from Marvin Gaye and Stevie Wonder to the Rolling Stones and Boz Skaggs. Then I got into writing songs and had success with things for Barry White, Rufus, Chaka Kahn and of course my own stuff – Ghostbusters was obviously a big break.

A.M.S.: *Is there anything you consider distinctive in the way you work?*

R.P.J.: I don't know about everyone else but I write to sounds. I have to go into the studio and hear the drums just like they are going to be on the record – I've got to hear the synthesizers, again just as they are going to sound on the record. Once I've got a framework I can formulate other things around that – I can't just sit down with a Linn like some people do. I have to have the sound EQ'd with reverb and effects added which is why AMS is so important.

A.M.S.: *What other reverb units do you use?*

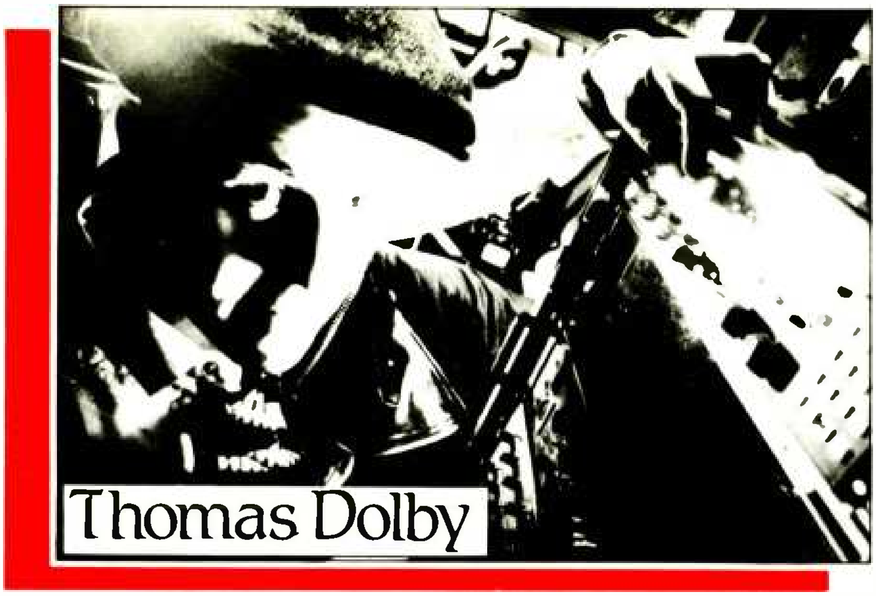
R.P.J.: Let me see, I've had a AKG spring for 9 years. I have a Lexicon 224 and the 224X and a big EMT but I've never really got into that. I like things where I can reach them and just punch buttons which is one reason why I decided to add the RMX 16. I love the sound of the A.M.S. reverb and the sounds I really like I can get quickly and easily. For that reason it's the system I use most of all – that and probably the 224.

A.M.S.: *Do you have any favourite programmes?*

R.P.J.: All the programmes sound real good but my favourite is the AMS Nonlin. It's so different – it's unique – yeh! AMS Nonlin I really love that one. The Reverse programme is nice too. I guess a plate or a plate programme will get people to say well that's reverberation – but these special effects programmes are real nice.

A.M.S.: *So what's next for you?*

R.P.J.: I enjoy being a solo artist/engineer and all I want to do is get in there and play with more buttons and gadgets and just experiment. I've heard a lot about DMX 15-80S DDL pitch changer and it sounds real interesting – I don't own one yet but my studios here are just choosing some new gear so who knows! Don't forget to listen out for my new album and 45 you'll definitely hear lots of AMS on them.



Thomas Dolby

Thomas Dolby seemed to appear from nowhere at a time when totally synthesizer based bands such as the Human League were enjoying the peak of their success. Unlike quite a few of the "totally electronic" bands, Thomas Dolby has survived and gone on to further develop his individual style. AMS caught up with him during a three month stay in Los Angeles where, amongst other things, he was completing work on a project with Joni Mitchell.

A.M.S.: *So here we are in the Hollywood Hills!*

T.D.: Yeh! I've rented this house whilst working here. The best thing about the house is not that it originally belonged to Jenny Agutter but that Steve McQueen's 50's pick-up truck is down in the garage in absolutely showroom condition.

A.M.S.: *How did your career develop?*

T.D.: When I was 14 I used to write the odd song on the piano but with not having lessons there was never any discipline to get good at it. Because of that I moved to synthesizers. People had just got past the long blond hair and cape stage and instead of individual bravado on a Minimoog, people like Brian Eno exploring different textures created by a synthesisers were beginning to influence popular music. Living alone in London during the Punk era meant that even though I'd got very good at writing and arranging quite sophisticated songs on the Portastudio that had just come out, I really wanted to play in a band. So I managed to get some session work with bands including Bruce Woolley, Lene

A.M.S.: *So how did the first album surface?*

T.D.: Doing sessions got me a bit of a reputation as a player which did open a few A & R men's doors. The first album was really just making a 24 track version of my demo material which I think caused it to suffer a bit as there were some things I just couldn't recreate. There is a lot going on in my songs and the fact that I write and direct my own videos gives me an opportunity to explain them better. The coverage given to the music and videos by MTV and cable here in the States gave me the break and it happened here in America before anywhere else.

A.M.S.: *Did you approach the second album differently?*

T.D.: Very much so, I don't think I was ever a part of the totally electronic sounding cult, but, people that liked those sort of bands would at least give me a listen and hopefully find something else in there. By this time I had used the DMX 15-80S as a sophisticated delay line, it was the first system with a good sound and character that meant you could match tape echo. Peter Gabriel and Kate Bush had just been through Townhouse studios using the Fairlight and at the same time "sampling" was everywhere and really hip.

A.M.S.: *So sampling and the Fairlight played an important role on your second album?*

T.D.: Yes they did and so did A.M.S. I write mainly on the Fairlight – however the Fairlight, as it stands now, seems to have the potential that the more you build – the smaller it gets if you know what I mean. So once I've done my arrangement I go back to the original sounds that I've sampled, store and edit them in the 15-80S and then trigger them from the Fairlight. That gives me far superior sound quality and perspective, longer samples and also very importantly more accurate control by being able to offset the triggered samples to get the right feel to the piece.

A.M.S.: *You aren't the first person I've heard mention perspective. How important is that to your music?*

T.D.: Perspective has been an enormous breakthrough. The creative energy in England that continues to build up seems to have gone into production rather than the raw commodity, but there are a group of English producers that are 2 or 3 years ahead of the rest of the world – and I think it's because of their use of perspective. When all you had was an echo plate the information you got was how far away you were from an instrument. Now with delay lines and units such as the RMX 16 you can create atmospheres and your instruments can come from anything from a small room to an empty lonely canyon.

A.M.S.: *Do you create these perspectives during recording?*

T.D.: Yes, my approach is very cinematic, I tend to use the RMX 16 during the recording process to build up the song as I don't like leaving everything to the mix. You can make mistakes this way introducing perspectives to a single track that don't work when taken with the whole song. In an ideal world I would have a huge rack of multiple everything such that the mix would be vocals from the multitrack and everything else running live.

A.M.S.: *Does that mean something like AMS AudioFile interests you?*

T.D.: AudioFile is very exciting, I could quite happily do away with my multitrack tape recorder because AudioFile would allow me to drop in and out of record, edit within a track and repeat phrases. It's fascinating and given it's my own view of the way things are going to go I think AudioFile is the first serious device to arrive and I'm sure it will have a big influence.

A.M.S.: *A final question. Is there any function on any piece of A.M.S. equipment you would miss most if you lost it?*

T.D.: No. I'd miss them all!

— *continued from page 18* . . .
band-aid. If you have a problem with imbalance at an input or output, simply bolt on a transformer and, *presto*, you've saved another box from hi-fi heaven.

Secondly, since they are more tolerant of miswired connections, transformers help in overall system reliability. Try shorting out your electronically balanced out put. While this shouldn't cause instant smoke on properly designed gear, you are using up one or more of its nine lives, making it more likely to fail from some other stress. Also, transformer-coupled inputs/outputs are not capable of passing DC; a nice thought when you're using DC-coupled power amps to reproduce those alpha waves found in today's popular music.

Secondly, since they are more tolerant of miswired connections, transformers help in overall system reliability. Try shorting out your electronically balanced out put. While this shouldn't cause instant smoke on properly designed gear, you are using up one or more of its nine lives, making it more likely to fail from some other stress. Also, transformer-coupled inputs/outputs are not capable of passing DC; a nice thought when you're using DC-coupled power amps to reproduce those alpha waves found in today's popular music.

Finally, the best reason for using transformers — and it has only a little to do with noise — is the abililty to block large common-mode potentials. The typical diff amp or electronically balanced input will only handle about 15V of common-mode garbage. This common-mode range can be increased by padding down the input and restoring it later, but with a subsequent reduction in dynamic range (because of the increased noise floor). Even with this increased common-mode range, a mis-wired stage-power circuit or lightning-induced spike (you don't need a direct hit) can still cause catastrophic failure.

Conclusion: Electronically balancing equipment is so inexpensive that I can see almost no reason for not doing it universally. (However, it still isn't free.) If you are sending signals over fairly short runs, the use of shielded, single-ended outputs with the simple diff amp as shown in Figure 1 should be sufficient. If you are working with live PA, all line-level snakes should be balanced (mike-level snakes already are).

Whether you need transformers or not is a judgement call. If the band and house mixer plug into the same extension cord, don't worry about it. If you get electricity bills from two different utility districts . . . worry! Finally, if you are dealing with telephone lines, or high-risk gigs (like protecting yourself when interfacing with somebody else's system) transformers are still the *only* way to go.

If you do decide to use a transformer, check out what is commercially available. Transformers are not at the point where their sound quality can be taken for granted. While, at their best, they can approach a high performance op-amp, the typically lower-priced transformer can degrade your system's sound. If you are in a cost-sensitive situation (and who isn't?) listen to a few different samples of what you can afford. Take care to terminate them with the same impedences, and run them at the same levels they will see in normal use. Maybe even listen to a sample of what you can't afford, and then make your choice.

For simple interfacing, I can think of no good reason to run the signal through even the best transformer. But, until everything is getting sent around on fiber-optic cables, there will be a place for transformers in professional audio. ■■■

Reading For Extra Credit

References 1, 2 and 3 go into great detail regarding grounding and shielding of all kinds or signals. Reference #4 is a good source of information about transformers designed for audio applications.

1. Henry W. Ott, *Noise Reduction Techniques in Electronic Systems* (John Wiley & Sons, New York, 1976).
2. Ralph Morrison, *Grounding and Shielding Techniques in Instrumentation* (John Wiley & Sons, New York, 1977).
3. Edward F. Vance, C *Shielded Cables* (John Wi New York, 1987).
4. Various Jensen Trans sheets and application no from 10735 Burbank Bou Holywood, CA 91601

News

KURZWEIL ANNOUNCES 50 kHz USER SAMPLING FOR 250 DIGITAL SYNTHESIZER

According to Bob Moog, VP of new product research, "The new 50-kHz enhancement has been tested extensively at the factory and in the field, and offers dramatically brighter and crisper sounds."

Starting in September, all shipments of the Advanced Sampling Kurzweil 250 unit, as well as the Sound Modeling Program option, will come standard with 50-kHz digital sampling. [An in-use operational assessment of the K250 can be found on page 194 of this issue — *Editor.*]

List price for the new 50 kHz Sound Modeling Program option is $1,995, the same as the previous 25-kHz version. Current owners of K250s with the 25-kHz version can upgrade to 50 kHz for $250 at an authorized Kurzweil service center.

MORE NEWS . . *continued on page 225* —

A PERSONAL VIEW

LET'S NOT FORGET THE SOUND:

Quality and Operational Advantages of Working with 35mm Mag Film, Compared with Tape-based Systems

by Murray R. Allen, president, Universal Recording Corporation

As the recording industry moves into the mid-Eighties, we are bombarded on a daily basis with all kinds of new technology. There are digital reverb systems that will allow us to create the acoustics of a room of unlimited dimensions around any sound that we record; there are synthesizers that will create just about any sound we want to put into that room of limitless size; and there are tape machines capable of recording up to 32 tracks on one piece of tape. And if that isn't enough, we can buy, rent or steal any number of timecode synchronizers to enable us to record an unlimited number of tracks of our synthesized sounds placed in the middle of our synthesized room.

I think this is all fantastic; some of the sounds being created today are really great in every meaning of the word. But there are some very basic principles being overlooked, and the technology of audio postproduction sweetening for video might well represent a good example of this possible obsession with technology at the expense of sound quality.

As many *R-e/p* readers will already be aware, when engineers began to record video pictures on magnetic tape, it became necessary to develop an addressing system that would accomplish the same purpose as the feet and frames information generated by sprocket holes on film. The resultant SMPTE timecode data records on a conventional audio track — or, the case of Vertical Interval Timecode (VITC), within the video picture itself — made it possible to locate electronically any point on any piece of magnetic tape to an accuracy of at least 33 milliseconds. It also became possible to lock together two or more tape transports in perfect synchronization, a function that opened the possibility of creating totally mixed tracks that would synchronize with picture without using the time-honored method of sprocketed magnetic film.

Those of us involved in both film mixing and non-sprocket recording were thrilled with this capability, for obvious reasons. In the late-Seventies the consoles used for studio recording were moving ahead technically faster than those designed for film re-recording, and the average, small rock and roll studio had more processing equipment than a larger film-mixing facility. The speed and difficulty of synchronizing sprocket film to tape, along with the questionable quality relative to magnetic film, contributed to the appeal of video sweetening using videotape rather than film.

Advantages of Mixing on Magnetic Film

However, we have all overlooked the good points of film mixing to embrace this new video-based technology. And times have changed. Seven years later, film mixing consoles are as well equipped as music consoles and, when it comes to panning features, they are superior. Every type of processor known to man will also be found in the film-remixing theatre. Film and tape can now be synchronized with no problems relative to speed or operational difficulty. In fact, when it comes to shifting tracks relative to one another, film-mixing techniques offer a superior advantage, since a mag dubber can be advanced a sprocket hole at a time to provide sync offsets in about 10-millisecond increments. Magnetic film stocks have also come of age. Because it is coated on a three- or five-mil base, mag film offers superior print-through relative to standard two-inch tapes. Also, 35mm magnetic film moves at speeds of 18 or 22.5 ips, depending on whether it is running at 24 or 30 frames a second. In addition, the frequency response of mag film is comparable in every way to regular two-inch tape.

Now let us get down to the reality of what happens in a mix. First we will examine what happens during a typical video-sweetening session. The client has mixed down his original 24-track music to four tracks of a new 24-track tape, along with timecode and 59.94 Hz video sync on two additional tracks. The track breakdown is (usually) rhythm, lead vocal, background vocals, and horns and strings. Additional tracks to be recorded on the new timecode-striped 24-track consist, typically, of sync dialog recorded on location, voice-over announcer recorded in the studio, and five sound-effects tracks.

The location sync sound normally comes out of the field on quarter-inch/full-track tape with a 60 Hz neopilot resolve tone recorded across the tape. Step #1 will be to transfer the location audio, using a machine capable of resolving the neopilot tone, to one channel of a two-track recorder, simultaneously recording timecode onto channel #2. (Make sure that your timecode generator is looking at the *same* clock source as your neopilot resolver.) Now line up your 24-track with your video picture; hopefully somebody knows at which timecode frame the music

...insist on a Jensen!

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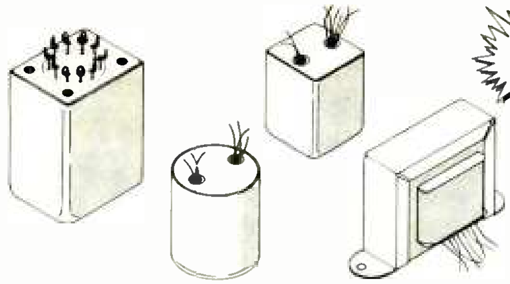

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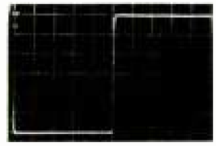

JE-16A
2 kHz Square Wave

INPUT TRANSFORMERS AND SPECIAL TYPES

Model	Application	Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20Hz Max Input Level[1]	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width[2] −3 dB @(kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Noise Figure (dB)	Magnetic Shield[4] (dB)	Number of Faraday[4] Shields	Package[5]	PRICES 1-19	100-249	1000
MICROPHONE INPUT																
† JE-16-A / JE-16-B	Mic in for 990 opamp	150-600	1:2	+8	0.036/0.003	−0.08/−0.05	230	−8	<1	1.7	−30	1	A=1 / B=2	65.25 / 71.73	43.59 / 47.92	30.07 / 33.06
JE-13K7-A / JE-13K7-B	Mic in for 990 or I.C.	150-3750	1:5	+8	0.036/0.003	−0.09/−0.21	85	−19	<2	2.3	−30	1	A=1 / B=2	65.25 / 71.73	43.59 / 47.92	30.07 / 33.06
JE-115K-E	Mic in for I.C. opamp	150-15K	1:10	−6	0.170/0.010	−0.50/+0.10	115	−5	<7	1.5	−30	1	3	44.84	29.95	23.39
LINE INPUT																
JE-11P-9	Line in	15K-15K	1:1	+26	0.025/0.003	−0.03/−0.30	52	−28	<3		−30	1	1	105.75	70.65	48.74
JE-11P-1	Line in	15K-15K	1:1	+17	0.045/0.003	−0.03/−0.25	85	−23	<1		−30	1	3	42.69	28.53	22.27
JE-6110K-B / JE-6110K-BB	Line in bridging	36K-2200 (10K-600)	4:1	+24	0.005/0.002	−0.02/−0.09	125	−12	<1		−30	1	B=1 / BB=2	63.98 / 74.05	42.75 / 49.47	31.37 / 34.13
JE-10KB-C	Line in bridging	30K-1800 (10K-600)	4:1	+19	0.033/0.003	−0.11/−0.08	160	−9	<2		−30	1	3	43.45	29.03	20.03
JE-11SSP-8M	Line in/repeat coil	600/150-600/150	1:1 split	+22	0.035/0.003	−0.03/−0.00	120	−9	<3.5		−30	1	4	168.39	112.50	77.61
JE-11SSP-6M	Line in/repeat coil	600/150-600/150	1:1 split	+17	0.035/0.003	−0.25/−0.00	160	−5	<3		−30	1	5	85.11	56.86	39.23
SPECIAL TYPES																
JE-MB-C	2-way[3] mic split	150-150	1:1	+1	0.050/0.003	−0.16/−0.13	100	−12	<1		−30	2	3	36.22	24.21	18.89
JE-MB-D	3-way[3] mic split	150-150-150	1:1:1	+2	0.044/0.003	−0.14/−0.16	100	−12	<1		−30	3	3	63.35	42.32	33.04
JE-MB-E	4-way[3] mic split	150-150-150-150	1:1:1:1	+10	0.050/0.002	−0.10/−1.00	40	−18	<1		−30	4	1	98.99	66.13	45.62
JE-DB-E	Direct box for guitar	20K-150	12:1	+19	0.096/0.005	−0.20/−0.20	80	−18	<1		−30	2	6	45.46	30.38	23.71

1. (dBu) Max input level = 1% THD; dBu = dBv ref. 0.775 V
2. With recommended secondary termination
3. Specifications shown are for max. number of secondaries terminated in 1000 ohm (typical mic preamp)
4. Separate lead supplied for case and for each faraday shield
5. Except as noted, above transformers are cased in 80% nickel mu-metal cans with wire leads.

PACKAGE DIMENSIONS:

	W	L	H
1 =	1 5/16" Diam.		× 1 9/16"
2 =	1 3/16"	× 1 3/16"	× 1 5/8"
3 =	1 1/8" Diam.		× 1 1/16"
4 =	1 1/2"	× 1 3/4"	2 1/2" w/solder terminals
5 =	1 5/8" Diam.		× 1 3/4"
6 =	1 1/8" Diam.		× 15/16"

NICKEL CORE OUTPUT TRANSFORMERS[6]

Model	Construction	Nominal Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20 Hz Max Output Level[7] (dBu)	across (n) windings	600 Ω Load Loss (dB)	DC Resistance per Winding	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width −3 dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Package	PRICES 1-19	100-249	1000
* JE-11-BMCF	Bifilar 80% nickel	600-600	1:1	+26	1	−1.1	40 Ω	0.002/0.002	−0.02/−0.00	>10MHz	−0.0	<1[9]	7	65.36	43.66	30.12
* JE-11-DMCF	Bifilar 80% nickel	600-600	1:1	+21	1	−1.0	38 Ω	0.004/0.002	−0.02/−0.00	>10MHz	−0.0	<1[9]	8	48.74	32.56	22.46
JE-123-BLCF	Quadfilar	600-600 / 150-600	1:1 / 1:2	+32	2	−1.1	20 Ω	0.041/0.003	−0.02/−0.01	>450 / 170	−1.9 / −4.0	<1[8]	7	64.57	37.71	26.02
* JE-11SS-DLCF	Bifilar split/split	600-600 / 150-600	1:1 / 1:2	+27	2	−1.0	19 Ω	0.065/0.003	−0.02/−0.01	>10MHz / 245	−0.0 / −2.5	<1[8]	8	46.38	30.98	21.37
* JE-11-ELCF	Bifilar	600-600	1:1	+23.5	1	−1.1	40 Ω	0.088/0.003	−0.03/−0.00	>10MHz	−0.0	<1[9]	9	30.21	20.18	13.93
* JE-11-FLCF	Bifilar	600-600	1:1	+20.4	1	−1.6	58 Ω	0.114/0.003	−0.03/−0.00	>10MHz	−0.0	<1[9]	10	23.66	15.81	10.91
JE-112-LCF	Quadfilar	600-600 / 150-600	1:1 / 1:2	+20.4	2	−1.6	29 Ω	0.114/0.003	−0.03/−0.01	>450 / 205	−1.2 / −3.2	<1[8]	10	26.68	17.82	13.08
JE-123-ALCF	Quadfilar	66.7-600	1:3	+26.5	3	−1.3	8 Ω	0.125/0.003	−0.04/+0.06	190	−4.6	<6[8]	8	44.09	29.45	20.32
JE-11S-LCF	Bifilar w/split pri.	600-600 / 150-600	1:1 / 1:2	+30	1 (sec)	−1.7	63 Ω	0.058/0.002	−0.02/+0.01 / −0.02/−0.05	>10MHz / 155	+1.1 / −4.1	<1[8]	8	44.09	29.45	20.32

6. Multifilar construction has no faraday shield; cannot be used as input transformer. All specifications are for 0 Ω source, 600 Ω load.
7. Max output level = 1% THD; dBu = dBv ref. 0.775 V
8. Source amplifier −3 dB @ 100 kHz
9. Source amplifier −3 dB @ 200 kHz
10. Output transformers are horizontal channel frame type with wire leads, vertical channel frames available. PC types available.

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PACKAGE DIMENSIONS:

	W	L	H	Mounting Centers
7 =	1 1/2"	× 2 5/16"	× 1 15/16"	2 13/16"
8 =	1 5/16"	× 1 15/16"	× 1 5/8"	2 3/8"
9 =	1 1/8"	× 1 11/16"	× 1 3/8"	2"
10 =	1 1/16"	× 1 7/16"	× 1 3/16"	1 3/4"

Prices shown are effective 8/1/85 and are subject to change without notice. Packing, shipping, and applicable sales taxes additional.

These charts include the most popular types which are usually available from stock. Many other types are available from stock or custom designs for OEM orders of 100 pieces or more can be made to order. Certified computer testing is available for OEM orders. Call or write for applications assistance and/or detailed data sheets on individual models.

jensen transformers
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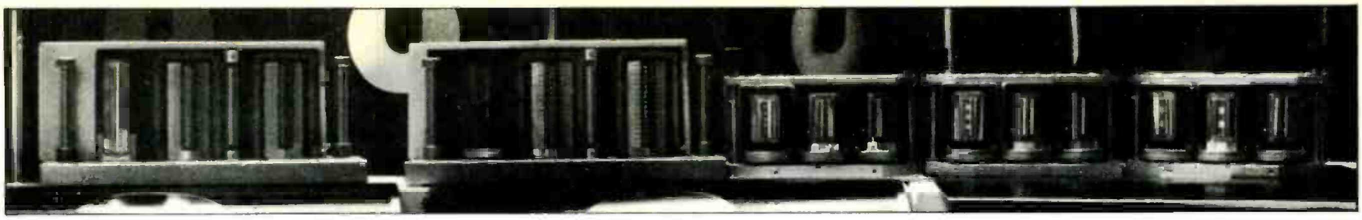


Tape and Track Formats (*L-to-R*): 24- and 16-track/two-inch tape, six-, four- and three-track 35mm mag film

QUALITY ADVANTAGES OF 35MM MAGNETIC FILM

track will be synchronized with the video picture! Best of all you will have an Edit Decision List, from which — using the newly created location sync two-track tape — the location sound can be laid onto the 24-track tape while constantly checking it with picture for sync.

The same procedure is followed for the voice-over announcer. Of course, when laying this audio onto the 24-track, you will not be checking for sync but for timing and dialog placement. If the timing is off, you may have to electronically edit the pauses to ensure correct timing. Next, follow the same procedure as above on each of the five sound-effects tracks.

Now you are ready to mix these 11 tracks. The music is secondgeneration, location sync is third-generation, as is voice-over announcer and sound effects. (As an aside one can record the announcer directly to the 24-track tape, while viewing picture; this method is a much better way to go, but quite a bit more costly.)

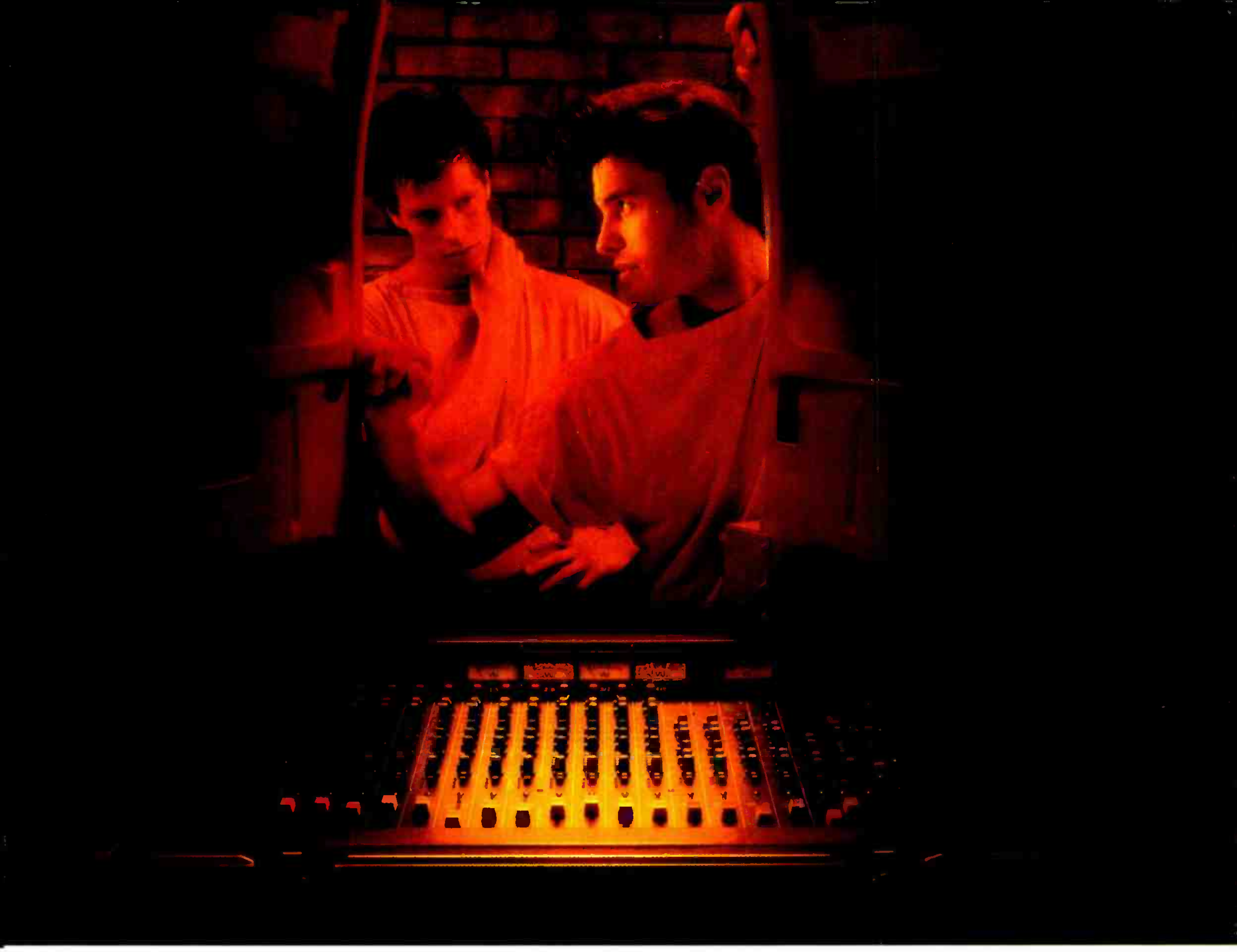
All of the above sound elements are mixed to a two- or four-track tape with audio on one or two tracks — depending on whether or not the program is monaural or stereo — with timecode on another track. In the case of four-track masters, it is desirable to also record 59.94 Hz derived from your in-house sync generator onto another track. This four-track audio master will be used to transfer (or layback) the final mono or stereo audio tracks to the master videotape.

You are now four generations down on the music, and five generations down on everything else. Some people like to master directly from the multitrack to videotape, a technique that can be okay if the resultant videotape is the one that's actually going to air without too many interim plays. (Remember that noise on one-inch C-Format videotape is comparable, at best, to a chrome audio cassette. I do not know anybody that would want to master their audio onto an audio cassette!) If there are any additional generations created on videotape, or if excessive plays cause dropouts on the audio track, the master audio can always be relayed.

Noise Build-up with Audio Tape

Let us discuss why it is worthwhile to conserve generations relative to analog audio. The first and most important consideration is noise, which is increased by every tape generation. In the above examples, the music would have picked up about 3.8 dB of noise prior to its transfer to videotape, while all other tracks individually would have picked up a total of 10 dB of noise prior to their transfer to videotape. Other considerations are increases in distortion, tape saturation, and a "dulling" of the general sound. These factors are influenced to varying degrees relative to the pro-

. . . continued on page 30 —



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Packing up for a gig. It's an important moment of truth for every musician. Within the safe confines of your studio, you've worked your music into shape. Polished it. Perfected it. Put it on tape. Now it's time to take it on the road. You're excited, keyed up. How will your music hold up under the hot lights and cold scrutiny of the outside world?

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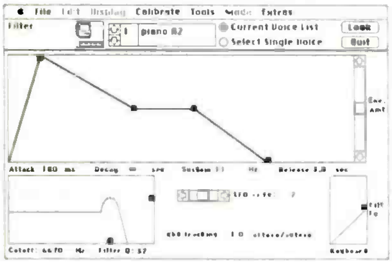

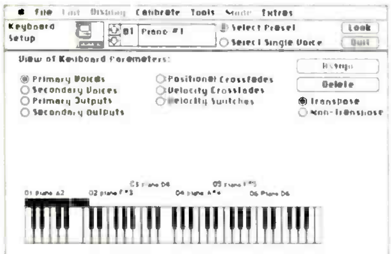

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At about one-third to one-tenth the price of comparable systems, the Sound Designer/Emulator II combination represents the best value in computer music systems. However, the system offers another advantage more important than money.

Most computer music systems are hardly user-friendly. User-indifferent is a better description: strange commands to memorize, confusing terminology and painfully slow operation have thwarted many musician's attempts to use this advanced technology.

You don't need unlimited patience and a Ph.D. to learn Sound Designer. Sound Designer takes full advantage of the Macintosh's simplicity—program functions are *visually* represented by icons (pictures). No cryptic commands to memorize!

The Emulator II/Sound Designer system is a valuable tool, whether you're scoring a film, adding sound effects to a video production or creating the sounds for your next hit. You'll find the system quite stimulating—to both your creativity and your ears!

Want to see the system in action? Send $29 to Digidesign (address below) for a 30 minute demonstration video (specify Beta or VHS). Like to know more about the Emulator II? Send $2.00 to E-mu Systems for a color brochure and a *very* impressive demo record.

Sound Designer requires a 512K Macintosh, 2 disk drives or an internal hard disk (recommended), and an active imagination. The Emulator II requires fingers.

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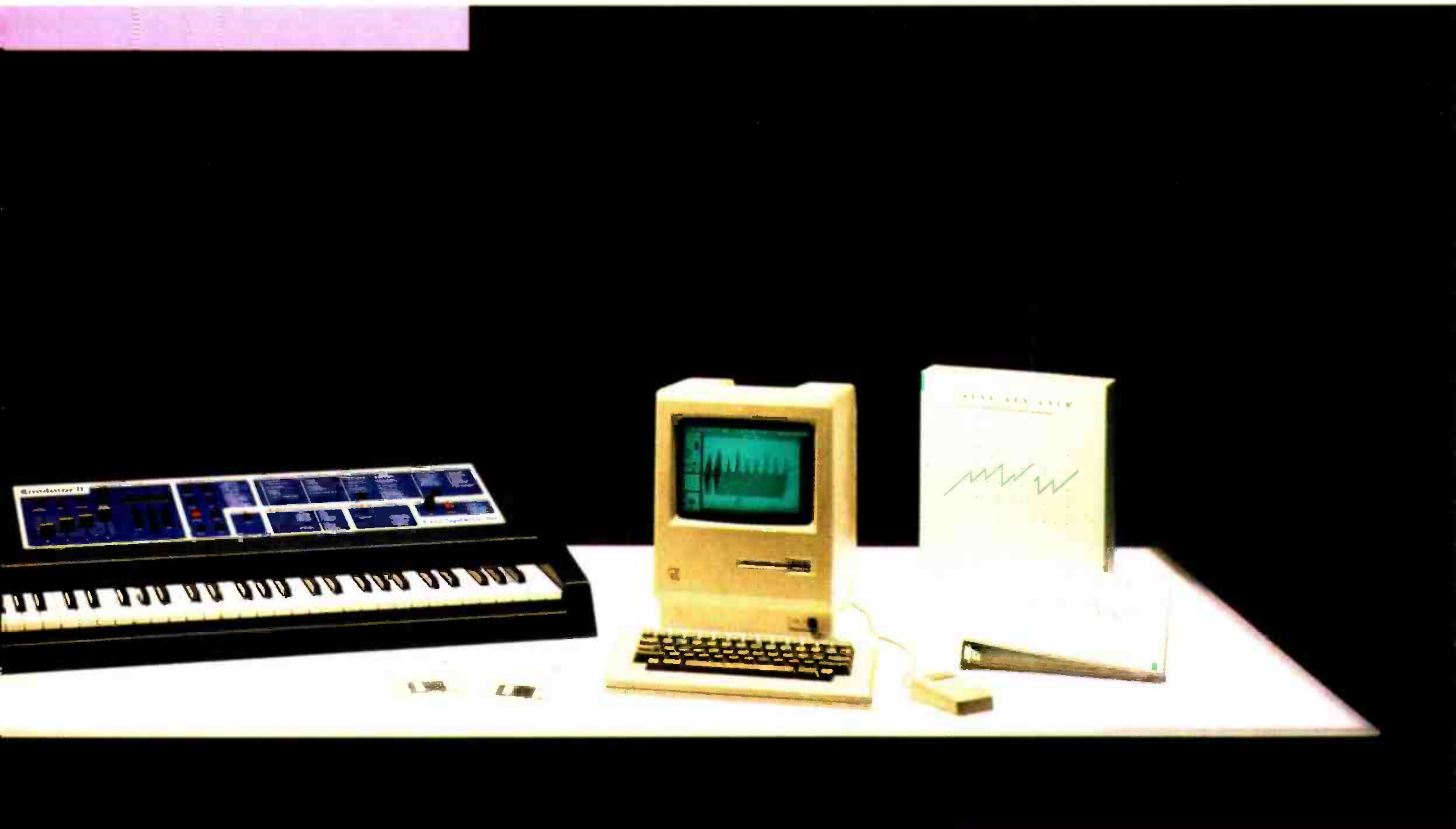

QUALITY ADVANTAGES OF 35MM MAGNETIC FILM

— *continued from page 26 . . .*

gram material, the competence of the recording and mixing engineers and, of course, the condition of the equipment being used. (Some engineers can capture more "punch" in their fourth-generation copies than others in their original; because of the many variables involved, I will not try to evaluate such factors.)

Another problem is wow and flutter. With current state-of-the-art equipment, one has to go many generations before exceeding the wow and flutter induced in the video transfer. Although technically this is a problem, in the real world of well maintained equipment, two generations more or less should not cause serious trouble.

Summarizing, the main concern of generation loss is one of additive noise.

Noise Build-up with Mag Film

Now let us go through the same process outlined above using film-mixing techniques. The original 24-track music tapes would be mixed to four-track 35mm magnetic film, the film recorder receiving its sync from the same timecode already recorded on the original 24-track for automated mixing purposes and picture sync during music recording. The four-track music mix would then be physically cut into proper synchronization with the picture on a standard Moviola. The location sync sound would be transferred to mag, utilizing a playback machine capable of resolving to the neopilot signal, and then physically edited to conform to picture. Similarly, the sound effects and voice-over announcer audio would all be transferred to 35mm magnetic film, physically edited and placed in the correct sprocket registration relative to picture. Although electronic and physical (cut-and-splice) editing take about the same amount of time, once the "splice" is made on film it can be physically laid in at high speed, whereas the electronic audio track on non-sprocketed tape must be laid in real time.

So, by the film method we are two generations down on music, sync, voice-over and sound effects; now comes the mix. We can master to a four- or a two-track tape machine, as well as a monaural or two-track film recorder. (Once again, we should try to avoid mastering to a C-Format VTR unless this master was going directly to air.) Once this audio master has be transferred to videotape, we are now four generations down on music, sync, voice-over and effects —the same result relative to music as with video sweetening, and an improvement of one generation over video sweetening relative to everything else. However, there is another factor to be considered.

The track width of a monaural film head measures 200 mils, and a four-track film head 150 mils per channel. In contrast, a four-track/half-inch tape head measures 70 mils per channel, while a two-track/quarter-inch head measures 80 mils per channel. As we all know, bigger track widths provide us a quieter and better signal, a fact that was proven when we moved to two-track/half-inch mastering. So, taking into consideration the above argument, the following will be true: assuming that a mono 35mm track is the quietest format, a four-track film head will be 1.25 dB noisier, a two-track/quarter-inch audio head will be 4 dB noisier, a four-track/half-inch audio head will be 4.5 dB noisier, and a 24-track head — which measures 37 mils per channel — will be 7.3 dB noisier.

All of which means that our original music transfer will be 2 dB quieter on four-track magnetic film than on a 24-track/two-inch tape. All our other tracks will have saved one generation by going via the film method, which is worth about 1 dB per track. When originally transferred to monaural 35mm magnetic film and mixed, these latter tracks will be 5 dB quieter than if the video-sweetening method has been used.

Using the video-sweetening mixing technique, the final mixed track prior to transfer to videotape will have an unweighted signal-to-noise ratio of about 52 dB. Using the standard film-mixing method, the comparable SNR will be 57 dB. In the real world of mixing, of course, some of these tracks will be brought up in level, thereby increasing noise; other tracks will be brought down, thereby decreasing the noise. Also, some tracks will be equalized, which might either increase or decrease noise. Other tracks will be compressed, thereby increasing noise, while "silent" areas might be gated to decrease the noise. Other factors affecting noise levels include variations in the specifications of record amplifiers, the tape's running speed, and the actual recording equalization curve used.

The unweighted signal-to-noise ratio of a C-Format VTR is about 56 dB. Let us assume that the "zero" operating level for such a video recorder is 8 dB down from the 3%-THD (Total Harmonic Distortion) level. Throughout the mixing process, we have been operating with our zero level at about 12 dB down from the 3%-THD point. Therefore, the noise floor below zero operating level for the video-sweetening master will be about -40 dB; the same parameter for the film-mixed master will be -45 dB. When transferred to videotape, the film master will be only 3 dB above the C-Format noise floor, while the video-sweetening master will be 8 dB above the same noise floor. Percentage-wise, the video-sweetening master is 78% noisier than the film-mixing master. When your local television station adds its own form of compression, this additional 5 dB of noise will create problems.

As you can see, it would be possible to go another generation on film and still be quieter than if the same mixing process had taken place on the regular audio tape used during video sweetening. Of course, in the hands of a good engineer, the video-sweetening technique represents a valuable and useful production tool. And, when we reach the stage at which only digital recorders are used, it will truly be a competitive tool. Eventually, *all* audio information will be stored in digital form on magnetic or optical disk and/or solid-state devices. When this happens, all of the above arguments regarding the noise and sonic advantages of mag film over audio tape will be meaningless. [For a full discussion of the impact that digital storage and random-access editing systems will have on film post production, see Larry Blake's article elsewhere in this issue — *Editor.*]

In the meantime, however, we should all make a gigantic effort to keep ourselves constantly aware of the effect that recording technology can have on the pristine quality of sound. If we allow our ears to accept less than the best, we are not properly serving our audio community. For those of us that plan to remain in this business long range, the stewardship of our aural credibility is of paramount importance. Whenever a new technology is added, or a new system is devised, or a new machine is tested, ask yourself . . . "What does this do to the sound?" ■■■

ETING REPORT

ıtions for Better Quality Cassettes"
. Seminar Organized by ElectroSound, San Francisco, August 1985

by Mel Lambert and Rhonda Kohler

For too long now, many audio professionals would argue, the prerecorded cassette has been looked upon as the "poor relation" of the music software market. There is no getting away from the fact that the Compact Cassette was introduced to the public several decades after vinyl releases had become firmly established in the majority of people's minds as the most convenient (and familiar) way of enjoying prerecorded music. (And, it must be conceded, the first cassette decks and tapes were of a very much lower quality than we are used to in the Eighties; such unfavorable, early experience might in part explain the public's reluctance to add yet another piece of hi-fi hardware.) In addition, the exceptional success of the Compact Disc has somewhat stolen the limelight from advances being made in both the analog disk and cassette manufacturing industries — one example of which, direct-metal disk mastering, is described in a feature article beginning on page 58 of this issue.

On the positive side, however, prerecorded cassettes now outsell vinyl releases, the 50% market share for Compact Cassettes having being passed just over two years ago. Also, the advent of "Walkman-type" portable cassette players, and radically improved in-car systems, has added to the growing impact of pre-recorded cassettes as a primary form of music software for a growing number of consumers.

But what of the technical advances being made in the design and operation of high-speed duplication equipment which, there is no denying, has radically increased the quality of prerecorded cassettes? Despite the fact that the majority of *R-e/p* readers are involved with the production of master tapes for the record- and cassette-buying public (plus corporate and audio-video clients), an overview of the cassette duplication processes — in particular the format of master tapes to be delivered to duplication facilities, improvements in tape formulations, and so on —can prove useful if the maximum quality potential is to be realized from the pre-recorded cassette medium.

Against this background, the panel discussions that took place at the recent "Applications for Better Quality Cassettes" seminar proved extremely interesting. While there is insufficient space here to fully report on each of the sessions, several themes and future developments came to light during the three-day seminar, held in San Franciso during mid-August, and which attracted almost 300 participants.

Organized by ElectroSound, and co-sponsored by Agfa-Gevaert, American Mulitimedia, Ampex, Audiomatic, BASF, Capitol Magnetics, Dolby Labs, ICM, IPS, JRF, Lenco, MRL, Saki, Sprague, Studer-Revox, and others, the seminar began with an overview session, entitled "The Art in Cassette Masters." Panel members comprised Gene Wooley, chief engineer and director of engineering, MCA Whitney Recording, Ed Outwater, director of quality assurance, Warner Bros. Records, Marv Bornstein, VP of quality control, A&M Records, and Steve Miller, former director of production, engineering and QC at Windham Hill Records, under the moderation of Sandy Richman, administrator of the XDR Program at Capitol Records.

To extract the very best quality from

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the high-speed duplication process, Richman offered, a facility needs to start with a high-quality master tape, and he placed particular emphasis on encouraging record labels — and hence the session producers — to provide digitally-encoded masters. Wooley conceded that certain producers and artists may not care for the sound of digital; in which case, he says, there should be a recommendation that a label offers both analog and digital masters to cover all eventualities. Miller also pointed out that, while he is a great proponent of the sonic advantages offered by digital recording, what needs to be stressed is the operational advantages to a cassette manufacturing facility when working with digital masters. During the preparation of loop-bin masters, the saving in wear and tear — and potential high-frequency losses — of an analog master can be reduced significantly if the master is a digitally-encoded videocassette.

A quick poll of the panel members produced the following rundown of master-tape formats received by their companies:

- Without exception, Capitol will make a digital copy of the original two-track master (either analog or digital) for use in preparing the loop-bin master.
- At MCA Whitney, 50% of country releases are mastered from JVC VP-900 digital format, and the remainder from analog masters; on pop-music titles around 35% of titles are received as a digital master. (Gene Wooley did point out, however, that in the future the company's duplicating facility plans to copy all tapes to digital prior to running-master preparation.)
- A&M's situation is similar to that of MCA, while both Warner Bros. and Windham Hill work exclusively with Sony PCM-1610 format masters.

Given the multitude of different digital formats, it was proposed by the panel that duplicating plants be prepared to handle PCM-1610, VP-900 and other digital-format masters.

Should duplication houses re-equalize or add dynamics processing to master tapes, in an attempt to produce a better quality product, the panel was asked? "No," Outwater stressed, "because that opens up a whole can of worms. Instead of re-equalizing, we would go back to the record company and get a better master tape; we prefer to transfer *flat*." Wooley agreed: "We need to ensure a flat transfer of a *good* master — there are many slave transports involved [during the duplication process], and we cannot adjust them all to correct for deficiencies in the master tape." It was also agreed that during the duplication of older archive and re-issued material, master tapes should be left unaltered, unless the artist "agrees to 'update' the sound," Wooley offered.

A member of the audience asked how each of the panel members ensured that the quality of test cassettes produced by a duplication facility matched that of production runs? Bornstein explained that A&M carries out batch checking of product delivered to record stores, while Outwater said that Warner Bros. effects the same kind of quality checks with production cassettes, to make sure that consumer product sounds the same as the previously approved test cassettes. According to Richman, Capitol uses the burst of special test tones recorded on the loop-bin master to check the quality of XDR pancakes, and ensure the integrity of the cassette transfer process; in addition, the company checks between two and three precent of the final product after the loading of pancakes into shells. "We prefer to check during the duplication process," he explained, "so that we can catch the problem *before* the tape reaches the cassette shells."

Next came a presentation entitled "Masters versus Cassettes — How do They Compare," by Dennis Staats, tape-duplication manager at Dolby Laboratories, during which he outlined some of the problems faced during the high-speed recording of music onto cassette tape. The 0.125-inch width of cassette tape, and resultant narrow track format, made it difficult to record short wavelengths, he pointed out, a problem that can be compounded by tape moving away from the heads at high duplicating speeds. Because of these and other factors, he offered, many duplication facilities felt that tape noise and distortion produced by the running or loop-bin master need not be considered, simply because all of the technical limitations of the duplication process can be attributed to the cassette medium.

This was untrue, he said, and proceeded to compare MOL (maximum output level), distortion and signal-to-noise figures for cassette formulations and the half-inch tape used typically for preparing loop-bin masters. At a 200 nW/m fluxivity level, a "typical" ferric-oxide cassette tape offers a noise performance of -52 dB, a 3% total harmonic distortion level of +5 dB, and a high-frequency MOL of -1 dB; the comparable figures for a half-inch, 7.5-ips bin master (referenced to 320 nW/m fluxivity) are -61 dB, +9 dB and +2 dB. Given the 6-dB increase in noise during the cassette recording stage, Staats considered that a 10-dB noise advantage should be offered by the half-inch master over the casssette formulation — which was barely the case with a 7.5-ips master.

Moving on to consider the figures for a *3.75-ips* half-inch master — the more popular format for duplicating facilities — the SNR, 3% THD and HF MOL drop to -58 dB, +8 dB and +2 dB, respectively. As could readily be appreciated, the noise and THD performance of such a bin master are pretty close to those of cassette tape. Consequently, Staats explained, Dolby had come to the conclusion that its HX (Headroom eXtension) process would be of particular use during the preparation of a 3.75-ips loop-bin master, to reduce noise and distortion.

He also added that the transfer of audio program from the two-track digital or analog master to the 3.75-ips running master was one that needs to be made with care, particularly for electronic and synthesizer-based material, for which the average-to-peak ratio may be in the region of 15 dB on digital masters. One way around the problem, he suggested, would be to specify that meters with fast rise times and peak-reading capability be used during the preparation of loop-bin masters.

The second panel session of the day, entitled "The Technolgoy of Cassette Masters," was moderated by Jim Roe, director of engineering at WEA Manufacturing, and included panelists Dennis Drake, studio manager/chief engineer of the Polygram Tape Facility, Pat Weber, technical director of MCA Whitney Recording, Paul West, director of national quality control and EMI-America studio operations, Kent Smithiger, national quality control manager, ElectroSound Group, Fred Layn, northwestern regional manager, Studer Revox of America, and Larry Schnapf, director of recording operations and facilities for RCA Records. Subtitled "A discussion about how to produce the best possible running master," the session addressed the question of whether a facility should utilize 7.5- or 3.75-ips loop-bin masters. According to West, the higher speed makes it "difficult to handle long lengths of tape," and leads to tape storage problems at a 240-ips loop-bin speed. Also, he offered that a master may not last as long at higher duplication speeds. "To me," he concluded, "[Dolby] HX holds the key to producing masters at 3.75 ips."

"The quality of 3.75-ips masters is sufficient for most material," opined Smithiger. "Seven-and-a-half ips may only be needed for classical releases, or for material with a wide dynamic range."

Asked to outline their company's policy towards the format and production techniques for loop-bin masters, the panelists summarized these as follows:

- Polygram utilizes PCM-1610-format digital masters and a system of tone burst (similar to the Capitol XDR technique) to provide an objective quality assessment at any stage during cassette duplication.
- MCA uses test tones and calibration tapes to verify the quality of a one-inch, 3.75-ips running master, and sets recording levels as high as possible before tape saturation.
- EMI uses a 1610-format master to produce a one-inch, 3.75-ips running master without additional equalization, limiting, etc. Also, test and alignment-level tones are added to allow a mastering facility to check azimuth, etc.
- ElectroSound uses fast-reading peak meters during the production of running masters.
- RCA *insists* that an LP accompany the master tape, so that the duplicating facility can check for EQ differences, etc. If there are any differences, the tape

will be rejected. Currently, the ratio of analog to digital master tapes used to produce the 3.75-ips, one-inch running master at RCA duplication facilities is 4:1.

• WEA uses a half-inch, 3.75-ips running master recorded with Dolby HX Pro circuitry.

Addressing the question of the projected lifetime of a loop-bin master, West related that EMI, through the use of test tones, has established 3,400 passes at the maximum life for the running master. In quantitative terms, he looks for a 1 dB dropoff at 16 kHz — assuming, of course, that there are no mechanical problems with the tape, including shedding and physical damage. According to Smithiger, ElectroSound manages to obtain around 2,000 passes from a master, "although the tape is usually destroyed before that time! Usually it fails because of tape stiffness, and operator error." The problem, West confided, was "finding a tape that works well at 3.75 ips, but that will also hold up at the 120-ips duplicating speed."

If the production of a running master can cause so many problems, a member of the audience asked, should it be mastered at a central location, rather that at the duplication plant? "Yes," Weber offered, "because it helps quality control, and also enables an artist to be present during the [transfer] process."

EMI/Capitol's mastering studio is located in Hollywood, West advised, "so that technical quality control is tightly monitored, and we have ready access to the artists." Drake conceded that a central facility made sense, but that "the acoustics and accuracy of the listening room was also *very* important — they should not 'color' the sound." The majority of the panelists were in general agreement with West and Drake's comments.

The second presentation of the day, "Mastering for the Media," comprised a discussion of the types of tape formulations that satisfy the demands of cassette duplication. The speakers were Dr. Andreas Merkel, technical training and customer support, Agfa-Gevaert, Warren Simmons, manager, professional audio products, Ampex Magnetic Tape Division, and Walter Derendorf, applications engineer, audio products, BASF.

Merkel began his presentation by comparing the 3% MOL and dynamic range of the original master tape, safety copy, 3.75-ips running master, running-master copy, and the final duplicated cassettes, from which he concluded that cassette-tape noise is the limiting factor in the transfer process, and that noise on the running master is "immaterial." (In fairness, he also contemplated the use of one-inch rather than half-inch tape for the 3.75-ips running master. While, because of increased track width, azimuth loss and phase shift were more critical for one-inch masters, head-azimuth adjustment can be set more easily with the larger width tape.)

To improve the signal-to-noise ratio range of a 3.75-ips running master, Merkel proposed that a 3180 plus 140 microsecond record/replay equalization curve be adopted — rather that the current 3180+90 — a change that would enhance the SNR figures by 3.5 dB.

Derendorf described the thinking behind the development of BASF's new 902 chrome tape for running masters which, at 3.75 ips, offers an MOL of 67.5 dB at 315 Hz, and a high-frequency dynamic range of 59.5 dB at 12.5 kHz. Since the new chrome formulation requires an additional 3 dB of bias — and also produces an additional 10 dB of HF headroom — Derendorf foresaw problems in providing sufficient bias

STOP PRESS: MARSHALL ELECTRONIC RE-APPOINTED U.S. DISTRIBUTOR FOR QUANTEC ROOM SIMULATOR

The December 1984 issue of *R-e/p* contained a review by Bob Hodas of the Quantec Room Simulator as modified and enhanced by the company's then U.S. distributor, Marshall Electronic. Shortly after the review was published, however, the Marshall-modified QRS, which featured hand-selected analog components, hand-trimmed DAC (digital-to-analog convertor) offsets, and several other proprietary enhancements, became temporarily unavailable.

As this issue was being prepared for the printers, Quantec and Marshall announced that Europa Technology is no longer serving as the QRS distributor for the U.S. As a result, the original, customized units again will be available exclusively from Marshall Electronic, Box 438, Brooklandville, MA 21022. (301) 484-2220 — *Editor*.

and record-level gain for conventional transports used to prepare the 3.75-ips running master. He did explain, however, that the BASF has achieved up to 4,000 passes from the new formulation.

Simmons began his presentation with an overview of the parameters of an "ideal" loop-bin master tape: it never sheds, never tangles, presents no change in electrical performance during production runs, lasts for 100,000 passes, and is low in cost. Coming back to reality, however, he summarized the cost versus quality trade-offs as follows:

- Low tangle factor, which necessitates the use of a high film-base thickness and a high conductivity backcoating;
- Low dropout activity, which can be improved by surface cleaning and correct slitting during manufacture, and the use of a tough binder system.
- Good short-wavelength performance, which requires the use of high-performance oxide coatings and/or a high level of oxide-surface gloss; and
- High signal-to-noise ratio and low electrical distortion, which can be achieved by using high quality oxides.

test

All of these parameters, he concluded, were available from Ampex 456 Grand Master. An added bonus, he related, was the fact that "limited experience [has shown] 456 will perform satisfactorily at 480-ips duplication speeds, because of its tough binder system."

During the seminar Richard Clark of American Multimedia invited attendees to a demonstration of the company's new bin-loop system, which allows 64:1 duplication while running a 7.5-ips master. According to Bob Farrow, chief engineer, the weakest point in cassette duplication is tension control in the high-speed playback of a running master. American Multimedia has modified an ElectroSound Model 8000 transport, including the addition of two vacuum servo systems to improve tape tension and handling. A vacuum transducer monitors the tape, and automatically adjusts the tension while keeping the tape correctly positioned. In order to show that the process accurately duplicates original material, a 7.5-ips master was made from a Compact Disc. After running in a 480-ips loop-bin feeding a slave, the resultant cassette copy was compared to the original. Several members of the audience were astonished at the remarkable similarity between the original and duplicated versions.

Highlights of the second day's proceedings included a presentation on tape performance by Kaus Goetz, manager of applications for BASF. Entitled "Tape Specifications — What are the Choices?" featured moderator Joe Kempler, director of technical marketing for Capitol Magnetic Products, and panel members Dr. Andreas Merkel of Agfa-Gevaert, Walter Derendorf of BASF, Helge (Kris) Kristensen, senior staff engineer, R&D Test and Evaluation, Ampex Magnetic Tape Division, John Hudson, technical services manager, Dupont, David Mills, director of marketing and sales, Pfizer, Donald Winquist, national sales manager, Recording Products, Hercules, Frank Diaz, technical director, Columbia Magnetics.

A panel discussion entitled "C-Os: What are the Choices?" touched on some of the questions confronting the C-O tests being conducted by the International Tape/Disc Association. Moderator Sam Burger, senior VP for CBS Records, and Henry Brief of the ITA explained that the tests being carried out in cooperation with 10 manufacturers and six duplicators are being made in an attempt to to reduce the angle error in shells, and thereby improve the quality of the pre-recorded cassette. The manufacturers involved in the ITA testing are Athenia Industries, Data Packaging Corp., Filam National Plastics, ICM, IPS, Lenco, Magnetic Media, Rainbow Manufacturing, Shape and Trans Am Industries, while the duplicators are Capitol Records, Cassette Productions, MCA Manufacturing, CBS Records, RCA Records, and WEA Manufacturing.

The final session of day two was a panel discussion on consumer expectations, and what can be done to improve the overall quality of the finished pro-

... continued on page 224 —

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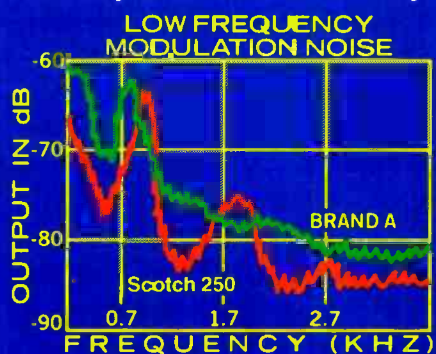

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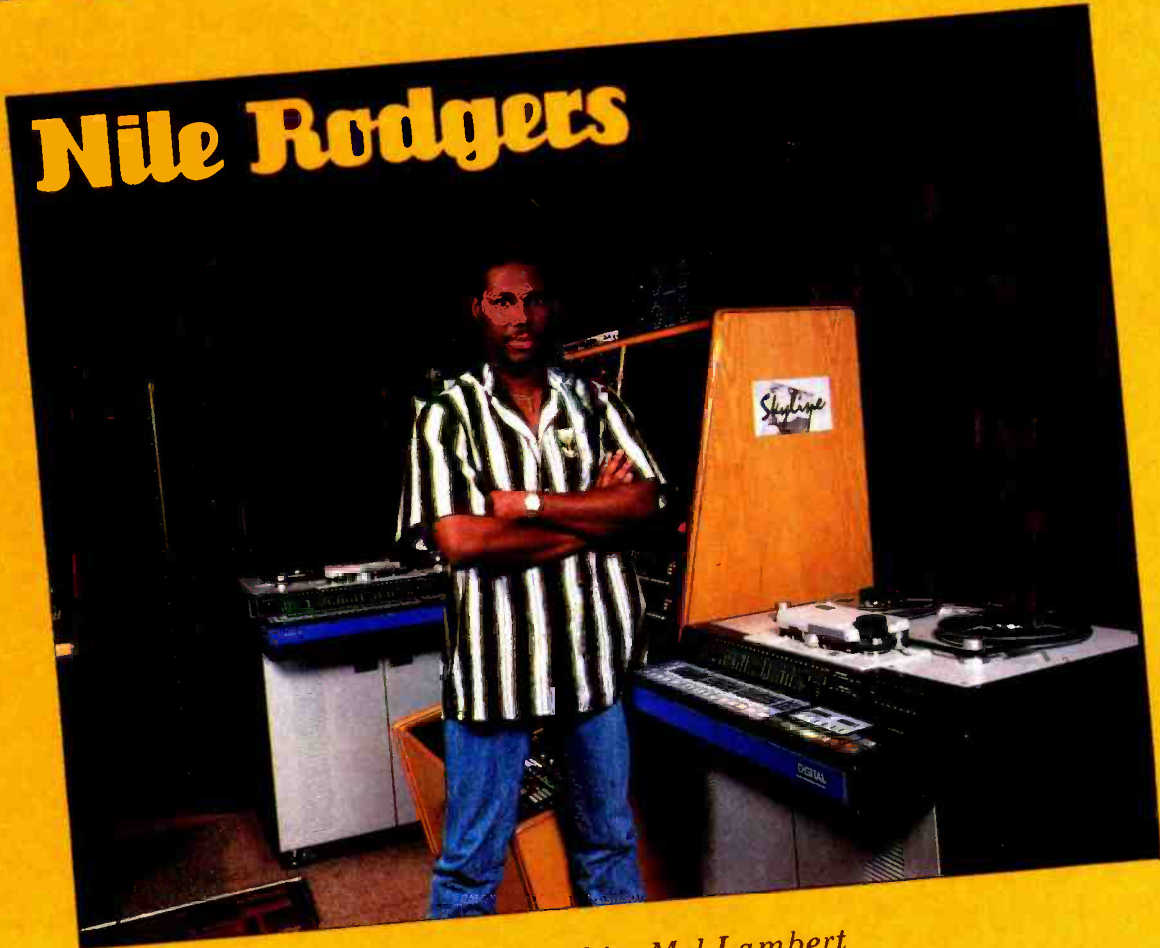
Nile Rodgers



Photography by Jim Curry/Jim Curry Associates

Interviewed by Mel Lambert

From his innovative sessions with Chic, Nile Rodgers has moved on to become one of the most prolific and successful producer/musicians around today. Over the last several years he has contributed his unique production style — infusing solid rhythm elements with strong lead and solo identities — to sessions with Sister Sledge (*When the Boys Meet the Girls*), Madonna (*Like A Virgin*), David Bowie (*Let's Dance*), Jeff Beck (several tracks from *Flash*), Mick Jagger (three tracks from *She's The Boss*), and his recent solo album, *B-Movie Matinee*. During that period, he has also found time to produce singles by Diana Ross ("Upside Down"), Duran Duran ("Wild Boys"), as well as several film soundtracks, including music for the upcoming movie *White Nights*.

R-e/p caught up with the busy producer during remix dates for the new Thompson Twins album, *Here's to Future Days*, working with first-call session engineer James Farber at Skyline Studios, Rodgers' new base of operations. Following the remix session, he was scheduled to begin a new album with Shenna Easton, followed by possible sessions with Philip Bailey and Duran Duran.

R-e/p (Mel Lambert): You have an enviable reputation as a session musician, as well as for your innovative work with Bernard Edwards and Chic. How did you first become involved with th production side of making records?

Nile Rodgers: Actually, it was out of necessity. When Bernard and I first put Chic together, we were concerned with being respected as players primarily, especially because it was right at the high point of the "Fusion Era" — so of course our role models were musicians. The problem was that any time we would go into the studio to make a record, because of the way we played, the producers we were working with would try and bring out that Fusion Element. Whenever we went into the studio, invariably we would come off sounding like "young John McLaughlins," or "young Chick Coreas." [Laughter] I was studying with a number of very good jazz and classical guitar players, so we were concerned with musicianship.

Unfortunately, we wanted to make a *commercial* pop record. As soon as we would start practicing, we would over-interpret our own songs, but we

Nile Rodgers

were just doing it for fun! The producers would pick up on that free-form playing, and take us in that "Fusion Direction," because fusion was happening, and they thought that was what we wanted to do. So we made a number of records and demos that were over-played and over-produced — Man, just a flurry of notes! It was just crazy! When we listen to it now we like it, because it was good stuff, but it was far from commercial!

Finally, Bernard went into the studio to do a disco song for a television show. They needed a B-side, so Bernard started jamming on the bass line that eventually became "Dance, Dance, Dance." They liked what he did so much that they said, "Man, this guy has really got something; we just ask him to play a groove, and he came up with something that we've never heard before."

R-e/p (Mel Lambert): Bernard set the whole character of the song with just the bass line?

Nile Rodgers: Right. As it came down, they thought that if Bernard had that much talent in a primal sort of way, I might be able to come up with something a little bit more "intellectual." They called me up and said, "Bernard did this on bass; what would you do on guitar?" I thought: "Now hang on wait a minute. They're gonna get Bernard and me to write this whole song, and produce it, but meanwhile we won't get any of the credit!" So I called Bernard, and he came over to my house. "Let me hear what you worked on," I said, and he played me the tape. "Why don't we do it together," I asked? But he and I were already in a band; we had Chic, but we didn't call it that back then. So we worked on the song and wrote "Dance, Dance, Dance." It was the first song that he and I had written together, because prior to that, I had written all the songs — they were much more jazz-oriented tunes, with thousands of chord changes and things like that. So the very first thing we wrote together became a monster hit.

R-e/p: So the production role was more a matter of being responsible for the finished sound? How to set up the instrumentation; the playing; how to record it; and how to blend the elements together in the mix.

NR: Yes. We sort of lucked out on those early Chic singles, and kept doing more and more. We started working in a friend of mine's recording studio; actually he was just the maintenance engineer. When the studio closed at two o'clock in the morning, we would sneak in after hours. We recorded the essence of our first album at that studio.

After we did "Dance, Dance, Dance" — with a full production — the record company said, "Well, what else do you have?" And boy, did we have a lot of other material! We said, "Glad you asked that question; listen to this." And we had hit record after hit record.

R-e/p: You have been reported as saying that, in both a playing and a production sense, you are "comfortable with any musical situation." Has that universality helped you to move into producing other, diverse styles beyond what what was your "personal music" with Chic? There are obvious and distinct differences in the musical directions and styles of, say, Sheena Easton, Mick Jagger, Jeff Beck and Madonna.

NR: Absolutely. For example, imagine — and this is *purely* hypothetical — if Mick Jagger had said to me: "I'd like to do a country and western track." What am I supposed to say: "Hey Mick, I can't hear it?" If I'm producing his album, I'd have to do it, and try to make it sound the best it could possibly be. You *have* to have some knowledge of all musical styles, even if it's only as a fan, because then you'll know how to achieve the end result. I have a very varied musical background: I played folk music for years; I played classical music for years; and I played jazz for years — and still do. I just never made records in those styles.

R-e/p: So you feel that it is important for a producer to have that overview during a session?

NR: No, it's not necessarily important for everyone; it's just important for *me.*

Having a diverse musical background has definitely helped me. That's not to say that producers who only produce Heavy Metal, for instance, are bad — they have that down. For me, unfortunately or fortunately, depending on how you look at it, because I'm such a "workaholic" I'd be very, very frustrated just working in one musical genre. A lot of the work I do never sees the light of day, but that's what fuels my fire — it keeps me interested in this whole musical lifestyle. I love this job; it's a *great* way to make a living!

R-e/p: I notice from the album credits of your recent productions that you also tend to play on a lot of the sessions. Do you try and keep your chops alive through the musical contributions you make in the studio?

NR: Yes, I play on every record I produce. On the Thompson Twins sessions [*Here's to Future Days*] I played guitar. In fact, I played more bass on the Sister Sledge sessions [*When the Boys Meet the Girls*] than any other album I've done, apart from my own solo album.

I started playing keyboards on the Madonna album [*Like a Virgin*] and she says: "Nile, if you're playing it right why are you calling these other people?" I suppose that I just never had the confidence to think that I could come up with a part, and then play it as well as these session guys. I'd always figure out the parts, and tell them what to do . . . well not always, because I work with excellent keyboard players. In certain situations when you're trying to interpret another person's compositions, you know there are certain elements they want to bring out and the players aren't hearing it. You just say: "Wait a minute, it just goes like this; can't you hear it? It's so simple." So Madonna said, "Well Nile, if *you* hear it why don't you just play it. It takes five minutes for you to play it, and it takes an hour and a half for you to explain it!" I said, "You're right; but I

"I love putting pressure on myself, and consequently on the artist too, because whenever a person has to [reach] for a note, it translates to the listeners as *emotion*."

CREATIVITY IN THE STUDIO: A CONVERSATION WITH NILE RODGERS' SESSION ENGINEER JAMES FARBER, AND SKYLINE SECOND ENGINEER SCOTT ANSELL

Since relocating his base of operations from The Power Station to Skyline Studios, New York, Nile Rodgers has been working almost exclusively with session engineer James Farber who, prior to joining Rodgers' staff, was employed as a staff engineer at Power Station. What was it like to engineer for such a creative producer, we queried?

"It's great," Faber enthuses. "After maybe nine months or so of working together, we're at the point now where we do not have to say things to each other anymore. I might turn around and look at him and say, 'Oh, I've got a great idea how to do this' And he'll say: 'Yeah, I hear. You just do it.' Nile knows exactly what I'm thinking, which is great. He is one of the most talented guys in the business, and it's a pleasure to be working with him.

"He gives me a lot of freedom. When things feel good to him he knows it. But when things are not feeling right, he'll say, 'Tell me what that track needs.' He *won't* say 'Add more 10-thousand cycles!' [Laughter] Or he might ask for it to be made a little brighter, or that the drum sound needs to be more 'aggressive,' and I know *exactly* what he means. He's not ambiguous; he doesn't use words like 'make it more *blue* or *orange*!'

"As far as mixing goes, he lets me get the balance to the point where it's getting towards being finished, and then gives me his input. He usually won't say too much about the mix he wants before I start, unless there's something he had in mind that we hadn't discussed. Usually when we overdub we have a good monitor mix going. I spend some time getting the monitor mix right, and we get a lot of the ideas from that while we're overdubbing. I always put effects on the monitors to try and make it sound as much like the finished record as possible when we're overdubbing, so that the new part will fit in. I also send the monitor mix to the musicians' headphones, rather than set up separate foldback busses. I haven't set up a separate cue mix in quite a while; one thing I *never* get complaints about is the headphone mix. Although sometimes when you're doing a huge session you need to have two different headphone mixes, so I'll set them up. But even for basics, 99 out of 100 times I'll use just one cue mix."

From what Nile Rodgers has to say in the main interview, it's very obvious that he is a firm convert to digital recording and mastering. What did Faber consider to be the operational and sonic advantages to working with digital?

"Just the fact that if you got the sound right at the time you recorded it, you don't have to re-equalize it and add a little treble when you come to the mix — the sound stays *exactly* the same once it's been recorded on the tape. Whereas if you keep using the same piece of analog tape, by the time you start mixing, the sound will be a little different. Plus you get tape compression — or whatever else happens when you hit analog tape with a hot signal —noise, wow and flutter, and distortion — it's *very* different when it comes back off analog.

"The main advantage to digital is that there's *no* noise, plus the fact that there's no such thing as a punch you cannot do. With a Sony [PCM-3324] you can rehearse a punch-in and then program it. The machine also crossfades when it punches.

"On this recent Thompson Twins album, we bounced up the basic tracks that were recorded on a 3M 32-track to a pair of synchronized 3324s, and ended up with a lot of spare tracks with the 48-track system. The band members would go out into the studio and sing one at a time. Maybe Tom [Bailey] would do four tracks of the melody, and then four tracks

Nile Rodgers with first-call session engineer James Farber (*left*), and Skyline second engineer Scott Ansell with track assignment charts and SSL Total Recall automation floppy disks atop Skyline's SL4000 console.

Mel Lambert

Nile Rodgers

always felt weird about doing that."

I've always had a talent for getting a sound out of any musical instrument. When I was very, very young I was able to mathematically figure out the patterns. I could play a clarinet right away, a saxophone, even a trumpet.

R-e/p: I'm almost tempted to say that your "instrument" is now the recording studio, because there are so many recording and production techniques you can use to extend your musical talents. The word "producer" doesn't get it for me anymore, simply because you are also a player, arranger, synthesist, and a whole lot more besides.

NR: Yeah! It's interesting that now we really can use the *entire* universe to produce sounds for a record. I have a guy working for me who basically goes around the city sampling sounds; for example, he goes down to the Brooklyn Bridge and samples the cars with different miking positions. We've also experimented with a lot of water techniques; recording things over water and under water. We don't have as much time as I'd like to have for experimentation, because I'm usually committed to a recording project. I'm going to take off about six months next year, just to do a lot more exploratory work.

I have all these sounds sampled into a [Sony] PCM-F1, which we use in my [New England Digital] Synclavier [digital synthesizer], or to fly into a track. Kevin, the guy who works for me, said that "we've compiled such an extensive sample library that we could sell it!" We've had offers for about 10 grand, just for our tapes. They don't seem that great to me, but when we start to analyze them — the full range of different grand pianos and other instruments; hybrid sounds that we sort of made ourselves — they're incredible. You'll hear a great example of this on the new Thompson Twins album [*Here's to Future Days*]. There is this really strange percussion instrument that looks like a drum set, but is all metal; it looks like a metal sculpture that resembles a drum kit. We came up with the most fantastic sound on the John Lennon song, "Revolution." We also used it at *Live Aid*. Now that sound is in my library.

R-e/p: There's a lot of original sounds that Jan Hammer programmed into the Fairlight CMI for the song "Escape", one of the tracks you coproduced for the Jeff Beck album, Flash. *And, like Jan's underscores for*

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CREATIVITY IN THE STUDIO — continued . . .

of his harmony. Then the other members would go out and add different harmonies; somebody would double the melody. We ended up with a *lot* of vocal tracks, and would bounce them down into two digital tracks. We would do that on one chorus, and then use the [New England Digital] Synclavier to 'fly' them into the rest of the song where we needed them. Sometimes we had as many as 20 tracks of vocals that we comped together. Sometimes we would treat the backgrounds a little differently by piping them out to the studio through a couple of those tiny, self-powered Radio Shack speakers, and miking them to pick up some room ambience."

For the Thompson Twins sessions, the pair of synchronized PCM-3324 mulitracks were linked via timecode tracks and a pair of RM-3310 remote control units, which make the pair behave as a single transport. Skyline's engineering staff has also managed to link the studio's SSL console automation system to the remote control units, so that transport functions can be operated from either the mixing position, or from the 3310s. Has that helped speed up mulitrack operations, we asked?

"Yes, because I can now run the machines from either place," Farber acknowledges." Although the butt on layout on the SSL [master multitrack control panel] is different from the Sony remote, I'm used to punching left-handed, so I use the 3310 [which are located to the left of the console], and which also has the rehearse function. So I always use the remote for punching, and also in the early mixing stages because it's faster for the two digital machines to locate [the slave to match its position to that of the master] via their control tracks than it is for them to go by timecode."

At which point in our conversation we were joined by Skyline engineer, Scott Ansell, who was acting as second engineeer/studio liaison on the Thompson Twins sessions. For an alternate viewpoint of working in the studio with Niles Rodgers and James Farber, we asked Ansell for the second engineer's persepective on "life in the control room."

"Ever since Nile came into Skyline about four months ago," Ansell confides, "I've really been the only person from the studio working with him, because he has a lockout from Monday thru Friday. My job is to look after the documentation for the [SSL] Total Recall automation, and to make notes of every piece of outboard that we might use, and how it's patched, to make sure that the whole mix is completely documented.

"That's one of the great things about James: he's got it *really* organized on this project. As well as the settings of every console knob and button that's stored onto floppy disk by the Total Recall, we have this system of noting what's coming in on each module and where it's patched from; all the subgroups; the sends and returns; and all the outboard-gear setting. We have these sheets [Ansell holds up several front-panel layouts of effects units that were generated by Skyline staff on an Apple Macintosh PC] so we can document every setting we use. We've got [Mac-generated drawings of] the Quantec QRS, AMS [DMX-1580], [Eventide] Harmonizers, Lexicon Prime Time and PCM-41, Marshall Time Modulator and Tape Eliminator, and many more. One thing the SSL doesn't memorize [under Total Recall] is the master section, so we have a separate sheet for making notes of the master-send settings, etc.

"We are also using an outboard Harrison 'consolette' [fitted with a pair of mono and six stereo PRO-7 modules] on the Twins' remix session, because the SSL [SL4000E 56-input mainframe with 40 4000-series and six 6000-series modules] doesn't have enough inputs. We are using the Harrison for returns from the various digital reverbs and delays, which are bussed across to the SSL's stereo returns. All those settings also have to be written up.

"This is the first project where I've actually been keeping the paperwork as we go along, instead of going through the patch bay and figuring out everything at four in the morning when the mix is finally done! Before we got the SSL we used to take Polaroids of the board; Total Recall works great."

How would you summarize Nile Rodgers' production technique? "It's really unique compared to a lot of the other producers that I've worked with. He's a musician; he concentrates on getting the track *and* the arrangement right. He's not in the control room watching every move saying, 'Well, I think you should try 3K instead of 4K.' He'll say: 'I need more "crack" on the snare drum.' When it gets down to mixing, he leaves and lets James and me do our thing in here. He comes back every few hours and gives us a little direction. You should see him when we're overdubbing — he's the easiest going guy to work with but, when he gets on an idea and he's moving, he wants to work!

"The first time I worked with him, Nile Rodgers was sitting on a stool playing guitar; I'm ready to run tape, and we're getting into it. I had an idea for the piece and opened my mouth to say something to him. Assistants are *always* taught not to say a word on a session! You are supposed to feel out the vibe, and see what it's like as far as offering suggestions goes. Nile was looking across, and going like, 'What, man?' So I said: 'I had this idea for a part.' 'Well, what is it? I'm always open to suggestions.' I suggested the part, and he dug it. There is a very open atmosphere for making suggestions and, being a second, you feel really good. I'm young and that's the thing I have to keep thinking about. James is an amazing engineer, but I'm very anxious to participate. You walk in with producers who don't have one hit record, who are doing demos all their lives and they treat you like you're the lowest of the low. Nile and James are *very* open to suggestions."

Nile Rodgers

Miami Vice, *the track is mixed real hot!*

NR: Jan is fabulous on that track; it's one of my favorite songs on the entire album.

The interesting thing about that track is that we also have to give credit to the engineer, Jason Corsaro. Jason was so in love with those sounds that it was really his idea to mix them at those drastic levels. As a matter of fact, that track became our test tape for checking out studios all over the world. You don't know how many studios we've gone to with that mix, and almost blown the speakers off the wall. Fried the UREIs!

R-e/p: Let's move on to talking about the studio environment. You've worked at a lot of different facilities, including Power Station, Atlantic, Media Sound, Hit Factory, Skyline and Record Plant here in New York, as well as Maison Rouge in London with Duran Duran. What attracts you to a particular studio?

NR: Basically, I think that any studio that's good has to have a good-quality console. I just happen to prefer the SSL now. I like to use a Neve to lay the tracks, and an SSL for mixing — although I don't mind laying tracks with an SSL. And a Sony PCM-3324 [digital mulitrack]. Of course, the echo that a studio provides is also important to me — especially after being at Power Station for all these years, and growing accustomed to their live chamber. We've successfully recreated a live chamber here [at Skyline Studios, Rodger's new base of operations] from the little gallery in front of the elevator. Unfortunately, you can only use it at night. The sound is incredibly smooth; the decay curve in there feels a lot better than what we enjoyed at the Power Station. I shouldn't say a lot *better*, because Chamber #2 was pretty amazing, but we've compared mixes and it sounds fine. . . . continued on page 51 —

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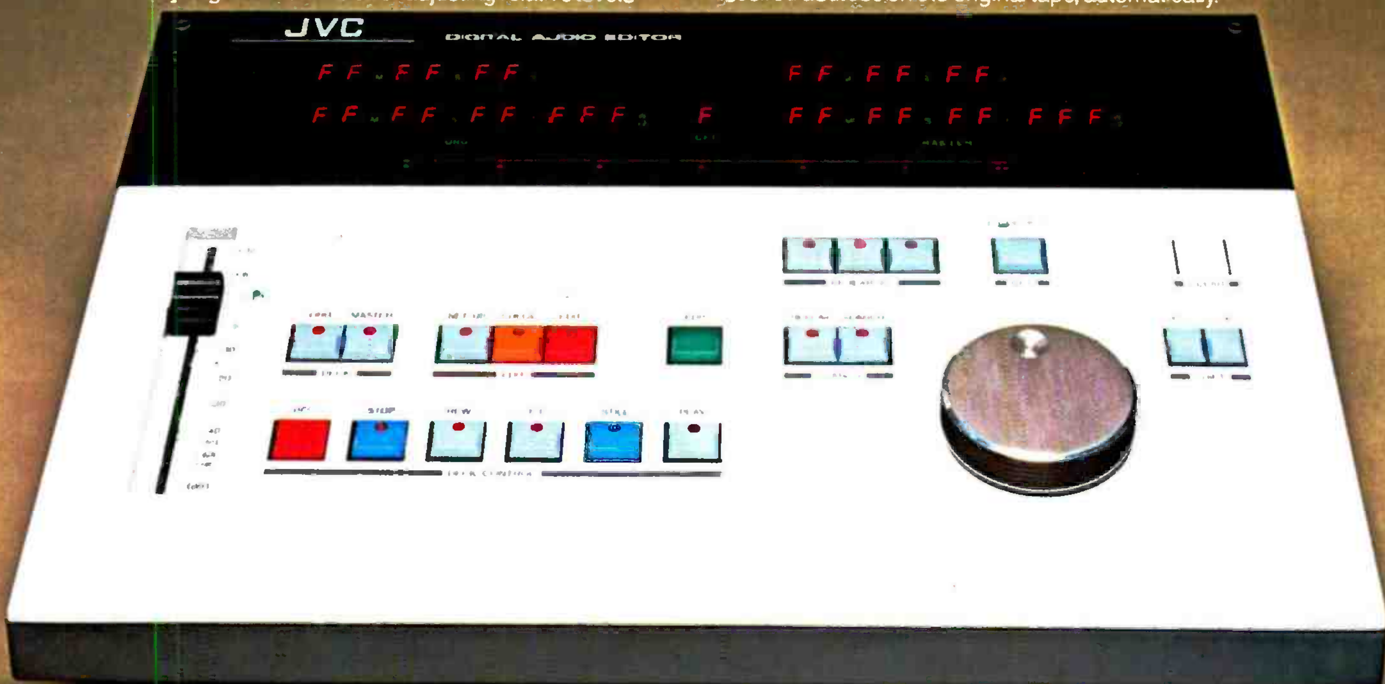

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Producer/musician Nile Rodgers in the studio with Laurie Anderson (*above*), Jeff Beck (*center*), Peter Gabriel, Adrian Belew and Rob Sabino (*top-right*), and Madonna (*lower right*)..

— continued from page 16 ...

Nile Rodgers

R-e/p: You use the live chamber on vocals?

NR: Vocals and mainly drums — any instrument that's in need of a very resonant high-frequency sound, but still the natural reverb that we get in the room. It adds a *nice* dimension to the record. However, I do love all of the digital reverb units; I haven't found one yet that we can't find some use for, even the cheap units. Even effects devices that people find pretty inferior can give you some *very* dramatic sounds. That's one thing I learned through working with people like Peter Gabriel.

I just finished a song called "The Terminator," which I had written for the movie *The Terminator II*. We tried to recreate the sound of Arnold Schwarzennegger smashing through the police station wall. But we couldn't get it right no matter what we used. "Wait a minute guys," I said, "turn on the microphone that's on my guitar amp." I ran out in the studio, lifted up my guitar amp and dropped it on the ground with the screaming reverb. Bang! It was perfect. Now *that* sound is also in my effects library.

R-e/p: Would you describe yourself as a technical producer?

NR: Not at all. I sort of stumbled over the techology. I get into a few pieces of equipment, and feel compelled to learn about them. But basically I feel that my strongest point is as a musician, arranger, orchestrator, and player — I can just hear something and say, "I don't think it's good, but if we do *this* it could be better!"

R-e/p: Do you do have a regular session engineer who looks after technical side of production?

NR: Yes. I work pretty exclusively now with James [Farber] who came here when I moved from The Power Station. I'm not an engineer at all; I'm only an engineer when I'm making demos!

R-e/p: You don't get behind the console, apart from maybe gain riding a lead vocal line, or something like that?

NR: I do touch it, but I think James is more than qualified to handle that side of things. What I really love to do is to see what *he* might come up with. If he thinks of something that is working I won't say a word; if, on the other hand, I do think of something that could be better I will say something. Although I might touch knobs every now and then, James doesn't come over and sit on the piano and play parts, even though he's a fantastic piano player! You *don't* want to distract a person, and I really don't like people who do that. If your engineer is working on something, and you don't know what's in their mind at that time, if you turn the knob and all of a sudden upset what they've been focusing on, it's *totally* unfair. In the same way, if I'm working on a guitar part and someone comes in and starts playing something else I'd get pissed; I feel that it must be the same way with an engineer.

I've sat behind the board and had someone else soloing other instruments trying to get a sound while I'm thinking about a specific sound — it's *completely distracting*! And I know a lot of people who do that; I don't think it's hip. They listen to the bass and say, "What if the bass was a little brighter?" Then they go over and start making it brighter, and meanwhile the engineer might be working on the drums.

I've been in the studio with a lot of my friends who are producers, and they do that all the time. I've learned to avoid doing that because, I guess, I've worked with great engineers for whom I have a tremendous amount of respect.

R-e/p: Do you communicate with James on a technical or on a musical level?

NR: The wonderful thing about James and I is that we seem to be made for each other, because he is an *amazing* musician; he's a phenomenal jazz piano player, and has played on our records already. So we have very good communication both musically *and* technically. We have an extraordinary affinity about other things outside of the studio too, and these things seem to help us with communication in the studio.

R-e/p: Why did you decide to move your base of operation from Power Station to Skyline Studios?

NR: When James and I were out in California working on the Sister Sledge album [*When the Boys Meet the Girls*], it was the first time I had a studio block-booked since I worked with Duran Duran in London. I said: "Man, this is *the* way to work." If we're doing a mix we know that the console isn't going to get torn down, and we know that there's no one in this studio who doesn't belong here. It's sort of uncomfortable if you're working with Mick Jagger, and mean-

Nile Rodgers

while there are lots of people coming in and out, because the studio has three different rooms. Sometimes fans don't understand that because you're concentrating on a lead vocal you can't be jovial, friendly and wonderful.

Working here in the garment district, people aren't so keen to come over and stand about at two o'clock in the morning. Also, there's no one in this studio who doesn't belong here. Skyline is just better for me.

I thought that the people I work with might find Skyline somewhat uncomfortable, because it's not nearly as cosmetically comfortable as the Power Station, or some of the other nicer studios. But I think the one thing that's great is that this studio gets us back to the basics: *The Record*. When artists come in here they feel that they're in a *real working situation*. This is a studio: we don't live here; we don't party here — we *work* here. Also, not to sound like an ogre, because we have a wonderful time here, but it's an atmosphere that's *very* conducive to work.

[Editor's Note: It transpired from my conversations with Nile that he has a lock-out booking arrangement with Skyline Studios, whereby the producer has exclusive use of the facility Monday thru Friday, the weekends being open for outside clients. In addition, the producer now uses the studio as a permament home for his personal-use Sony PCM-3324 digital multitrack, and also bases his production company at the studio — ML.]

R-e/p: I seem to recall that you first worked with the Sony PCM-3324 digital multitrack during production of some Peter Gabriel tracks for the film Gremlins. *What was your initial reaction to recording on the 3324?*

NR: To be honest with you, I was *totally* blown away! After recording for the last three or four years on the best equipment available, to hear the difference between the Sony and the Studer A800 — and man, I *love* A800s — I was ruined. I couldn't believe it when they took the machine back. Later, Jason [Corsaro, engineer] put up the analog tapes, and I said, "What are you doing, man? What *is* this, a new concept you're exploring?" He looked at me, and said, "No, what are you talking about?" I said, "What *is* all that high-end hiss? Do you have a tape echo going, and we're just hearing the tape?" He said, "Nile, that's the *multitrack*; that's what it sounds like." I couldn't believe it.

For this new Thompson Twins album, they were working digitally in Paris [with a 3M 32-track Digital

Mastering System], and when they came to Skyline they were working on the Sony. We made some rough mixes to analog two-track, and were playing them for the record company. When the record heads left, Alannah [Currie] — the girl in the Thompson Twins [with Tom Bailey and Joe Leeway] — says, "Nile, what were these terrible mixes you were just playing? What was all that 'shhhhhhh' — what was all that hiss?" Which was amazing! Even she realized right away, because the band were so accustomed to hearing playback from the digital multitracks and [Sony] PCM-1610 digital stereo masters.

R-e/p: You don't hear any "funnies" in the top-end, with cymbals, or very high-frequency harmonics? Some people say that the one thing they don't like about the sound of digital is that, because of the low sampling frequency being used, the A-to-D processors have to roll off the HF too steeply — the so-called "20-kHz brickwall."

NR: If that has a real place in the field of music , fine. That's their choice. I'll tell you: in *my* music, I've never heard anyone complain about that [top-end loss]. I've *never* seen a kid standing in line at the store saying, "I would've bought the record, but when that cymbal cut off at 20k . . ." [Laughs at the absurd image he has conjured] It's totally ridiculous!

In my world, I want my records to be as technically as perfect as possible; I love it when I listen to my CD player and hear my records coming back but, I tell you, I'm not nearly that sophisticated. They sound unbelievable, compared to what I used to listen to. If I had a percussion outfit or something like that — say maybe 70% of the record had cymbals — and digital made it sound lousy, then I *might* get pissed off. But digital sounds *dynamite* to me, and I honestly don't hear it.

R-e/p: How you prepare for a session? Do you spend time in rehearsal with the band or artist, listen to their previous projects, or start with a fresh ear?

NR: I think most of the artists I work with are incredibly surprised at how nonchalant I am about their records. It's not because I don't care: as a matter of fact, it's the exact opposite. It's *because* I care so much that I want to be just as excited as they are about their music; I want to feel like I'm in the band, and I'm learning the song for the first time. I love to pick it apart and put it together with everybody: I *love* the creative process. Anybody can rehearse a song and play it back. But the learning of the song and putting it together while working together as a band — as a unit — *that's* what my strength is. It's the easiest thing in the world to go into rehearsals for an hour, and then play the song back. That's what I love to do — in the studio. *Not* outside of the studio.

The only time I've ever done preproduction with a group was for one song, "Wild Boys," with Duran Duran. And it's just because I was there. I would've rather they just worked on the song by themselves and, once they got it down and felt comfortable with it, had called me up and said, "Let's go into the studio and record." Usually, I don't even want to hear the demo! I listened to a Thompson Twins demo once, and they kept calling me back. I almost didn't do the project, because I was so nervous that they wanted answers from me, and I wasn't prepared to give them any. I listened to the songs one day at the Power Station, and felt so confident that we'd make a good record in the studio that I didn't have to listen to them again.

R-e/p: What, for you, forms the foundation of a track? Is it the drums and bass, or a more complex combination that you regard as being the "bedrock"?

NR: I guess it's the drums, bass, and guitar — maybe because I'm a guitar player, I always want interesting guitar parts. Sometimes I put the melodic portion of the rhythm section in the bass; I like melodic, moving bass lines — probably from working with Bernard for so long! I like to have very static guitar parts: while the bass has major melodic movement, the guitar is just basically giving a rhythm feel. That's not written in blood though; it could change at any point in time.

R-e/p: Do you tend to try to keep as many of the basics that you track, because there's often a coherence to them, since the band was playing together?

NR: No, not really. I think it's just something that happens. I'm honestly not conscious of anything that I do when it comes to making a record. I really do look at each song *individu-*

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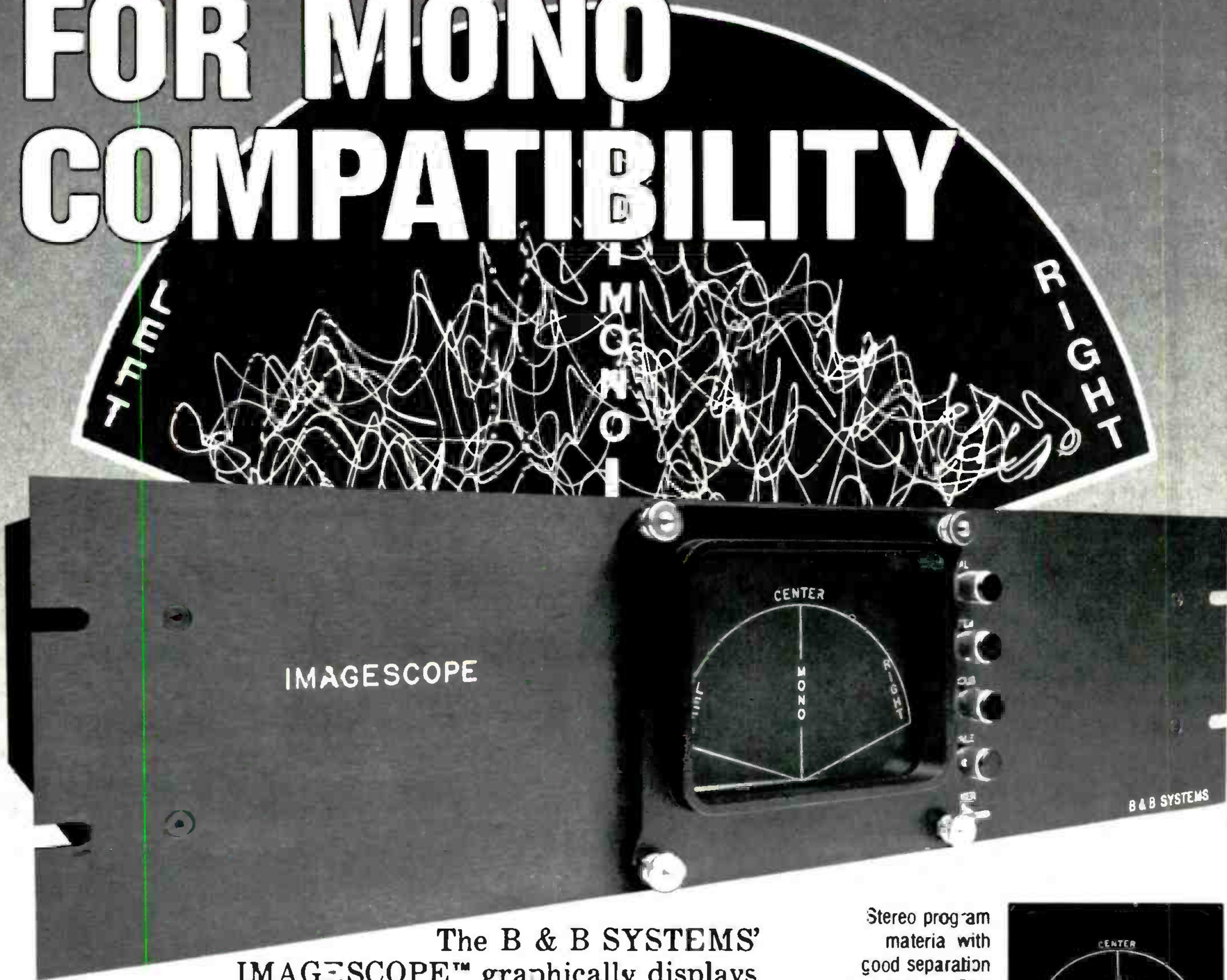

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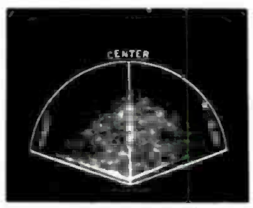

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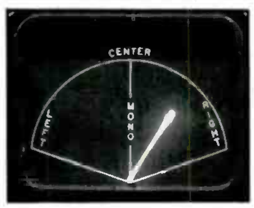

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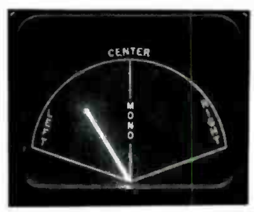

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ally. If something is out of the "norm" for me, I listen to it very carefully before I make a final decision.

R-e/p: You may change your mind, and go back and do something else.

NR: Usually I make my decisions right there on the spot; it's very rare that I change something, because I usually like to live with my decisions. If I do something that's not working, then I'll put on a different part to make the previous part work. I think that comes from working in a band. A person may have a part that they just love, but you've somehow got to do something to accent that part, or help it along. You just don't say to the player, "No man, this part sucks! Get out of here!" Instead, you try to do another part that brings it out, and actually makes the part more *musical*.

R-e/p: Do you make much use of MIDI-controlled synthesizers and electric percussion?

NR: I *love* MIDI. It cuts down some of the time that I would spend layering parts. The only problem with using MIDI is that sometimes I tend to make the textures too "thick" right away — as opposed to doing it one instrument at a time, and somehow leaving some of the space free. Instead of doubling a part *exactly*, you only double certain portions of the part to give texture to the composition. We use the [Friend Chip] SRC Box to slave our MIDI sequences to time-code, and it's fantastic.

R-e/p: You also have a New England Digital Synclavier II digital synthesizer which, I seem to recall, is equipped with the polyphonic sampling option.

NR: Yeah, I just happened to get the first Synclavier with that option; I guess because I had the dough at the time!

I use the Synclavier for a lot of different things. I just wrote a song arrangement for Sheena Easton's new album. I did the whole thing — bass, drums, horn parts, vocal pads, the "oohs and aahs," that sort of thing — on the Synclavier. I just push the button and it starts up. You can change the key, change the tempo, slow it down, speed it up, change the meter, do rubato moves — anything we want. Also, the stereo options inside the Synclavier are far more complex than the console's stereo features; it's like every track has a [dynamic] panning function. In other words, you can move in certain increments across the stereo spread; you just pick up each portion of the sound and move it throughout the pan spectrum. That's pretty unique!

The polyphonic option allows you to sample any sound that you want, and then simultaneously play back as many notes as you have memory for. It gives you more interesting sounds. Let's say you sample one hand clap. Traditionally, when you have hand claps in a drum machine, you have basically that *one* sound. If I take my fingers and do this [taps several fingers on the keyboard], it sounds like audience applause, because you have the cardinal pitch, and various overtones around it [from the polyphonic playback]. The substance of the sound is changing as you hit several notes of the keyboard, because once you move it around [in pitch] the combined sound gets fatter or higher. It *sounds* like an audience, rather than multiple, individual handclaps.


Berry Berenson

R-e/p: Let's move onto some specific productions, for example, the Madonna album, Like a Virgin.

NR: The main thing that I tried to do on that album was to give Madonna more artistic credibility, to bring out more of her musicianship. Which means that I did a lot of the tracks with a live rhythm section. It was *very* important for me to make her feel like she's developed to the next stage. Her first album was all drum machine; you could have done it in a control room. Whereas on her second album, the studio played a tremendous part, with live strings and the rich sound of a real orchestra, with people making mistakes. But that's what creates the *live* feel!

R-e/p: Okay, let's move on to the David Bowie album, Let's Dance. *David Bowie worked with Tony Visconti for a long while, he produced himself, and he's produced other people. Was it a surprise for you when he wanted you to co-produce that album?*

NR: Absolutely. I didn't even return his calls for about a week, because I didn't think it was actually David! I thought the construction workers that were working on my house were making jokes: "Hey, Mr. Rogers, David Bowie's on the phone!"

R-e/p: How did the sessions go with him?

NR: Unbelievable! We finished the album in 21 days; in actuality, it was 19 days — I didn't even show up for work the last two days, because I knew it was finished. For those extra two days, he just had to play it for the record company, because we did that record without him being signed to a label. The fact that it was a spec deal [released eventually on EMI] helped create that magic; the fact that we were doing it on our own and didn't have anyone to answer to.

David's album is the first time I ever did what they call "head charts." The horn players were all sitting there, and I said: "I want you to do that, exactly." "What," the guys said? So I went over to the chart and wrote down the voicing I just did. I said, "*That's* what I want you to play." They said, "Sure!" [Sings] *Bip! Bip!* [Waits] *Bip!* [Waits] *Dey, doot, dey, doot, doot.* That was it. And David started singing parts that he had, like [sings] "Let's dance" *doot doot* "to the sound they're playing on the radio" *ooh, bop, ooh, bop, ooh bah bah.* That was the first time I had ever done a record like that: so very laid back; incredibly nonchalant. Now ... I don't worry about whether or not I have the charts written out the night before, because I know I could write them on the spot with the guys, and it'll be fine.

With Madonna, I wrote everything out, because I really wanted her to feel like we were moving into a different phase in her life. I thought it was very nice for her to go in there and see a full string section, and for her to see charts labeled with her name as well as the instrumentation. To see a score written out, and the string part marked to wait for her; I just know that made her feel good.

Once she said, "Nile, I can't do this." And I said, "Of course you can, Madonna, because *we're* following *you*, you're not following us." Those little things are what the artist remembers, and they are the high points of the record, even though you never get to talk about it.

R-e/p: And the Jeff Beck album, Flash? *I understand that one difference between Jeff's album and, say, the Madonna sessions, was that most of the songs were worked out in the studio.*

NR: Yes, Jeff was completely unprepared; all we had were very fragmented ideas. I had basically written

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what would have been a whole album, if we had wanted to go that route. My concept behind working with Jeff was for him to re-form the Jeff Beck Group because, as great a guitar player and musician as he is, I think that Jeff is really appreciated more in the commercial arena when he "steps out" from a band.

R-e/p: How did you get that incredibly distant guitar sound on the song "Ambitious?" Was there a live chamber on that track?

NR: No, we used the room. There were two amps set up at Power Station in stereo: a Seymour Duncan amp and a Fender Concert. Both were pointed up to the dome ceiling in Studio C, which is quite high — maybe 18 or 20 feet tall. The mikes are on a pulley system, and we put them all the way up at the top, with the amps on an angle so that the sound is bouncing off the walls — one wall is glass, the other one wood. We were getting a series of subliminal delay times before it gets up to the microphone mounted in the ceiling. It's subtle, and you really only hear it clearly when the guitar is soloed.

And on that track I'm also playing a very static guitar part that was taken direct. So you hear my direct part, Jeff's ambient parts with all the delays, and it makes for an even more distant effect; you feel the part in time, right in front of you, and you hear the delays. But they're *not* programmed delays; because it's natural the delay is changing all the time.

R-e/p: Your own recent solo release, B-Movie Matinee, *was a concept album that tried to capture the nostalgia and mood of those trashy Saturday-morning movies we flocked to see in our youth. Like the Jagger album, you recorded it over several months. Was it very difficult for you to maintain a sense of continuity?*

NR: Very difficult. And I'll never do that again: I'm gonna start treating *myself* the way I treat everyone else. And it's tough, too, because I get overly involved emotionally in every record I do. Like, I'm working this weekend, but I'll let these guys take off. I could use a couple of days off, too — I'm exhausted! Even when I finish here, I have to go home and write some songs.

R-e/p: The Thompson Twins album that you've just finished [early August], Here's To Future Days, *was started in Europe on a 3M DMS digital 32-track. Back here you bounced those tracks up to a pair of timecode-synchronized Sony PCM-3324 digital multitracks, and overdubbed solos and vocals. Apart from your own personal-use 3324, I understand you rented a second machine for the dates from Electric Lady Studios, New York. What was it like working double digital 24-track?*

NR: It's absolutely wonderful. The quality difference really shows in the vocals; listen to the difference in the vocals between the new Thompson Twins album, and, let's say, the Sister Sledge and my last few records. The difference is *amazing.* On a couple of these Thompson tracks, we have as many as 130 voices that we bounced down a number of times on the 3324.

R-e/p: Do you have an overall sound of the record in your mind before the production gets underway?

NR: No, not at all. It only develops once I get into the studio, start listening to the songs and working on them.

Usually when I listen to a song for the very first time, I start thinking of parts. And I always say to myself that if I can think of that while listening to the demo tape, then of course I'm going be able to think of it when you put the guitar in my hands! And that's what I feel very comfortable in doing. I like being a studio musician, I guess because I've done that all my life. So, I don't even think about the song.

I love putting pressure on myself, and consequently on the artist too, because whenever a person has to strain for a note, it translates to the listener as *emotion.* If I'm struggling to get to a part, when I finally get it right it feels fantastic! The sense of accomplishment; there's nothing that can replace that. In the studio you're dealing with people who have carved out a place in the musical world. How do you keep them inspired? You have to create obstacles every day, because these people are accustomed to getting over obstacles, or overcoming hardship. When a person is a star, *they've* made themselves stars; they don't just luck up! Usually, these are people who have studied for years, tried and failed, tried and failed, and finally they've made it.

R-e/p: I once had a conversation with Tony Visconti, who said that he'd find it very difficult to work with new bands or soloists. As you may know, Tony plays practically every instrument, and has worked extensively with Thin Lizzy, Bowie, Hazel O'Connor, and dozens of established acts. He felt that he would have a hard time producing somebody who couldn't play an instrument at least as well as he could. You wouldn't have a problem with that?

NR: None whatsoever. I worked with a group called INXS. To most people's minds, the band was relatively new, and still haven't broken big in America. To me, it was just like being in Chic again. They are great musicians, and I didn't care if somebody had said that the guitar player couldn't play as well as me. Because the record is *their* energy, and *their* individuality — that's what makes that band sound great.

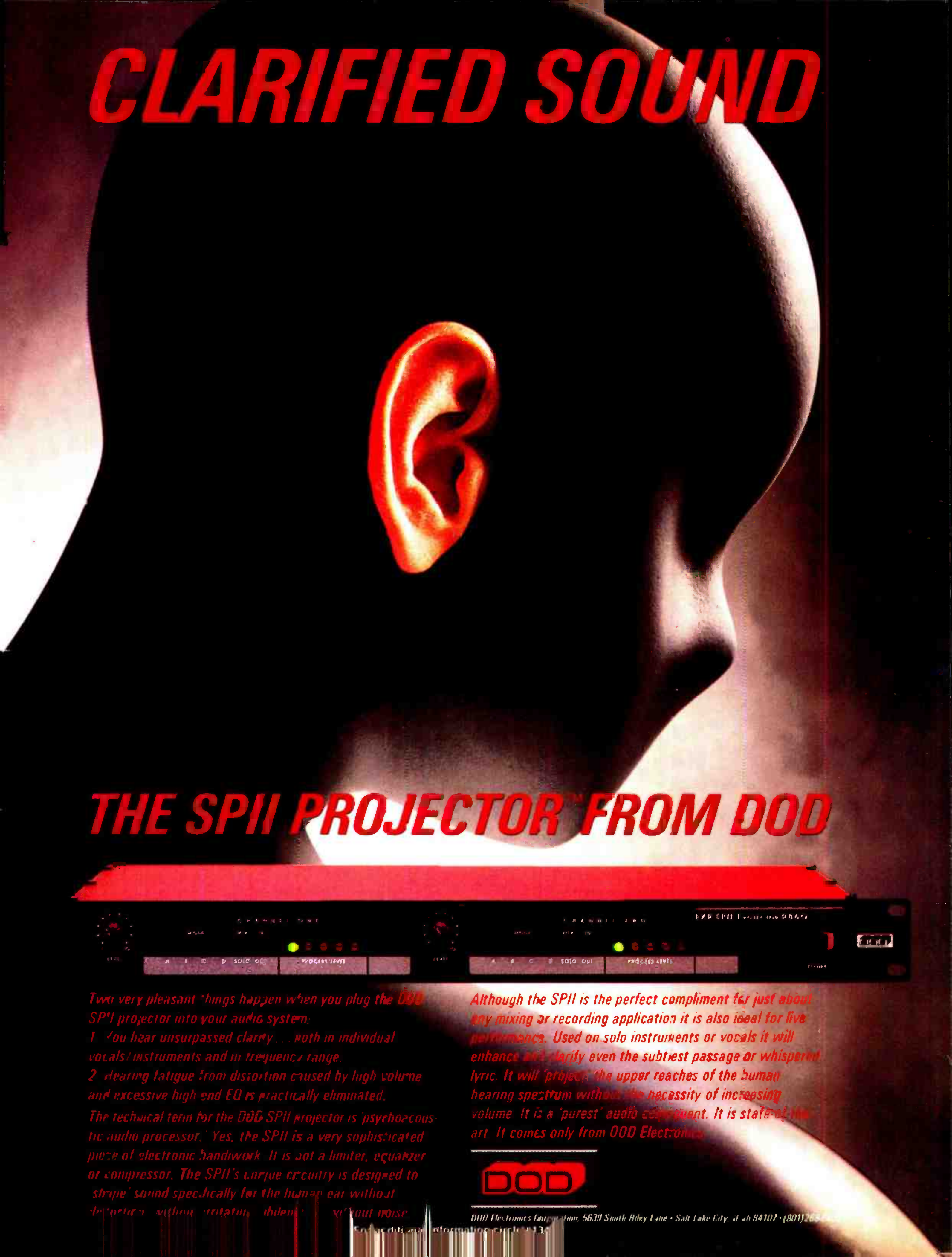
I understand Tony's frustration; I know exactly what he means. Just like Madonna said, "Nile, if you can play it, what are you doing explaining it to the guy for? Go in there and *play* the part." But the other side of the coin is that this is a *band.* These are kids who have struggled and fought and stuck together through all kinds of adversity to make it. Now that they've made it, you can't take that away from them; you can't just say, "Gimme that damned guitar. Let me show you how to do it."

You *have* to be sensitive to that side of an artist's creativity, because it's their career and future. A lot of times I'm working with people and the engineer will know that I could play the part better that the band member. But I always think back to the times I was in a band, and Bernard would do that to me, because he can play great guitar. With Chic that was okay, because we were in the band, and I could go over to the drums and say to him, "No, play *this* part." But if the producer had walked in, grabbed my guitar and played it, I might feel a little bad!

I work with vocalists in the same way. They might say: "What do you mean by that, Nile?" "In the second part," I'll say, "don't go [reads the word 'tonight' in various ways]." Sometimes your explanation is very complicated and confusing; you know what you hear, but the vocalist doesn't know what you're calling the second part. Every musician, every producer has a *completely* different language in the studio. I remember Bernard once said to Diana Ross that she was "Under". "What do you mean by 'under'," Diana asked? "I think you're under the track — flat." "*Flat!*" She freaked out. You *have* to be tactful. Maybe I should always give people a glossary of *my* terms!

R-e/p: So, to summarize, there's no rules, except the person should know where you stand, and you should understand where they stand.

NR: Yes. And the more difficult a person is — when I say "difficult" I usually mean "egotistical," or what I prefer to call "confident" — the happier I am, and the better the record comes out. Because I love that interplay between the artist and myself. I really *need* that energy; the more you have to say, the better a record's going to be. ■■■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DIRECT METAL MASTERING

Improving the Quality of Disk Cutting and Album Manufacturing

by Rob Tuffly

To many recording engineers, the disk mastering process is often considered a creative art form, with the cutting engineer successfully making subtle but vital enhancements to the final mix. The mark of a good disk mastering engineer is that he or she knows by experience what kind of changes will occur during the transition from tape to the final release medium.

In essence, disk mastering can be looked upon as much as a "philosophy" as a technical skill. The transfer of recorded information to a master lacquer — or to copper, as with Direct Metal Mastering — requires that the mastering engineer be conversant with both the recording *and* the disk mastering aspects of engineering. Their job forms the very core of the record-manufacturing process.

To understand this somewhat complex process, and to bring up to speed those readers unfamiliar with conventional disk-mastering techniques, let's trace the progress of a master tape from cutting through plating, and finally to record pressing.

The cutting engineer's first job is to choose a lacquer disk suitable for cutting. In a good batch, between 10 to 20% of the blanks may be rejected. Many mastering engineers estimate that up to 75% of all flaws on a record result from the use of a bad or damaged lacquer. This degree of critical requirement is easy to comprehend if one remembers that a lacquer blank comprises a 36-mil (0.036-inch or 0.914mm) aluminum substrate, on which is sprayed a nitro-cellulose (lacquer) coating of about 7 mils (0.007 inches or 0.178mm), containing platicizers, resins and modifiers used for adhesion, cutting and plating purposes. Since the lacquer coating is akin to a paint layer, it must be treated with extreme care, beginning foremost with the quality of equipment available at respective cutting, plating, and pressing facilities. [For a comprehesive review of disk-mastering philosophies, see the article "Disk Mastering: The Misunderstood Link," by Bernie Grundman, published in the February 1985 issue of *R-e/p — Editor.*]

Once that master tape — either analog or digital — has been loaded on the cutting room's replay machine, the engineer will listen to playback through the transfer console, setting rough levels, checking for out-of-phase material, and re-equalizing the material if necessary, followed by compression and overall limiting. The signal normally passes through an eliptical equalizer, which filters out-of-phase low-frequency information and converts it to mono. (Otherwise out-of-phase low-end signals result in excessive vertical motion of the stylus cutting a deeper groove, which is often difficult to track on playback, and reduces the overall music capacity of the record.)

From the transfer console, the signal is passed to the disk cutting amplifiers and to the cutterhead, and also to the computer that controls operation of the cutting lathe. Normally, the disk-cutting computer adjusts the groove pitch and depth according to the dynamics of the material being mastered. To enable the computer controlling the lathe to anticipate loud transients, or bursts of low-frequency material (both of which require increased groove spacing or depth, or both) the cutting system is fed with a preview signal, which arrives before the program signal. This identical, anticipatory signal can be provided by a dedicated preview head fitted to the replay tape machine; for digital mastering, a digital delay line is used to provide the cutting signal, with a direct feed providing the preview.

Following one or two (or possibly more) reference cuts, the mastering engineer will make any additional EQ and compression changes. (It might be interesting to note that although the DMM process centers on mastering onto copper material, the producer can still cut and play a 12-inch copper disk for a reference master.) Once the session producer has approved the reference lacquer, a final master is cut and sent to the plating facility. Time is often of essence in this critical stage of record production.

"Optimum time to plate a disk after cutting is within one hour," offers Greg Kuhn, manager of quality assurance at Sheffield Matrix Labs, Santa Monica, Calif. "Subtle, high-end losses occur if the lacquer sits around unpro-

Color photography by Rich Vogel

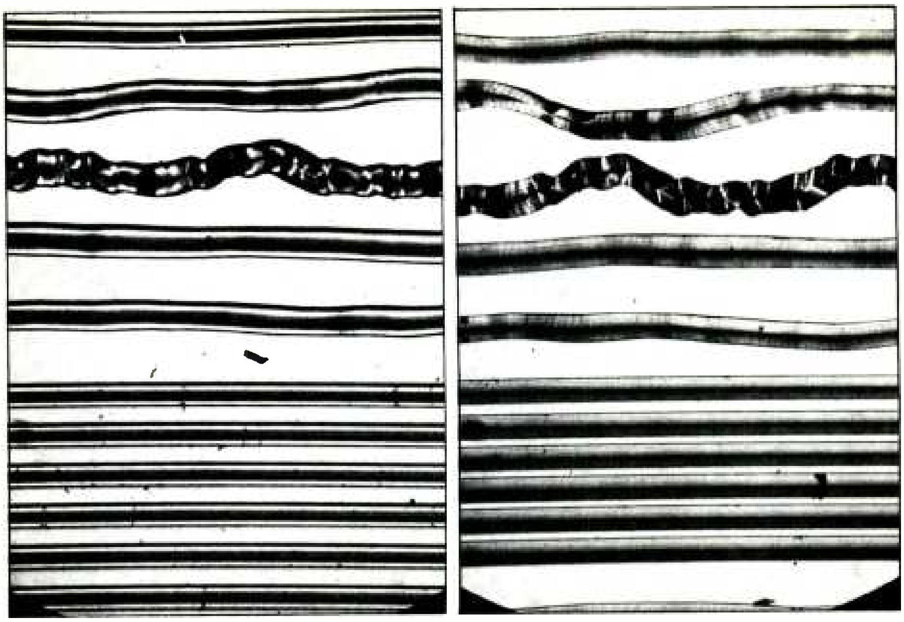

AMPEX

cessed, in addition to pre- and post-groove echo. After a week, the losses and lacquer degradation in general would be too great to plate a good master."

Prior to the plating process, the master lacquer is thoroughly cleaned, and the edges and center hole sanded to enhance adhesion of a silver-nitrate solution. Then, after receiving a very thin coating of a highly refined silver, the lacquer is lowered into a pre-plating nickel sulfamate bath through which flows a low electrical current. During this electro-chemical process, a layer of nickel slowly deposits onto the surface of the master, resulting in the initial silver coating taking a "mirror" effect. "After the preplating," Kuhn continues, "the master is washed again, then placed into a high-speed, nickel plating bath until about a pound of nickel is deposited on the disk."

Separation of the deposited metal master from the master lacquer provides a "father," but destroys the original lacquer. After a solvent and cleaning solution are applied to the father, between five and 12 nickel "mothers" can be grown in a similiar way to that described above. When separated from its mate, the mother can be played on a phonograph to check for any ticks, clicks, or similiar anomolies generated from the silver or nickel-plating processes.

After approval of its quality, the mother is then returned to the nickel bath for stamper plating. To increase the life of a stamper, and to decrease oxidation, some facilities will sometimes plate a thin layer of chrome onto the 10 to 20 stampers made from each mother.



Photomicrograph of mother produced from lacquer master (*left*) and from DMM copper master (*right*), showing improved transient response resulting from the "feather-edged," non-heated Direct Metal Mastering cutting stylus.

The final step in record manufacturing falls to the record pressing plant. Here, the stampers — one for each side of the single or album — are placed in a mold shaped with the basic record profile, steam-heated vinyl at a temperature of approximately 300 degrees F (149° C) injected into the mold, and the two halves pressed together for about 25 to 50 seconds, followed by 10 to 20 seconds of cooling. (During this stage, the labels are incorporated into the album center.) The finished records are then inspected, and sleeved.

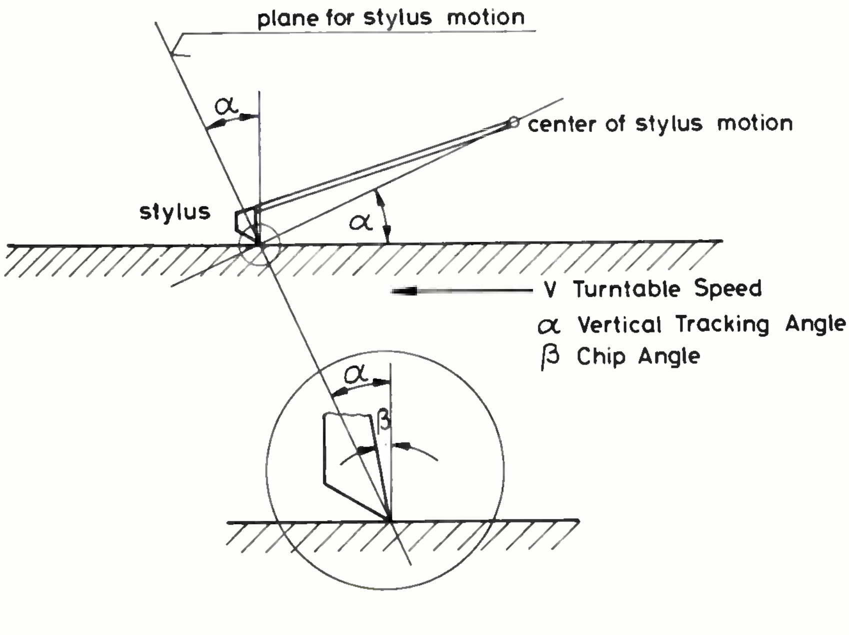
Since the processes described above are automated, and susceptible to the introduction of surface noise, pressing techniques vary at each facility. Some typical inherent noise characteristics introduced during the manufacturing process include "rumble" (low frequency noise), "ocean roar" (from lead-in grooves), and "swishes" (again from the lead-in grooves). All these episodic degradations are pretty much common in every plant, and dealt with in equally episodic ways.

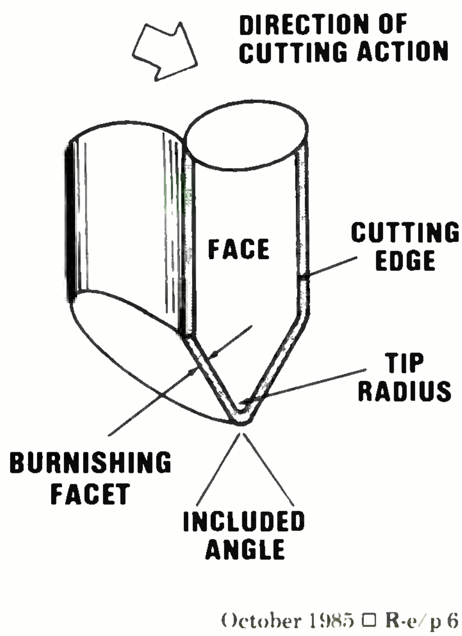
An Alternative: DMM Technology

Direct Metal Mastering has existed for just over five years, with 15 facilities around the world currently utiliz-

. . . continued on page 64 —

DIAGRAM OF CUTTING STYLUS ORIENTATION TO SURFACE OF BLANK, SHOWING RELATIONSHIPS BETWEEN VERTICAL CUTTING AND CHIP ANGLES. AT RIGHT IS A CLOSE-UP OF THE DMM CUTTING STYLUS.





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Recording consoles sold today were designed over four years ago. They are patterned after designs that are a decade older. Think about how recording has changed since then. There were no synthesizer virtual tracks, no music videos, no MIDI, no VITC, and no requirement to better the noise and distortion of digital recorders and CDs. Today split monitoring consoles have gone the way of the brontosaurus, with in-line consoles soon to follow. What contemporary studios need is something completely different.

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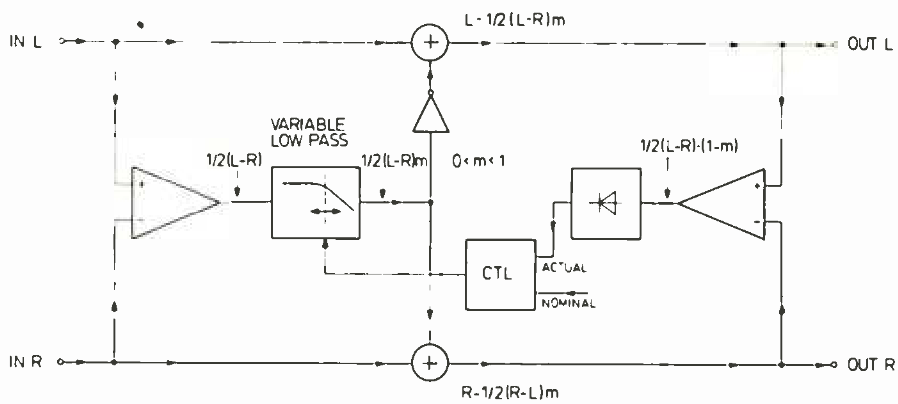


BLOCK DIAGRAM OF NEUMANN VAB-84 VERTICAL AMPLITUDE LIMITER

ing DMM-equipped cutting lathes. It has been estimated that, to date, approximately 60 million records have been pressed from DMM-mastered material.

Its advent, however, is new for the American recording industry. Europadisk, an independently-owned cutting facility based in New York City, is the first record processor within the U.S. to fully manufacture, master, plate and press under the DMM license. Jim Shelton, Europadisk co-owner with Christian Lach, says that his facility has always been a record manufacturer, and "our decision to utilize DMM came from a natural progression. The choice of DMM was because the system is better — much better than lacquer technology. Technically, DMM sounds cleaner with a musically wide dynamic range.

"From our point of view," he continues, "we have now joined the Europeans in disk technology; pretty consistently they made better records than the Americans for a variety of reasons, not the least of which is their constant R&D. DMM results directly from trying to improve the LP record."

A jointly developed product of Telefunken-Decca Schallplatten GmbH (Teldec), West Germany, working in conjunction with Georg Neumann GmbH (which is 25% owned by Teldec), DMM technology is based on the ability of a modified Neumann lathe to cut onto a 4-mil (100-micron) thick coating of copper layered onto a 300-mil stainless steel substrate, rather than a conventional aluminum/lacquer disk. Following the cutting of a DMM copper mother master, the plating facility can directly plate nickel stampers or fathers, bypassing the expensive, and flaw-provoking, silver-plated father stage, and the ensuing mother-nickel step.

The advantages claimed by DMM can be summarized as follows:

- The elimination of all lacquer-related problems;
- Substantial improvements in high-frequency and transient response;
- 15% more playing time per side;
- No "horns" or plating anomolies;
- The virtual elimination of recuts; and
- Because two plating steps can be eliminated, faster stamper production is possible.

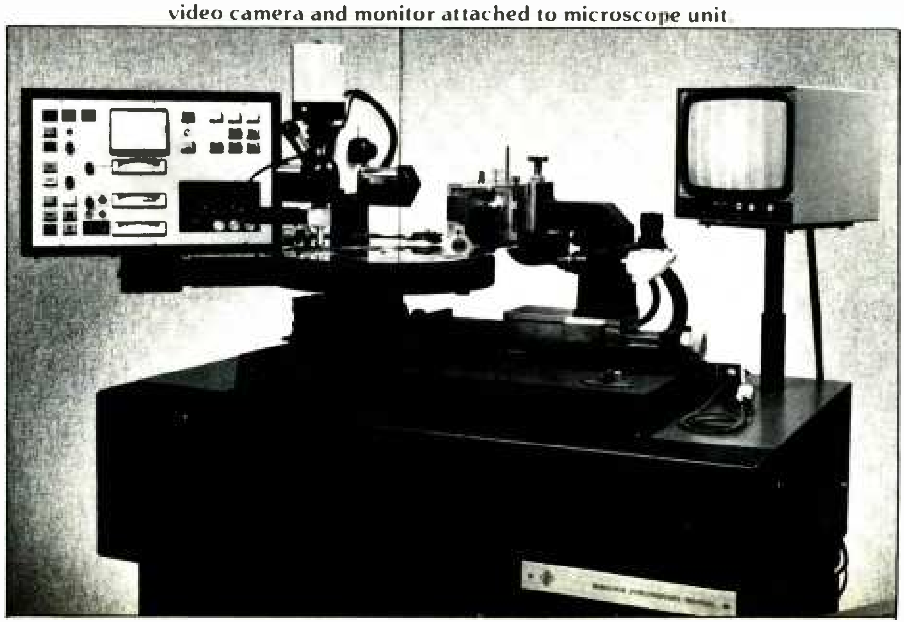
DMM utilizes the same, basic mastering technologies as conventional cutting facilities. Europadisk uses a Telefunken M15A quarter- and half-inch replay machine with preview heads linked to a Neumann SP-79B console. The latter features options for routing the cutting and preview signals through Neumann U473 combined compressor, expander and limiters, and Neumann OE DUO three-band equalizers, calibrated in 1-dB steps. A pair of JBL 250 TIs serve as monitor loudspeakers. For replaying digitally mastered tapes, the facility uses a Sony PCM-1610, and Studer DAD-16 digital delay to provide the delayed cutting signal.

After the signals pass through the transfer console and outboard gear, they are then fed to the lathe and cutting rack via a VAB-84 vertical amplitude limiter, which replaces the formerly used eliptical EQ. During the cutting stage, the Neumann VMS-82 lathe's computer translates the preview signal into vertical (groove depth) and lateral (groove width) signals for control of pitch and depth. In essence, pitch concerns itself with the "sum" of the left and right channel, while depth looks at the "difference" of the those same channels; each wall of the groove holds discrete left- and right-channel information.

By applying the sum/difference equations on the stereo preview signal, the frequency-dependent VAB-84 unit blends left and right information to mono, at a rate of 6 dB per octave below a dynamically determined frequency — usually 150 Hz. By acting dynamically, the VAB-84 only "blends" the signals when necessary to limit vertical stylus motion, thereby preserving the stereo image. (The use of fixed-frequency blending in conventional cutting is a serious drawback of the current, eliptical EQ.) The VAB-84 unit limits vertical excursions in steps of 0.4 mils, from 1.2 to 4.0 mils over the basic groove depth (usually 1.8 mils), with the low-frequency compression ratio fixed at 20:1. In addition, the VAB-84 insures longer playing time on the disk, and ultimately gives stampers a flatter backside, which significantly reduces rumble on the final pressing.

The signal then passes to the VC-82 Vertical Tracking Angle Processor, which electronically generates — and compensates for — the cutting-stylus angle. The development of the VC-82 results from the knowledge that the process of cutting a groove into a copper surface causes more stress on the cutterhead and stylus than mastering on lacquer. Therefore a new

A complete Direct Metal Mastering System comprises a Neumann VMS-82 cutting lathe, SAX-84 cutter head and stylus assembly, and optional video camera and monitor attached to microscope unit

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approach was sought to provide optimum cutting with reduced Frequency Intermodulation Distortion (FIM).

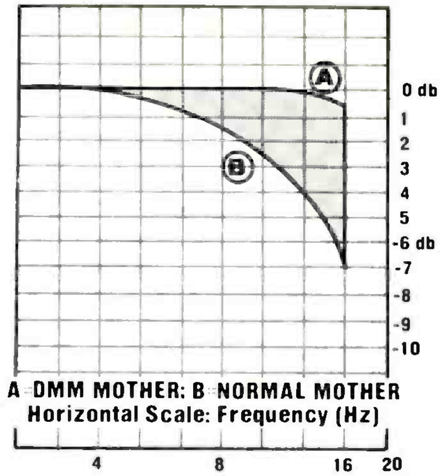
To understand the situation further, consider that, many years ago, the recording industry established an IEC standard tracking angle of 20 degrees (±5 degrees) off perpendicular for the sake of consistency in consumer cartridge design. Most lacquer mastering lathes cut at this standard 20-degree angle. Subsequently, most consumer cartridges also track at that same angle to decrease FIM within the groove. However, if a cut was made into copper at that same 20-degree angle, because of the vertical component of the force vector, the cutting stylus would literally eject from the groove, accomplishing only a surface scratch. As a result, Teldec adopted a zero-degree stylus cutting angle (90-degree angle to the record plane). This reduces the subsequent cutting force required to a sufficient level so that DMM uses normal cutter-amp power. (In other words, the rotational force of the turntable is directed against the face of the stylus, attempting to break it, rather than push it away from the record at a 20-degree angle.)

But, since a consumer cartridges still tracks at 20 degrees, the grooves need to be cut at that particular angle; otherwise playback will be heavily distorted. To compensate, the VC-82 adds a tracking-correction component to the cutting signal. Put another way, DMM electronically "tilts" the signal to a 20-degree angle, so that the resulting groove has a negative, 20-degree component of FIM introduced into the signal. When played back at the normal, 20-degree tracking angle, the two variables cancel each other, creating low FIM for the consumer.

Next in the signal path lie the BSB-84 high-frequency limiter — which, when necessary, limits high frequency transients for better trackability of consumer styli — followed by the CAT-84B left- and right-channel cutting amps, capable of delivering 550 watts per channel; the SAB-84B signal processing pre-amp; and the SEL-84B circuit-breaker system. By means of this latter unit, should the maximum temperature of the cutterhead drive coil exceed 200 degrees Celsius, or more than one ampere of current flow through the coils, the cutterhead automatically disconnects from the power amps, avoiding a possible cutterhead burnout.

For reference monitoring, the rack houses a PUE-84 pick-up cartridge equalizer amplifier, a MWS-84 monitor playback selector, a MSA-84 monitor equalizer, and two LOV-84B monitor power amplifiers, providing 185 watts per channel.

It should be noted that conversions are available for existing SAL-74B Cutter Drive Logic units to enable DMM cutting. In most cases, modifications to the circuit cards are needed, and the former TS-66 Tracing Simulator is replaced with the VC-82.



Comparison between recording losses at small wavelengths for DMM and normal mothers.

The VMS-82 Cutting Lathe

Since the Neumann VMS-82 lathe is used to cut into metal, its design necessitated a more sturdy format.

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First, a new carbon-vein vacuum pump acts much like a rotary engine with an eliptical chamber, and is serviced by a two horsepower motor. The pump extracts up to 15 cubic meters of air per hour, removing the dense, copper "chip" during the actual cutting process. Since the weight of the copper chip varies with groove depth and width, the vacuum pressure is regulated by the depth-control circuitry through a series of valves. A second purpose for the vacuum is to hold firmly the 24-ounce copper blank onto the turntable.

With conventional lathes, the force relating to stylus/disk interactions were thought to be expressed in terms of linear equations. But, due to the additional force needed to cut copper blanks, it was discovered that a quadratic equation — involving the square of the former linear equation — is actually correct. As a result, a new suspension and depth-control computer was developed to deal with the forces generated by the cutterhead as it cuts into the copper surface. Also, because of the forces generated, the new suspension system uses rigid, friction-free pivot bearings to support the DMM cutterhead, rather than the conventional ball bearings previously used.

Other features of the VMS-82 lathe

TECHNICAL SPECIFICATIONS FOR A DMM COPPER MASTER

The following data were obtained with a Shure V15-V pickup. In addition, these values refer to single channel recording at 280 mm (11 inch) recording diameter. The values for smaller diameters conform to playback theory.

Frequency Response: 20 Hz to 20 kHz, ±1 dB, referring to the entire cutting radius up to the innermost allowable modulation diameter. In contrast to lacquer masters, there are no losses applicable up to a wavelength of 10 microns. (See accompanying figure in main article.)

Frequency Intermodulation: Less than 0.3%. F_1 = 315 Hz; F_2 = 3.150 kHz; $U_1:U_2$ = 4:1; recording peak velocity V_{total} = 5 cm/s referring to RIAA equalization for 1 kHz.

Total Harmonic Distortion: Less than 0.5%; DIN 45503; F = 1 Hz; recording peak velocity V = 5 cm/s.

Difference Tone 2nd Order: D_2 is less than 0.1%; F_1 = 1 kHz; F_2 = 1.2 kHz; recording peak velocity V_{total} = 5 cm/s.

Signal-to-Noise Ratio: Greater than 70 dB, typical 75 dB; A-weighted DIN 45633, IEC-179; referenced to a peak velocity equaling 10 cm/s.

Spectral Distribution of Noise: See accompanying figure in main article.

Pre- and Post-Groove Echo Attenuation: Greater than 65 dB with 10 microns of land between grooves; 75 dB with 150 microns of land between grooves; F = 1 kHz; recording peak velocity V = 8 cm/s single channel.

Recording Losses for Small Wavelengths: See accompanying photography in main article.

Transient Response: As above.

DMM STANDARD DATA

Maximum Recording Level: Identical to lacquer.

Maximum Groove Width During Modulation: 180 microns.

DMM Cutting Stylus: Utilization period of 20 LP-sides per polishing; life expectancy is a minumum of 50 repolishings.

Storage Times: Copper Blanks: three months; after mastering: "unlimited."

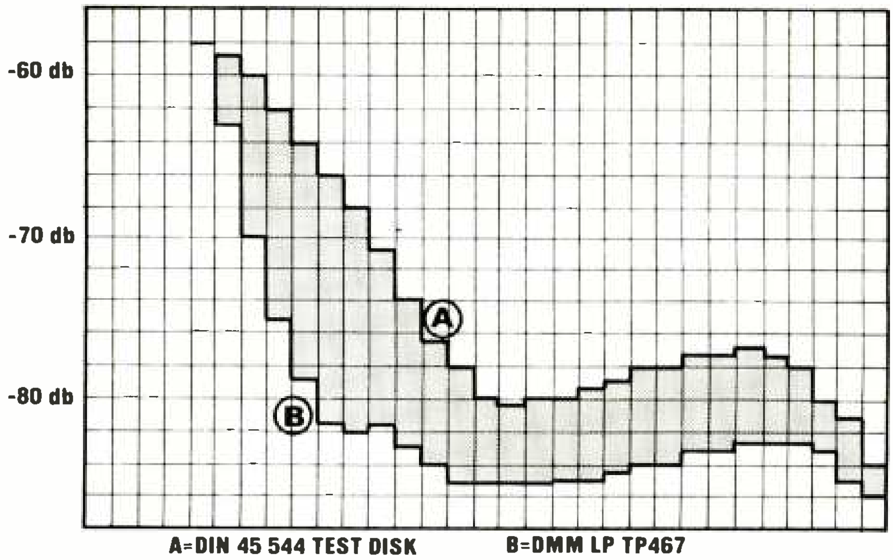
Matrixing: Number of stampers by direct plating: minumum 20; by three-step plating: equivalent to standard processing.

Groove Integrity: "Lack of horns is avoided in DMM cutting, therefore maintaining a high degree of groove integrity." □□□

For additional information circle #142

include a crystal-controlled turntable motor and hydrodynamic oil bearing, which provide, it is claimed, a 12-dB rumble improvement. In addition, a new DC-servo control keeps the turntable at a constant speed despite the increased cutting forces involved, and reduces dynamic flutter to an "unmeasurable" level.

Finally, a real-time digital processor was built into the lathe to control pitch and depth which, in turn, increases the number of grooves per side of an album. By means of a digital delay and a vertical dependent pitch control, the processor stores the preview signal of one-half a turntable revolution prior to cutting. According to magnitude and time relationships, the processor extracts the left-hand signal, the right-hand signal, and the base pitch (the basic groove width set by an engineer prior to cutting). These three signals are summed by the processor, and an intermediate peak pitch control signal (IS) calculated every 1/16th of a turntable revolution. The processor then calculates the least amount of land space necessary between grooves to ensure a successful cut. (In essence, the processor is "sampling" the signal, and then storing, summing, and calculating the next groove position. In many cases, space between grooves can be reduced to nearly zero.)



Comparison between spectral distribution of noise for DMM-produced test album and conventional vinyl test disk.

According to Russ Hamm, president of Gotham Audio, Teldec's agent for DMM distribution within the U.S. and Canada, the latest DMM development is a modified depth-of-cut board. Serving as a type of error-correction device, the new addition increases groove depth for wide-excursion lateral information. This use of the device will allow a playback stylus to ride in the normal position of the groove, rather than being driven out of the groove by the "pinch effect" during wide excursions, which is particularly problematic with low-quality tone arms commonly found on cheaper, domestic systems.

Since the depth corrections are only neccessary for short, lateral signal bursts, the devices described above allow the system to cut with a smaller basic groove depth, instead of the normal practice of cutting deep, basic grooves for security purposes. Smaller basic groove depth gives a greater dynamic range, and increased playing time.

Gotham Audio says that DMM conversions for a Neumann VMS-80 lathe — a new cutterhead suspension, a new turntable equipped with the enhanced vacuum system, and new turntable drive motor being required — costs approximately $40,000. A complete cutting system, comprising the VMS-82 lathe, SAL-84 cutting amps, and SX-84 cutterhead, costs approximately $140,000.

The SX-84 Cutterhead

One of the unique aspects of the DMM system design is the SX-84 cutterhead, which uses samarium/ cobalt magnets. Because "chatter" (a stick/slip effect) results when a lathe cuts any substance — be it lacquer, copper or even wood — Teldec found that ultrasonic modulations in the 80-kHz range were advantageous when cutting into a copper master, resulting in an extremely smooth groove wall. In turn, these modulations keep the mechanical loading of the cutterhead structure and electrical power requirements within reason.

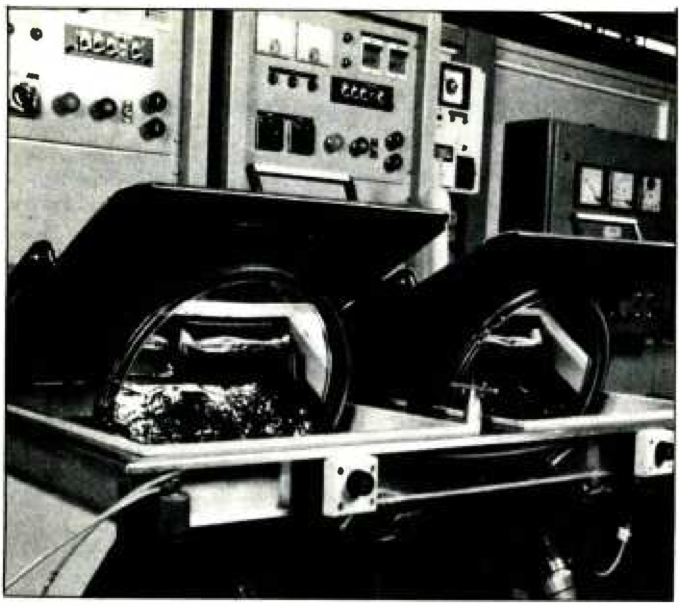
The stylus mount holds a diamond rather than a sapphire stylus found in most lacquer cutting heads. Also in contrast to conventional cutting styli, the DMM stylus contains no burnishing facets; instead, it is a "feathered-edge" jewel. The burnishing facets, which are required to cut lacquer, result in a "self-erasure" and "blurring" of high frequencies. Moreover, the DMM stylus cuts without the heat needed to lower the surface noise for lacquer cutting.

With conventional lathes, melted lacquer can cling to the heated sapphire stylus, meaning that — under the worst circumstances — a mastering stylus could last only two passes per session. With DMM, the diamond stylus cuts copper with a highly polished surface, guaranteeing a high signal-to-noise ratio, and results in a longer stylus life of at least 20 passes per diamond polishings, and sometimes as high as 100 cuts. (Diamonds can be repolished at least 50 times.)

A direct cause of plating problems and high-frequency loss when mastering into lacquer are horns, partially caused by a heated stylus, but alleviated to a certain extent by burnishing facets. (The horns look like ridges on the groove edges.) With a feathered-edge, diamond DMM stylus, on the other hand, these horns are said to be eliminated in DMM mastering. Plating plants no longer have to remove these artifacts by polishing and buffing the mothers, a process that can be destructive to the overall sound quality.

The combination of two primary features — cutting into metal and the control system of the lathe — allows the grooves to be layered more closely together. With DMM, up to 40 minutes of material can be recorded on one side of an album while still maintaining inner-diameter high-frequency response, and recording levels equal to those of lacquer-based technologies.

One of the main advantages offered by DMM technology is the ability to control the production of the copper blank. It is interesting to note that melodia, the Russian state-owned record company, is reported to have adopted DMM technology to free itself from having to depend on the world's primary supply of lacquer material. Inherently, the electroforming process used to produce the copper blank results in a perfectly homogeneous material, eliminating all impulse-type noises. In addition, there are no cutting "swishes" caused by unmixed lacquer resins, nor lacquer "pinholes" (air bubbles) or imbedded dust particles. When Europadisk manufactures its amorphous copper blanks, they are stored at zero degrees F, and have a life expectancy of three months at that temperature. After mastering, the disks have an "unlimited" shelf life,



During preparation of the DMM copper blank, a stainless steel substrate receives a 4-mill coating of pure copper (*left*). At the matrix-preparation facility, a nickel stamper is "grown" directly by electrolysis from the copper mother.

and the amorphous copper surface returns to its original, crystalline state.

Licensing and Cost Advantages with DMM

To be licensed for DMM technology involves the signing of an agreement with Teldec in order to use the company's technology and trademark, and the payment of a one-time, start-up fee. In the U.S., Europadisk is protected by the Teldec trademark, and is licensed to use the DMM patents for record manufacturing.

In mid-September, a second DMM system had been installed at Sterling Sound, New York City. Owner Lee Hulko states that he has purchased a Neumann SAL-84 cutting rack, VMS-82 lathe and SX-84 cutterhead, and that his facility is licensed for disk mastering only; he will obtain copper blanks from Europadisk. In addition, a third DMM system, the first such system on the West Coast, is scheduled for delivery by mid-October to Amigo Studios, North Hollywood, CA. Owner Chet Himes says that his existing Neumann disk-mastering system will be upgraded to handle DMM technology by incorporating enhanced electronics to the existing SAL-74 cutting rack — bringing it up to the SAL-84 circuitry — plus adding a new VMS-82 cutting lathe and SX-84 cutterhead. Amigo will be licensed to master DMM copper masters, with Sheffield Matrix Labs plating the masters for album production.

Other facilities, CBS Records and RCA — both located in New York — are reported to be testing the DMM process. The first major pressing plant to meet the approval of Teldec DMM standards, Warner Bros. Speciality Record plant in Pennsylvania, is geared up to begin manufacturing DMM disks in the near future.

Europadisk actively employs a combination of DMM licenses, including one for the disk mastering process (a $25,000 start-up fee to Teldec), for the plating process and for the record-pressing process. The fee for the record pressing process is based on the short-term cost savings effected by using DMM technology. Any of the licenses, or combination of the licenses, may be obtained individually from Teldec. In other words, if one facility operates under one license, it does not necessarily need another to operate under the DMM trademark. However, to place the "DMM" logo on an album jacket, the project must be mastered, plated, and pressed at DMM-licensed facilities.

After mastering a copper blank at a DMM-licensed facility, the client has the option of either finishing his product through final pressing under the DMM license and trademark, or he may go to other, non-DMM licensed facilities to complete the album, thus foregoing the identifying logo.

For this reason, Europadisk has two ways of plating a DMM-mastered disk. The first way is to take the DMM master and keep the processes "in-house" through plating and pressing. However, if the client wants to take his DMM master to another non-DMM licensed facility, he must first let Europadisk create a "sub-father," from which the subsequent nickel mothers may be taken to another facility. The copper master never leaves Europadisk, which is a stipulation within Teldec's contract concerning any DMM-licensed facility.

When speaking of cost advantages of DMM, three major points can be specified: First the exisiting plating and pressing facility will have no major capitol investment when plating and pressing DMM products. In essence, the facility may use the same equipment that it currently possesses.

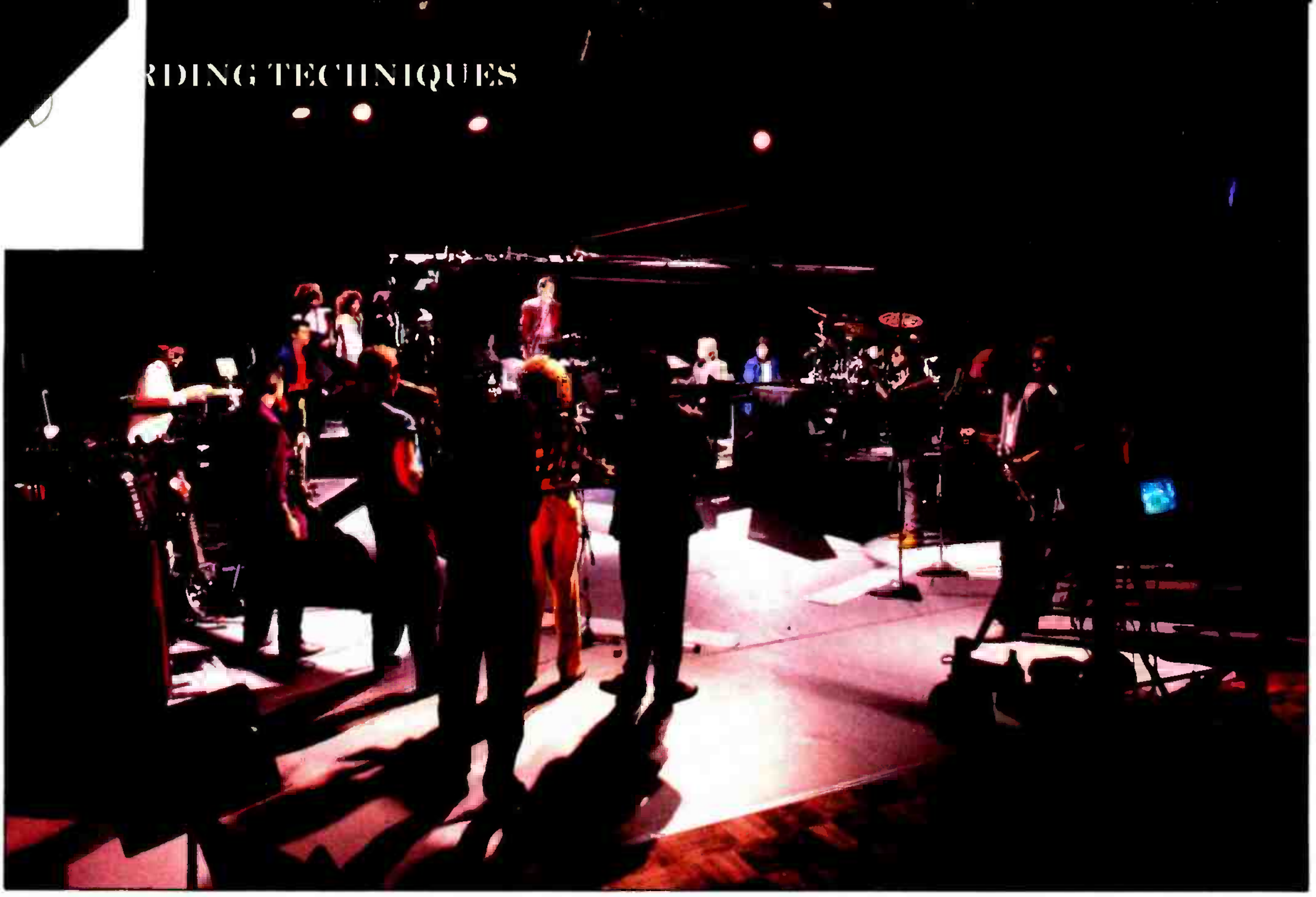
Secondly, DMM technology eliminates the difficult, and expensive, silvering process, which is the major source of quality defects and a primary cause for most recuts. It is estimated that a large plating facility could save up to $100,000 per year by eliminating the silvering stage.

Finally, concerning jazz, classical, and other limited release product of under 50,000 pressings, stampers can be produced directly from DMM masters, resulting in a typical savings of $100 per side.

Perhaps the cost advantage explains why DMM technology is taking hold so well in Europe. Noble is the goal which enhances the ability to limit, or actually eliminate, some of the extraneous problems related to record manufacturing. To many observers, DMM is the natural "evolutionary" step for black vinyl records. And it looks set to offer the quality edge over lacquer mastering for many years to come. ■■■

Acknowledgments

My thanks to Joe Gastwirt, independent cutting engineer, formerly of the JVC Cutting Center, Hollywood; Gary Rice, co-owner of Future Disc Systems, Studio City, Calif.; and Michele Stone of KM Records for providing background information; and Russ Hamm of Gotham Audio for technical data on DMM.



Photography by Barbara Kraft

DIRECT TO TWO-TRACK

Capturing the Energy of a Live Session

by Jeffrey Weber

The guitarist was in the process of ripping off a great solo; it was burning. Everyone in the control room silently held their breath. Some wanted to go home after nine hours of guitar solos. Others wanted the soloist to come through with this last four-bar section, so that they could assemble the various other parts of the solo, recorded earlier that day. Continuity was an obvious concern. When the solo part was completed, the rush of expelled air was evident; it was a wrap. The producer noted he was only three hours over budget. It could have been worse. He began to smile, knowing the evening was at its conclusion. The guitarist, after listening to the segmented playbacks with satisfaction, started to tear down. Until the solo was put together during remix, it would be difficult to get the sense of an emotional flow, but the player remained confident that, when pieced together, his solo would be seamless.

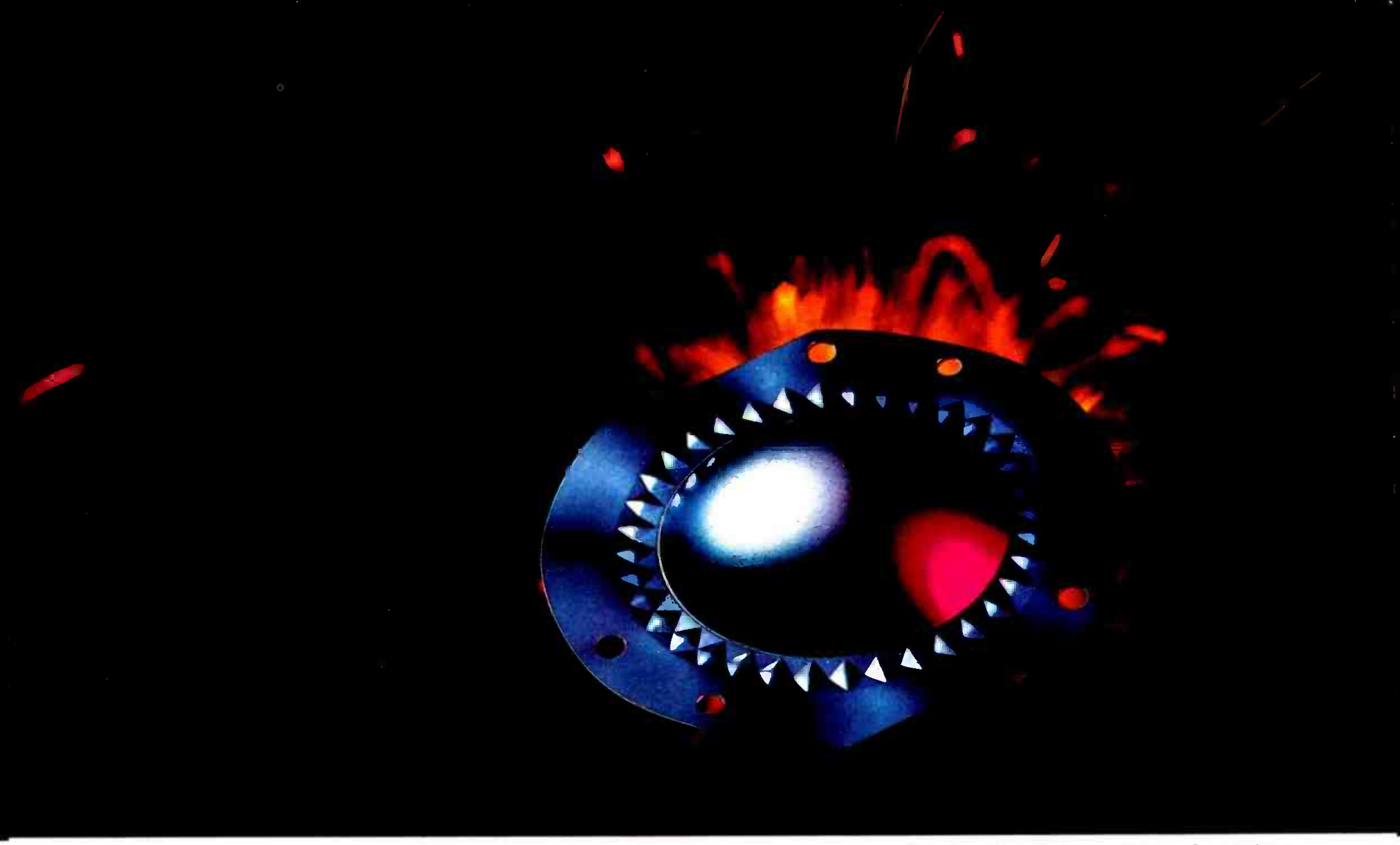
The producer checked and noted that he was scheduled to record more overdubs the following day. A wave of fatigue hit him as he gathered his gear and walked down the hall towards the exit sign. The adjacent studio was open, and he noticed some familiar faces. He stuck his head in the control room and saw it packed with musicians listening to a playback. There was an unusual energy coming from the monitors as well as from the band in the room — they were up, smiling and intense. He learned that what he was listening to was a live two-track digital recording, and that the entire album had been completed over the last two days. A second wave of fatigue hit him. As he nodded goodbyes to his friends, he marveled at the punch and vitality of what he heard, and wondered about the process, especially the part about completing an entire album from start to finish in just two days of session time.

Everyone, at one time or another, has experienced the fatigue of a multitrack project. The above, semi-fictional — or should I say semi-factual — scenario represents two distinctly different theories of recording music. Having recorded both in the multitrack medium and live to two-track, I am convinced beyond any doubt that recording direct to two-track is more emotionally satisfying, sonically superior and financially more feasible than a multitrack effort. This article will explore the advantages, applications and common resistance to

— the Author —

Jeff Weber is an independant producer specializing in live two-track, digital and direct-to-disk recording. He has over 32 projects to his credit, almost all of which were recorded live to two-track. This year alone, in addition to the Scott Page sessions described in a companion sidebar, he has produced albums for McCoy Tyner and Jackie McLean, guitarist Grant Geissman, pianist David Benoit, guitarist Pat Kelley, and saxophonist Joe Hackney.

In the past, Weber has made albums with a wide range of artists, including Toni Tennille, Tom Scott, Maynard Ferguson, Nancy Wilson, Chick Corea, Stanley Clarke, Lenny White, Tim Weisberg, Mongo Santamaria, Lalo Schifrin, Freddie Hubbard, Patrice Rushen and Kenny Burrell, among others. His albums have received four Grammy nominations.



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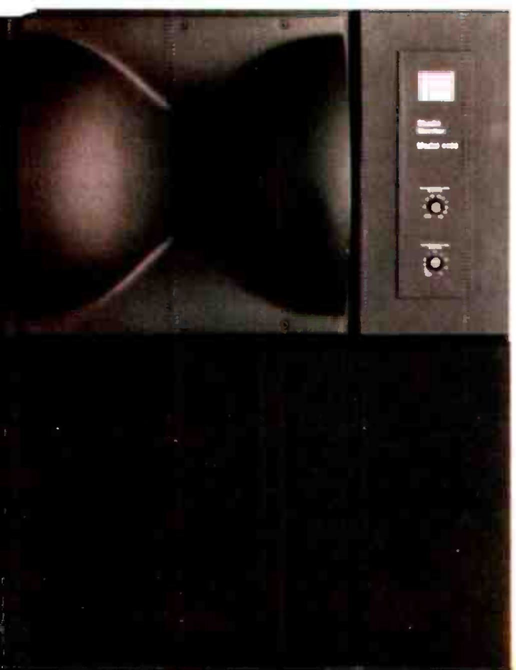

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The author's digital sessions include *Desire* by Tom Scott, mastered to Soundstream . . .

this the direct-to-two-track recording philosophy.

The main goal of live, two-track recording is not simply to make another record, but to create an *event.* I have found that the emotional bond between a performer and the listening audience creates a desire for the listener to buy the artist's recordings. When that listener attends a live concert, his or her aural sense is augmented by sight; it is often this latter factor, as expressed by body motion and facial expression, that supplies much of the emotion to a song's rendition.

The function of live, two-track recording is to achieve the heightened emotional sense of a live concert without the aid of sight. The stores are filled with thousands of records one can *listen to,* but I wonder how many of these one can *feel.* When vocalists and instrumentalists play together, the bilateral merging of energies creates a musical force that is almost euphoric. Each player serves to encourage the other and that, in turn, continually fuels the energy of the session. It is that type of contagious, emotional energy that is so often lacking in most recordings. I have noticed that a female singer, for example, will actually change the phrasing of a line, depending upon whether she sings to pre-recorded tracks through headphones, or surrounded by musicians. The emotional intensity of live recording creates a belief — and if the listener *believes* the song and *believes* the singer — then the listener becomes *involved.* This involvement translates into sales.

. . . *Night Plane* recorded to 3M DMS/Sony PCM-1610 . . .

Sonic Quality Advantages

While added emotion is by far the most important attribute of live, two-track recording, the sonic superiority of the medium enhances the emotional dynamics. With multitrack recording, the two-inch reel is mixed down to a quarter- or half-inch master. Initially the tracks are detrimentally affected by being run back and forth across the tape heads while building a tune. To this is added the fact that analog tape does not have the greatest high-frequency memory. Over the course of a week or two, the harmonics and subharmonics of an instrument or vocal track have a tendency to soften and become "cloudy." And it is these harmonics that give character and personality to an instrumental or vocal. Dynamics are also affected in this fashion: peaks become "round," and the "punch" is softened. Not to be ignored is the ever-present tape hiss which, in varying degrees, accompanies all analog tape, and increases during the subsequent mixdown and playback sessions. Although such problems will disappear with the widespread use of digital multitrack recorders, at this time their presence cannot be deemed a major factor.

The same problems that beset a two-inch reel also invades the effectiveness of the quarter- or half-inch master. The use of a digital two-track during the mixdown process alleviates such second-generation-type problems.

When recording live to two-track, the first noticeable improvement is the absense of the two-inch multitrack reels and their attendant problems. If a digital two-track recorder is used as the primary storage medium, all normal tape problems are eliminated.

Because the console feeds the two-track directly, I can record to a variety of formats simultaneously, using separate pairs of master outputs from the console. I like to give myself a number of options, and record to analog quarter-inch, 15 ips; quarter-inch, 30 ips; half-inch, 30 ips; and, of course, digital — either JVC VP-900 or Sony PCM-1610 processors with companion U-Matic VCRs. Based on the music, we then decide which medium best conveys the emotional impact we're trying to get across.

Believe it or not, it's not always the digital recording we choose in the end.

. . . Toni Tennille's *More Than You Know*, recorded to JVC . . .

I recently did a live-to-two-track album with Toni Tennille, titled *More Than You Know,* for which we felt that the analog master beat digital in terms of richness. For that record, a collection of standards from the Thirties and Forties, we wanted a more romantic feeling which, we considered, was best conveyed by analog tape.

Microphone Techniques

On many live-to-two-track projects, I've found digital recording and tube microphones to be an ideal combination. Tube mikes have a wonderful accuracy and fullness and, when combined with the digital medium — which has been accused of being harsh and strident in the top-end — the results can be punchy, tight, accurate and listenable. If you're recording a digital project with a lot of horns, for example, tube mikes can help you compensate for any unnatural stridency. Given the subtle nuances of the music interacting with the microphones and the acoustics of the room, it's nice to be able to choose among several different tape formats after the album has been recorded. This is one area where live-to-track can offer more flexibility than multitrack.

The setup for an actual live-to-digital recording date is similar in most respects to that for a multitrack ses-

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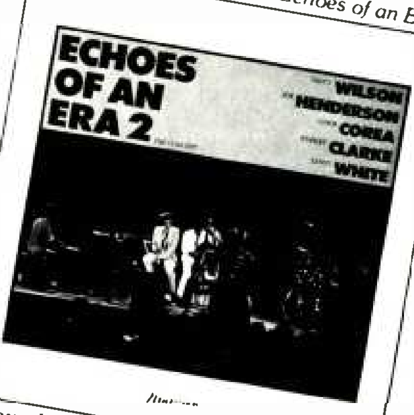

. . . *Echoes of an Era 2* to Soundtream two-track.


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Digital technology holds forth the promise of theoretical perfection in the art of recording.

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Naturally, the ultimate success of digital hinges on the integrity of the engineer and the recording process. But it also depends on the correct choice and placement of microphones, quite possibly the most critical element in the recording chain. This can make the difference between recording any generic instrument and a particular instrument played by a specific musician at a certain point in time.

The exactitude of digital recording presents the recordist with a new set of problems, however. The sonic potential of total accuracy throughout the extended frequency range results in a faithful, almost unforgiving, recording with no "masks" or the noise caused by normal analog deterioration. As digital recording evolves, it places more exacting demands on microphones.

Ribbon microphones are a natural match for digital because they are sensitive and definitively accurate. *The warm, natural sound characteristic of a ribbon mic acts as the ideal "humanizing" element to enhance the technically perfect sound of digital.*

Beyer ribbon mics become an even more logical component of digital recording due to an exceptional transient response capable of capturing all of the nuances and dynamic shifts that distinguish a particular performance without the self-generated noise and strident sound generally attributed to condenser mics.

The range of Beyer ribbon microphones.
From left to right: M 500, M 160, M 260, M 130

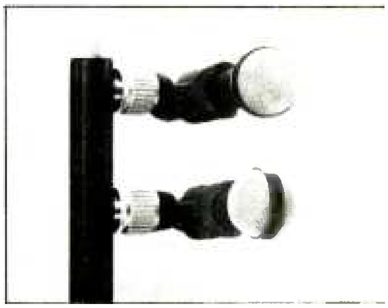
Beyer is committed to the concept of ribbon microphones. *We manufacture a full range of ribbon mics for every vocal and musical instrument application.*

The Beyer M 260 typifies the smoothness and accuracy of a ribbon and can be used in stereo pairs for a "live" ambient recording situation to record brass and stringed instruments with what musicians listening to a playback of their performance have termed "frightening" accuracy.

Because of its essential double-ribbon element design, the Beyer M 160 has the frequency response and sensitive, transparent sound characteristic of ribbons. This allows it to faithfully capture the sound of stringed instruments and piano, both of which have traditionally presented a challenge to the engineer bent on accurate reproduction. Axis markers on the mic indicate the direction of maximum and minimum pickup. This allows the M 160 to be used as a focused "camera lens" vis a vis the source for maximum control over the sound field and noise rejection.

Epitomizing the warm, detailed sound of ribbon mics, the Beyer M 500 can enhance a vocal performance and capture the fast transients of "plucked" stringed instruments and embouchure brass. Its diminutive, durable ribbon element can also withstand extremely high sound pressure levels.

The Beyer M 130's bi-directional pattern enables the engineer to derive maximum ambience along with clean, uncolored noise suppression. Two M 130s correctly positioned in relationship to each other and the source can be used as part of the Mid-Side miking technique. The outputs from the array can be separated and "phase-combined" via a matrix of transformers to enable the most honest spatial and perceptual stereo imaging — sound the way we hear it with both ears in relationship to the source.



Given the high price of critical hardware used in digital recording, the relative price of microphones is nominal. *Realizing that microphones are the critical sound "source point," no professional can allow himself the luxury of superficial judgements in this area. Especially when one considers the value of ongoing experimentation with miking techniques. For this reason, we invite you to acquaint yourselves with the possibilities of employing Beyer ribbon technology to enhance the acknowledged "perfection" of digital recording technology.*

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— continued from page 77 . . .

DIGITAL TWO-TRACK

sion. Perhaps the biggest difference is that sound leakage presents less of a problem — it's natural, and I'm inclined to go with it if it doesn't intrude. Acoustic isolation in a multitrack situation is of greater importance, since if facilitates the movement of each instrument into an artificially created musical perspective with all of the other instruments. In live-two-track, however, that sound perspective is created as it is recorded, with the result that the need for microscopic isolation of individual tracks disappears. I still favor moderate isolation though, in order to give definition to each instrument; the listener should be able to pick out any particular part, and follow it all the way through the song.

As with any recording session, miking techniques for the live-two-track format is based on the nature of the music itself. On a rock project, I'll multimike the drum kit, for example, to capture the energy and attack of the sticks hitting each different drum head. On a large orchestral date, if we're mainly after a rich, round texture, some overheads and a few spot

SESSION EXAMPLE: SCOTT PAGE AND MEMBERS OF TOTO RECORDING A DIRECT-TO-DIGITAL SESSION AND SIMULTANEOUS VIDEO SHOOT AT GROVER HELSLEY RECORDING.

Jeff Weber's most recent project involved the production of a live, two-track digital and 46-track analog recording with saxophonist Scott Page, during a simultaneous six-camera video shoot. Page, a veteran of numerous tours with Toto, Duran Duran and Supertramp, has developed what many would consider to be a unique perspective for creating music. Along with his writing and playing partner, Tony McShear, Page felt the need to create an "audio image that would be totally integrated with a visual image." The pair perceived that in most Music Videos there is often a distinct separation between what takes place *visually* and what occurs *aurally*. Page and McShear wanted to create an in-studio environment conducive to inspiration and "spontaneous combustion," as they describe it, rather than total acoustic isolation and cleanliness.

The goal was to create a musical moment of unparalleled "heat" and to capture it in all formats — audio and video. Having heard Weber's previous two-track recordings, in addition to having worked with him in the past, Page and McShear believed that the producer would be able to capture and preserve the audio "heat" they desired.

For the audio/video recording, a lighting gantry and stage area were assembled in Studio A at Grover Helsley Recording (the old RCA Complex, Hollywood). The band was set up on various levels of risers in a semi-circle surrounding Page, and a complete stage-monitor system supplied by Schubert Systems Group allowed the band to avoid the use of headphones to hear one another. The band consisted of Scott Page, saxophone and recorder; Tony McShear, vocals, guitar, keyboards and percussion; Jeff Porcaro, drums, Simmons Electronic Drums and E-mu Systems Emulator II; Bob Glaub, bass; Lenny Castro, percussion and Simmons; Steve Lukather, guitars; Cal David, vocals, guitar and

At the Neve console during the direct-to-digital session (*L-to-R*): producer Jeff Weber, head engineer Franz Pusch, Paul Ray, Shep Lonsdale and Dan Voss, Jnr. (in rear). Pictured *right* are the featured artist, saxophonist Scott Page, and The Brunettes vocal backing group.

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SESSION EXAMPLE – continued . . .

electric sitar; Bill Payne and Chris Boardman, keyboards; The Brunettes, background vocals; and the Heart Attack Horns.

The seven-song recording session was completed in two days. A total of three consoles were necessary in the control room to handle the numerous vocal, instrument and direct-inject inputs: the studio's main Neve Model 8078 board, plus a pair of Schubert Systems Group custom 24-in/eight-group consoles used for subgrouping keyboard and drum inputs before routing to the Neve console. The live stereo mix was recorded on a JVC VP-900 digital processor and U-Matic VCR supplied by CMS Digital of Altadena, CA, while backups were made to a Threshold-modified Nakamichi DMP-1000 EIAJ-Format digital processor and half-inch VCR, an Ampex ATR-102 quarter-inch, 15 ips two-track, and a half-inch VHS HiFi format.

Five engineers were utilized on the session, led by Franz Pusch: Shep Lonsdale, Paul Ray, Chris McNary and Jeff Cowan. The associate producer for audio was Dan Voss.

The two-track mixes are intended for a eventual Compact Disc release, with the 46-track analog tracks destined for later remix against the edited video. In addition to the normal studio microphones, the mikes attached to each video camera were routed to various multitrack inputs; these will be integrated into the final audio mix during video post production. □□□

JVC VP-900 digital processor and U-Matic VCR supplied by CMS Digital.

DIGITAL TWO-TRACK

mikes may be all that's necessary to cover the drums.

Capturing room ambience is one of the prime objectives in this type of live recording. I like to rely on two AKG C-12s or C-12As placed way up high in the room. I often combine the output from these mikes with a natural chamber, in order to achieve the ideal sense of room ambience.

A live-to-two-track date generally will take up every available console input. Along with the microphone inputs, some instruments are taken direct, just as they would be on a multitrack session. Each outboard effect will be put on its own channel and bussed to the correct instrument at the correct time. Occasionally, I'll have to use a separate submixer, but I can usually get by with a 36 × 24 board. All mixing is done live while we're recording, most settings and levels having been determined in advance, and we know where there will be solos and other sections that have to be pumped up.

After a live take, we'll all get together in the control room for a playback, and make an evaluation on perspectives and relative levels. The musicians will see where they made their mistakes on our side of the glass. Within a couple of takes — never more than four — we've got it on tape. The performance we capture on two-track is, to my mind at least, far more vital than what one can get from isolated, individual multitrack overdubs. A great *performance* is our perfection.

Financial Advantages

Apart form the sonic and artistic advantages of live, two-track recording, the dramatic financial advantages are romantic enticements to any record company. As a conservative estimate, an average multitrack project takes 10, five-day weeks to complete. With studio and engineer costs averaging $2,000 per day, it would take $100,000 to complete the project, a price that does not include tape stock, musicians, fees, producer fees, rentals, arrangements, cartage, etc. Two-inch multitrack tape running at 30 ips costs $600 per hour — if you can find it for only $150 per reel!

With two-track digital recording, on the other hand, tape costs are $35 per hour of recorded music. Each record project requires only two days of studio time: we spend about eight hours each day recording four to five tunes. Studio time, tape costs and engineer fees rarely exceed $5,000 for the entire album, and yet nothing is sacrificed in terms of quality. Every aspect of the recording is state-of-the-art.

Orchestral and Jazz Sessions

Live, two-track recording is an obvious asset to classical music, since orchestras rarely record by sections.

The improvizational nature of jazz also lends itself easily to the medium. What most people do not consider is how the direct-to-two-track medium can serve the raw energy of rock and roll. I have found that live, two-track recording of rock sessions have a punch and a vibrant raw edge that multitrack simply cannot achieve. People dismiss two-track rock by stating that there is a loss of creativity and experimentation unless multitract techniques are used. Actually, there is plenty of room to experiment; it just takes place *before* the recording date rather than during the mix, as is usually the case with multitrack projects.

An incredible amount of advance planning goes into a live-to-two-track record. In order for any music to meet my recording criteria, it must have a wonderful left-to-right stereo spread, a sense of depth — with the instruments placed in front-to-back perspective — and a carefully arranged frequency spectrum. I therefore begin a project by going through the arrangement for each song, and create a schematic diagram showing where each instrument is going to appear in the overall sonic perspective. Such a schematic forms the basis for predetermining panning, EQ settings, outboard effects, echo, etc.

As part of the preparation process, I'll often bring the artists into a small studio, and do some preliminary recording, just to see how things sound. This gives the players a great warm-up for the actual date, and lets us do a lot of experimentation. If we feel a guitar sound is lacking something, we can try any effect or piece of outboard gear to get what we want. Everything is carefully noted, and the effect will be added to the instrument during the final recording, rather than later during the mix. Basically, anything that can be done "post" tracking can also be done "pre" recording.

Any excess number of parts can be sampled and synchronized with today's modern equipment or, if one needs four keyboard parts, four keyboard players can be utilized — it only serves to heighten the energy of the session.

It used to be that certain effects, such as inverted looping or reverse cymbal effects, just could not be done in the live, two-track format. With sampling keyboards and other pieces of recording technology, virtually any effect for any type of music can be prepared for live-to-two-track sessions.

Besides studio recording, live concerts can also be recorded and, if there is film or video being shot simultaneously, a direct two-track feed can be sent to the video truck, thereby virtually eliminating post-audio sweetening. Live TV broadcasts or simulcasts immediately become state-of-the-art in terms of audio, and necessary audio sweetening costs diminish.

The unnatural breakdown of music into its component parts and the eventual re-assembly of those parts in a layered, artifical manner simply does not have to be done. Working direct-to-stereo means that music can be recorded in 95% less time, for 75% less money, and can easily sound up to twice as vibrant as a multitrack recording. At the end of each two-track recording I notice three natural occurrences: I do not hate the players (and *vice versa*); I still love the tunes; and we have all experienced a great deal of enjoyment. What more could you ask of a session? ■■■

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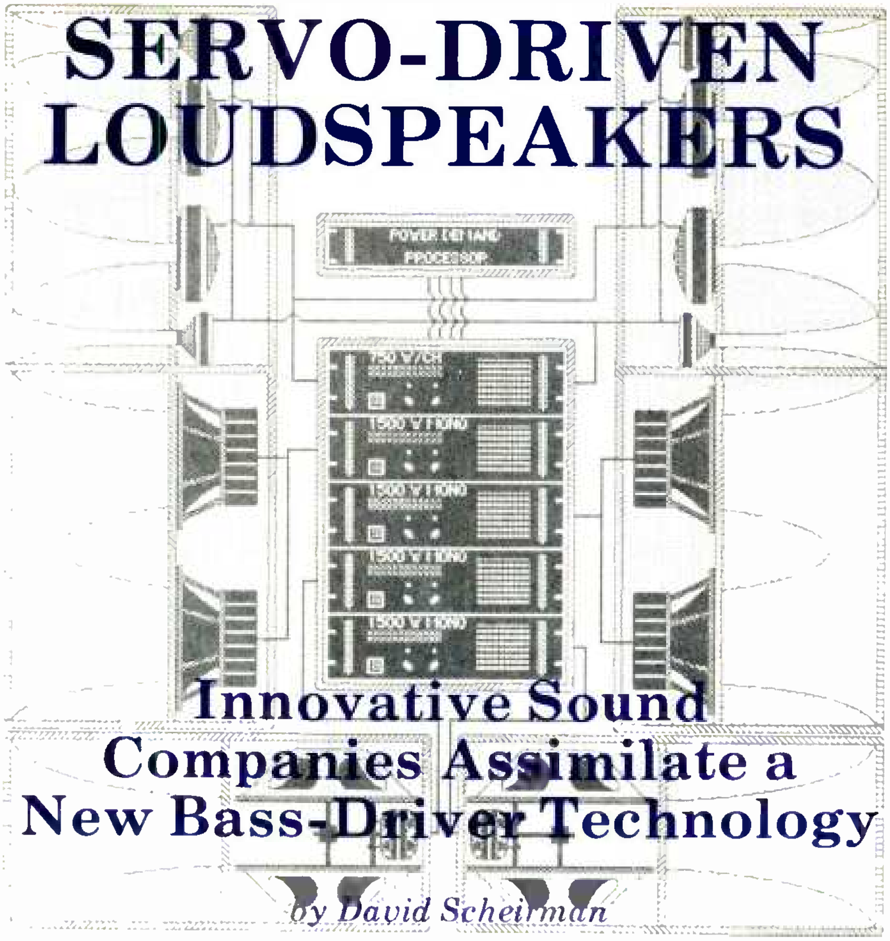
SERVO-DRIVEN LOUDSPEAKERS



Innovative Sound Companies Assimilate a New Bass-Driver Technology

by David Scheirman

All photography by David Scheirman

Few inventions traditionally seem to bring skeptics right out of the woodwork. In any industry, be it audio or aeronautics, it is usually difficult to change the direction of the mainstream of progress in a given field. Established, conservative factions have been known to denounce new technologies when they first arrive upon the scene. (In the early 1900s, a noted English scholar staked his reputation on a published statement that man would never fly a heavier-than-air machine . . . it was mathematically impossible. Less than a year later, the Wright Brothers flew at Kittyhawk.)

One of the most challenging problems faced by portable sound-reinforcement design has been the development of low-frequency generators that are compact enough for easy travel, yet offer audio fidelity rivaling traditional, large-bass cabinetry.

Two decades ago, the RCA-type folded-horn "W-bin" represented the commonly accepted compromise between full-length, gigantic bass horns and the practical considerations of mass production. As transducers were developed that could handle greater amounts of power input, direct-radiating bass enclosures came into their own. Today, a survey of typical touring and installed sound systems will show a mixture of different types of low-frequency cabinetry to be in use. One thing seems probable, however: the design parameters of paper-cone, voice-coil transducers have nearly approached their working limits with known materials. Since bass enclosures comprise the greatest amount of mass in any given full-bandwidth sound reinforcement system, a further reduction in the size of today's standard systems will require a different technology.

New Technologies

Younger, aggressive sound companies are often the first to try out unproven technologies. While a majority of rental firms attempt to duplicate those systems made popular by industry "giants," and model their own efforts after what has gone before, some entrepreneurial companies are willing to take a chance on something totally new, when it appears that an innovative tool, product or process may be a real problem-solver. In the sound-rental business, this usually means regional, "new-generation" companies located in a major market area, and competing for a finite amount of available sound system contracts.

Pace Sound & Lighting (New Orleans) and Pyramid Audio (South Holland, IL) are two such companies. Both firms are directed by young, aggressive managers intent not only upon gaining a lion's share of their respective regional markets, but who look forward to national touring and installation work. These companies are but only two of the growing number of regional rental concerns trying to produce the maximum amount of audio output from the lowest amount of loudspeaker system mass. Smaller systems mean lower labor and transportation costs, easier storage, and a competitive edge.

Pace Sound and Pyramid Audio have no relation to each other in a tangible business sense, yet the two companies have something in common. Each has developed its own propriety mid/high cabinet that is a modular unit intended to be used with a new bass technology: servo-drive loudspeakers. Before we look at the two firms and their respective needs and applications, a brief look at this new type of loudspeaker device may answer some questions about what it is, how it works, and why these regional sound companies are staking their future growth on it.

Servo-drive Speakers: A Brief History

Before the current, familiar loudspeaker was invented, only acoustic horns existed for sound "reinforcement." Early inventors mated a small compressor to the acoustic horn, and achieved truly high sound pressure levels from a very small device. This approach was left in the dust, however, by the rapid acceptance of early voice-coil loudspeakers. The earliest loudspeakers were surprisingly sensitive (due to their relatively small moving mass), and enclosures were typically quite large. (Have you ever closely studied a 50-year-old sound system behind the movie screen of an abandoned theatre?)

As heavy-duty loudspeakers were developed, a greater amount of moving mass was involved in the speaker cone. The efficiency of the speaker unit dropped, requiring more power amplification. Larger voice coils were developed that increased input power capacity. Bass reflex cabinets came into vogue. Nearly everyone had forgotten about the old, experimental compressor/horn hybrid devices. Nearly everyone except Thomas Danley, an electro-acoustic researcher with Intersonics, Incorporated, an Illinois-based firm that has developed a "sonic levitator" used during on-board Space Shuttle experiments, among other things. It is the company's Sound Physics Division that has developed the SDL® line of loudspeakers currently being marketed.

"Truly deep bass requires either a big enclosure, or a large cone area," explains Danley. "If you increase the moving mass, you are going to move more air. In today's new sound systems, smaller boxes make more sense. What we have done is take the old idea of coupling a 'compressor' to an acoustic horn. We move the speaker cones with servo-drive, brush-commutated low-inertia motors that are mechanically connected to the heavy-duty cones with a driveshaft and a rotary-to-linear motion converter." Figure 1 shows the basic principle of the SDL units.

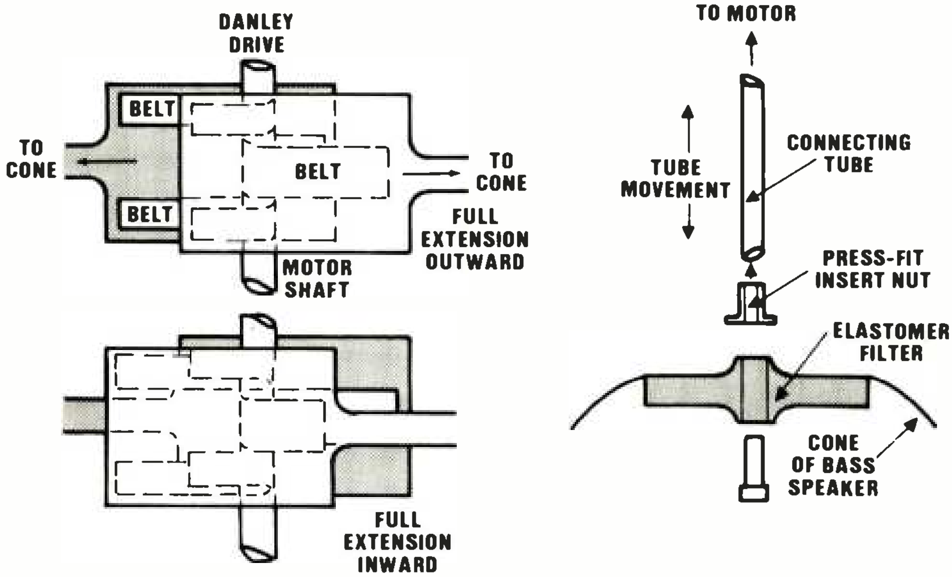


Figure 1: A rotary-to-linear converter drives the speaker cones in the Intersonics SDL units. An electric motor's power is transferred to the cone with a "drive shaft."

Intersonics' SDL Series sub-bass systems theoretically have enough power to offer more than twice the amount of cone excursion as traditional, voice-coil loudspeakers. As the cone excursion increases, so does the unit's acoustic output. Company engineers estimate that only one-eighth the enclosure volume is taken up when compared with a traditional bass loudspeaker enclosure for a given frequency range and acoustic output.

"Our TPL-3, with its two, 15-inch speaker drive units, is able to replace as many as four large horn-loaded double-15 bass cabinets," Danley offers. "The potential space and transportation cost savings for the sound-reinforcement industry should be obvious.

"The TPL-3 measures only 45 by 45 by 22½ inches, and will develop up to 116 dB at 22 Hz with a 32-volt RMS input when measured at three feet from the effective source center." A sectional view of the Model TPL-3 is shown in Figure 2.

Pretty serious claims, indeed. And claims that could have serious consequences for traditional bass cabinetry, if the servo-drive technology is sonically acceptable to discriminating sound reinforcement companies. I have had the opportunity to listen to two different sound systems featuring the new technology. While some well-known sound companies have used the units on a trial basis for special subwoofer effects — notably, Stanal Sound with Neil Diamond, and Sound On Stage with Huey Lewis and the News — both Pyramid Audio and Pace Sound rely on the units as the *sole* reproduction units for *all* audio frequencies below 80 to 100Hz.

Case Study #1: Pyramid Audio

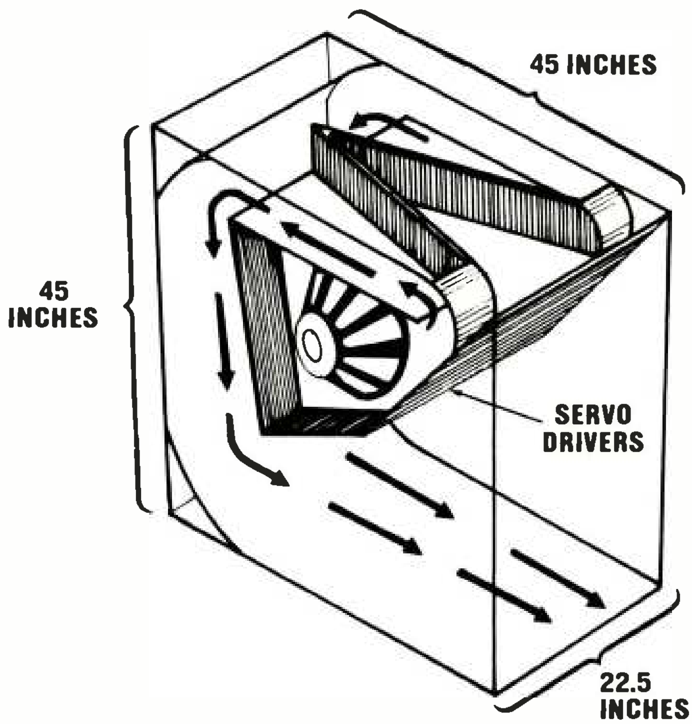
"We've put together the most compact, most efficient and most cost-effective audio system on the planet," states Pyramid Audio president Rob Vukelich (Figure 3). "I have shown photographs of the 'Wonderbox' system to people, and they think we are joking. They don't believe we can get the results that we claim from such a small speaker system. All I can say is that they have to come out and listen to it."

Vukelich has been interested in physics since high school, and was awarded a scholarship in that field to attend the University of Illinois at Champaign-Urbana. He turned out to be an "A" student, but was reportedly kicked out of the physics program for being a "rebel." In the decade since 1975, Vukelich has built up Pyramid Audio from a garage operation. The company now boasts two, 16-track recording studios, a 24-track studio, and a retail showroom housed in a 15,000-square foot building facility.

"We had been looking at compact cabinet designs for some time," Vukelich explains. "When the Intersonics SDL units came along, I realized that this was it — the low-frequency drive unit upon which we could base a truly revolutionary concert speaker system. The Wonderbox system will offer 33,000 clean watts of power in a package small enough to fit in an 18-foot straight truck."

The "Wonderbox" that Vukelich

Figure 2: The Intersonics Model TPL-3 measures 45 by 22½ by 45 inches deep. Two 15-inch heavy-duty speaker cones are loaded into a twin-throat, aerodynamic folded horn with a modified hyperbolic flare rate.



SERVO-DRIVEN SPEAKERS

refers to measures 30 by 27½ by 22½ inches, and is loaded with Electro-Voice components. Eight pairs of the boxes use only 60 inches of truck space (Figure 4). The box is all horn-loaded and tri-amplified, and houses two, 12-inch speakers, two aluminum-diaphragm compression drivers, and two titanium-diaphragm compression drivers. All components are housed in a single enclosure that features foam-dampened, moulded fiberglass horn flares. Frequency response is said to be 80 Hz to 20 kHz, when used in conjunction with the Power Demand Processor (PDP).

The Power Demand Processor is a continuously-variable, microprocessor-controlled, four-way electronic crossover with protective circuitry and active equalization for the optimization of power-band responses. The unit contains equalization memories for several combinations of Wonderboxes in varying acoustical environments. (*Would you like an automatic transmission and cruise control with your new PA system, Sir?*)

One "stack" of a 30,000-watt system comprises four Intersonics TPL-2 enclosures and four Wonderboxes. The three-way cabinets are built to the same face dimensions as the TPL-2s for an easy truck pack, and are intended to be suspended from simple flying hardware such as the Genie hoists favored by lighting technicians (Figure 5). Compressed air bottles are used to raise and lower the towers, as shown in Figure 6.

Actual-Use Situation: In June of 1985, this writer attended a Concert Showcase of the Pyramid Audio Wonderbox system, which was held in New Orleans, during the NAMM trade show. Vukelich and his crew had set up the system in an 800-seat theater for use by a Canadian rock group that goes by the name of Clearlight. The five-piece ensemble features a tribute to Pink Floyd as part of the program material, complete with high-level synthesizer parts.

Two of the TPL-2 units were set up in the aisle of the theater on each side near the stage, and Genie-lift towers hoisted one pair of the Wonderboxes on each side. "One bass box per side would be more than enough in here," explains Vukelich. "We usually team up one bass box with two, three-way boxes. Here, I wanted to make sure that everyone could hear these things really work!" Figure 7 shows the pair of TPL-2 cabinets.

Rob Vukelich needn't have worried. During the group's performance, the impressive low-frequency tones swelling up throughout the venue actually shook doors and windowglass out in the lobby. Dispersion did not seem to be a problem, either. Kick drum, bass guitar and synthesizer lines seemed to be issuing forth from a 40-cabinet, traditional arena system rather than the small boxes visible near the stage.

Figure 3: Pyramid Audio president Rob Vukelich (*left*) and partner Mike Acklin, beside the compact Wonderbox.

"We are experimenting with special-purpose subwoofers that are capable of reproducing frequencies to 15 Hz at high pressure levels," notes Tom Danley of Intersonics, who was present at the event. "Remember, this is a brand-new technology. We have only scratched the surface."

Such a system, with its simple two-point hanging package, high-powered Crest amplifiers, and brand-new Neotek consoles is certainly a step ahead of many companies that attempt to compete with Pyramid for local rentals and regional tours. Even so,

. . . continued overleaf —

Figure 4: Pyramid Audio's Wonderbox measures 30 by 27½ by 22½ inches (H×W×D), and weighs 180 pounds when loaded with two speakers, two two-inch compression drivers, and two one-inch compression drivers.

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Figure 5: Up to four Wonderboxes are easily suspended from the portable Genie hoists traditionally favored as ground support for lighting instruments.

SERVO-DRIVEN SPEAKERS

Vukelich is finding it difficult to break into the national touring market with the system. The reason? Production managers that have not heard the system refuse to believe such a small package will really live up to its developers' claims.

Case Study #2: Pace Sound

Glenn Himmaugh and Peter Shulmann are partners in Pace Sound & Lighting, based in New Orleans. PS&L is a diverse regional sound systems contractor with experiences and capabilities stretching from festival-sound reinforcement to multi-track recording and permanent installations. When the company was awarded the contract in early 1985 for the New Orleans Jazz and Heritage Festival, the partners decided that the time had come to build a new sound system. Not just more of the same traditional cabinets, but something *new* . . . smaller, lighter, more powerful, and one that made use of their own ideas regarding enclosure design.

"The bottom-end of a large portable system is *always* the tough decision," explains Himmaugh. "Your decisions in that frequency band not only affect the sound, but the overall size and portability of the whole system as well. When we first heard the Intersonics servo-drive products, we knew that it could meet our needs. We didn't have the time to experiment a lot on our own, as we had only weeks to put everything together."

Pace purchased an initial four of the TPL-3 enclosures, and coupled them with a proprietary box that was produced in-house and labeled the T-3 (Figure 8).

The T-3 unit comprises a trapezoidal, 14-ply birch enclosure loaded with JBL components. A pair of Model E-140 15-inch loudspeakers are

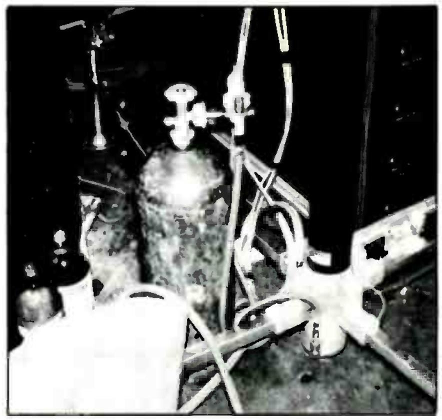


Figure 6: Bottles of compressed air are used to raise the Wonderboxes into the air. The entire suspension assembly is compact enough to be used even in clubs and small theaters.

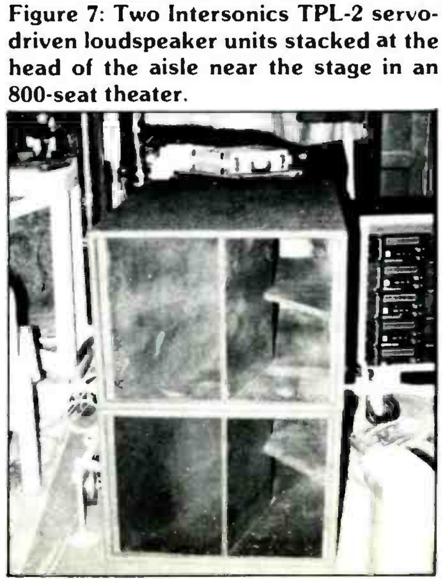
mounted onto a fiberglass horn section; each cabinet also houses a two-inch Model 2445 compression driver loaded on a proprietary, flat-front midrange horn. A Bi-Radial compression tweeter completes the package. Two of the T-3 enclosures rest easily on a single Intersonics TPL-3 unit (Figure 9).

"Besides the big festival, we had a major convention booked," Himmaugh recalls. "We had about a month's worth of time to pull it all together. The Jazz Festival is an important event for everyone in this city, and we wanted to do a really good job."

Pace debuted the new system at the festival, using a total of 10 T-3 boxes and four TPL-3 enclosures for outdoor crowds that approached 50,000 persons. The event drew a total of 250,000 persons over 10 days, with over 500 different musicians performing on eight stages, including such regional favorites as the Neville Brothers, Doc Watson, and Albert King.

"We used to do that event with

Figure 7: Two Intersonics TPL-2 servo-driven loudspeaker units stacked at the head of the aisle near the stage in an 800-seat theater.

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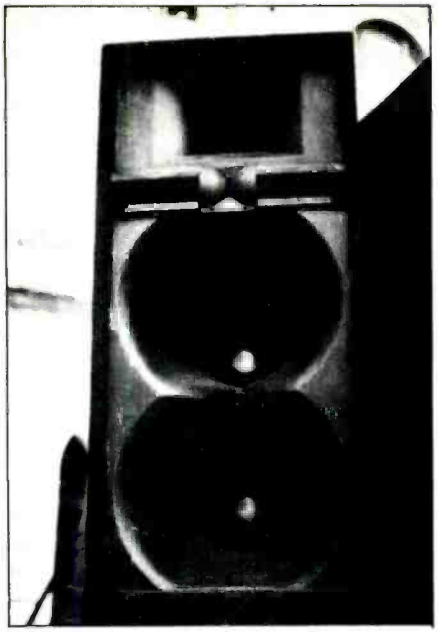

Figure 8: Pace Sound's proprietary T-3 cabinet with two JBL E-140 speakers, a JBL 2445 two-inch driver, and a Bi-Radial compression tweeter.

SERVO-DRIVEN SPEAKERS

sound wings that were 12 feet high and 10 feet wide," Himmaugh says. "This year, they were only *five* feet wide. We put the subwoofers under the center of the stage, and used a *lot* of power for the system. New Crown Microtech 1000s gave us a good bit of headroom, and the new T-3 cabinet projects really well. The highlight of the whole thing though, was that the servo-drive speakers really worked. One TPL-3 replaces eight, 15-inch speakers in horn-loaded boxes. It is a tremendous savings in time and space, and the sound is noteworthy. You could hear the low end all over the New Orleans Fairgrounds."

Actual-Use Situation: While I did not hear the Pace system at the above-described event, I was able to observe it in action outdoors at a private party that featured two separate stages for approximately 3,000 persons. Each of four sound wings housed a single TPL-3 and two T-3 enclosures. The site was New Orleans' Louis Armstrong Park, and the program material included rock music and a society orchestra. At approximately 100 feet from the stage areas, the sub-bass units were audibly the most obvious part of the speaker system. Extremely low frequencies were reproduced, giving the whole affair a feeling not dissimilar to what an outdoor "disco" might sound like (Figure 10).

Pace's proprietary T-3 enclosures represent an interesting and effective first attempt at proprietary loudspeaker systems. The trapezoidally-shaped cabinets make the assembly of compact hanging arrays quite simple. While no dedicated processor as yet has been developed for the cabinets, the unique system seems to be giving the firm its own competitive edge in negotiating for sound-system rental contracts.

"There are lots of pre-built, pro-sound enclosures available on the market today," remarks Glenn Himmaugh. "Those are fine for somebody just purchasing a couple. It was definitely worth the time and trouble it took to get our own enclosures happening. It was not such a large capital investment, and we were able to build something that fits our needs exactly. Putting together a large sound-reinforcement system today requires some *very* careful thought and advance planning. With a large system, *any* mistakes are multiplied beyond belief."

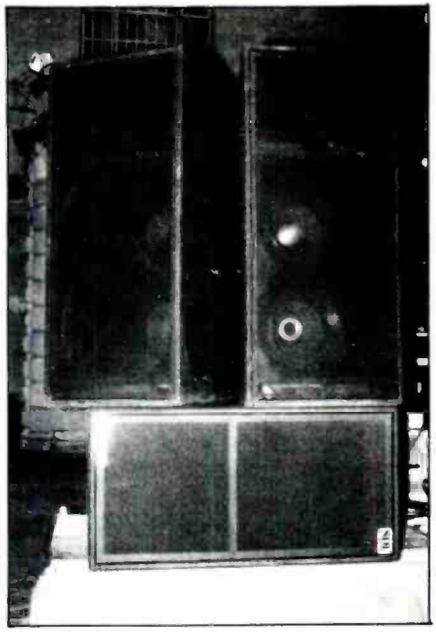

Figure 9: A pair of T-3 enclosures rest eadily atop an Intersonics TPL-3 servo-drive cabinet.

Conclusions

The two regional sound-rental firms profiled here are both successfully negotiating the Eighties. This decade, however, offers a distinctly different

Companies mentioned in this article:

Intersonics, Inc.
Sound Physics Division
425 Huehl Road, Unit 11A
Northbrook, IL 60062.

Pace Sound & Lighting, Inc.
2504 Bayou Road
New Orleans, LA 70119.

Pyramid Audio, Inc.
450 W. Taft Drive
South Holland, IL 60473.

Figure 10: Shown outdoors in New Orleans Louis Armstrong Park are (*left-to-right*): Peter Shulmann and Glenn Himmaugh of Pace Sound & Lighting, and Thomas Danley and Tom Melvern of Intersonics.

economic climate for growing sound companies than did the last one. Both Pyramid and Pace have looked to a manufacturer to solve what has been perceived by many as one of live sound's biggest challenges with portable systems: How to reproduce full-bandwidth musical material without having to hire and entire trucking fleet to get the loudspeaker system to each temporary site.

It is perhaps interesting to note that the solution to both companies' problem in that respect came not from one of the major loudspeaker manufacturers, but from an innovative, little-known firm with a high amount of technical expertise in a specialized field. While Pace Sound and Pyramid Audio rely on JBL and Electro-Voice, respectively, for loudspeaker components with which to load their own proprietary, custom-designed enclosures, servo-drive units from Intersonics were eagerly acquired as soon as they came to the attention of those two sound companies.

Hardly two years ago, a technical paper was presented at the 74th Convention of the Audio Engineering Society, October, 1983, by Thomas J. Danley, Charles A. Rey and Roy R. Whymark of Intersonics, Incorporated, titled, "A High Efficiency Servo-Motor Drive Subwoofer." Since that time, the idea has gone from prototype stage to the realization of a marketable, problem-solving tool that is changing the way some regional sound companies design their systems. This development would appear to represent a good example of how a new technology gains a foothold in the ever-changing business of concert sound. ■■■

*Pre-print #2043 (E-8), available from the Audio Engineering Society, 60 East 42nd Street, New York, New York 10165.

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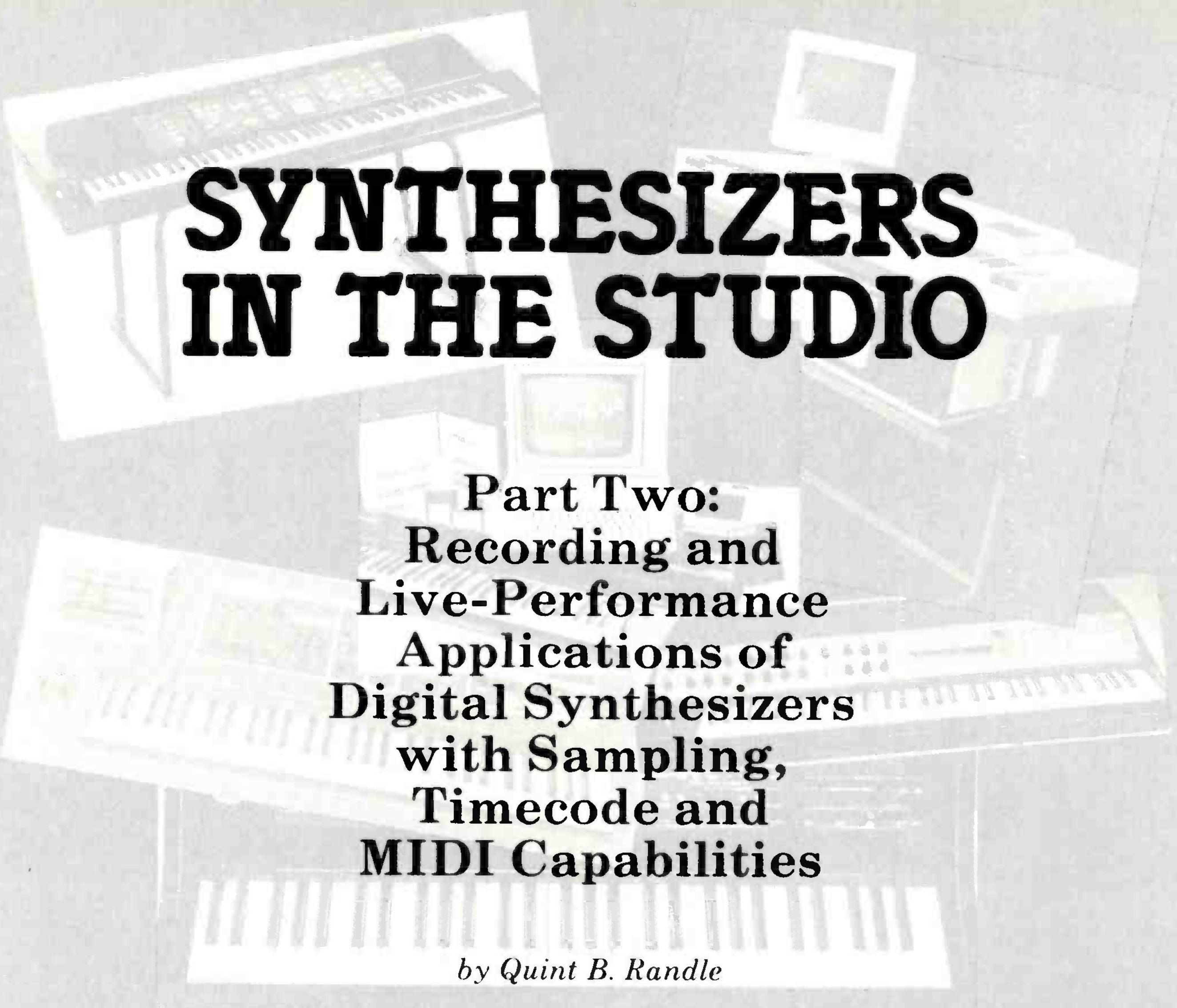

SYNTHESIZERS IN THE STUDIO

Part Two: Recording and Live-Performance Applications of Digital Synthesizers with Sampling, Timecode and MIDI Capabilities

by Quint B. Randle

With the advancement of keyboard-oriented synthesizers, sequencers and related devices comes a complexity that does not readily lend itself to portability. As emphasized in Part One of this article, published in the August issue of *R-e/p*, long gone are the days when a session player simply walked into the control room with his Sequential Prophet IV under his arm, and asked the engineer where he wanted it set up. Today, a well-equipped session player may charge as much as $200 in cartage just to get his or her array of synthesizers and outboard gear from point A to point B. Due in part to this increasing complexity, and today's overall music trend towards synthesizer-based sessions, a number of studios are slanting themselves towards a keyboard-oriented clientele.

But beyond this growing emphasis on keyboards in the studio, there are a number of session players who have found it to their advantage to build their own personal-use multitrack facilities. In other words, instead of bringing the keyboards to the studio, why not bring the studio to the keyboards? Somewhat of a switch has occurred — the session player no longer has to lug equipment from studio to studio, session to session; now the client need only walk into the player's studio with multitrack tape in hand.

In addition to the "standard" 16- or 24-track and mixing console, the heart of such player-owned studios is usually some type of a sampling computer system: a Fairlight CMI, Ensoniq Mirage, New England Digital Synclavier II, Kurtzweil 250, E-mu Systems Emulator II, etc; the rest of the facility's synthesizer setup seems to revolve around use of such digital synthesizers.

All in all, the new synthesizer technology has had a tremendous effect on the way songs, jingles, and live performances are conceived and produced. The remainder of this article profiles several players, engineers and producers that are using the new technology extensively to make their musical lifestyles more efficient, productive and creative.

TERRY FRYER — Colnot-Fryer Music

Terry Fryer and partner Cliff Colnot, of Colnot-Fryer Music, Inc., are best known for their commercial jingles for such clients as McDonalds, Nutra-Sweet, RCA, Computerland, Pizza Hut, Levi's and Doritos (the latter being the commercial with the catchy hook, "D-D-D Doritos"). The two put together their studio in downtown Chicago during the summer of 1983, with advice from such synthesizer experts as Robert Moog, Tom Rhea, and David Luce.

Having had his own 24-track facility since 1980, it was at the time of starting this new studio that Fryer

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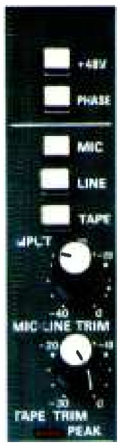
Both boards have separate inputs for mike, line and tape. To let you mix any vocal, instrument or tape track you want.



Pushbutton Input Selection

And if you need to isolate any part of the mix, just hit the stereo-in-place solo switch. And the tracks you want will come up in stereo.

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purchased his Fairlight CMI. According to the Clio Award-winning producer his choice of synthesizer system was based on sound quality and flexibility: "The Fairlight was the first high-end system that I heard actual sounds coming out of that would be useful, " he recalls.

While the Fairlight CMI may have won out as the studio's sampling device, it has not replaced Fryer's need for other synthesizers, and he says they are all used fairly evenly. "The sound of the [Sequential] Prophet oscillator is one of those sounds that, when you need it, there's *no* other way of getting it. I think the same applies to Oberheim, Roland, PPG and the [Yamaha] DX-7," he offers.

As a jingle writer and studio owner, Fryer says the MIDI and the other new interface technologies have affected him in several ways. "One thing the MIDI has done is to allow a sequencer to become a *composing* tool," he confides. "Before, a sequencer has been one of those devices that made particular types of sounds; played certain parts; or made you write a certain way, etc."

But now, the producer continues, because of MIDI a player can assign one device to make drum sounds, another to do a bass line, another to play a rhythm part, another to play a lead line, and so on. "You can experiment with parts and see how they sound together. And, if they are clashing, the writer can simply change a part; I have no extreme, vested interest in something that has been put down to tape," he points out.

Continuing, Fryer explains that although obvious changes in textures, via MIDI control, are now possible, and that "sounds have evolved quickly" through the ease of programming synthesizers, manufacturers are also progressing with manipulative features that are not *directly* related to sound itself — factors such as aftertouch, sustain pedals and breath controllers. "But in a negative direction," he adds, "if I MIDI together two or three instruments, it's a major problem to hook up a sustain pedal or breath controller to all of them. If I want to use a breath-controller, I'm limited to using three of these six instruments [in my studio]; if I want to use some kind of sustain pedal, I'm limited to another couple of them."

Fryer recalls another situation that most players and engineers involved with sequencers and drum machines can relate to: "It's one of those minor nuisances that makes you want to kick your sequencer down the stairs," he states calmly, before explaining that when he uses an off-tape sync code to drive his Garfield Electronics Dr. Click synchronization system — which is providing the pulse train for the sequencer or drum machine — he almost inevitably forgets to change the unit's drive clock from internal to external.

"Countless times I sit there and punch-in to record and wait . . . [the unit] is out of sync with what's going on, and you're trying to figure it out in your head. We built a box that takes either the click generated [by the Dr. Click] or the off-tape click, depending on whether or not the tape machine is in record mode. It's real slick; you don't have to worry anymore!"

Concluding, Fryer says the overall effect of the new technology has been to increase his productivity. "With instruments that have internal memory capabilities, you can actually have an on-going project — you just store your sounds on a disk, and can build up the sequence. You come back to it a couple of days later, and it's all still there — you don't have to start over again."

Terry Fryer's 24-track facility is centered around a Fairlight CMI with sampling options, linked via MIDI and other interface schemes to a variety of analog, FM-synthesis and digital synthsizers.

ED FREEMAN — Quantum Leap

The recording studio conventionally has been thought of as a place to take a piece of music that has been composed previously, and to then capture its performance on tape. Yet, as time goes by, recording and keyboard equipment are becoming more and more of a compositional tool — as opposed to strictly a performance tool — a development that has allowed set limits of creativity to be approached and challenged. Along with executive producer and Record Plant owner Chris Stone, Ed Freeman and a host of synthesizer players/programmers — collectively known as Quantum Leap — recently challenged those boundaries of creativity through an experimental album project called *The War on Attitude.*

Freeman, a veteran producer, and currently operations manager at Motown's Hitsville Studios, Hollywood, explains his original concept for the project: "The music was going to be largely improvisational, and it was going to be improvising with computers [sequencers]. But it evolved into something that was much more composed, much more static than had originally been planned." The finished album displays the current capabilities of digital recording equipment, as well as keyboard and synthesizer technology — in addition to stretching creatively from a compositional standpoint.

At times, as many as 28 synthesizers, nine programmers and six recording engineers were simultaneously employed to create the sounds of Quantum Leap. Some of the equipment used included: Sony PCM-3324 digital multitracks, PCM-1610 stereo digital processors, and DAE-1100 digital audio editors; Oberheim DSX sequencers, OBX-As, Xpanders, and more synthesizers from Yamaha, Sequential Korg, Roland and Kurzweil. In some passages, Freeman points out, a number of these synthesizers were combined to formulate sounds with the richness of a Stradivarius — yet with the bizarre quality of sounds never heard before.

"Most of the music was written on a DSX sequencer, which I felt was the most advanced sequencer available at the time," says Freeman of the recording sessions that started in October last year, and finished in March.

. . . continued overleaf —

The members Quantum leap (*above: left to right*) include Jim Mandell, Blake Lewin, Susan Seamans, Drew Neumann, Anne Graham, Ed Freeman, Marc Mann and Eric Rasmussen. Shown *3right* is the band's keyboards collection at Record Plant Studio M, Hollywood.

"But last year was 'pre-history' when you're talking about technology," he adds. In most instances, the playback/performance configuration of the sequences and synthesizers was set up to include one or more DSX, whose signals were sent into one or more Oberheim Xpander, and then out to any number of other synthesizers being used.

The reason for this configuration was that DSX sequencers were not MIDI controllable at the time, but employed control-voltage gates for triggering. "The Xpanders convert CV-gate information into MIDI. So, in every case, we were using the Xpanders as CV-to-MIDI converters — and beyond the Xpanders was a variety of synthesizers, depending on what sound we wanted to get at the time."

"I would never do it that way again," Freeman continues. "There are some theoretical advantages [to the use of CV gates] that may transfer into technical advantages; I think CV gates are less prone to some problems, especially [processor time] delay, but the advantages of MIDI are a *thousand* times over CV gates."

As it turned out, MIDI-generated processor delay generally was not a problem throughout the project. In most cases, Freeman and crew did not chain synthesizers through MIDI Out, Thru, etc.; instead, MIDI splitter boxes were utilized to string synthesizers together.

In addition to the sound-layering possibilities available through MIDI, Quantum Leap set out to exploit the track bouncing and overdubbing capabilities of digital recording to further create and manipulate individual sounds. "We were looking at what was possible on a digital machine that [could not be achieved] on analog; one of the first things is that you can do a lot more generations on digital than you can on analog. We managed to do some very, very complex sounds, which required a good deal of bouncing and sound shifting. Some songs have as many as 60 tracks on them."

The finished album is difficult to listen to all the way through, Freeman admits, and he hesitates to even call it "music," simply because there aren't really any verses, choruses and melodies *per se*. "I was doing a lot of things at the same time, not only experimenting with equipment, but with harmonic structures, composition techniques, and so on. We weren't told what not to do, and we all came out the wiser for it." he concludes.

PAUL FOX — Session Player

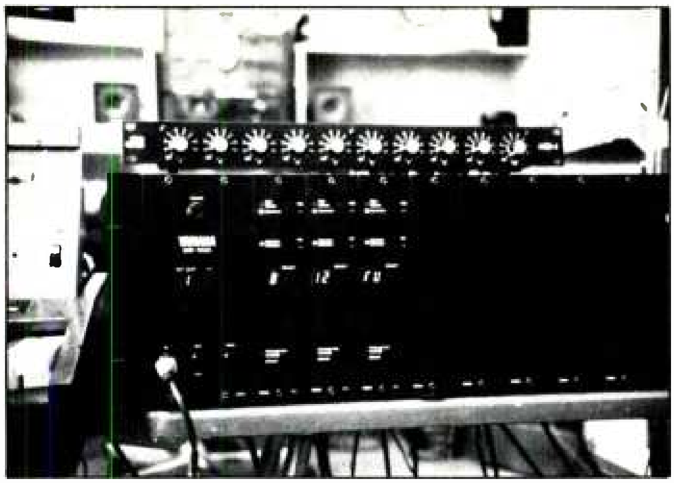
Musician Paul Fox and manager Rick Stevens went into the studio business together a little more than a year ago. While Stevens owns most of the keyboard and recording equipment, the studio is housed at Stevens' Suma Entertainment Group headquarters, West Hollywood. Fox has recently been doing session work for Barry Manilow, Jeffrey Osborne, Sara Vaughn, and is credited on a soon to be released Pointer Sisters album.

"Basically we set up the studio as an all-synthesizer writing studio, and then started getting into making it as state-of-the-art as we could," says Fox of the current facility. "We sought out the cleanest board for recording synthesizers that we could — for the amount of money we had — and to purchase the outboard gear that would serve all our requirements. Basically, we designed it around the *synthesizers'* needs."

The equipment that Fox and Stevens settled on was an AMEK TAC Matchless board and an MCI JH-114 transformerless 24-track. "The board is very clean for synthesizers," Fox confides. "There's a choice of a few boards in that price range, but this one is pretty much the cleanest one that we found."

Fox also points to a modification

Paul Fox' keyboard collection includes an E-mu Emulator II, Roland Jupiter 6, Oberheim Expander and Yamaha TX-816 rack, linked via a J.L. Cooper MIDI Patch Bay.



Close-up details of what Paul Fox describes as forming the "nerve center" of his keyboards-orientated studio. Pictured *left* is a J.L. Cooper MPB-1 MIDI Patch Bay, mounted on a Yamaha TX-816 rack, while *right* is a Yamaha PC and a Garfield Doctor Click sync system.

that has been made to the board for keyboard-oriented work: "When you start using a lot of different synthesizers you start eating up a lot of tracks. We have a modification to allow you to monitor from the monitor section what's already on tape, while still bringing up more synthesizers through the input faders — in this way, you don't lose the information on tape while doing more overdubs."

Other outboard gear includes an AMS RMX-16 digital reverb, BEL digital delay, two Roland SDE-3000 processors, dbx Model 160 compressor, Eventide Harmonizer, and several delay units. For monitors, Fox and Stevens are sticking with Yamaha NS-10Ms, which they feel provide a good standard reference for almost everybody in Los Angeles.

Instead of having his more than 17 keyboards and sequencers set up close to the patch bay, or relying on the use of direct boxes, Fox has provided patch points throughout the room housing the keyboards, console and tape machine. "Everything is line-in," he explains. "With 15 terminals on each box that can be used as inputs or outputs, there are ways of sending clicks, codes, delays, etc., around the room. If I want to send a click, for instance to a Dr. Click, and the source is on the other side of the room, you can send it down the patch bay and out one of the boxes."

A device Fox considers to be a real "lifesaver" is the J.L. Cooper MIDI Patch Bay, "I'm using the non-programmable one, because for now I just need to be able to switch things on and off quickly." With the MIDI Patch Bay, Fox can use eight different keyboards, or sequencers, as masters to control/program as many as 10 other MIDI-capable devices. "I have modular synthesizers, like the Oberheim Xpander and the [Yamaha] TX-816, and they're located at different places in the room. I have to be able to jump on a keyboard and be able to program. I can go from the Emulator II, Jupiter 6 or DX-7 as the master keyboard, or different sequencers; and it saves having to be switching from MIDI Ins and Outs all over the place. Basically, you hook up your MIDI and just roll."

CRAIG HARRIS — Sound Composer

Craig Harris is a prime example of today's session-player-turned-studio-owner. Specializing in voice effects, his movie credits include *War Games*, *Footloose*, *The Last Starfighter* and *Twilight Zone*. Harris' album work has included playing on sessions for Michael Jackson, Donna Summer, Tom Petty and the Heartbreakers, and Manhattan Transfer, although probably his most universally recognizable track has been the vocal effects on the bridge of Michael Sembello's "Maniac," from the movie *Flashdance*.

About 3½ years ago Harris purchased a New England Digital Synclavier with the concept of becoming a Synclavier session player, "running around town to different studios," as he puts it. Although he had a four-track at that time for pre-production purposes, it soon became apparent that an eight-track machine would enable him to actually produce commercials which, at the time, made up a large percentage of his work. "Within two months I was never going out-of-house," says Harris of the decision to move up to eight-track.

"At that point, it became very clear to me that if I had a 24-track machine, I would never have to charge cartage or costs like that, and have my equipment tied up in transit eight hours a day." Since the purchase of an Ampex MM-1200 24-track, Harris says, "in three years of recording commercials and other projects, I've only gone out once, to record a big-band sound. Other than that session I've pretty much done it all here."

Harris' studio, located in his San Fernando Valley home, houses a Soundcraft board, Ampex 24-track, and 32-voice Synclavier system, which he recently updated with Polyphonic Sampling (16 voices with a total sample time of 250 seconds), eight multichannel stereo outputs, and SMPTE interface card. (The latter allows the in-house Cipher Digital/BTX Softouch System to control the Synclavier's 32-track sequencer and Linn 9000, as well as the Sony VCR, MM-1200 24-track, and PCM-F1 digital system.)

One Synclavier capability that Harris uses extensively is that of re-synthesis. Using as an example the tune, "Rock House," from Manhattan Transfer's current album, he explains the process. Having sampled one of the singer's voices into the Synclavier, "you take the voice and carefully mark where the waveform is changing throughout a given word, and the computer [the Synclavier's Digital Equipment Corporation VT100 video terminal and controller system] will then analyze each one of the points and load them into the synthesizer. I then use the FM synthesizers in the Synclavier to re-synthesize the sound.

"It works amazingly well [when you need] to take voices out of their range, elongate parts of the words, twist them around and get strange vibratos, etc. Plus, it's an effect in itself, because the result doesn't sound like the original sample."

The concept also works very well with sounds that originate from synthesizers, Harris continues. "The other thing that I do a lot of is [sam-

. . . *continued on page 104* —

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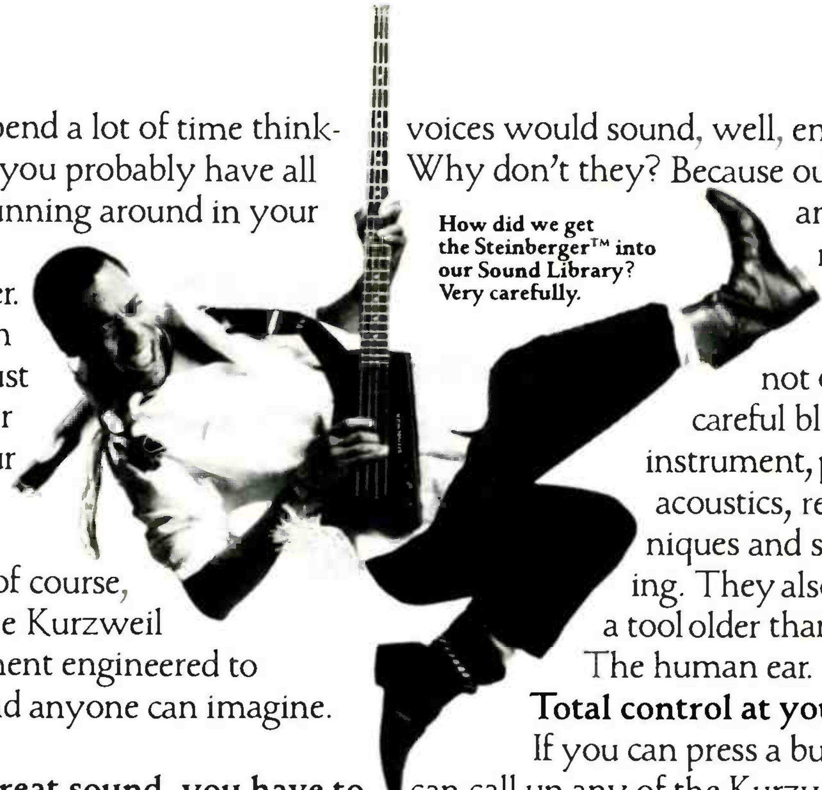

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— *continued from page 101 . . .*

DIGITAL SYNTHESIZERS

ple] the raw sound of a squarewave or sawtoothwave — take your favorite synthesizer — and re-synthesize those. That's basically what re-synthesis is set up for doing: you just take the output and feed it through a filter and do the filtering yourself."

Harris tends to use the Synclavier to perform this task, or feeds the raw wave sample into his ARP 2600 synthesizer. He feels that such a capability has helped him to side-step problems of MIDI communication between various synthesizers of different manufacture; the Synclavier's on-board sequencer is employed when sequencing is needed.

In trying to accommodate the "I-don't-know-what-I-want-but-I'll-know-it-when-I-hear-it" attitude that Harris generally gets from producers while working on a movie soundtrack effect, the composer has designed what he calls a "master-monster" patch bay for voice processing. To create a typical voice effect, he may use two or more pitch-shifters, Vocoders, pitch- and envelope-followers, noise gates and several synthesizers. All of these devices are in one master patch that involves hundreds of individual patch cords. "Basically what I have is a 'portable' patch bay. I can change whether the EQ is before or after the Harmonizer, for example, or the Harmonizer is connected before the voice goes into the Vocoder, and afterwards, whether I want to harmonize the other effects, etc.

"Unfortunately patching 15 to 20 boxes together to create one effect is not applicable to most people's situations. The funny thing is that, after patching in all those effects and boxes, they [the producers] want it to sound as 'natural' and 'organic' as possible!"

Craig Harris' keyboard collection at his 24-track facility includes a 32-voice NED Synclavier digital synth with Polyphonic Sampling option and timecode interface card.

This comment leads Harris away from the technicalities of computers and other devices, and into the plain world of communicating musical ideas to non-synthesizer programmers. "The big problem with film people is they have no language to communicate with as far as sound is concerned. So you kind of have to set up the ground rules — a language with which they can talk to you. That's something that's important to anybody, whatever you're doing. You have to meet them with what they're comfortable with. You have to come up with definite things they can identify and understand," he concludes.

SHEP LONSDALE — Toto's Engineer

With the price of sampling systems taking somewhat of a downward turn, the technology is available not only to those studio musicians able to afford the price tag of high-end systems, but also to Top-40 musicians playing local nightclubs. In addition to this, major artists are beginning to use sampling keyboards on tour, in lieu of playing live against pre-recorded tapes being played back through the house and monitor PA systems. Toto recently completed a tour supporting its new album, *Isolation,* during which several E-mu Systems Emulator II synthesizers played

Shep Lonsdale, Toto's in-house production engineer.

horn parts (on the tune "Rosanna" in particular), synthesizer parts, background vocals, and percussive effects.

According to the group's in-house engineer, Shep Lonsdale, Toto considered sampling as representing the only sensible way to perform live what they had done on the album — "without bringing along a million extra players." Lonsdale recently engineered the soundtrack for the movie *Dune*, and had worked with Traffic, the J. Geils Band and the Doobie Brothers before taking a "staff" position with Toto a few years back.

"The same day the Doobies broke up," he recalls, "Toto asked me, 'Why don't you quit the Doobies and come work for us'?"

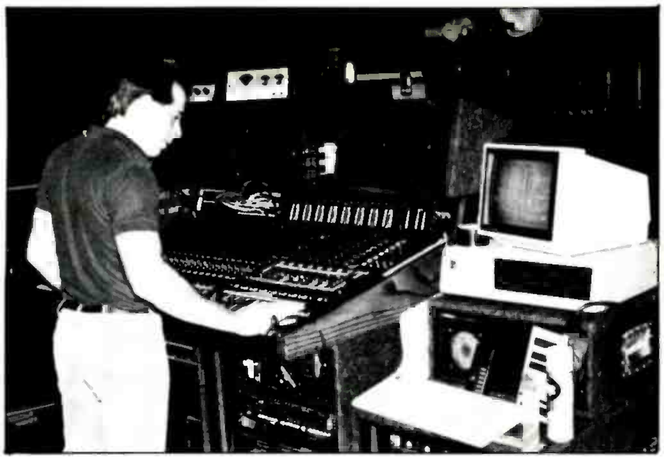
Lonsdale points out that although many groups have been taking along a two- or four-track tape machines on tour to serve as an additional backup "musician," Toto did not feel totally comfortable with the idea. "What's the point of the band being there if you've got all this [material] going on tape? We figured it would be much better to put down the sampled parts, and at least play them ourselves." Previously the group had never used tapes but, as Lonsdale suggests, had never before tried to pull off live what the group had planned to do on its latest tour. [For a full rundown of The complex keyboard monitoring and sound-system design for Toto's recent world tour, see David Scheirman's article in the August issue of *R-e/p — Editor*].

"The whole organization is very aware of what we are doing on the road, because it's the same people handling live sound as in the studio. So, we even went to the extreme of doing special re-mixes right after mixing a particular tune for the actual album. At the time we didn't know exactly how we were going to use it, but we had a whole backup reference of mixes that were to be experimented with in a live situation."

In most instances, an entire hook, horn line, etc., was keyed off one note. In other words, instead of sampling the sound of a horn section into the Emulator, and then playing a particular lick, the *entire* lick was sampled into the keyboard. And, rather than taking the parts from the studio 24-track masters, or the special mixes mentioned above, and directly sampling them into the Emulator II, a Sony PCM-F1 digital processor was employed as a go-between. "I took everything to an F1 and did things like 'one-pass stereo'; 'one-pass mono'; 'one-pass with effects'; 'one-pass without effects.' We were then able to experiment with what we had on the F1 by going into the Emulator to see actually how much information we needed to make it work.

"It gave us a lot of flexibility, because we ended up with a log of all the parts that would be difficult to reproduce live. And, because we took the F1 with us, anytime during the tour we could update and amend any of the samples we had; we just whipped out the F1 and redid the sample." There was no loss of sonic quality either, Lonsdale adds: "Digital-to-digital is just fine; the Emulator loved it."

The main advantages of using sampling in a concert setting, Lonsdale



During Toto's recent World Tour, MIDI-equipped keyboards were controled from an IBM Personal Computer (shown *left* at the stage-right keyboards mix position) and Compaq PC (*right*) running custom software developed by Ralph Dyck.

DIGITAL SYNTHESIZERS

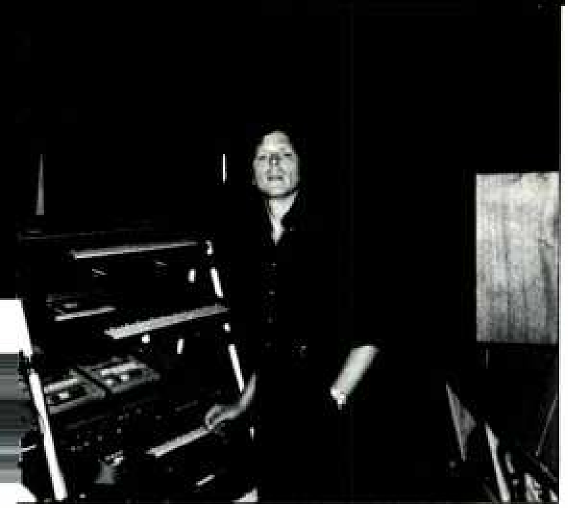
adds, is that it allows greater freedom than playing against pre-recorded tapes provides. "When you use tapes, you're stuck to the format of the tune. You can't stretch it out; you can't let the tune 'breath'; you've got to do it the *same* way every night, and it get's real dull; that's *not* what live music is all about."

According to Lonsdale, Jeff Porcaro's drum timing was meticulous, with the result that there was no problem with tempos alternating from one night to the next. "Could your average listener tell that they were parts that were sampled and not actually physically being played?" Lonsdale questions himself. "No, they couldn't tell at all."

MICHAEL BODDICKER — Keyboardist

It is safe to say that keyboardist-synthesist Michael Boddicker is one of the busier session players currently working on the West Coast. His album, jingle and soundtrack credits include musical work with Lionel Richie, Quincy Jones, Mattel Toys, *Back To The Future* and *Real Genius.* The following comments detail his particular uses, likes and dislikes of the various keyboards that comprise his multi-instrument session setup:

- **Yamaha DX7:** "I have two DX7s that have been modified to [provide] eight banks of memory each. They also have been modified to raise the



Session keyboardist Michael Boddicker at Sound Castle, L.A.

output gain so that they are a little hotter than line-level, because certain models don't have the internal output-level pot turned up enough.

"They have great percussive color and great digital sounds. When I need richness in a sound I don't use them; however, when I need real *crisp, biting* sounds, I use the DX7. Essentially what I use them for is to duplicate 'colors' that I might normally get on a PPG. I use one of my DX7s as a touch-sensitive master keyboard."

• **Roland Jupiter-6:** "Most of the time I use the Jupiter-6 as the master keyboard, even though it's a non-touch-sensitive keyboard, and I get the same velocity response all the time. It's also at the right height in my rack!

"The Jupiter-6 is also one of the instruments I find myself using more and more, particularly on orchestral colors — it works great for those sorts of sounds. It also has more filter effects than just about any other synthesizer I know of except the Oberheim Matrix 12, which I'm in the process of getting."

• **Roland Jupiter-8:** "I can use the -8 as kind of an auxiliary analog synth. It has a richer tone color than the Jupiter-6, but one of the major things that I use it for is triggering; I have had the envelope generators modified so I can fire them with an external signal. I have also had the memory modified to provide three times as much memory capacity. It also has arpeggiator output, through MIDI, which is something I use it to provide in the studio."

• **E-mu Systems Emulator I:** "I still use the Emulator quite a bit because I have such an extensive library of sampled sounds. Even though the sound bandwidth isn't as great as I would like it to be, it is a very useful instrument, and I don't foresee taking it out of the rack. I tried to transfer a lot of the sounds to the PPG and to the Emulator II, and they take on a different color."

• **E-mu Systems Emulator II:** "Of all the instruments I have right now, The Emulator II is the *most* exciting, because of the new software developments and what not. I've been able to create a really extensive —and exclusive — library of sounds that gives me such a scope. Besides natural instruments, I have samples of sounds loaded into the system that you just wouldn't be able to do with natural instruments."

• **PPG 2.3 with Waveterm:** "The PPG really crunches up the top-end of a digitized sound — you can hear the aliasing on the high frequencies. And that gives the keyboard a *very unique* color. Even though that's not what you really want from a sampling synthesizer, I think that you can afford to have a toy that makes that 'weird noise' — the PPG makes some real 'electronic' sounds."

• **Roland JX-8P:** "I'm amazed; I've been using the 8P on a Lionel Richie session for some bass color [instead of a Minimoog]. I also use it for real top-end, bright bell colors. I know it may sound like I'm contradicting myself to say the 8P sounds real bright and bassy, but on the top-end it's real 'sparkly' and as close to the DX7 or a PPG sound from an analog synth that I've found."

Other keyboards Boddicker still finds very useful include an ARP String Ensemble and Minimoog.■■■

STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE

Northeast:

□ **KAJEM RECORDING** (Gladwyne, Pennsylvania), claimed as Philadelphia's only Solid State Logic-equipped facility, has added an **AMS RMX-16** digital reverb system and **DMX 15-80S** digital delay and sampling unit to its outboard rack, plus a **Summit Audio** Tube Limiter, **Ensoniq Mirage** sampling synthesizer, **Rocktron HUSH II** noise reduction, and a **Lexicon Prime Time II** digital effects unit. *1400 Mill Creek Road, Gladwyne, PA 19035 (215) 642-2346.*

□ **CRYSTAL CITY TAPE DUPLICATORS** (Huntington, New York) has added new **Otari DP-85** slave transports to its **DP-7500** high-speed cassette duplication system. The new slaves are equipped with Dolby HX Headroom Extension circuitry to improve high-frequency response of the duplicated material. According to **Douglas Young**, the facility's VP of marketing, the new equipment will double Crystal City's daily production capability. *48 Stewart Avenue, Huntington, New York 11743 (516) 421-0222.*

□ **SADDLE RIVER MUSIC** (Paramus, New Jersey) has relocated its facility, and acquired a **New England Digital Synclavier II** Digital Music System with 32-track memory recorder, 50-kHz sampling rate, and SMPTE timecode capabilities. In addition, an **Otari MX-5050 Mk III** and **MB-5050** half-track have been linked to the facility's new **Soundcraft Series 200** console. Partners **Neil Fishman** and **Harvey Edelman** say that their facility awaits added work geared towards music creation and radio/video production. *105 Fairview Avenue, Paramus, NJ 07652 (201) 843-3880.*

Southeast:

□ **SHEFFIELD AUDIO-VIDEO PRODUCTIONS** (Phoenix, Maryland) has added **AMS RMX-16** and **Lexicon Model 200** digital reverb systems, the latter for use in the facility's new digital audio remote truck, equipped with a **Sony PCM-3324** DASH-Format 24-track. Recent modifications to Studio A's control room include the addition of more **RPG Sound Diffusers**. *13816 Sunnybrook Road, Phoenix MD 21131 (301) 628-7260.*

South Central:

□ **TIM STANTON AUDIO** (Austin, Texas) has upgraded from one-inch/16-track format to two-inch/16-track operation with the purchase of a new **Sony/MCI JH-24/16** multitrack with **Autolocator III** remote control. Recently added reverb systems comprise an **Ursa Major 8X32** with remote, and **ART 01A** units. Other new equipment includes eight **Gatex Audio** noise gates, four **dbx Model 160X** limiters, and a **Roland SDE-300** digital delay. *West Fifth St., Suite 103, Austin, TX 78703 (512) 477-5618.*

Midwest:

□ **SWEETWATER SOUND** (Fort Wayne, Indiana) has added an **AMEK TAC Matchless** console and **Soundcraft Series 760** 24-track, and is offering a special introductory offer for recording time this month of $40 per hour. The studio's synthesizer and drum machine array includes a **Kurzweil 250** (reportedly the only studio in Indiana to own such a digital sampling keyboard system), **Yamaha DX-7**, **Roland JX-8P**, **Moog Memorymoog**, **Linndrum** and **Yamaha RX-11**; all instruments are available for

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use by visiting engineers and producers at no extra charge. A **Fostex B-16** half-inch/16 track also is scheduled for installation in the facility's Studio B later this month. *2350 Getz Road, Fort Wayne, IN 46804 (219) 432-8176.*

☐ **KOPPERHEAD PRODUCTIONS** (North Canton, Ohio) has completed a major expansion of its facility, and now offers 24-track recording in a new room designed by **John Storyk**. New control-room equipment includes a **Sony/MCI JH-24** multitrack with **dbx** noise reduction and **Autolocator III** remote control, and a 32-input, transformerless **Soundcraft Series IIB** console; outboard equipment includes a **Lexicon 224** digital reverb, **EMT** echo plates, **AKG**, **MicMix** and **Orban** analog reverb units, **UREI LN1176**, **Audio-Design/Calrec Vocal Stressor**, **dbx Model 161** and **162** compressor/limiters, **Valley People Kepex** and **DynaMite** noise gates, **Eventide Harmonizer** and **Instant Flanger**, **Orban Stereo Synthesizer** and **Parametric Equalizer**, plus microphones from **Neumann, AKG, Sony, Shure, Sennheiser** and **Electro-Voice**. The company will continue to offer 16-track recording sessions with a **Tascam 90-16**, as a low-cost alternative for demo sessions, while custom music production with a **New England Digital Synclavier II** digital synthesizer system also will be offered by the company. *935 Schneider Road NW, North Canton, OH 44720 (216) 494-8760.*

KOPPERHEAD — refurbished control room

☐ **SOLID SOUND** (Ann Arbor, Michigan) has acquired the following audio-for-video post production equipment: an **Adams Smith 2600** modular timecode/synchronizing system which consists of generators, readers, synchronizers and character inserters; **NEC 2505, 1905** and **1305** monitors; **Otari MTR 12 Mk IV** multitrack; and a **Sony** custom-Alphatized 5850 U-Matic VCR. *P.O. Box 7611, Ann Arbor, MI 48107 (313) 662-0667.*

☐ **BEAT 'N TRACK** (Pittsburg) has added an **MXR** digital reverb and audio exciter to its outboard gear rack. *111 LeBouf Drive, New Kensington, PA 15068 (412) 339-0814.*

Mountain:

☐ **BREEZEWAY STUDIOS** (Waukesha, Wisconsin) has purchased an **AKG C-24** stereo tube condenser microphone, **Sony PCM-F1** stereo digital processor, **Klark-Teknik DN780** digital reverb, **API** pre-amp and **Model 550A** outboard equalizer, **Tannoy NFM-8** studio monitors, **Aphex Type B** Stereo Aural Exciter, **Studio Technologies AN-2** Stereo Simulator, and a **Shure SM-7** dynamic mike. *363 West Main Street, Waukesha, WI 53186 (414) 547-5757.*

☐ **GRAMMIE'S HOUSE** (Reno, Nevada) is a new residential, 24-track music recording and audio-for-video studio scheduled to open in late October. According to co-owner **Robert Forman**, "We planned this new facility as a 'resort studio' that would be close to the Los Angeles recording community, but which would provide the feeling of getting away from the pressure of the Coast — yet be only an hour away [by air] from L.A. The building — which we have re-modeled and furnished to look like a colonial-style house — has three bedrooms for clients, with a fourth now under construction. We also have arranged with the nearby MGM Grand Hotel

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to offer competetive rates to us for people that cannot be accommodated in the house. We also plan to offer in-house catering for lock-out sessions." Forman's partners in the venture, which has a reported total budget of $2 million, are **James Nicholson** and **Sig Rogich**, who has strong connections with the local and national advertising community. The new facility features a **Chips Davis** acoustic design that utilizes the principles of a "Live-End/Dead-End type" environment. The 650-square foot control room boasts a 56-input mainframe **Solid State Logic** 6000E console currently fitted with 48 channel modules, **Studio Computer** fader automation, **Total Recall** and **Three-Machine Synchronizer.** The new SSL console is described as the first such board to be delivered to a facility in the Western States. Other control-room equipment includes a pair of **Studer A800** 24-tracks for 46-track sessions, and two **A820** mastering machines equipped with half-inch heads; a digital mastering machine is also under consideration.


GRAMMIES' HOUSE — new residential studio

Outboard gear consists of **Lexicon 224XL** and **Advanced Music Systems RX-16** digital reverbs; **Eventide SP2016** delay/special-effects processor, plus **H910** and **H949 Harmonizers; Drawmer** noise gates; and an extensive collection of vintage microphones. The facility also boasts a **Fairlight CMI Series III** digital synthesizer with sampling, timecode synchronization and MIDI interface options. The studio area measures 1,200 square feet, and contains separate piano and drum booths, plus a utility isolation room. Covered Foley pits are provided for audio-for-video and -film sweetening sessions. One wall of the studio is covered with RPD absorption panels to provide a diffuse reflection pattern. The control-room design also features RPG difffusors and polycylinders mounted on the rear and side walls, respectively, to provide a "dead-end" behind the mixing position. A second, smaller studio measuring 800 square feet is also currently in the planning stages, and will be designed for smaller music-recording sessions, and jingle commercials work. *1515 Plumas, Reno, NV 89509. (702) 786-2622.*

Southern California:

☐ **UNITEL** (Los Angeles), a video post-production facility, located in Los Angeles, has installed four new **YNH ESC-02** mixing consoles in its pair of on-line video editing suites, off-line suite and telecine bay. **Newt Bellis**, the facility's West Coast president, explains that he selected the new boards because "they are very flexible, compact [and] computer-interfaced, [allowing] us to stay current with such imminent changes in the industry as stereo broadcasting." The modular designed, rackmount desk comprises two, 32-by-12 routers, and accepts six stereo audio sources for separate recording and monitoring, plus automatic crosspoint pre-sets, LED readouts, built-in routers, three-band equalization, and RS-232 computer interface — used for remote audio level control. The company's L.A. facility is the most recent addition to the existing Pittsburg and New York operations. The consoles were supplied by Audio Intervisual Design Systems, which has also equipped **Pacific Video** with a YNH board. *5555 Melrose Ave., Studio G, Los Angeles, 90038 (213) 468-4606.*

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□ **EVERGREEN STUDIOS** (Burbank) has replaced several, low-frequency drivers in its exising **UREI Model 813** monitors located in Studio A with **Cetec Gauss Model 3588** 15-inch coaxials and **4583A** 15-inch woofers. The facility's head of engineering, **Marc Gebauer**, says that the updated speakers "sound better than anyone imagined. The top-end is much smoother, [with a] tighter bottom-end." When audio consultant **George Augspurger** re-tuned the studio after the speakers were installed, Gebauer commented that "the room EQ controls were virtually flat." The facility is now designing its own cabinets for the Gauss loudspeakers — which will be bi-amped — and plans to replace the Studio B monitors with Gauss components and custom cabinets. *146 North Golden Mall, Burbank, CA 91502 (818) 842-5163.*

□ **POST GROUP SOUND** (Hollywood) opened its new sound department on August 1, offering post-production audio services for film and video. The two new studios are fully "floated," and acoustically treated with a copyrighted "undulating full-wall acoustic diffusing system" designed by **Jeff Cooper Architects, A.I.A.** The studios themselves were designed by Post Group engineers **Tamara Johnson** and **Peter Cole**. A modified, 48-track **Neve 8128** desk (with **NECAM** automation) resides in Studio A, and a **Neotek Elite** board controls Studio B. **Otari MTR-90** 24-track, **MTR-20** four-and two-tracks (reported to be the first Otari center-track timecode machines to be delivered in this country) are shared by both studios, plus **CMX 340** machine control throughout the entire facility. Outboard gear includes **Adams-Smith** synchronizers, **UREI 813B** studio monitors, **Quantec QRS** Room Simulator, **Lexicon Super Prime Time** and **Model 224** digital processors, **Dolby** and **dbx** processing, and **SONY BVH-2000** and **BVU-800** video recorders. *6335 Homewood Avenue, Los Angeles, CA 90028 (213) 462-2300.*

POST GROUP — new Neve 8128 console

□ **SMOKETREE STUDIO** (Chatsworth) has installed a new **Neve 8078 A** 76-input console linked to the facility's existing two **Studer** A800 24-track machines. *Chatsoeth, CA (818) 998-2097*

Northern California:

□ **FIG TREE PRODUCTIONS** (Aptos) plans to open its MIDI-based composing and recording studio to outside songwriters and musicians for developing arrangements and recording song demos. Previously, the facility was primarily for in-house productions. The keyboard-oriented studio is based around a 36-track sequencing/editing software system running on a **Commodore SX64** computer. Other insruments includes a **Yamaha KX88** Master Keyboard Controller and **TX-7** synthesizer, **Roland Juno-106** synth, **Sequential Six-Trak** and **Drumtracks** drum machines, plus **Korg** and **Roland** synchronizing and interface units. Recording hardware is based around a **Fostex** four-track, **Tascam** two-track, **DeltaLab** digital delay, **E-V/Tapco** spring reverb, **UREI** limiters, and **TOA** monitor loudspeakers. *P.O. Box 1634, Aptos, CA 95003.*

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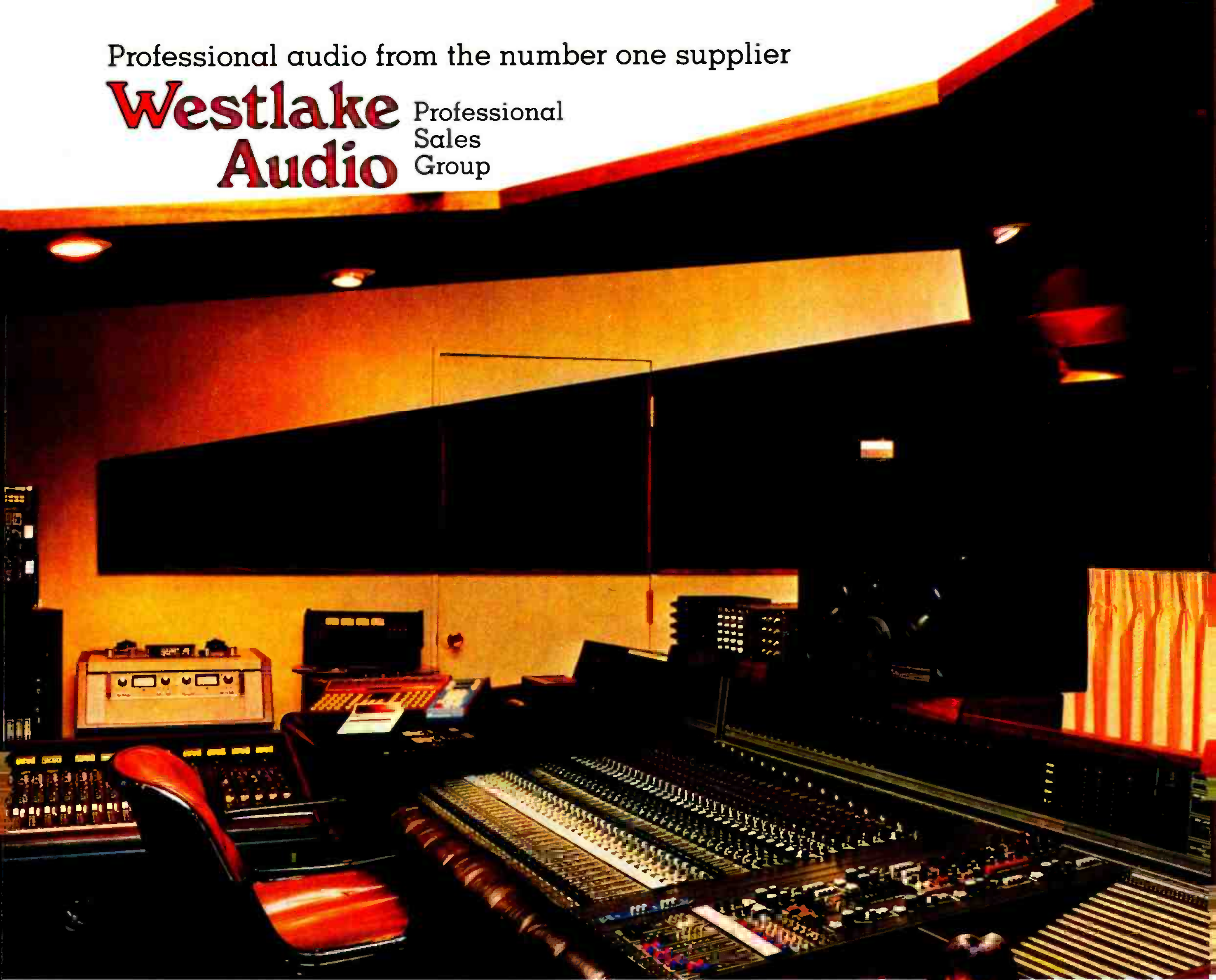

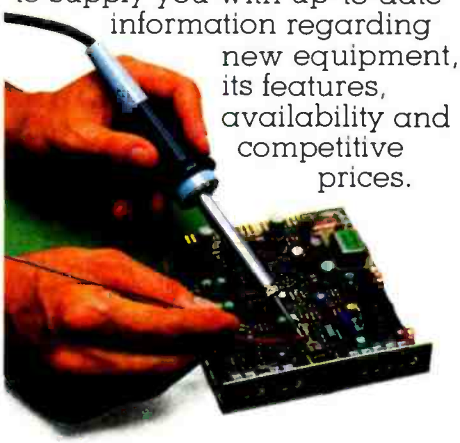

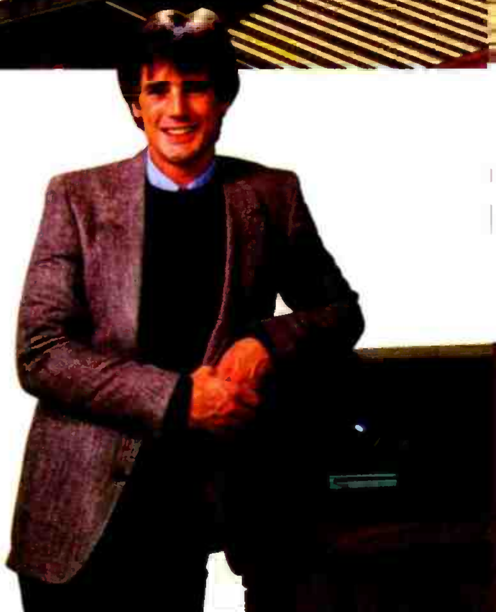

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Including Production Recording and Random-Access Editing

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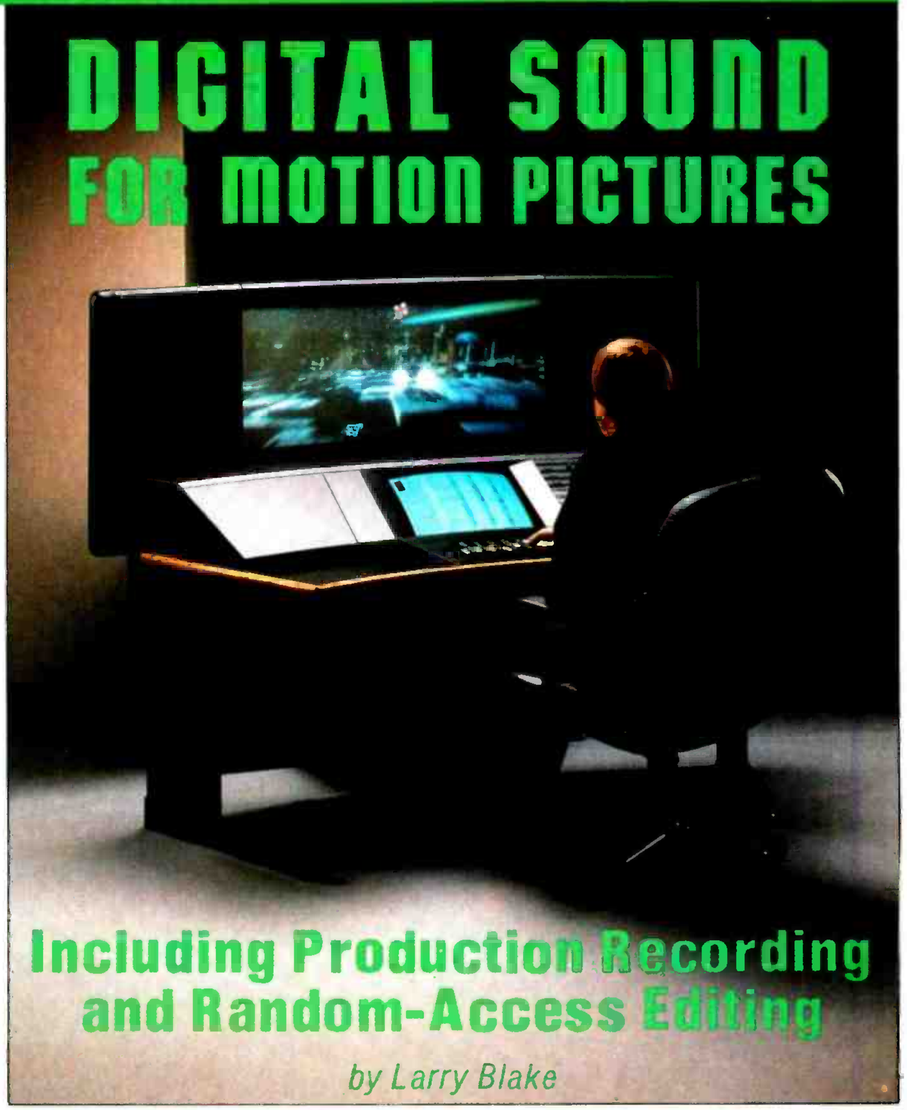


The Droid Works SoundDroid digital sound editing and mixing system.

This article will discuss current developments in digital sound technology as it relates to film production recording, and its potential for random-access editing. In addition, an overview of the previous uses of digital recording in film post production will be provided. Part two, to be published in a subsequent issue of *R-e/p*, will focus both on the use of digital multitrack recorders in post production, and current developments in digital playback systems for motion-picture theaters.

Although the first multitrack digital recorders, manufactured by 3M, were being delivered to recording studios during the early months of 1979, they were not used in motion picture post production at Hollywood facilities until 1982. An exception was the digital recording of motion picture scores; the music for such films as *The Black Hole* and *Annie* was recorded with the 3M Digital Mastering System, with conventional 35mm analog mixdowns used in the final film mix.

Probably the first film to utilize a completely digital soundtrack was the rescoring of Walt Disney's *Fantasia* in early 1982 (described by this author in the October 1982 issue of *R-e/p*). The new score, conducted by Irwin Kostal, was recorded and mixed by Shawn Murphy on 32-track 3M Digital Mastering Systems at CBS Studio Center in Studio City, CA, in January of that year, with subsequent post-production at Evergreen Studios and Walt Disney Productions.

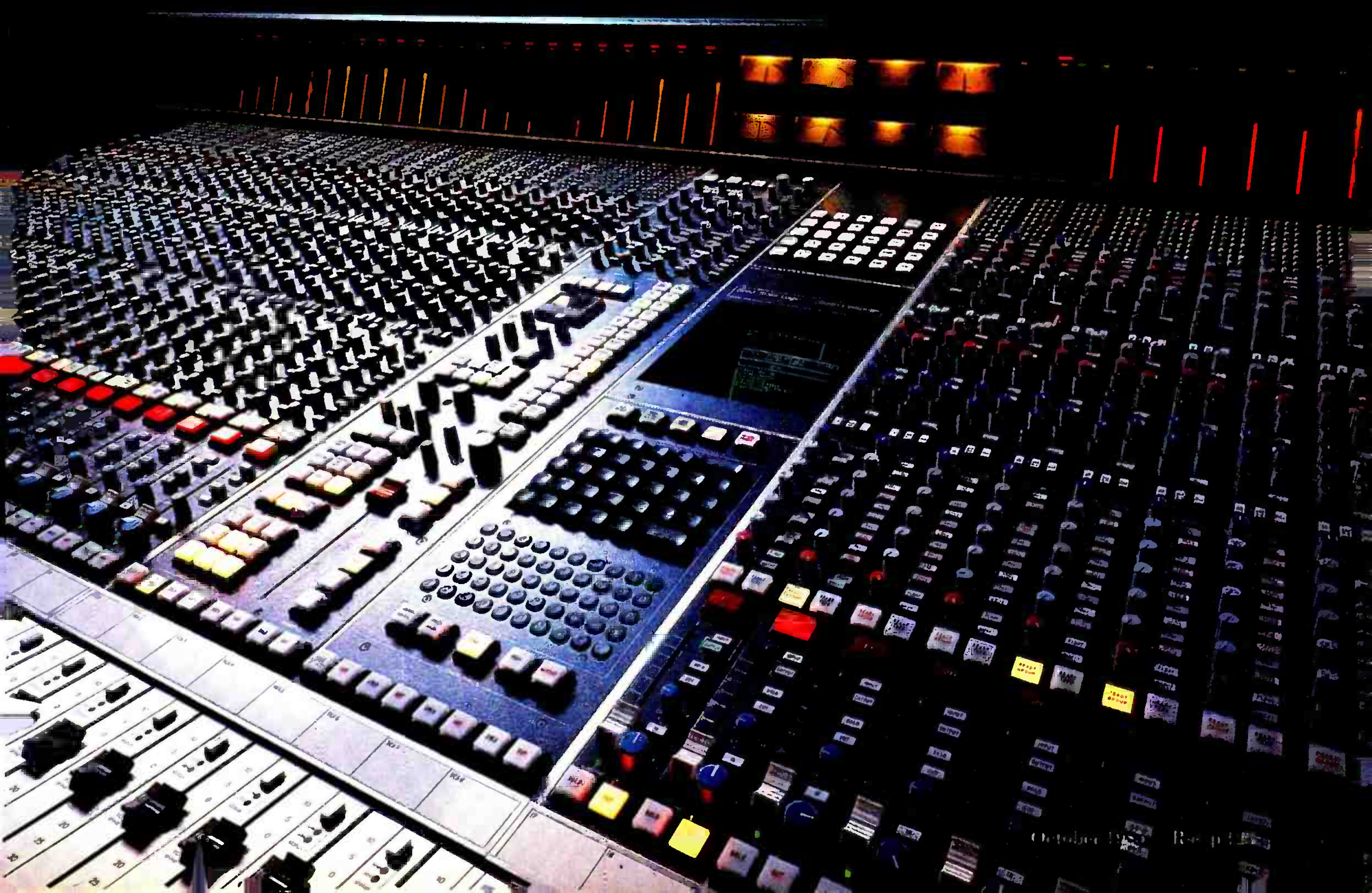
The primary release of the rescored *Fantasia* was in 35mm, Dolby-encoded four-track magnetic prints that were sounded from a 35mm four-track printing master. In other words, what was heard in theaters was all-digital plus two analog generations.

The digital re-recording of *Fantasia* was made possible in part by R&D that previously enabled Disney's trio of 3M digital multitracks to be used in the re-recording and music recording of all of the films prepared for the EPCOT Center, Florida. Since 1980, Disney has digitally recorded the scores for almost all of its feature films, including *Splash*, *Country*, and *The Black Cauldron*.

In 1984, three films were released that had used Sony PCM-3324 24-track digital recorders during post-production. Sound for two of these films — *Digital Dream* and *Metropolis* — was completely digital up to the analog print, while *Stop Making Sense* was recorded 24-track analog, the digital multitracks being used for all post production up to the final Dolby Stereo optical negative. (It might interest some readers to note that the Dolby Stereo Lt-Rt printing masters for *Stop Making Sense* and *Metropolis* were Dolby-encoded on the digital multitracks to allow for a 1:1 transfer during the subsequent shooting of optical negatives, without having to encode the tracks being replayed from the digital master. The 35mm Dolby Stereo prints of *Digital Dream*, on the other hand, were sounded from an analog 35mm Lt-Rt mix.)

Digital Dream was produced at the new Glen Glenn sound facility in Hollywood, utilizing Sony digital multitracks for all music, Foley and ADR recording, plus re-recording. (Although portable Nagra analog tape machines were used to record production sound, all dialog was later replaced in ADR.) Sound effects were recorded on Sony PCM-F1 processors, and bumped up to PCM-1610 format in order to interlock the resultant ¾-inch U-Matic cassettes with the Glen Glenn PAP (Post Audio Production) system that had been modified for use with the digital machines.

Giorgio Moroder's rescoring of the Fritz Lang classsic, *Metropolis*, was recorded and pre-mixed using the three Sony digital multitracks at the producer's Oasis Studio in North Hollywood, CA, with four-track LCRS (left-center-right-surround) pre-mixes being checkerboarded onto eight tracks of the digital edit master. Next, the two pre-mixes were combined into a four-track master by bouncing up onto the digital dubbing master. According to engineer Brian Reeves, although all mixing at Oasis had been done utilizing a discrete center speaker while monitoring through the Dolby Stereo 4-2-4 matrix, it was decided that a film dubbing studio



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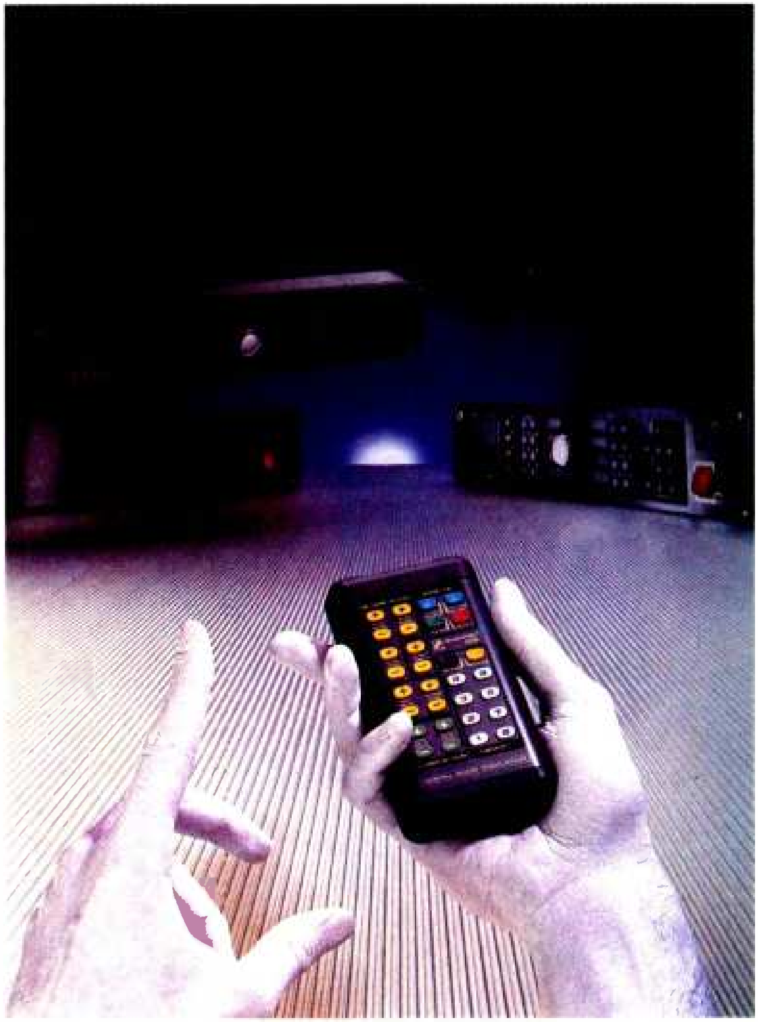

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Quantec GmbH, Sollnerstr. 7a, D-8000 München 71, Tel. 089 / 7 91 40 41, Telex 5 23 793 would provide the most accurate simulation of a movie theater, especially in regard to the level of surround information. As a result, the Dolby Stereo Lt-Rt printing master was made at Todd-AO Stage A, Hollywood.

The use of Sony digital multitracks for the Talking Heads' *Stop Making Sense* concert movie was confined to the post-production stage, with the analog 24-track masters being transferred, with timecode offsets, to a digital dubbing master. Minimal premixing was done by recording engineer Joel Moss at this point, it having been decided to leave balancing of the final mix at Warner Hollywood to Moss and dubbing mixer Steve Maslow. The Dolby Stereo Lt-Rt was bounced up on the digital edit master.

Late 1984 saw the first use of the Mitsubishi X-800 digital 32-track at The Burbank Studios (TBS) for the re-recording of the film *Body Double*. The music score by Pino Donaggio was recorded on an X-800, and mixed down to a four-track 35mm editing copy, without timecode. After he had rehearsed the edits on 35mm mag, music editor Richard Stone noted their location in feet/frames from the Academy Picture Start mark and then transposed them into SMPTE timecode numbers. These in/out points were later used to create a digital edit master by copying the digital four-track mix (which had been bounced up on the original multitrack tapes) to another X-800, which in turn had been pre-striped with SMPTE timecode for each reel.

This digital edit master reel went to the final mix at TBS Stage 5, where the music master mix was recorded on four-track analog 35mm mag, along with four-track dialog and four-track effects "splits." This single mag-film generation was used because there was insufficient room on the X-800 for all 12 tracks of the final mix. Additionally, it was felt prudent to keep the final mix elements (four-track dialog, music and effects) on the same medium, especially in the event that the final mix had to be cut to conform to picture changes.

The X-800 *was* used, however, to record the Dolby Lt-Rt printing master, combining the three, four-track splits. This mix was then transferred to a Mitsubishi X-80 digital two-track for convenience in transferring to the optical negative.

The Mitsubishi X-800 also saw extensive use in the music recording of Francis Coppola's *The Cotton Club*. Although prescoring for the musical sequences was made to 24-track analog, all of the music heard in the final film was recorded digitally. Approximately half of the vocals were post-synced, with the other half being transferred from the 24-track analog masters to the digital mulititrack tapes.

Recording engineer Tom Jung premixed the music onto 24-track analog in preparation for the final mix at Zoetrope Studios in Napa, CA, thus providing sound designer Richard Beggs with an average of 12 tracks per song. These tracks were also "strung off" onto three-track 35mm mag elements for ease in editing, and "slipping" of sync during the final mix.

Digital Recorders in Production

For the past 15 years, almost all films shot in the United States (and most of the world, for that matter) have used Nagra ¼-inch portable tape recorders made by Kudelski S.A. of Switzerland. Although these trusted machines owe their ubiquity partly to high quality (73 dB signal-to-noise ratio at 15 ips), the unit's high reliability perhaps figures more strongly in their universal appeal. (The standard comment is that "you can throw a Nagra off a cliff" and expect it to work.) The QGX2-60 crystal in a Nagra 4.2, the standard mono recorder, is accurate to within 10 parts per million (0.864 frames per hour). Since the magazines of most 16mm and 35mm cameras hold a maximum of 11 minutes of film, the Nagra's crystal is more than accurate for even the longest, continuous take.

What, then, does digital recording have to offer the world of production film sound? Despite the Nagra's excellent technical specs, sound editors and re-recording mixers must still deal with tracks that are seriously under-recorded (peaking at -20 dB on a Nagra modulometer), with dialog buried in tape hiss. Or recorded too hot, with excessive distortion. In some instances, this is not quite the fault of the production mixer, since the rushed atmosphere on film sets today often precludes time for a rehearsal. Also, soft-speaking child actors present a big problem, especially when they are talking to adults in a scene. All of the situations listed above make gain riding (and, perhaps more importantly, gain *anticipation*) an essential skill that can be learned only through much experience.

Another problem that exists in spite of the Nagra is tape print-through. The public (and, many times, even the re-recording mixer) is unaware of the careful frame-by-frame handiwork that dialog editors perform on production tracks. A recent informal poll of experienced dialog editors revealed

PRACTICAL APPLICATIONS OF THE SONY PCM-F1 IN FILM RECORDING

At a John Denver concert shoot in Russia last Thanksgiving, veteran recording engineer Roger Nichols employed a Nakamichi DMP-100 digital processor and VHS Hi-Fi VCR combination to record a total of five tracks on one piece of videotape. "I used the video as the stereo digital [PCM] channel," Nichols explains, "and the Hi-Fi channels for stereo audience; the standard [linear] track contained camera slates.

"Because I was trying out the VHS Hi-Fi deck for the first time, and wasn't sure if the Hi-Fi heads were going to mess with the video 'image' containing the digital audio, I ran a standard Beta deck as a backup." This past June, confident that the system would function properly, Nichols and Denver returned to Russia, and Nichols took with him only the Magnavox portable VHS Hi-Fi deck.

Nichols' DMP-100 had been modified with an internal 60 Hz-referenced crystal, and thus was running without sync connection to the pair of 16mm crystal-sync, 24 fps film cameras. Back in the U.S., the 40 hours of camera negative was transferred to video via a Rank-Cintel flying-spot scanner. During a telecine transfer, 24-fps film is effectively slowed down to the NTSC color-field rate of 59.94 Hz (23.97 fps). The standard method of achieving audio sync, when shooting film for video post-production, is for the camera operator to pan down at the beginning of each camera start and shoot a few seconds of character-generated timecode displayed nearby on a video monitor. Thus, all cameras will have the same timecode location at a given point in a song. This technique also allows audio to be transferred during telecine, since the rough mix (usually recorded on half-inch, four-track, with stereo audio, 60-Hz sinewave and 30-frame timecode) contains the same timecode.

However, since Nichols' shoot in Russia did not employ SMPTE timecode in any form during its production, the telecine transfers were made silent, with the timecode numbers recorded on the videotape (one-inch type-C master and ¾-inch U-Matic for off-line editing) bearing no relationship between cameras.

At his home in California, Nichols mixed down the four tracks of audio — VHS Hi-Fi stereo audience and separate PCM-encoded Denver vocal and guitar tracks — to two tracks on a second PCM-F1. "I was playing back [the original tracks] at 59.94, and the second digital processor was recording at 59.94. The DMP-100 that I was recording on was in the 'genlock' mode, synchronized to the 59.94 crystal of the playback deck. In this manner, it was in fact synchronized to the F1 I was playing back [the concert tape] on."

. . . continued overleaf —

PRACTICAL APPLICATIONS OF F1 — continued . .

The EIAJ-Format mixdown tape was made to ¾-inch U-Matic cassette for convenience in post-production. Again, since there was no timecode used during the shoot, the code recorded on track #2 of the U-Matic EIAJ sound master bore no relationship to the timecode on the videotape transfers. "We used the video coming out of the playback deck as sync for the timecode generator," Nichols recalls.

Nichols had used the standard "linear" analog track to record camera slates with the DigiSlate system. Similar to the bloop light system used by documentary sound/-camera teams over the years, at the beginning of a camera start the operator swings over to an assistant holding up a small slate. When a button is pressed, a bulb is lit at the same time that the DigiSlate sends a "bloop" tone to the recorder. The resultant sync point is easily found, and many systems include an LED readout for numbering takes. This slate information — the bloops plus the vocal slates; for example "camera two, roll four" — was transferred to analog track #1 of the ¾-inch EIAJ master.

At Compact Video in Burbank, the ¾-inch and one-inch videotapes were striped with sound from the ¾-inch EIAJ sound master by finding the proper timecode offset, using the bloop light at the beginning of each camera slate as sychronization reference. "It was perfect, and never went out of sync throughout the whole concert," says Nichols. "The digital processor was locked to house sync [59.94] during the transfer. Therefore, the only variable in the system was the frequency of my crystal in my DMP-100 when I recorded the concert, versus the frequency of the crystal in the 16mm cameras."

After the off- and on-line edit sessions, the U-Matic EIAJ master sound videocassettes were locked to the one-inch master (again after finding the necessary timecode offset) resulting, effectively, in first-generation analog audio.

In the fall of 1982, just after the introduction of the Sony PCM-F1, Nichols had recorded 28 concerts for John Denver using a pair of PCM-F1s. Since genlock modifications were not readily available for the F1 at this time, Nichols was unable to route the video out of one F1 into the other.

Nichols' solution was to "take the clock out of one F1 and run it into the second, so they were synchronized. I also had a timecode generator which was looking at the F1-generated video, and using it as sync for the timecode that was going on the analog tracks of both Beta VCRs. Because there's no Beta II machine that you can lock up [for frame-accurate editing], I transferred the tape, using the copy mode, to ¾-inch F1, regenerating the timecode."

The four tracks recorded during these earlier concerts were apportioned slightly differently than for the Russian shoot: one track for the vocal mike, two tracks for guitar, and one for audience. "I used the delay of the audience that bled into the vocal mike to fake a stereo audience track," Nichols explains.

When using a VHS deck to record with a PCM-F1 (or any PCM processor), Nichols advises that the deck should be modified to disable the dropout compensator, "so that the F1 can do all of its own error correction. Beta machines have a switch on the back marked 'PCM.' Many VHS machines don't have it, so their internal dropout compensator is still active [when using an F1]. So if there is a dropout on the tape, it [the VCR] substitutes a whole line, which makes the errors even worse. In most VHS machines you can wire a jumper to disable the dropout compensator."

Modifying a PCM-F1 Combination for Field Recording

A production mixer bringing a PCM-F1 on a movie set in 1985 probably attracts the same attention that was caused 20 years ago by the sight of a Nagra III. One difference,

Roger Nichols with his four-channel recording system, shown in close up above (*L to R*): Sony eight-input mixer, Nakamichi DMP-100 processor, Magnavox VHS Hi-Fi VCR and ART 01 digital reverb.

DIGITAL FILM SOUND

that rare indeed is the production track that *doesn't* have bad print-through problems.

Reducing noise and distortion, and eliminating print-through, would be a tangible benefit even when the digital master was transferred to non-Dolby 35mm mag stripe, whose signal-to-noise ratio and specs don't come close to those of the Sony PCM-F1, currently the most popular portable digital processor.

(It has to be stated for the record that the biggest problems faced by production mixers — getting the mike in the right place and having a quiet set — can be avoided almost entirely with cooperation from the director, assistant director, and cinematographer. *No* recorder can solve the problems created by these people!)

There is clearly some room in location sound for the benefits offered by digital recording, and indeed many production mixers have begun experimenting with digital processors, mainly the Sony PCM-F1. However, there has been some resistance in Hollywood to the use of EIAJ-Format processors because the units (and their companion half-inch VCRs) were not *designed* to form a rugged, professional film production channel. Probably *the* most controversial and misunderstood stumbling block regarding the acceptance of such digital processors (including, in addition to the Sony F1, the Nakamichi DMP-100, and the 14-bit JVC VP-101 Technics SV-100, and Sansui X-I Tricode) is the fact that the internal sync capabilities of most units were not designed for standard double-system film work (i.e., both the camera and recorder running on internal crystals referenced to 60 Hz).

When a crystal-controlled Nagra is operating, a white cross appears on the front panel to tell the mixer that the 60 Hz Neopilot track (13.5 kHz FM track on a stereo Nagra) is being recorded to tape; a second cross indicates presence of correct speed and power. All professional motion-picture cameras employ motors governed by an internal 60 Hz-referenced crystal, and many of them sound a warning beep when the camera is running off sync speed (24 fps).

During transfer of a ¼-inch Nagra tape of production sound to 35mm single stripe for editing, one of three sync modes is possible: 1, no sync connection, with the Nagra self-resolving (phase-locking the Neopilot track on the playback tape with the 60 Hz crystal) and the mag machine running on its own internal crystal; 2, driving the mag recorder from the Neopilot track using a synchronizer such as the Magna-Tech 93; or, 3, modifying the speed of the replayed ¼-inch tape to follow the crystal reference in the dubber. The latter two methods guarantee a proper transfer, while the first technique is virtually foolproof unless the crystals on both the production and transfer Nagras and the mag recorder are too high or too low in reference frequency. Even when the errors add up, a sync error would probably only be detectable in a long take. This is how films stay in sync.

Synchronizing an EIAJ-Format Processor

In the "record" mode, when using a standard half-inch consumer videodeck (such as the Sony SL-2000), a U.S.-bought Sony PCM-F1 feeds NTSC 59.94 Hz composite video into the "video input" of the VCR. During playback, the F1 processor locks to incoming video, which is governed by the 59.94 Hz field rate of the crystal in the video playback deck.

Which brings up the most obvious issue with regard to the sync capabilities of an unmodified EIAJ processor: It Is Not Referenced to 60 Hz!

We are again back to the issue of the internal crystals fitted to the processor and VCR: Is 59.94 Hz acceptable? The answers from experienced mixers and engineers was a "definite maybe."

The "definite" stems from the concept that if the F1/VCR combination records and plays back with reference to the 59.94 Hz, even though the camera utilizes a standard 60 Hz crystal, picture and sound will match because they both obey the same "clock on the wall." One second is one second.

David Smith, chief engineer of Editel, New York, has made extensive tests on the F1, researching its use in video editing. "When you are recording, the *F1* is the master; in playback, the playback deck's *crystal* is the master. If the playback deck and the F1 are both close to each other; it's pretty good.

"In television post-production, where we use the F1 frequently, we have to pull it up about one frame every seven-to-10 minutes if it isn't synchronously locked to the master video generator. If you are going to shoot three- or four- minute scenes, or 30- or 20-second clips, it works fine. If you are going to do the sound for a half-hour scene, however, you would have to accept an external timebase."

Richard Topham, Jr., general manager of Audio Services Corp. of North Hollywood, is not only one of the largest Nagra dealers in the world, but also rents the PCM-F1 for film use, and has sold over 1,000 of the Sony processors. Although all of his rental F1s are modified to accept external sync, he recommends using the F1 on its internal 59.94 Hz crystal and has had good results with it. "I know it's right and I stand behind it," he says. "There are a lot of guys who think you *have* to [resolve to] 60 Hz because you are dealing with film. I do not believe that this is true. It doesn't matter if it's 59.94, as long you *resolve* it to 59.94."

The most obvious way around the problem would seem to be to print a crystal-generated, 60-Hz sinewave on the VCR's longitudinal analog audio track, and then use this signal to drive the mag recorder during transfer. David Smith has utilized such a technique with a Sony SL-2000 Betamax deck, and reports that it works fine. It should be noted, however, that it is possible that the replayed sync signal will contain a significant wow and flutter content. If the phase-locked loop in the synchronization system tracks accurately during transfer, the mag copy might also be full of wow and flutter, just like the VCR. This problem could be avoided altogether by taking up one of the two PCM audio tracks to record the sync signal, but that is a compromise that many mixers would not like to make.

The other two methods of achieving 60-Hz sync on location involve either installing a 60-Hz referenced crystal

PRACTICAL APPLICATIONS OF F1 — continued . .

however, is that while the Nagra did not offer any substantial quality perks over recording on 35mm mag, it was (and is) much more portable and convenient than even the smallest mag film recorder.

On the other hand, while the PCM-F1 offers potential for increased quality, it is a consumer unit, and one not designed to be "thrown off a cliff." Furthermore, the features that are taken for granted in a Nagra, such as balanced mike inputs, low battery consumption and crystal sync, must be jerry-rigged in an F1.

The speed with which an F1/SL-2000 combination eats up the NP-1 NiCad packs (20 minutes on the F1, one hour with SL-2000 VCR) has led mixers to seek other sources of power. While AC is an obvious answer, accessibility can be a problem, the power cord being one more hassle to deal with.

One common solution is the modification of a 12-volt auto battery for use with the F1/SL-2000. Audio Services makes 12-volts a stock item available to mixers, and claims that they will run a digital production channel for 12 hours.

One of the main reasons why powering is such an issue with production mixers is that it is important to keep the "numbers up" on the video deck, to allow the transfer person to find the "print" takes. When using an F1, mixers "zero" the timer on the video recorder at the beginning of a tape, and write the start-of-take times for each tape on the log sheet. (It is essential that if a Nagra is used as back-up, separate logs be used for the F1 material and the Nagra, to avoid confusion by sound editors as to what track they are dealing with. On the subject of logs, it is probably a wise idea to note whether the F1 was running off its standard, 59.94 Hz internal crystal, or whether a 60 Hz-based system was used.)

The mixer only has a few seconds warning that the batteries are about to go, and not only might everything grind to a halt in the middle of a take, but the numbers will be wiped out. Nagras don't need any timing system because transfer recordists are used to fast winding the ¼-inch tape with the head lifters defeated, listening for either a 40 Hz tone under the voice slates before a take or the two "pips" after a take.

Jacques Nosco, who has used an F1/SL-2000 combination on many films and commercials, uses an Audio Services 12-volt battery while recording, and switches to the NC-1 NiCads between takes to keep the timer active while saving the large battery. Although 12 hours would seem to be enough time for one day's shooting, Nosco notes that "in the film business, the days get longer and longer later in the week, and on Friday you could be working until two or three in the morning. You put in 16 or 18 hours and at those times when it is the roughest is when the battery may fail. Then you're running around looking for AC at the worst times."

Any 12-volt battery will do the job, and Rich Topham of Audio Services remembers that he had one client "who wanted to use the F1 on his boat to record seals. We made him a cable that clip-leaded to the battery of the boat." While most production mixers in features use outboard mixing boards with their Nagras, there is frequently a need to mike directly into the machine. The F1 shows its consumer roots again with its ¼-inch unbalanced mike inputs. Topham says that many people who use Schoeps mikes with an F1 for stereo sound effects recordings jumper pins 1 and 3 on the XLR connector to "more or less balance it.". The only difference is that you don't have transformer isolation, although I haven't had any problems with ground loops on battery operation."

Possibly the most important tip concerning the use of digital production recording is to make sure that it is *used*, and not left on the shelf while the Nagra ¼-inch back-up is transferred. Jacques Nosco neatly avoids this by handing in his Nagra tapes a day later, after the F1 tapes are transferred to mag for dailies. □□□

DIGITAL FILM SOUND

in the F1, or modifying it to accept an external 60-Hz sync source.

An easy way to obtain a 60-Hz F1 is to purchase a PAL/SECAM version. As it turns out, PA/SECAM F1s come with the necessary peripheral circuitry, and the ability to alter sampling rate for both crystals is integral to the unit. However, the rates are not *switch*-selectable, and must be modified to be so. Although the PAL F1 runs at 50 fields, the sampling frequency is 44.1 kHz, which is a multiple of 60 Hz (as opposed to 44.056, which relates to 59.94-field NTSC color video).

Among the companies that offer modifications of Sony digital processors is Audio+Design/Calrec, of Bremerton, WA. Although PAL operation is part of the modification they make to the PCM-701ES, the company's Tom Gandy notes that "if you want an F1 to run at 60 instead of 59.94, you should call up Sony Parts, lay down a few bucks for a crystal and pop it in; it's a five-minute job. You need a service manual to see which circuit to pull, but it's clearly labeled."

The second way to achieve 60-Hz crystal sync is to modify the F1 to accept an external sync input. David Smith says that his machines "take 60-cycle sinusoidal sync and convert it to vertical drive component of video: squaring it, making it the proper level. The width must be adjusted to 10 lines, which is 640 microseconds. I then do an external video lock to this 'phony' video that has been made from the 60-cycle sinewave. You will then get 60 cycles and a sampling frequency of 44.1 kHz."

When working with F1-encoded material that has been recorded at 60 Hz — either referenced to internal crystal, or an external 60 Hz source —the mixer must be sure that the playback deck used during transfer can accept external sync. David Smith explains: "The F1 is going to spit out [during recording] 60 fields, 30 frames at [a sampling frequency of] 44.1 kHz, while the playback deck will be at 59.94/29.97 at 44.056. You will get a time slide — it will play back slightly slow — and a pitch shift.

"We modify the machine that is playing back the F1 material so that it runs at the *exact* 60 cycles that it was recorded at. In other words, the playback machine's internal crystals are all at the 59.94 field rate; we have to up it to 60 by feeding in an external source of 60 into the machine. Then you don't have to put 60-cycle sinusoidal on the longitudinal track to get a perfect playback."

Audio Intervisual Design, Los Angeles, offers two levels of modifications for the F1. Stage one allows the unit to lock to incoming composite video , while not affecting normal crystal NTSC operation. The second step enables the F1 to lock not only to composite video (60 or 59.94 Hz), but also to "just about any sync source you might have. This includes vertical drive, squarewave, sinewave, etc.," according to AID's Mike Novitch. A 60-Hz crystal is also installed, in addition to an RCA jack, which provides a sinewave output from the reference frequency. Whatever sync is coming into the machine the F1 is referenced to is now going out, so if anyone else needs to see it, you have that option."

Two switches allow conversion choice between internal 59.94 Hz and external sync, which has three options: 60-Hz crystal, external 60 Hz, and external composite video.

Despite his high regard for the quality of the F1 processor, Audio Service's Rich Topham recommends using a Nagra as a backup. "The F1 is too new and mixers have to cover themselves," he offers. "*I'd* like to have a $7,000 machine sitting next to a consumer video deck and PCM processor."

Some of the problems presented by an EIAJ-Format processor can never be "solved" in its present form: for portability, the unit has to be connected to a half-inch consumer VCR, and neither machine is film-ready. One cannot throw a F1/SL-2000 over the shoulder, for example, and run with it during a shoot without fear of accidentally hitting the wrong switch or power running down. Also, off-tape monitoring is not possible and winding back to listen to a just-recorded take is not as convenient as it is with a Nagra.

All of which leads everyone to the same question: Who will come out with the first "digital Nagra"? There is no official word from Kudelski — or, for that matter, any other major manufacturer — that they will make this happen in the near future.

The industry (and much money) awaits the first company on the block with a portable, one-piece, "bullet-proof," sync-ready digital recorder.

RANDOM-ACCESS SOUND EDITING

The potential sonic benefits of digital sound effects and production recordings are minimized — some might even say eliminated — when the digital tracks are transferred to 35mm mag stripe and then re-recorded four times before reaching the final analog print. An "easy" solution would be to utilize a system based on digital 35mm recorders which, presumably, would feature an analog guide track to allow editing on Moviolas, flatbed editing tables, and sync blocks. While this capability would take care of the quality/generation loss issue, it would do *nothing* in regards to what is probably the biggest problem faced by sound editors: tight deadlines, abetted by frequent picture changes.

Digital 35mm tracks would still have to be *manually* synced, edited, and leadered. Dozens of elements would have to be re-cut every time the director or editor decides to make a change. If two feet of picture are added at the 400-foot, three-frame point from the start mark, every sound unit must reflect the update or sync will be lost 400 feet and four frames into the reel.

Along with this recutting process comes the rewriting of the cue sheets, with perhaps a half-dozen assistant editors working overtime coordinating the whole show. (The record world should count its lucky stars that it has no analogy to this problem.) Multiply this headache by the 12 reels in an average film and you have many talented sound editors wasting much of their time just keeping up with changes.

The recutting problem outlined above is one reason why digital random-access sound editing one day will completely replace the venerable Moviolas that have served sound editors for over 55 years. ("Random-access" is partly a misnomer, because digitized sounds would be pulled from Winchester hard disks or optical disks, etc., and not from random-access memory; it will be many years, if not decades, before RAM becomes inexpensive enough to store the large quantities of digitized sound effects needed during an edit/mix session.)

Not only would sound be able to remain in the digital domain, but the labor- and time-intensive manual steps involved in 35mm feature sound effects editing — auditioning ¼-inch tapes, transferring selected sounds to individual mag rolls, editing sounds to fit the picture, leadering and labeling the sounds onto reels, and writing up cue sheets — would be condensed dramatically.

Random-access editing would allow one person to audition, say, all of the explosions in a library, edit the chosen effects to fit the picture (perhaps adding digital processing), and print out a cue sheet in the time it would normally take to audition the sounds (assuming the library was cross-referenced on a computer!).

With a random-access system (as we can best envision them today), there will be little difference between the hardware used in sound editing and mixing. The chosen explosion, for example, will never actually be copied from the original disk auditioned by the sound editor. Instead, only when that reel is pre-mixed will the sounds be re-recorded, placing them in accordance to the timecode-based Edit Decision List (EDL) created by the sound editor. If the editor will not be mixing in house, the required digital tracks could be transferred to a digital multitrack for playback at the dubbing stage.

Since it is presumed that both picture *and* sound editing will be connected to a central computer database, such mundane but time-consuming chores as revising cue sheets and conforming the audio to match picture changes will present little problem.

Such freedom is more complex and time-saving than might be imagined. Perhaps it can most clearly be stated that *all* material recorded for the film — production dialog, sound effects, music, Foley and ADR — along with library sound effects, are always readily accessible in any order at any time to anyone. Thus, if the sound editor wants to modify, for example, a car crash pre-mix at the time of the final mix, this can be accommodated readily, even though the picture cut for that reel has changed through *three* later versions. Today's technology would require a few hours of editing (conforming the original cut tracks to the current version) and mixing time — while trying to remember all of the EQ settings, etc., used during the pre-mix, which may have taken place weeks ago.

It should be made clear that the problems concerning today's analog 35mm-mag-based technology are also present in any digital recording format that would be stored in any serial medium — digital multitrack tape, digital 35mm mag film, digitally-encoded videotape, etc.

SoundDroid

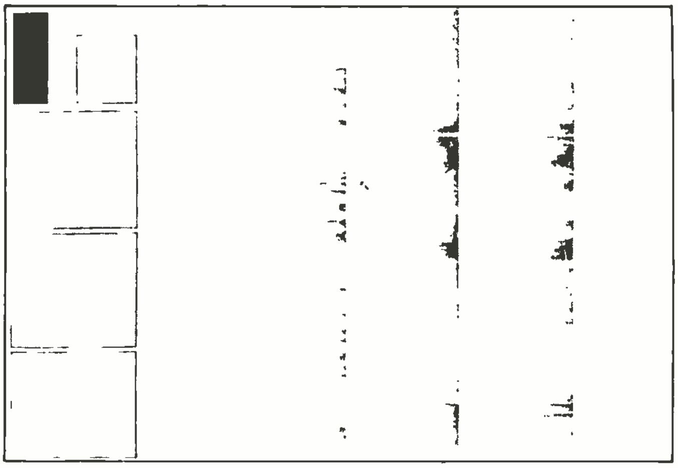
One of the most eagerly anticipated events at the 1985 NAB Convention in Las Vegas was the introduction of the Lucasfilm/Convergence SoundDroid sound editing and mixing system. Since the last *R-e/p* update on digital sound research at Lucasfilm, project leader Andy Moorer and his staff of six have concentrated on the "front-end user interface." The system is currently using a touch screen laid over a high-resolution graphics VDU, which is a slight change from the trackball configuration used in the companion Lucasfilm/Convergence EditDroid picture editing system.

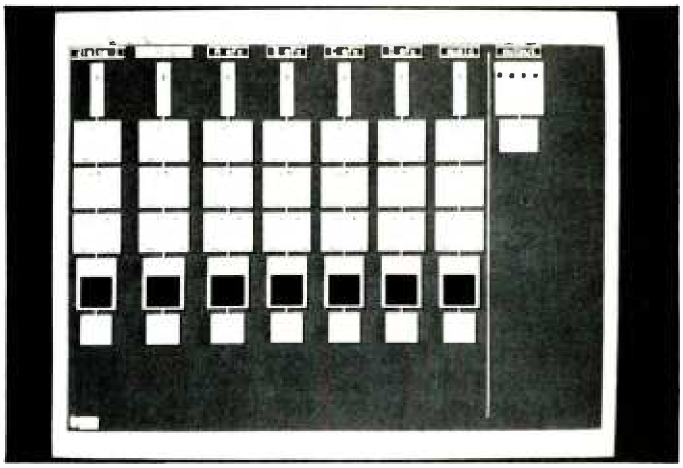
The touchscreen "gives the user a more direct way of interacting with what's going on on the screen," Moorer offers. "Rather than reaching off on the side, you point directly at it. But you can use a trackball or mouse with the system."

The 1024- by 800-pixel bit-map screen has three basic formats. The Electronic Cue Sheet, not only lists footages, but also displays action description and dialog. The Meter Screen allows for patching of three-band digital equalization, reverb, and panpotting, while indicating the signal flow. A recent improvement to the Meter Screen is the concept of "pages," allowing the user to immediately jump to eight other channels, as opposed to the scrolling feature used earlier. Moorer notes that "people were asking for faster console response, and we were willing to give up a little flexibility. The paging feature allows us to bring up eight more faders in a flash." Finally, the Library Screen accesses the database of sound effects and production recordings.

The current SoundDroid hardware configuration consists of two Motorola 68010 microprocessors — including one for the control computer — two Mbyte of main memory, the hi-res graphic display, and a fixed Control Data 825-Mbyte Winchester hard disk that holds up to two hours of 16-bit, 48 kHz mono sound. The second 68010 is located in the ASP (Audio Signal Processor), which controls up to 16 DSPs (Digital Signal Processors) and includes the control panel/console.

While the basic ASP contains one





SoundDroid's Electronic Cue Sheet screen (*left*) indicates audio modulations, start/stop points, and slate information for three tracks simultaneously. Panning, EQ, and reverberation are handled by the Meter Screen (*right*), and are controlled by eight assignable knobs.

DSP, allowing 16 channels of digital audio to be processed in real time, an ASP can handle up to 256 channels simultaneously when connected to the full configuration of 16 DSPs. Each DSP, in turn, can control up to 16, 825-Mbyte disks, thus allowing instant access to a staggering 512 hours of mono digital audio.

Since having access to 256 tracks might be considered overkill for even the most elaborate Lucasfilm soundtrack, one might wonder why such capabilities were built into the system? The answer lies in the concept of a facility that would share the disk drives among many SoundDroid editing and mixing stations.

At the present time, the on-line working store utilizes the 825-Mbyte CDC hard disks, which are cheaper, faster and more reliable than the 300-Mbyte drives "equipped with interchangeable disk packs" that the SoundDroid team has been using since the beginning of the system's development. The fixed disks are "dual-ported," and would be connected both to a SoundDroid station and to a transfer-room robot arm changer/archive machine. Thus, while the mixing theater is using the "A" disks, the "B" disks are assigned to the transfer machine and can be loading for the next session or the next reel, or used for archiving yesterday's work.

The DSPs and the disk drives that they control can only be used with their asssigned SoundDroid; the dubbing stage can't "borrow" a few disk drives during a busy reel *unless* the lending station is not in use.

Later on, during final mixing, with most of the editing stations not in operation, the SoundDroid mixing station might commandere half of the DSPs in a facility. The other half might be used by an editing station to pre-mix and to conform tracks to picture changes.

Hand-in-hand with the issue of how many channels can be processed in real-time is the question of how many tracks are on-line, and how the off-line material will be archived — a problem that has to be addressed by all random-access editing/mixing systems.

Moorer feels that the solution to the long-term archival storage problem is 10-inch, write-once optical disks (a product also made by Control Data) and which hold an hour of stereo information on each of the platter's two sides. "The price compares favorably with digital master machines. We can buy, off the shelf, a robot arm changer [made by FileNet] that will hold 64 platters, and can insert the disks into any of four players," Moorer says. The changer, which functions in much the same way as a juke box, can retrieve and load an optical disk, and hig "play" within 15 seconds of receiving a command.

Because the response time of the optical disks is slow (they can record and playback in real time only), and because they are only capable of ste-

SoundDroid mixing console, featuring GML/Penny & Giles moving faders.

reo transfer, their use will be limited to the off-line storage of sound effects and production tracks. If an optical disk needs to be accessed during an edit/mix session, the first audition would be recorded on the hard disk for future instant access. Multichannel material, such as pre-mixes, final mixes or music recordings, can also be archived to — or transferred from — optical disks, albeit only two tracks at a time. Streaming tape or multitrack digital recorders might come into use at these latter stages of post-production, although even then the CDC 825-Mbyte drives would be limited to simultaneously transferring 10 tracks of digital audio at a time.

Moorer notes that the media cost of the optical disks — approximately $175 per platter — is fairly high, although this figure is expected to drop considerably when production is up to speed.

The *modus operandi* described above suggests that the SoundDroid does not become cost-efficient unless a system approach is taken; a basic SoundDroid costs approximately $200,000, including synchronization capability and a single disk drive. This money will provide only an eight-in, eight-out digital recorder/console/editor. No extensive mixing would be possible; a SoundDroid re-recording station, incorporating basically three SoundDroids consoles connected to a single ASP, is currently under development.

The system (perhaps "facility" is a better approach) has been emphasized by Andy Moorer since the beginning of the Lucasfilm Digital Audio Project in April 1980. The price of a SoundDroid facility built from scratch would be approximately the same as a similarly equipped state-of-the-art analog facility. (Moorer relates that in the new Lucasfilm Sprocket Systems building on the Skywalker Ranch in Marin County, CA, there will be only *two* 35mm mag machines!)

It is presumed that such an all-digital facility would also include EditDroid picture editing stations. SoundDroid was designed from the start to interface with the EditDroid "in terms of message formats, communications standards, and the type of computers that we would be using," says Moorer. "EditDroid has the same basic computer system, operating system [Unix], and is programmed in same language [C] using the same database management system and Ethernet protocol. We can pick up a [Edit Decision] list from EditDroid and edit directly to it: location of scene changes, whether they are cuts or dissolves. And, when picture is re-cut, conforming of all sound material is semi-automatic."

Moorer notes that there are three levels of software in the SoundDroid: "There's the front-end user interface; then there's what we call the 'real-time monitor' — which actually runs in a separate computer — that handles things like disk scheduling and the loading and unloading of microcode. The third piece is console response and automation. Now we are rewriting part of the user interface, and starting to rewrite the automation. We got a lot of good ideas and comments at NAB."

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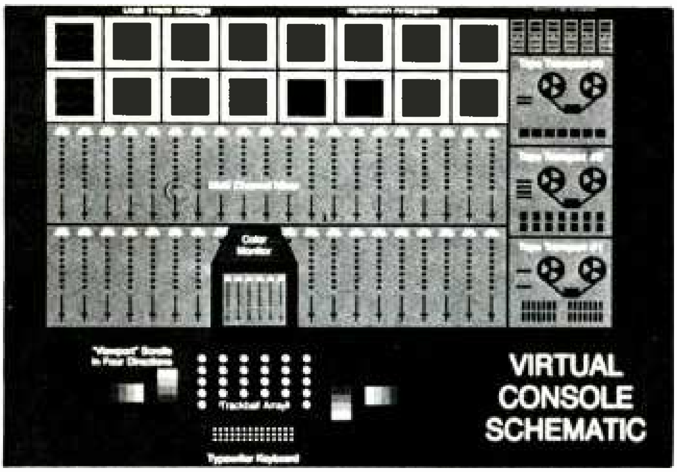

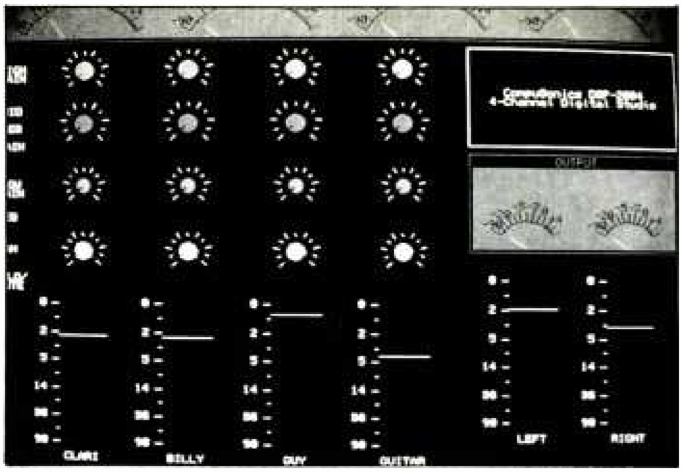

Schematic (*left*) of the "virtual console" design of the CompuSonics DSP-2000 Series mixers/recorders. At *right* is the CompuSonics console screen, with all of the "knobs" and "faders" controlled by trackballs. These screen graphics are in full color.

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The SoundDroid staff is currently in the process of developing user interfaces in the mixing domain, including patching, mixing and monitoring. "We have set up standard patches for most of the mixing desk functions, like equalization, panpots, reverberation," Moorer explains. "We've also got graphical front ends for the loop and doppler-shift programs."

The SoundDroid sound-editing program, now called SD (originally "FMX" and "EdiSon"), allows eight audio tracks to be manipulated simultaneously. The tracks can not only be viewed on the high-resolution display, but also mixed on the GML/Penny and Giles motorized fader system.

Moorer and his staff have experimented with a set of touch-sensitive, six-inch Farenstat control strips made by Tasa Electronics. "It's entirely capacitive — touch sensitive — so that nothing actually moves. The good point is that there is no nulling problem; you just pick your finger up and put it somewhere else. The bad points are that the resolution is not as fine as you get with regular sliders. Also, to get the same resolution on a Farenstat that you get with a normal fader — from 'silence' to 'wide open' —the distance would have to be longer than six inches, which would mean that you would have to push it down with three strokes [to achieve the same control range]. Also, since there is no mechanical feedback, you are relying entirely on your ears. The guys at Sprocket Systems have built into their fingers what a 3-dB movement feels like; it's absolutely automatic and I'm a little hesitant to break that training."

SoundDroid saw its first use in the processing of certain effects for *Indiana Jones and the Temple of Doom*. *Amadeus* utilized the SoundDroid to help clean up about three minutes where Salieri is talking over quiet music passages. The noise on the production track interfered with the clarity of the music. Moorer remembers that he calibrated the system "using samples of the noises [between words] to let it know what the noise energy of each band was. It would then automatically set the gate thresholds."

Alpha testing of SoundDroid for the sound editing of a feature will begin this Fall. The SoundDroid will be used for all standard sound editing: the picture editor's work track will be loaded into the system after copying onto optical disks. Dialog clean-up and splitting of tracks, along with standard sound effects editing, will be done totally on the digital processor.

Because the current channel capacityof the prototype system is limited to 16, the final mix will be performed on Lucasfilm's Neve 8128 analog console, playing back from analog copies of the cut and edited digital tracks.

SoundDroid will be shipped to beta test sites in December. Mary Sauer, director of marketing for the Droid Works, notes that "there is more interest for the beta units than we can meet." The first production models will be shipped toward the end of the first quarter of 1986, she says.

Both the EditDroid and the SoundDroid are manufactured by The Droid Works, a joint venture of Lucasfilm Ltd. and Covergence Corporation that will handle their marketing, sales, and maintenance. The systems have built-in diagnostic routines that can be run over modems for remote troubleshooting, bringing a standard practice found in the world of mainframe computers to the film sound industry.

CompuSonics DSP-2000

CompuSonics was founded by David Schwartz in 1982 with the intent to design and manufacture a floppy disk-based digital record/playback unit for the consumer market. Armed with a prototype of the DSP-1000 consumer unit, $750,000 was raised in a public stock offering, with another $1.5 million added in April 1985.

When development systems for the floppy-disk consumer unit "started looking more and more like like very nice professional systems," says John Stautner, CompuSonics vice president, "we decided to market those as well, in addition to developing software for professional hard-disk applications." In May 1984, CompuSonics formally introduced both the consumer DSP-1000 and the professional DSP-2000 Series of random-access digital recorders/editors/mixers.

Vitello & Associates, a film and TV sound editorial company located in North Hollywood, CA, received the first CompuSonics DSP-2002 two-channel editing/recording system in November 1984. The basic $35,000 unit contains a CPU module with Motorola MC68000 microprocessor and TI TMS320 digital signal processors. The 16-bit system operates at a sampling frequency of 50 kHz, other standard rates being optional. Sounds stored on the unit's 140-Mbyte hard disk are accessed via a keyboard and 12-inch monochrome monitor displaying menu-driven software. The system is modular and can be upgraded, with the addition of plug-in modules, from the two-output DSP-2002 to a four-channel DSP-2004 system with trackball mixing console and 19-inch color graphics monitor.

One of the first assignments Paul Vitello gave his new system was the creation of sound effects for 125 episodes of the animated stereo TV series, *Voltron*. Although in early episodes the sound effects were layed in "on the fly," CompuSonics introduced a SMPTE timecode interface to enable sound effects to be triggered at specific timecode locations. Another recent software development allows more than one stereo pair of tracks to be built for a given timecode location, thus creating a "playlist" that allows an editor to simultaneously lay multiple stereo pairs separately onto mag film or a multitrack. This facility is in contrast to the DSP-2002's original design, which allowed only two tracks to be created and replayed at one time. Stautner says that the software is "continually under development, and we add new features and programs to it all the time and send out upgrades."

After the CES Convention in June 1984, the staff at CompuSonics became aware of interest in the system being generated by broadcasters, as well as for film and video post-production. The 3.3-Mbyte "SuperFloppy" disk drives currently used in the DSP-2000 units will soon be expanded to 6.6 Mbytes for the professional broadcast version of the DSP-1000 scheduled for introduction this winter. The 6.6-Mbyte floppy disk is capable of storing and replaying up to one minute of mono sound, which is considered long enough for station IDs and commercials; with Compusonics' data reduction techniques (described below) entire singles could be stored on a 6.6-Mbyte disk.

When the consumer DSP-1000 is finally released, it is expected that the capacity of the SuperFloppy will be 25 Mbytes, to accommodate the longer recording times of albums. Both versions of the DSP-1000 can be connected via an RS-232 serial port to a personal computer, allowing editing and even noise clean-up from the keyboard. CompuSonics plans to furnish software written for the IBM PC and its compatibles.

The Music Workspace module utilized in a basic DSP-2002 contains a SuperFloppy 3.3-Mbyte disk drive and one 140-Mbyte hard disk, with room to add three more drives. (The SuperFloppy is primarily used in the DSP-2000 Series for mastering to the DSP-1000 format, and to load in applications software.) Up to seven additional Music Workspace expansion modules can be added, each containing a minimum of two and a maximum of four 140-Mbyte hard disks.

Using, as a rule of thumb, the fact that 10 seconds of mono audio can be stored per megabyte (assuming a 50 kHz sampling rate and normal error-detection overheads), each formatted 140-Mbyte hard disk holds up to 10 minutes of stereo sound, yielding 40 minutes of storage time per Music Workspace module, and up to a maximum of 5.3 hours of stereo sound on-line simultaneously with a fully loaded system.

However, these figures do not assume the use of the patented CompuSonics CSX™ data-reduction scheme, which allows the SuperFloppy to serve as a viable recording medium. CSX analyzes the time, frequency and amplitude content of the incoming audio signal after it has been broken into a maximum of 128 bands. A short-term "model" of the signal is built using the filters, and the parameters stored by the system. The model adapts itself to the changing audio content, and is updated every 10 milliseconds.

Data reduction can be introduced either during or after a recording, and Stautner notes that "you can reduce up to a factor of two without any loss of data at all; after expansion, all of the bits are still there. If you compared the reconstructed signal with what you originally recorded, there would be *no* difference. This technique is called the 'loss-less' data reduction algorithm, and it is important to note that the amount of reduction is program dependent; in some cases we have seen it go up past a factor of three, and in others it is less than two."

Further data reduction — up to a factor of eight — does involve the loss of data bits, although Stautner emphasizes that the amount of data reduction, or its use in the first place, is at the discretion of the engineer at

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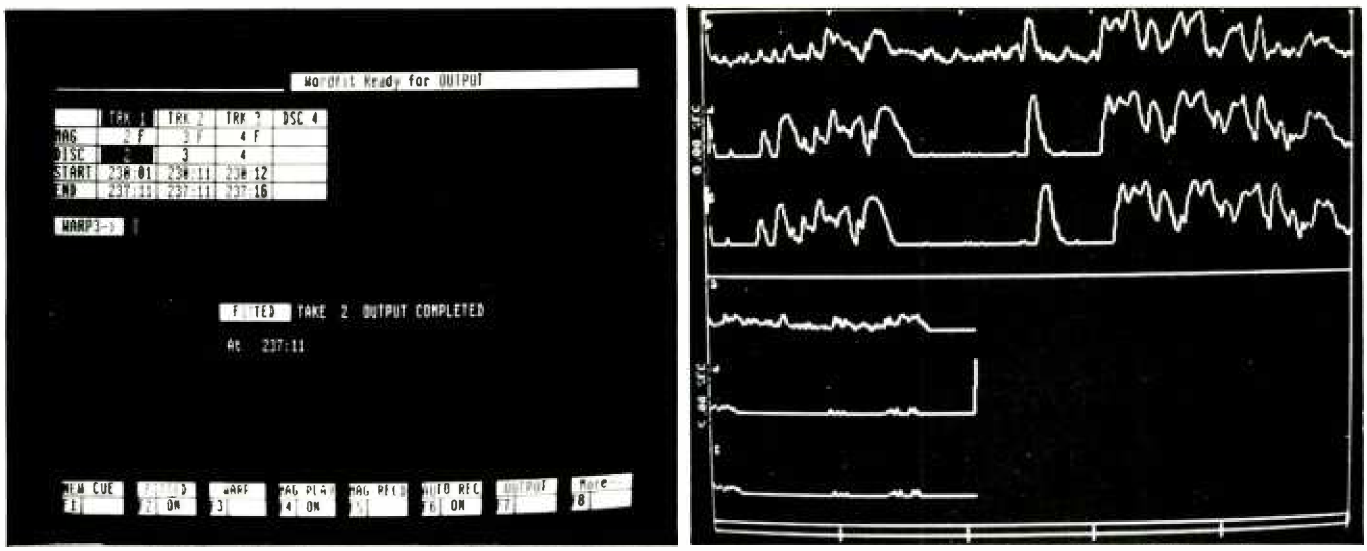

Wordfit ADR system screens. *Left* is the operating menu; at *right* are traces of the guide track (top), fitted track (center), and unfitted track (lower). One screen width is equal to five seconds; the next two seconds of all three tracks are on the bottom half of the screen.

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all times. "It's just like a tape-speed knob; if you put it on 15 ips, you get high-quality recording, but you get less time. On 7.5, you get less quality, but more time."

The current method of archival storage in a DSP-2000 system is by means of 500-Mbyte streaming tape drives manufactured by MegaTape and costing $9,500. The unit can run at two speeds, the slower speed providing back up for four channels of 16-bit, 50 kHz audio in real time, or two channels at twice real time. The company encourages the slower speed because "it's easier on the tape."

By October of this year CompuSonics says that 10 DSP-2000 systems will have been delivered, all but three of which are the two-channel DSP-2002 editor/recorder. The firm reports receiving several orders for larger consoles: one each for the 4×4 DSP-2004, 8×4 2008 and for the 16×4 2016. Stautner reports that the company is "concentrating on the two-channel systems, because they have a shorter delivery/lead time."

The DSP-2004 trackball mixing array is equipped with five rows and six columns of trackball controls, all capable of being assigned in various configurations according to the application program. The color screen translates trackball movements into the plan view of a familiar-looking "console," with slider faders, panpots, EQ knobs, VU meters, etc.

In larger models (DSP-2008, etc.), notes Stautner, the user can " 'scroll' the control panel on what we call a 'virtual console.' In other words, if you have only four channels on the screen, you can 'move' down the console according to a legend at the bottom of the screen. Instead of me getting up and reaching over, I 'roll' over to it."

The trackball mixers range from the 4×4 DSP-2004 at $50,000, to the 64×16 2064, which costs $400,000. All units come with one hard disk, and each 560-Mbyte Music Expansion Module costs $25,000. The software included in all DSP-2000 Series units comes with one-year free upgrades.

Wordfit ADR System

Production dialog for motion pictures produced in the U.S. is replaced almost exclusively using the ADR (Automated Dialog Replacement) system. The actor watches a projected image of the scene and hears in a headset the "guide track" of the original production recording that will be replaced, with three "pips" counting down to the beginning of the line. In some facilities, video is used for playback and audio is recorded on a multitrack; most studios, however, still use

35mm picture playback and three- or four-track 35mm mag recorders. In both cases a sync-pulsed ¼-inch tape is always running as backup.

Regardless of what medium is used to record the looped lines during an ADR session, copies of individual tracks are "strung off" onto 35mm stripe for fine-tuning of lip sync. It can be safely stated that some amount of "massaging" of sync — a frame here, a sprocket there — is always needed, the precise amount being dependant upon the skill of the actor and the amount of time the ADR editors have to prepare the tracks for dubbing. Thus, a small army of ADR editors will often be required during post production of a film that has a large percentage of the production track replaced in ADR.

Probably the first application of digital random-access techniques for the ADR process is the Wordfit system, designed by Dr. Jeffrey Bloom and Nick Rose, of Digital Audio Research Ltd. of London. (The system's co-inventor, Garth Marshall, is no longer with the company.) In a nutshell, the Wordfit system compares the guide track to the looped line being recorded, and tries to edit the replaced dialog to "mod match" the original recording. The idea is that after the ADR session, little correction will have to be made in terms of sync. In addition, the onus of achieving sync is partly removed from the actor, hopefully allowing him or her to concentrate on performance rather than timing.

When the actor says a line during an ADR session, it is recorded digitally (16-bit, 32 kHz sampling frequency) onto a Winchester disk, in addition to the studio's standard multitrack or 35mm mag recorder. As the actor is speaking, the Wordfit processor performs spectral analysis both on the guide track and on the track being recorded. Analysis — and recording of the looped track onto the Winchester disk — begins when the mag record mode is activated.

Results of the spectral analysis are compared in one of the system's two computers, which tries thousands of different trial aligments of the "local spectra." The program will try to align one pattern to another to a tolerance of ±10 milliseconds, or the rough equivalent of a sprocket hole in 35mm mag (96 sprockets/second). The length of the track is reduced or extended by making an average of 10 edits per second on the "unfitted" (normal) recording, which is saved on the hard disk. No pitch-shifting of elements is employed, since the Wordfit system stores a list of the edit points that are necessary to create the matched version. (The "fitted" version is not recorded on the hard disk, only the calculated edit points.)

When the spectral analysis and editing are completed — after the actor finishes delivering the line but usually before the picture and track have been rewound for playback — the ADR editor has the choice of listening to the unfitted track (normal), or to the fitted version, which the system constructs from the edit list. While the fitted version is auditioned, the editor has the option of recording it on the mag recorder. Incidentally, the fitted version cannot be recorded on mag at the same time as the actor is speaking because the processing must allow for instances in which the actor may be late in reading a line. (It cannot process what it doesn't have!)

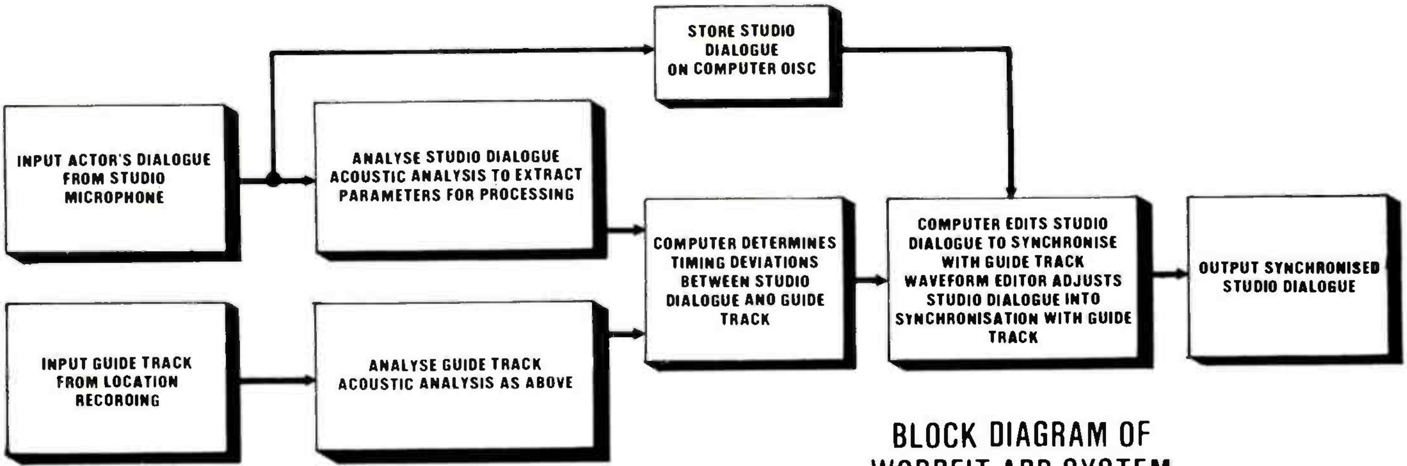
The 168-Mbyte Winchester hard disks available in the just-released production models of the Wordfit system hold the software plus approximately 35 minutes of 32 kHz-sampled mono audio. Up to nine takes of a line can be held on the disk, with the control screen indicating if the fitted or unfitted version has been recorded on the magnetic recorder, and what takes have been recorded to hard disk and mag.

The Wordfit system can analyze guide tracks that contain a large amount of background noise, which is an important consideration since background noise is often the main reason for a scene being looped in the first place.

As the saying goes, there is no free lunch, and Wordfit comes with certain limitations and caveats. In light of this there are four types of processing "warps" that can account for most situations encountered in looping. These settings depend upon both the background noise and the number of people speaking.

Warp 1 is intended for guide tracks containing little background noise, and little overlapping dialog. Because of this, Warp 1 can correct lines that are between -0.5 and +1.0 seconds out of sync. Warp 2 is less flexible, and places slightly more responsiblity on the actor for sync correction. Warps 3 and 4 are more "stiff," and therefore will find more use in high-noise situations or with multiple speakers.

Since, according to Wordfit literature, Warp 4 "should be reserved for situations in which it is not desirable to alter the timing of the replacement dialog too much," this configuration should prove useful for revoicing a part with a different actor or, as is the case during foreign dubbing, with a different *language*. (Although, this technique has only been tried experimentally, Bloom notes that "in prac-



BLOCK DIAGRAM OF WORDFIT ADR SYSTEM

tice it has been found that Warp 4 is possibly the most consistently useful because it copes with a wide degree of acoustic conditions in the guide track. In practice, Warp 4 has been left on for the majority of the looping sessions.")

Bloom notes that "there shouldn't have to be a selection of the warp modes, but we have found that with the state of our understanding of how to do time alignment, it is more useful to let the operator do an initial assessment of the guide track for background level and number of people speaking."

Among the situations where Digital Audio Research advises caution is when a deliberate pace change in reading might be desired. Since the processor "looks" to the guide track for timing, the system should not be used in these instances.

Wordfit has been in routine operation in London since early 1984, primarily at Mayflower Film Recording, Ltd., where it was used in the looping of such films as *Dune*, *A Passage to India*, and *The Killing Fields*. Its first use in the U.S. was in March/April 1985 at Directors Sound, Burbank, for *The Goonies*. Universal Studios took delivery of the first production unit in September of this year. Cost of the installed system is currently $91,000.

Bloom reports that Digital Audio Research is currently developing a "low-cost, interactive digital sound editing system, incorporating Wordfit's editing technology." Demonstration models should be available by the first quarter of 1986.

Other Random-Access Systems

• The forerunner of all random-access sound editing tools is the **ACCESS** (Automated Computer Controlled Editing Sound System) system, developed for Neiman-Tillar Associates (now TAV Sound), Los Angeles, in January 1977. The hardware was designed by Bill Dietrick of Mini-Micro Systems, Anaheim, CA, with software written by Jim McCann. In 1981 a second system was installed at the Sound Shop in New York, and both sytems have utilized 12-bit resolution and 50 kHz sampling frequency.

New updates to the system include 16-bit resolution and stereo capability, in addition to storage capability on optical disk. Current systems utilize disk drives with removable 200-Mbyte disk packs. Dietrick estimates that an ACCESS system sold today incorporating the above updates would cost aspproximately $350,000. [See the October 1982 issue of *R-e/p* for more details on the ACCESS system — *Editor.*]

• Although **Soundstream, Inc.** is no longer active (in early 1985 it was purchased, along with its parent company, Digital Recording Corp., by a Canadian corporation), the firm's digital recorders and random-access Digital Editing System are currently

RCA engineer Tom MacCluskey at the Soundstream Instant Access digital editing system, with classical producer Thomas Z. Shepard.

alive and well at RCA Studios in New York.

The Soundstream Instant Access system can edit up to eight audio tracks stored on a 300-Mbyte disk pack. Direct digital transfer is available for Sony PCM-1610, JVC VP-900, Mitsubishi X-80 two-track, 3M M81 DMS four-track and, of course, the Soundstream two-, four-, and eight-track recorder. Only three of the Soundstream Instant Access systems were built; aside from the RCA system, one currently resides with the Canadian company that bought Soundstream, and the third at Sonapress in Germany.

During the company's final days in Los Angeles at Paramount Studios, Soundstream completed development of software to provide SMPTE timecode lockup. Although the software has never been used in production, there is the possibility that that capability may be revived at RCA Studios. "The editing system, because it is limited to eight tracks, is most often used for classical music," says Tom MacCluskey, RCA staff engineer who was previously Soundstream's general manager; he has worked with the editor since its inception. "A number of classical producers have thought about recording eight-track on a digital multitrack [Sony PCM-3324 or Mitsubishi X-800], making three or four passes on the tape. We would transfer directly off the multitrack digital tape, eight tracks at a time, into the computer and then edit it."

The edited tracks can either be transferred back to the digital multitrack, or mixed directly to two-track from the Soundstream editor at RCA.

• **The SYSTEX System** currently being marketed by Gotham Audio Corporation, New York, utilizes 330-Mbyte disk drives and a Motorola 68000 microprocessor. The system, which is based on the Digiphon 450 multitrack, random-access recorder developed by EMT-Franz of West Germany, utilizes 16-bit linear resolution and a 48-kHz sampling frequency, and can store up to 60 minutes of mono or 30 minutes of stereo per disk pack.

In its current configuration, Gotham is aiming SYSTEX at the broadcast market, as it is with the EMT-Franz Model 448 Digital Storage System, which stores effects on a 5¼-inch floppy disk cartridge holding up to 50 seconds of mono audio.

• The **Advanced Music Systems AudioFile** digital storage system (not to be confused with the proposed Soundstream AudioFile playback card) utilizes a Winchester disk drive that holds up to an hour of 16-bit/48 kHz audio. The system is currently configured for eight outputs, and, with a built-in SMPTE timecode interlock capability, can be used for such film sound tasks as ADR/Foley recording and sound-effects creation. The unit is expected to go on sale this Fall. ■■■

TIMECODE APPLICATIONS

Larry Blake is currently preparing, for possible inclusion in the February 1986 issue of *R-e/p*, a comprehensive overview of the use of SMPTE timecode in audio/video/film production. He would like to hear from anyone that could help him shed light on this important and often misunderstood topic. Contact him c/o the *R-e/p* office, whose address is included on the Contents page —*Editor*.

All photography by Christi Carter © 1985 One Pass, Inc.

AUDIO/VIDEO RECORDING OF BUDDY RICH AND HIS BAND "LIVE ON KING STREET"

Utilizing the SQ/Tate Two-Channel Surround-Sound System for Album, Compact Disc, and Videodisc Release

by Paul Broucek

April 3rd, 1985, marked something of an engineering "Superevent" at the King Street Studios, San Francisco, as Bogue-Reber Productions, in association with One Pass Productions, taped an audio/video concert special, *Mister Drums: Buddy Rich and His Band Live on King Street*. The soundtrack for the special, featuring the SQ/Tate surround system, was recorded and mixed live to six different tape transports, including two digital formats and four of the best recorders that analog technology currently offers. More than just an impressive array of technical hardware, the *Mister Drums* special was an audio event with a level of direct manufacturers' support and participation rarely seen outside of an AES Seminar; "state-of-the-art" high-tech was definitely the standard for this audio/video production.

Few productions are planned and presold as completely and diversely as the *Mister Drums* special: the concert video is licensed to Pioneer Artists for LaserDisc release (with digital audio via the new Pioneer 900 series players, and other units to come); Sony Corporation for world-wide Super-Beta Hi-Fi, VHS Hi-Fi videocassettes, and Video-8 with digital sound; and to both the Bravo Entertainment and Discovery cable networks as a pay-TV special. An agreement also has been reached with the People's Republic of China National TV network for airing the special later this year — a first for an American Jazz concert video. In addition, Mobile Fidelity's new label, Cafe Records, will be issuing two separate releases from the concert audio in both digital Compact Disc and analog vinyl versions.

— the Author —

A producer, film composer, musician and audio engineer, **Paul Broucek**'s credits include sessions with Aretha Franklin, Jimmy Buffett, Rod Stewart, The Go-Go's, Ultravox, Psychedelic Furs, Sparks, Tanya Tucker and Tom Scott, as well as soundtrack work for *Apocalypse Now* and *Flashdance*. He has served on staff at both the L.A. Record Plant and at Francis Coppola's American Zoetrope Studios. In 1984, after a stint as general manager for the Plant Studios/Sausalito, Broucek set out on his own to start-up a new company, M2 Productions. In the past year he has worked on *Hollywood Halloween*, a rock-video collaboration between Broucek and filmmaker Jon Poll, and the score for an AFI production *Voice in Exile*.

The special began life as the brainchild of producer Gary Reber, whose credits include SQ/Tate soundtrack production of David Bowie's *Serious Moonlight* concert video, and the *Dolly Parton in London* HBO special. Reber, a jazz enthusiast of the highest order, approached Steve Michelson and Scott Ross at One Pass Productions, one of San Francisco's leading

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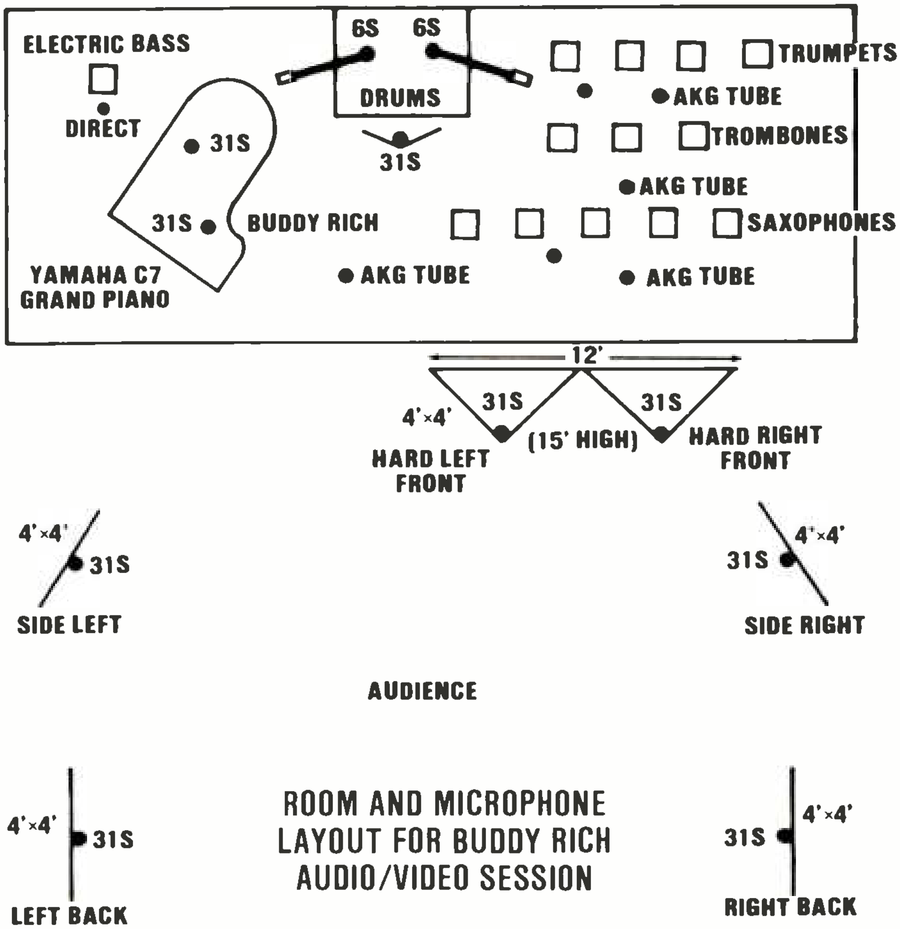
video operations, to produce the Buddy Rich special on the facility's newly completed 30,000-square-foot King Street Soundstage Complex. The plan was to make *Mister Drums* the first special in what will hopefully be a long series of state-of-the-art jazz and classic pop concert videos.

It's hard to imagine a better or more prestigious choice for the first *Live on King Street* production than a jazz artist of Buddy Rich's caliber. Even with 40 years experience as a big band leader, Rich's current crew is as fresh and exciting as any going in the jazz world today. A 15-piece outfit — four trumpets, five saxes, three trombones, electric bass, acoustic piano and drums — the band features Steve Marcus on tenor and soprano sax. For the *King Street* special, the band played through two, 50-minute sets — the "Channel One" and "West Side Story" sets — before a live studio audience. Each of the sets was centered around its respective title tune, with some great solos by Steve Marcus and, of course, Buddy Rich himself. The sheer energy and excitement generated by the horn section alone seemed to project an upbeat feel to the entire production crew.



ROOM AND MICROPHONE LAYOUT FOR BUDDY RICH AUDIO/VIDEO SESSION

Production Philosophy

Planning for the special began nearly a year ago and, from the start, Ken Rasek had been selected as the project's mixing engineer. Based out of Chicago, Rasek has an extensive background in mixing both electric and acoustic music live to stereo. He also understands the nuances, as well as the potential pitfalls, of mixing for a stereo surround-sound format such as SQ/Tate. Rasek had worked with Gary Reber in 1982 on the first live SQ/Tate broadcast featuring Devo at the Beverly Theatre, Los Angeles. The *Mister Drums* project proved to be the perfect opportunity for the pair of them to continue their work together.

Instead of bringing in an existing mobile audio truck, Reber and Rasek decided to go with their idea of creating a "living-room-type" control room. When putting together a mixing studio from the ground up, there is a greater freedom to carefully pick and choose each piece of equipment in the recording chain. Reber began approaching the manufacturers he considered to be the best for his needs, with the idea of being involved in a production that is dedicated to showcasing the finest hardware that modern audio has to offer.

Judging from the turnout at the shoot itself, the equipment manufacturers liked this idea. In the tape-machine field alone, Sony, JVC, Studer, Nagra and Ultramaster were all represented; Barcus-Berry, Monster Cable, Lexicon, Yamaha, JBL, Electron Kinetics, Lenco and Stax Professional also participated. Sound Genesis of San Francisco, and Leo's Professional Audio of Oakland, California, helped Bogue-Reber Productions with the sheer logistics of coordinating the various "pieces" of the control-room puzzle. A real sense of camraderie existed among all of the participants — it was great to see the manufacturers themselves supporting the project, and in the recording of this state-of-the-art soundtrack.

Gary Reber is anxious to discuss the "philosophy" behind the choice of audio equipment. His philosophy

A sound control room was custom built at the King Street studios for the Buddy Rich shoot. From left-to-right: Nick Latimer and Joe Van Whitsen of Discovery Network, and John Caden, head of CMS Digital.

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SQ/TATE MATRIX ENCODED SURROUND-SOUND SYSTEM

We asked producer Gary Reber the obvious question: Why such a complete commitment to working in the SQ/Tate format?

"Because the technology works," he replied. "The SQ/Tate technology is the one system that truly delivers the promise of a live playback experience. It has the natural depth and dimension that our ears hear. Conventional stereo just can't recreate the same sense of 3-D."

What makes the SQ/Tate system different from the various "quad" formats of the mid-Seventies?

"Several key factors. The various competing 'quad' formats were rushed to the market before the encode/decode process was completely worked out. Front-to-back separation was as low as 3 dB in some cases, while the Tate system decoding of SQ provides better than 35 dB of interchannel separation.

"Also, stereo/mono compatibility is not a problem with the Tate system: the encoded signal behaves exactly like mono or stereo when heard on home systems, yet the same signal becomes surround-enhanced when a Tate system decoder is added in the playback chain of a four-channel audio system. There is no 'planned obsolesence' inherent in the system's consumer applications.

"I've been producing exclusively in the SQ/Tate format since 1980, and co-produced the very first live SQ/Tate radio broadcast in 1982. I had been eagerly following the 'quad' developments in the early Seventies, while still working as an Economic Development Planning Consultant and teaching in the graduate program at UC Berkeley. I was glad to see 'quad' sound expand and flourish with the success of surround-sound concepts in the film industry during the late Seventies. By the time I began producing I knew I wanted to record in surround stereo. I also knew that SQ/Tate was *the* surround format."

What was your SQ-Concept for the Buddy Rich Project?

"We went for a natural recreation of the live experience. We kept the normal spatial relationships of the sound on the King Street stage: the band up-front left to right, wrap the environment of the 'club' in the rear channels, and mix the audience in perspective.

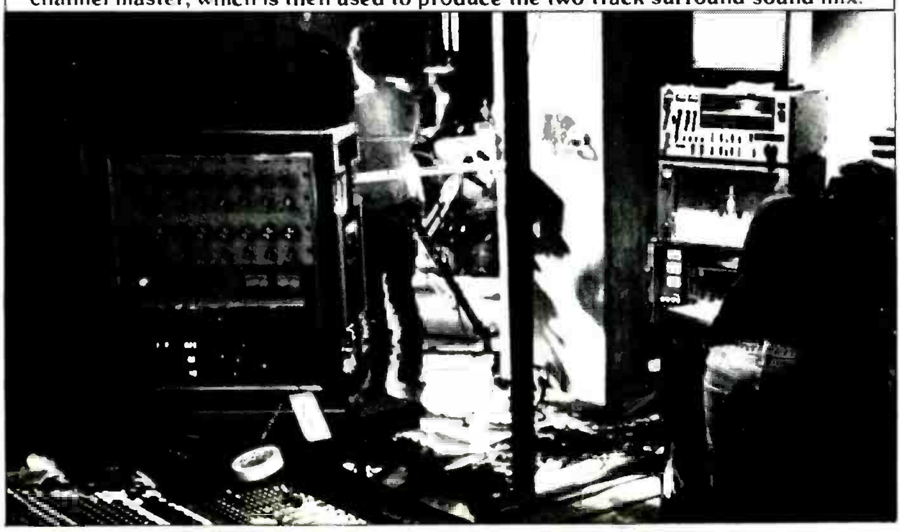
"Like all production decisions, use of the system has to be appropriate to the spirit of the music being produced. SQ/Tate allows you to position any given signal in a 360-degree spherical soundfield. The Buddy Rich soundtrack is all depth, dimension and separation, but there are no solos being panned around the room. Now, if I was doing a Pink Floyd recording, a much more aggressive use of the system would be appropriate!

"One of the potential Pitfalls in mixing to the SQ/Tate format is that you can't pan anything in the exact center-back position, or you will lose it completely due to phase cancellation when you go to mono.

"Regarding additional equipment costs for an SQ/Tate production; the only special equipment you need is the actual position encoder console, which is currently a rental item. The professional model will be available for sale to the industry in 1986. Additionally, you would need speakers and amplification for rear-channel monitoring. Storage of the two-channel encoded program can be either digital or analog.

"If you have the kind of project which involves a multitrack mixdown, of course it would take the additional studio time to make placement choices, particularly with the rear channels. Obviously you now have a whole new world of dimensional choices to make that just don't exist with conventional stereo or mono. □□□

Seen here to the right of the main Yamaha mixing console is the rack-mounted, 16-channel SQ/Tate Position Encoder Console, used to pan sounds into the four-channel master, which is then used to produce the two-track surround-sound mix.



stems from a *total* commitment to digital audio and, even more importantly, to Reber's long-standing involvement with the SQ/Tate stereo surround-sound system. The Tate system is a 4-2-4 matrix licensed through CBS, Inc., as the companion to the company's original SQ encoding design. (As discussed in an accompanying sidebar, the SQ matrix is an encoding/decoding process that allows pan positioning within a spherical and 360-degree symmetrical soundfield. The four-channel mix of left-front, right-front, left-rear and right-rear are matrix-encoded to produce a pair of "transmission channels" for album or videocassette release, and then decoded in the home to provide four surround-sound replay signals.)

The objective of Reber's approach, as he states it, "is to put the listener *there*. I want to give the consumer a true live concert with all of the depth and dimension that goes with the live experience. Our equipment choices — particularly with microphones — were all made while keeping in mind the need for extreme accuracy and transparency in order to maximize the benefits of recording digitally and in the Tate surround-sound format."

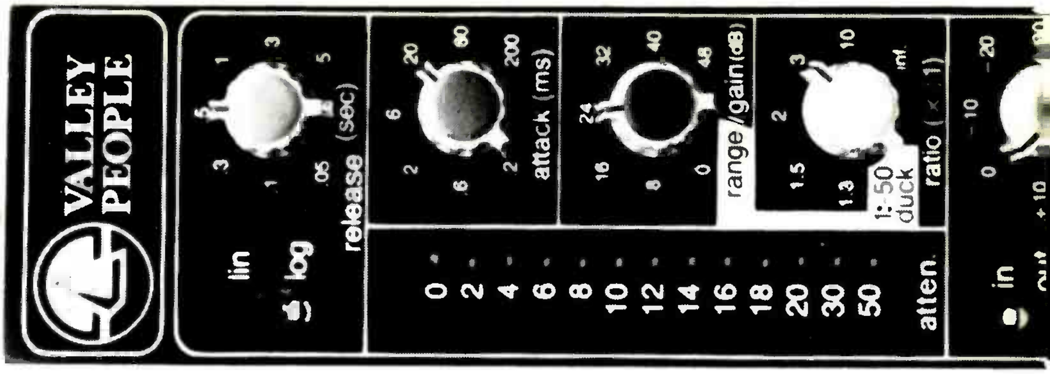
Microphone Selection

From the very start, it became readily apparent that Reber's approach to

AUDIO PRODUCTION EQUIPMENT LIST

- Yamaha 2000 Console
- Electron Kinetics Eagle 7A power amps
- JBL 4435 studio monitors
- STAX SR Lambda Pro Electrostatic Earspeakers
- Lexicon Model 200 digital reverb
- Barcus-Berry BBE/202 Signal Processors
- TATE/SQ Position Encoder Console
- Fosgate Research SQ/Tate Model 101A decoder
- Sony PCM-1610 digital processor
- JVC DAS-900 digital processor
- Sony BVU-800 U-Matic video recorders
- Studer A810 two-track recorder
- Nagra IV-S TC two-track recorder
- Nagra T-AUDIO TC two-track recorder
- Ultramaster half-inch two-track recorder
- Sony 701-ES digital processor
- Nakamichi DMP-100 digital processors
- Sony BETA HI-FI VCR
- JVC VHS HI-FI
- Crown PZM-31S microphone elements
- Crown PZM-6S microphone elements
- Countryman Isomax II condenser mikes
- AKG The Tube microphones
- Monster Cable Prolink audio cables
- Monster Cable Hi Performance microphone cables
- Nakamichi The Dragon cassette recorder
- Monster Cable Soundex acoustical panels

□□□



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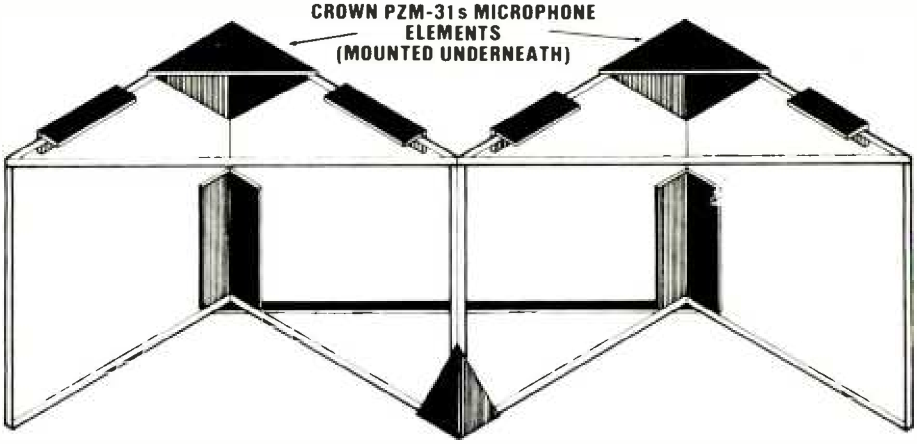
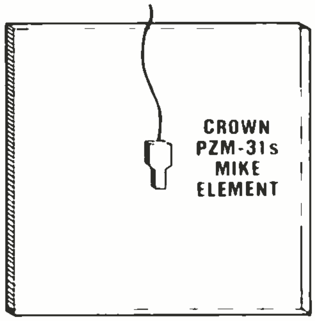




ABOVE: CLOSE-UP DETAIL OF STEREO ARRAY USED DURING BUDDY RICH AUDIO/VIDEO SHOOT. THE ARRAY MEASURES 12 BY FOUR FEET, AND IS MADE UP FROM 4- BY 4-FOOT PLEXIGLASS SHEETS.

LEFT: SINGLE-SHEET ARRAYS MAKE UP FROM THE SAME SIZE PLEXIGLASS.

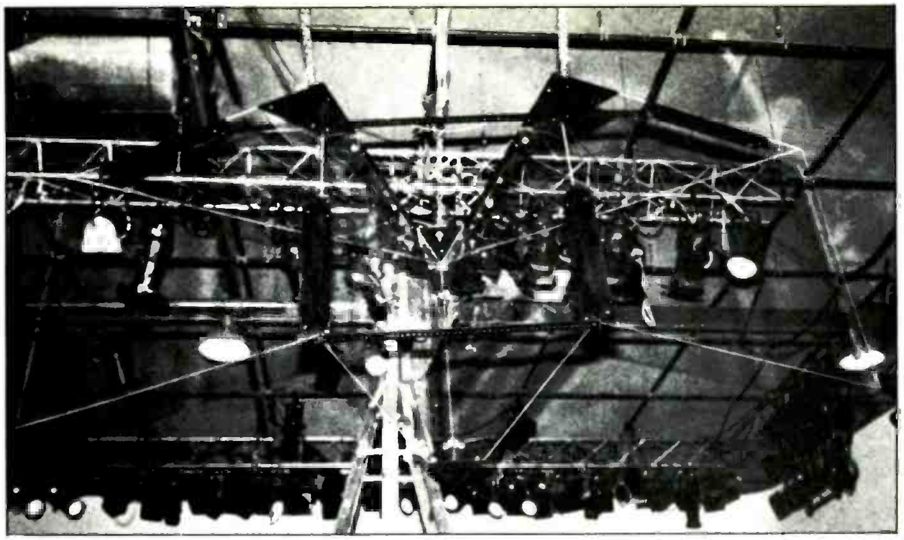
the project, particularly regarding the choice and application of microphones, was far from conventional. *R-e/p* was invited to the King Street soundstage the day before the shoot, while the crew readied the various technical elements. In the center of the floor was a large, geometrically-shaped plexiglass construction. A closer look revealed a specially designed housing for a pair of Crown PZM-31S microphone elements — definitely a contender for the *Guinness Book* as the world's largest stereo microphone! This plexiglass array, and several others, was designed by Reber and Vince Motel, the project's PZM applications engineer. Ken Wahrenbrock, the acknowledged "father of the Pressure Zone Microphone™" concept, also lent his support and advice to the array designs.

The main stereo PZM array consists of two, three-sided "V-shaped" units braced together. All of the plexiglass was ¼-inch thick, and treated with a silicon compound to make the surface as hard and as reflective as possible. The four largest sheets were four-foot-square sections mounted at right angles, and had triangular pieces of plexiglass mounted on the top only. The PZM elements were mounted in the upper corners, where the bottom of the top section meets the right angle. The completed array measured 12 feet across when fully assembled. Once positioned, the entire unit was flown approximately 10 feet above the floor and centered in front of the band's horn section.

In addition to the stereo array, four single-sheet PZM arrays were positioned above the floor, and at the four outside corners of the audience area. These single plexiglass sheets are also four-foot square with PZM-31S elements mounted in the center of each sheet. The outputs from these additional microphones were assigned via the SQ/Tate matrix to left-side, right-side, left-back and right-back, respectively.

Miking for the drum kit was also handled by a combination of plexiglass and PZM elements. Opting for a three-mike setup, Reber and Motel this time used a smaller V-shaped array placed directly in front of the kick drum. Mounted within the V-angle was another Crown PZM-31S element. The overall drum sound was handled by a pair of parabolic dishes each holding a PZM-6S element, and positioned in a standard drum-overhead configuration using two AKG baby-boom stands.

The Yamaha C7 acoustic grand piano was miked using a pair of PZM-31S units mounted at the high and low position of the piano's harp, while the electric bass was taken direct. Four AKG The Tube microphones were placed within the band itself, in order to capture an intimate feel for



The main stereo array suspended in front of the Buddy Rich Band was fabricated from four, four-foot by four-foot sheets of plexiglass, with two Crown PZM-31 elements mounted in the top apex corners.

the various trumpet, trombone and sax soloists. Countryman Isomax II condenser microphones were placed on the flutes used by three of the sax players for occasional parts.

Mixing Environment

One Pass Video's 45-foot Mobile One served as command center for the seven-camera Ikegami shoot, while the audio crew set themselves up in a control room adjacent to the sound-stage itself. The control room was acoustically transformed with Monster Cable Soundex Acoustical Panels to approximate the size and sound quality of an average consumer's living-room. Next to the control room was a production "theater" setup that housed the director's on-line video and a full, decoded SQ/Tate surround playback system. (The "theater" presentation was arranged in an attempt to help keep the control room proper free of extra bodies, as anything else.)

The main mixing console was a Yamaha Model PM-2000 with 24 inputs, eight subgroups, and four mains. Linked to the Yamaha was a special 16-channel SQ/Tate system "position encoder" console, which is necessary to create the final "live-to-two-channel" encoded surround-sound mix. The encoder console is equipped with 360-degree, fixed-position panpots that allow the engineer to assign individual input channels within the surround-sound environment. (The SQ/Tate two-channel encoded mix is completely compatible with conventional stereo and mono playback systems.) For the *Mister Drums* mix, only the outputs from the two single-sheet PZM arrays located at the back of the audience area, plus a slight amount of digital reverb from a Lexicon Model 200, was encoded to the rear channels of the surround matrix.

Due to the nature of mixing live-to-tape, Ken Rasek was forced to work quickly and efficiently. One of the most interesting features of Rasek's mixing approach was that he basically used little or no equalization or signal processing. In addition to the Lexicon Model 200 digital reverb, the only other piece of processing gear used on the session was a Barcus-Berry Model 202 connected to those input channels covering the Buddy Rich rhythm section. Applied just ahead of the SQ/Tate encoder, the Model 202 was used to enhance the signal transparency and the transient definition via a patented circuitry which, essentially, deals with automatic EQ and phase correction.

The SQ/Tate mix was monitored through a Tate Professional Decoder that then fed the left-front, right-front, left-back and right-back signals through Electron Kinetics Eagle 7A stereo amplifiers powering four JBL 4435 monitor loudspeakers. Monster Cable covered the entire audio wiring requirements for the sound-track production. It's interesting to note that Ken Rasek himself used Stax Ear-speakers exclusively to monitor the mix in progress.

Reber selected a Lenco Model 600 distribution amplifier system to handle feeding of the final mix to the six main tape machines, plus the myriad of Nakamichi digital processors and analog cassette recorders used for the producer's reference copies. CMS Digital, of Altadena, Calif., provided Sony PCM-1610 and JVC DAS-900

Production staff involved in the "Mister Drums" audio/video shoot included (*left* photograph, from *L to R*): Gene Shiveley of CMS Digital, Fred Layn, northwestern regional manager of Studer Revox America, Adam Reed of Barcus Berry Electronics, Gary Reber, and Paul Stubblebin of Monster Cable. Pictured *right* is Ken Rasek, soundman for the session.

digital audio processors coupled to a pair of Sony BVU-800 U-Matic video recorders. On the analog side, Studer provided an A810, while Nagra supplied both a IV-STC and T-Audio TC model two-tracks. Of particular interest was the Ultramaster two-track analog machine, a unique hybrid that features half-inch, 30 ips recording only. John Curl and Dave Wilson designed the machine's custom electronics, which they have mounted on a Studer A80 chassis. 3M Scotch tape was used for all systems: 250 for analog, and Color Plus Super High Grade cassettes for the various video-based digital formats.

The Show

It might be easy to get the wrong impression about production for the *Mister Drums* special, which was *much more* than a technical exercise. When it came time for taping, it was all Buddy Rich's show. He is big band, swing and be-bop all at once, and that's the impression one gets before he even plays a note. The man Sinatra calls "the world's greatest drummer" has often been accused of being as conceited as he is talented. If so, that conceit was only represented by Buddy's complete commitment to the quality of the *Mister Drums* production. He was patient and relaxed throughout any technical detours or delays, seemingly as much a fan of the crew as they were of his music. More than anything, there was an underlying feeling of fun when the band set itself up on stage. Buddy displayed a keen sense of humor, directing a steady stream of one-liners around the bandstand and throughout the stage.

The sound stage had been set up in an intimate Jazz-club fashion, with 20 or more round tables seating well-dressed studio audience. Bright neon outlined the roomy bandstand with a "BR" neon logo positioned squarely above the drum riser, which consisted of transluscent glass bricks, a clear plexiglass top and white lights underneath — a design that, at times, gave the illusion of Buddy almost floating on a bed of light.

Throughout the night's taping Buddy Rich propelled his band through an exciting ensemble sound, and punctuated his way around some very hot horn soloing. But, of course, the real treat was when Buddy himself would take a solo. By the end of the first set Buddy sweated his way into a sustained snare roll that should definitely keep his legend alive for another 40 years.

During a subsequent visit to One Pass Video's post-production facilities during the editing process, audio quality of the session sounded superb. Even in the standard stereo mode, the sense of dimension provided by the SQ/Tate System, to this writer at least, was impressive. Channel separation is extremely wide, which provides the listener with the clarity to enjoy the production crew's excellent use of ambient miking. You cannot help but admire a producer like Gary Reber, who has a complete vision of what he wants, and is willing to go through the sometimes difficult process of trial and error. He fully embraces the technology available today, and anxiously awaits the best new technical developments of the future. Reber is not interested in breaking any rules; he just wants to set new standards. ■■■

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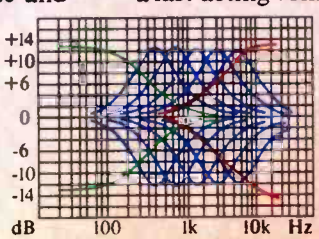

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MIDI RECORDERS: MYTH OR REALITY?

An R-e/p Guide to MIDI Data Recorders and Sequencer Software Designed to Run on Personal Computers

by Stephen St. Croix

Photography by Nick Griffin/Griffin, Hamilton & Thompson

Many of us are living today, sort of artificially, thanks to the present state of technological advancement of medicine; these thousands of people would not be alive if it were not for artificial means of some sort. Many of us are also playing music today, sort of artificially, thanks to the present state of technological advancement in computer hardware and software; these people could not be creating as high a quality commercial product if it were not for artificial means of some sort.

I am one of those persons who composes and plays great music, just as long as massive amounts of computer and multitrack technology are there to help. I have had simple data transfer and sync programs operating with my equipment for years, as have several other people, but it has never been enough. Then came MIDI.

With the advent of a Musical Instrument Digital Interface we, as engineers and artists, were finally faced with what the musical-instrument industry has been threatening us with for years: data standardization; or for players: equipment from different manufacturers that can talk to each other. As limited as MIDI is (compared to the speed and flexibility of some of the older computer-interface standards), the fact that the interface was designed to do exactly what we want done in the musical environment makes it a powerful tool that can greatly improve productivity.

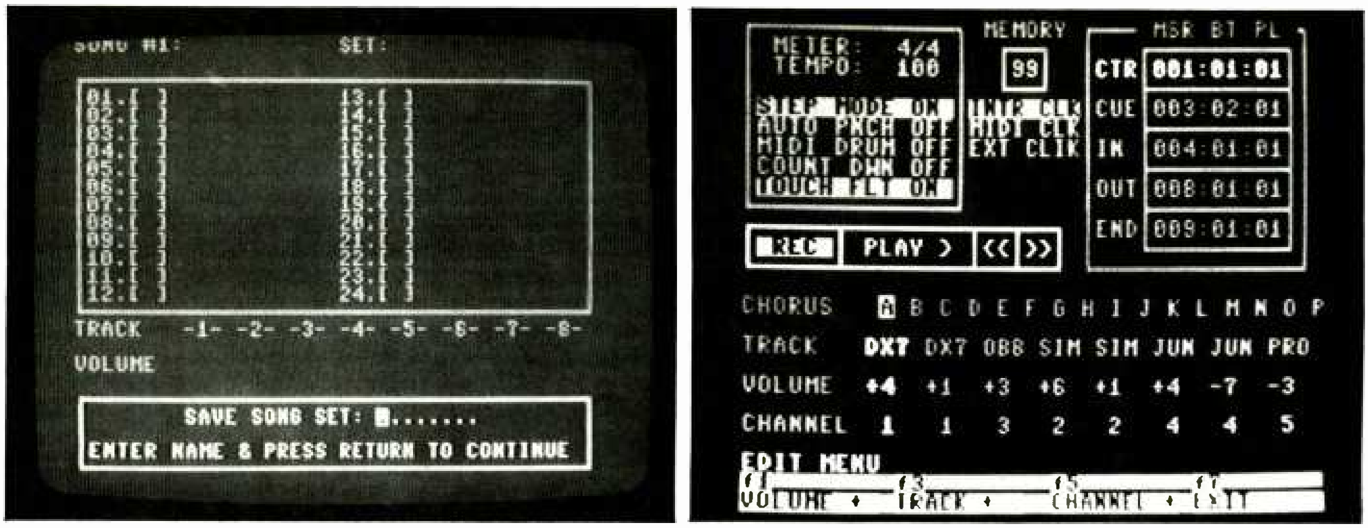
Linking together synthesizers with simple Note-On and Note-Off data came first, then the locking of drum machines and additional synthesizers. Simple synchronizing to and from tape, and the transfer of velocity, bend, pressure and other variables followed shortly afterwards. Finally, there appeared serious data transfer, such as bulk store and System Exclusive commands, which allowed great stuff like playing your drum machines from your favorite MIDI-equipped keyboard, assigning any drum sound to any key, and having real-time velocity control. Then came enhancements such as saving voices to floppy disk, rather than those mystery cartridges. The MIDI evolution became graphically obvious when voice editing appeared for the ubiquitous Yamaha DX-7, in the form of the DX-PRO package from Kevin Laubach.

But artists who felt that computers could aid them in composing were still divided into three groups: those who made use of the limited dedicated sequencers available; those who developed their own MIDI interfaces and recording/editing software; and those who gave up in frustration and sat down to wait.

Technology moves very rapidly once a market for it develops, and MIDI recorders are no exception. We now find ourselves in an interesting position. In just a few months we have moved from a position of waiting for the first MIDI data recorder to appear in non-crash dress, to suddenly having so many available that it became impossible to compare them.

I began looking into the new MIDI data recorders for the third time many months ago, and found it almost impossible to collect all of them together for comparison and evaluation. Although a very large and complete music store near Washington, DC, had several of the MIDI software packages on its shelves — plus the necessary hardware interfaces for linking the controlling personal computers to the MIDI-equipped keyboards — it found itself in a similar position: there was simply too much to wade through. Since many studio engineers and producers might also be facing the same problem, *R-e/p* has decided that it might be timely to publish this in-use field comparison and overview of some of the more innovative MIDI data recorder software.

When I began comparing the various packages available for the Apple II, Macintosh and IBM PC (plus PC-compatibles), I thought that I would simply sit down for a few days and write a tune on each one, thereby learning all that I needed to know about each system. No. Instead, I discovered that even though I had been



All displays © 1985 by Syntech Corporation

Screen displays for Syntech Corp.'s Digital Studio II software. Shown *left* is an eight-track song setup menu, while *right* is a menu for controlling the master "multitrack" controls, MIDI assignments, mode functions, and sequencer data.

MIDI RECORDERS

hearing about these packages for some time now, less than one-quarter of them actually existed in a useable form; of these, only a few actually implemented all the advertised features. The entire field seemed to be just coming into existence. Further, and sadly so, of the packages that I actually did manage to try out, most were under-developed, under-researched, or under-powered. It seemed that it was just too soon to dispose of the 24-track, much less write this article.

Despite such reservations, I went ahead with the evaluation project, and was assembling a very negative overview of the whole endeavor when I called one of the publishers of a MIDI recorder in order to ask why the system seemed so immature. The answer that I received was interesting. It seems that the program was, in fact, very immature, and that a new version was coming "real soon." I explained my situation, and asked if my "real soon" could be now. The new version arrived the next day, and the difference was so remarkable that I thought it only fair to call all of the companies involved, and ask for their newest versions. Almost all of them *did* have newer versions, either instantly available or in beta stage (under development).

As these finished and experimental versions began arriving, a totally new picture developed. The MIDI data recorder programs had matured so much over the three-month period I had been examining the older ones, that one thing became very clear: the entire comparison had to be done over. Some of these companies must really have listened to their first customers, because the new programs were great.

There is an impressively wide selection of features currently offered, and the systems represent a spread from simple multitrack sequencers to powerful multitrack MIDI data recorders with real-time and step editing (features that allow creative minds with out-of-control fingers to create controlled music). Some systems allow you to treat them almost as if they were real mechanical multitrack tape recorders with additional features, while others are "modular" or phrase oriented, building songs from short basic phrases much like drum machines. The more elaborate systems even allow you to clean up horrible timing and other errors, and then go through and "rehumanize" (reintroduce smaller errors) to each note by hand.

Each approach clearly has its own merits and power. I finally chose more than one system so that I could transfer between them, and use my favorite features of each (remember, there is no actual audio recorded, so there is no signal-quality degradation).

There seems to be five types of potential users for MIDI recorder software or dedicated hardware packages:

- The person who just wants a good sequencer.
- The player who wants to use a computer as a tape recorder; to jam into, record tunes and be able to overdub and punch.
- The person who might not be the most flawless player, but who wants a system so he can play a little loose, even miss a few notes, and then go in and time correct and fix the bad bits.
- The guy who wants to build songs from sections, like he would on a drum machine.
- The pro who needs to do ultra-precise production work within exact time constraints, where a need for repeat phrasing may exist but the ability to edit each note independently afterwards must be provided.

— the Author —
Stephen St.Croix is a real neat guy who studied to be a welder, but at the last minute decided to be a rock star. The transition is not yet complete.

A few of the currently available systems deserve highlighting here, since they were clearly exceptional. No doubt there are other very good programs out there, but I found that the four systems described in greater detail in the remainder of this article to be the most impressive of the ones that actually showed up in time to be included in the first part of this overview (several promised packages never showed). A companion sidebar contains detailed descriptions of these and other software packages designed to run on various personal computers. This writer would be interested in hearing from other companies that are developing MIDI software, for possible inclusion in part two of this article, to be published in a subsequent issue of *R-e/p* write me c/o the magazine.

Syntech Music Digital Studio II

Tape recorder emulation systems are the most natural for a musician to use, and initially the most rewarding. Such programs generally allow you to pick one of several tracks on which to record polyphonically, with little or no complicated setup. You may then overdub on individual tracks and bounce between them all you want. MDS II, however, offers such direct simplicity, and more.

So far, this program is definitely *the* one for the Apple II; it is very fast and the features work. While only eight tracks are provided — but this is the most I have seen on the Apple II — you can set up a total of 16 sequences. Each track is channel assignable. MDS offers solo and mute for each

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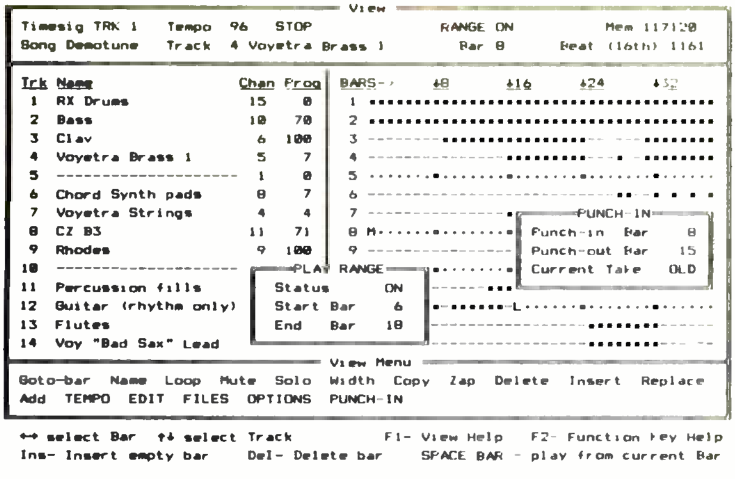
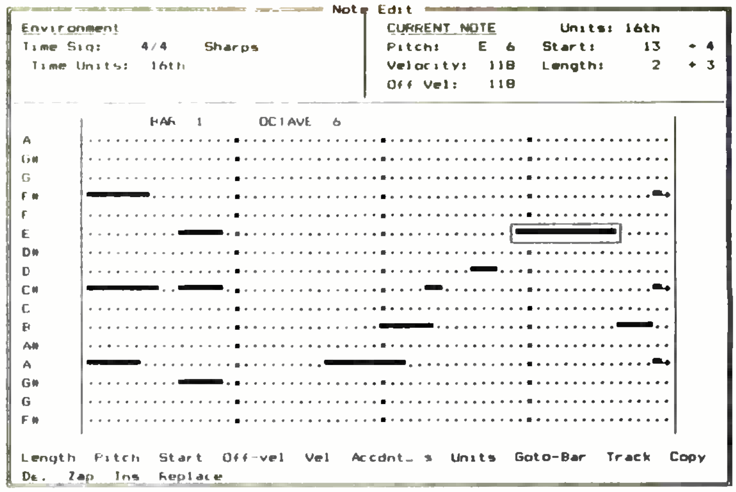


The VIEW screen shows up to 72 bars of 22 tracks (expanded display mode, not shown). '▪' and '-' indicate full & empty bars; 'L' and 'M' indicate Looped and Muted tracks. Sections of a track may be quickly "cut & pasted" by highlighting the bars you want to move, copy or delete. Also shown are "popup" windows for automatic Punch-In & Out and for playing a range of bars.



The expanded NOTE EDIT screen is showing a bar of 4/4: time is left-to-right, pitches are vertical, and '▪' indicates the major beats. The symbols show drum strokes & fader events as readily as Notes. This bar contains F# and C# minor chords, a short melody, and a chord tied to the next bar. Staccato & legato articulation is clearly shown. The upper window shows Pitch, exact Start time & Length and Velocity of the selected note for easy editing.

All displays ©1985 by Octave Plateau Electronics

Screen printouts from Octave Plateau's Sequencer Plus, REV 2.0 MIDI recorder/sequencer software, showing the View Screen for up to 72 bars of 72 tracks (*left*), and expanded Note Edit Screen for a bar of 4/4 music (*right*).

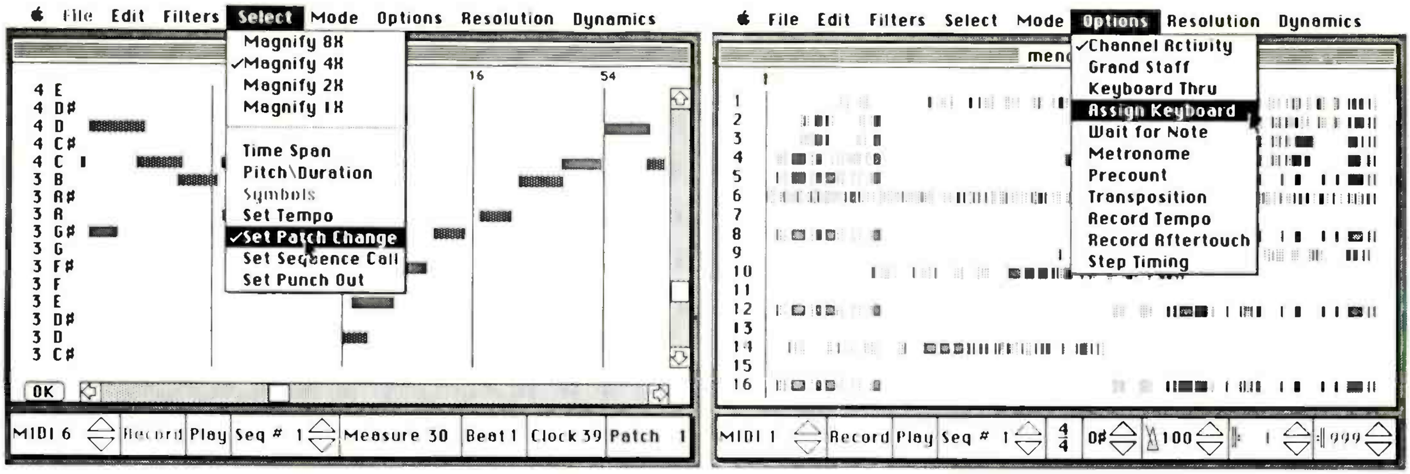
MIDI RECORDERS

track, along with track bounce, track shift, program change editing, and controller filters. (Time shift enables sliding of an entire track backwards or forwards by very small amounts, for special effects or to compensate for a synthesizer's MIDI lag.) The program allows you to store sequences and whole songs to disk; sequences can be loaded and appended to build larger, complete songs. This means that you can keep a library of standard drum parts, for example, to use in many versions of a song.

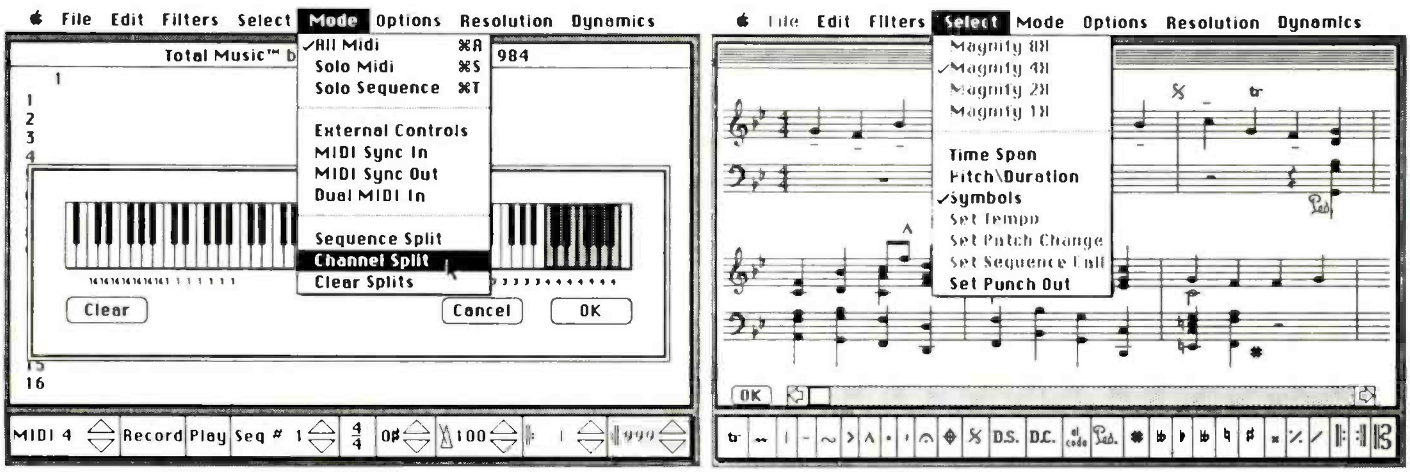
Merely listing the features of these programs would take a lot of space, and doesn't mean too much. What *really* matters is how each one does its job, and Music Digital Studio II is easy to use and to learn — it works. The auto correct works well, which is rare. You can punch on-the-fly or in step mode. You can step enter notes; you can even step by clock pulses to tear apart a chord and fix a bad note. (Remember that MIDI is a *serial* interface, so even if you manage to slam down five keys in a chord at exactly the same time, the data is sent and recorded as a series of note-on/off pulses one after the other.)

There are a few points that I didn't like about the software, but I understand that they are related to limitations in current hardware standards. This system supports Passport-type hardware and, in fact, a much nicer MIDI interface card sold by Syntech was included with the package. However, the software does not support the Roland MPU-401 interface card; as a result, even though there a visual metronome is featured, no audio metronome is available. When confronted with this omission, the company stated that its expects people to have a drum machine synchronized to the interface card, a configuration that would provide a perfect metronome. (Which seems like a reasonable answer.) Further, unless you are actually recording, there is no MIDI-Through capability. The system will not make use of additional memory cards fitted to the Apple II, so use those filters.

The owner's manual is very good, and includes a command listing with such nice instructions as "Z" for "Zero all Channels Pitch and Mod Wheels." (A few days spent with most other MIDI software programs leaves you



Southworth Music Systems' Total Music for the Apple Macintosh enables editing of note durations and velocities, patch and tempo changes, plus other MIDI data (*left*), while a main screen provides an overview of the activity on all 16 MIDI channels in a sequence (*right*).



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Other screens of Total Music let the user split the keyboard into an unlimited number of segments, and synchronize the program to an external MIDI device (*left*). In the Grand Staff mode (*left*), a user can display and print what has been played in standard music notation.

looking for just such a command.) Those of you with one disk drive will like the fact that, once you have booted the system diskette, you can remove it for the duration. I will be covering the company's other software packages in part two of this article, including what seems to be a very nice version of the above mentioned program designed to run on the the Commodore 64.

Octave Plateau Sequencer Plus; REV 2.0

Frankly, I wasn't expecting too much from a program that calls itself a "sequencer," while all the others are screaming that they are "multitrack recorders." The first version of Sequencer Plus that I received was actually kind of nice; though, it was better than I had expected, but nothing amazing. This program runs on an IBM PC or compatible, with control of screen attributes for each particular computer. While the program worked, it had some limitations that were a bit frustrating, and some quirks that were a little annoying. The second version, however, is *incredible* — the limitations and quirks are gone. You get no less than 64 tracks to play with; and editing that is well thought out and powerful . . . very powerful.

The first revision also had 64 tracks, which brings up a point that I feel strongly about. Most of the other companies that publish MIDI programs with only a few tracks talk about that fact being unimportant, because you can bounce down. But, since you can't take the tracks apart again after you bounce, global editing on one of the bounced overdubs becomes impossible. If, for example, you decide at a later time that you want the arpeggios before the chords to be on a different MIDI channel, you can only make the change if they were residing on their own track(s). Because of this restriction, I prefer to have lots of available tracks (64 is lots) with solo and mute.

You can see as much as 72 bars of 22 tracks at one time; full cut-and-paste editing is now supported; pop-up windows get you directly to functions from anywhere in the program; individual tracks or whole songs can be saved; external sync and "locating" MIDI song position are also supported. Lots of nice punch commands exist, along with shuttle ("play range").

When the program is playing back, the screen moves across the music so that you can always see what is going on, and where you are. This feature is not a gimmick: the ability to see where that klunker is when you hear it is valuable!

The owner's manual is in a nicely executed three-ring binder, and is well written, but doesn't really say much when you consider the power of this program. And it doesn't have to, a fact you learn surprisingly fast. The IBM PC function keys are very well used, and all the help that you could want, both general and specific to your current command, pops up on screen with one keystroke. The program will install on a hard disk attached to an IBM PC-Series, which makes life a lot easier.

Sequencer Plus does not offer music notation, but the people at Octave Plateau are working with other companies to establish a standardization

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of MIDI song-file formats. If this happens, you will be able to instantly transfer songs from one company's package to another. If this does not happen, watch for conversion programs to appear.

Southworth Music Systems Total Music

This program, designed to run on a 512-Kbyte Apple Macintosh, should be called Total Shock. To make one little computer work this hard, and do this much, is almost cruel. Total Music is the most complete MIDI package that I have ever seen. Those of you who believe that programs for the Macintosh should be as "Mac-like" as possible will fall for this one in seconds. Notes, bars, whole sections of music can be cut, pasted, transposed and more by defining them with mouse-placed brackets. In fact, I used this program for two days without the Mac keyboard even being plugged in.

You have to envision this program as being multidimensional: it provides a matrix of 99 sequences by 16 MIDI channels, and allows total movement and editing power within this matrix by means of graphic screen manipulation of each note's timing, duration, velocity value, and more. Total Music supports two types of real-time punch, three types of timing correction and, of course, full cursor editing.

At the time of writing this article the program was still under development, but it should be on the street by the time the issue appears on your desk. I found the people at Southworth Music Systems to be as helpful and responsive as could be. After playing with my first disk, I had some questions and suggestions about greatly expanding the data editing capabilities of the MIDI controller. Less than one day later they called back and informed me that these items will be included! One will now be able to tag any note with the mouse, and ask for a display of *all* MIDI information that is related to it, in addition to the normal editing screen display. Such upgrading has happened a few more times since then, including multiple ways of viewing the immense data matrix with similar responses. Total Music is now a *true monster*. [According to Paul Lehrman of Southworth Music Systems, every attempt will be made to include these and other features suggested by the user — *Editor*.]

The mouse-driven, on-screen buttons, classic Macintosh pull-down menus, and drag bars make it possible to do most things intuitively, but not all. The owner's "novel" runs just short of 200 full-size pages. I read it cover to cover, and the text is actually interesting! In addition to teaching you how to use the system, the manual contains notes and pointers on MIDI and several associated products. Parts of the it read like notes from a friend who has discovered a way to do something better, or maybe a strange thing that happens if you try to do certain things with certain pieces of gear. They help.

The manual contains no reference section, however. Since there are no on-screen help pages, it would be very nice to be able to immediately find the proper area in the book for support when the need arises. Maybe this will be corrected by the time Total Music is released.

The release version of the program will enable registered users to back up the master diskette — which, considering the mysteries of the Mac mind, is very important. The program is keyed to the hardware; however, if you buy two systems, *don't* mix them up.

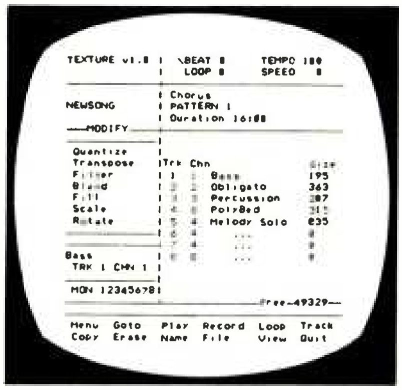
The MIDI hardware interface card, wisely, does not draw power from the Mac's PSU. The interface has four outputs and *two* inputs that are all active simultaneously. Did I mention that Total Music provides full conventional musical notation, on-stave editing and very good score-sheet printing?

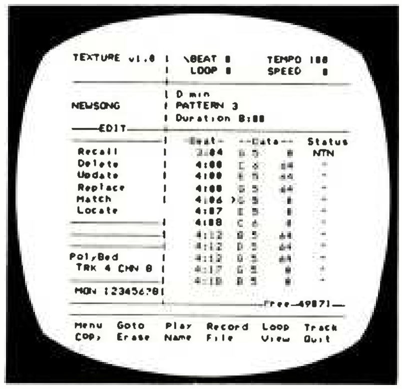
Cherry Lane Texture

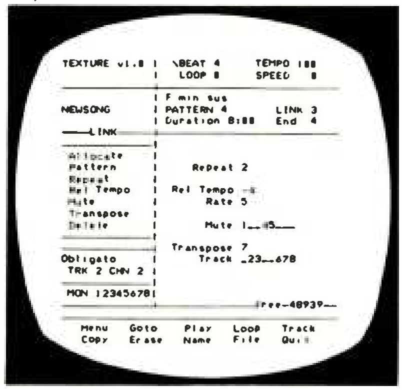
This program was one of the first to reach the market, and it is the first revision that I cover here. (The next revision of Texture is almost ready, however, and should be covered in the next part of this article.)

Texture is also quite different from the majority of other MIDI software. Written by Roger Powell, a member of an exclusive club (being both a musician *and* programmer) the program is designed around what he calls a "Modular" approach to song building. Systems like this are structured to work in a similar way to drum machines — you build up little sections and assemble them, but with modifiers like key changes and differing combinations of track timing or "rotation." Such a programming approach allows extensive manipulation of small phrases — such as repeated short four-bar loops — then calling a bridge loop, then another verse loop, and so on.

With Texture you can mask overdubs so that different ones are brought up with different loop sets. The advantages offered by such a compositional technique include the ability to construct songs very quickly; to store song segments for use else-







All displays • 1985 by Cherry Lane Technologies

Examples of screen displays from Cherry Lane Technologies' Texture software (*left-to-right*): Pattern Overview of Modify Menu; MIDI Overview of Edit Menu; and Link Menu.

where; and minimal use of memory (since these systems don't actually copy the looped song segments, but merely call the same one up over and over again.) The bad part is that songs built in this way may tend to be "cold" or over structured, since each repeated section is identical, with no unique errors or "slop" (expression).

The owner's manual teaches you about the rules of modular recording; you learn about tracks, patterns and links. There is even a short tutorial that walks you through a tune as an example. A quick reference card included with the software is excellent, and is all you need once you've learned the system.

The program is structured so that a song may have up to 64 links taken from a library of up to 64 patterns. There are eight tracks, each of which is MIDI-channel assignable.

Step editing is good, and all of the standard features are there. Although Texture is not a graphics-oriented program in its present revision, having learned how to read the screen you discover that a lot of data is there at one time. Editing is by direct manipulation of MIDI data values. Much to my surprise, I found that I became very fast at editing within a very short period of time. A lot of different types of note and controller data could be massaged very quickly. The new revision is promised to be "days away," and I look forward to updating you on its features in part two.

Concluding Comments

There are several factors that I became aware of while evaluating some MIDI recorder systems, of which I have chosen to spotlight these four packages; some may seem obvious, and others may just save you a day's work. The MIDI data streams produced by Bend and Mod Wheel controllers, plus Aftertouch (pressure), can be very dense, use up massive amounts of memory, and make a song file quite large in not very much time. Furthermore, the production of high-density information can actually "clog" the MIDI data stream, and cause audible slowing or lagging of parts of the music, or even strange errors. Some synthesizers, including the Yamaha DX-7, buffer the data in such a way as to grossly aggravate the clogging problem. It is for this reason that most of the systems offer "filters" that allow stripping of such data. (Sequencer Plus actually allows you to "thin out" the data without losing it.) Use them; it makes *no* sense to load things down with 20 Kbytes of pressure information, if the synthesizer assigned to play back that track can't use it!

MIDI programs that emulate tape recorders generally use more memory than block editors, because the latter play back segments many times, and transpose them, etc. Some systems will allow you to "rush" or lead tracks by one MIDI clock pulse at a time (called track shift), in order to compensate for older synthesizers with MIDI retrofits, and slower ones like the DX-7 that can lag behind as much as 150 milliseconds.

Your disk drive is always sitting there waiting — save your work *often*. All of these systems are new, and you never know when you might discover the magic sequence of events to send the program and your music to Mars forever. (I managed to find a way of doing just that on almost every software package.) Even if this doesn't happen, a power glitch might. In the words of one of the owner's manuals: Use it or Lose it.

If you use MIDI Thru as you record, *be sure* that it is not returning to the keyboard that you are playing. Otherwise, if it does you are in for a strange suprise with every one of these programs, and even without them.

My thanks to Washington Music Center in Wheaton, Maryland, and Robert Levin and John Chase there, without whose help this article would have been very, very short. ■■■

In Part Two: Hardware MIDI recorder systems such as the Yamaha QX-1 and the Roland MSQ-700 and -100; plus a look at other software packages (and the new Harley-Davidson FX-ST Soft tail). A list of the companies offering MIDI recording software and hardware is included in an accompanying sidebar. Given the speed at which things change in this industry, there is a good chance that the listing is incomplete. I invite any companies that we have omitted to contact the *R-e/p* for inclusion in the next part of **this article.**

ANNOTATED LISTING OF
MIDI DATA RECORDERS AND SEQUENCER SOFTWARE
FOR VARIOUS LOW-COST PERSONAL COMPUTERS

HYBRID ARTS, INC.
11920 West Olympic Boulevard
Los Angeles, CA 90064
(213) 826-3777

MIDITRACK II

Runs on: Atari 800XL.
Interface: MidiMate interface cards available for different synthesizers, plus Cherry Lane interfaces.
Tracks: 16.
Key Features: Sequences can be up 6,500 MIDI events in duration; 16-track overdubbing, punch-in/out, autolocate, full MIDI Channel assign, velocity encoding, pitch and mod-wheel recording; program-change recording; transpose and quantizing; step editing; a variety of sync in/out interfacing; three entire recording saved per floppy disk; visual and audible metronomes. (MidiTrack III, with a 12,000-note capacity is scheduled for release in the near future; price has yet to be announced.)
SRP: $349.00, including MidiMate interface, cables and user's guide.

MUSIC DATA
844 Wilshire Boulevard
Beverly Hills, CA 90211
(213) 655-3580

MIDI SEQUENCER

Runs on: Commodore 64, Apple II+ and IIe.
Interface: Passport Design-Compatibles, or company's own interface.
Tracks: 16 tracks, with 16 unique sequences per track; polyphonic.
Key Features: 8,500 MIDI events; "ease of operation — everything you need is on the screen at all times"; song selection capability, each track having its own independent length; looping capability; each set of 16 tracks can be assigned to any 16 MIDI chan-

MIDI SOFTWARE ... continued —

nels; programmable tempo per section.
SRP: $150.00; interface $125.00.

MUSICWORKS
16 Haviland
Boston, MA 02115
(617) 266-2886

MEGATRACK

Runs one: Apple Macintosh.
Interface: Company's Midiworks MIDI interface.
Tracks: Unlimited via 32 MIDI channels.
Key Features: Capacity dependent on memory size of Macintosh (1 MByte of memory is equivalent to 72,000 notes or events); unlimited overdubs; mouse operation; "ease of use."
SRP: $150.00; Midiworks interface $100.00.

OCTAVE PLATEAU ELECTRONICS, INC.
51 Main Street
Yonkers, NY 10701
(914) 964-0225

SEQUENCER PLUS

Runs on: IBM PC, or compatibles; operates under MS-DOS.
Interface: Roland MPU-401, and company's OP-4001.
Tracks: 64.
Key Features: 60,000-note capacity; user interface; menus styled after Lotus 1-2-3™; menu-driven operation.
SRP: $495.00; OP-4001 interface $295.00.

OPCODES SYSTEMS
1040 Ramona
Palo Alto, CA 94301
(415) 321-8977

MIDIMAC SEQUENCER

Runs on: Apple Macintosh with 512 Kbytes of memory.
Interface: Any available MIDI interface, including own product.
Tracks: 10 tracks per individual sequence, but with replay of previous sequence while recording another; 32 tracks available for simultaneous replay.
Key Features: 48,000 MIDI events (24,000 notes); utilizes Apple mouse and pull-down menus; "fast and highly interactive"; configured for live-performance applications; MIDI keyboard controls pitch of replayed sequence.
SRP: $150.00

PASSPORT DESIGNS, INC.
625 Miramontes Street #103
Half Moon Bay, CA 94019
(415) 726-0280

MIDI/4 PLUS AND MIDI/8

Runs on: Apple II+, IIe, IIc and Commodore 64.
Interface: Company's own interface, available with or without tape-sync capability.
Tracks: Four and eight, respectively.
Key Features: 7,000-note capacity; auto-correct, punch-in/out, fast forward/rewind modes; sequence chaining; sync-to-tape (with suitable interface); sync to MIDI and drum machines; real-time editing; tempo control; can be linked to Polywriter music-printing software.
SRP: MIDI/4 plus $99.95, MIDI/8 $149.95 (upgrades for existing MIDI/4 software $35.00); Apple IIe/II+ interface with drum and tape sync $199.95, interface with just drum sync $149.95; Commodore 64 drum/tape interface $169.95, drum-only $129.95. (Also available is a MIDI Pro interface for the Apple IIc, IBM PC and Apple Macintosh; $249.95.)

ROLANDCORP US
7200 Dominion Circle
Los Angeles, CA 90040
(213) 685-5141

MSP (MUSIC PROCESSING SYSTEM)

Runs on: IBM PC, or compatibles; requires IBM standard color-graphics card (but will work with monochrome monitor).
Interface: MPU-401 plus MIF-IPC interface card.
Tracks: Eight, with unlimited merging.
Key Features: Up to 12,000 MIDI events with 256-Kbyte memory, and up to 65,500 with 640 Kbytes; built-in editor enables modification of sequence data down to a single note; transcribes a sequence into standard musical notation for output to a standard dot-matrix printer; handles all MIDI commands; built-in data "filter" to strip out pitch-bend and mod-wheel information and conserve memory; each track can contain information from up to 16 MIDI channels; "highly interactive user interface"; internal-sync, MIDI-sync and FSK/tape-sync.
SRP: $495.00; MPU-401 $200.00; MIF-IPC $110.00.

MUSE (MIDI User's Sequencer/Editor)

Runs on: Apple IIe, IIc, II+ (with 64 Kbytes of RAM) and Commodore 64.
Interface: MPU-401 plus MIF-APL interface card for Apple II+ and IIe; no interface card needed for Commodore 64; custom-designed interface available from J.L. Cooper Electronics for linking MPU-401 to Apple IIc.
Tracks: Eight, with unlimited merging.
Key Features: Up to 6,000 MIDI events; use of joystick enables "point-and-click" access to all functions; automatic punch-in/out facilities; chain mode enables entire tracks to be built out of smaller phrases; editing functions are used to delete, insert, and copy any portions of any track; controller "filters" similar to MPS software.
SRP: $150.00; MPU-401 $200.00; MIF-APL $110.00.

SEQUENTIAL
3051 North First Street
San Jose, CA 95134
(408) 946-5240

MODEL 964

Runs on: Commodore 64.
Interface: Model 242 MIDI Interface Cartridge.
Tracks: Eight.
Key Features: In excess of 40,000 notes; reads all MIDI information, including velocity, after-touch, pitch wheel and program changes (but excluding System Exclusive); autocorrect mode.
SRP: $99.00; Model 242 $99.00.

SOUTHWORTH MUSIC SYSTEMS, INC.
Box 275, Route #1
Harvard, MA 10451
(617) 497-7522

TOTAL MUSIC

Run on: Apple Macintosh with 512 Kbytes of memory.
Interface: Company's self-powered interface.
Tracks: 1,584 (99 sequences by 16 MIDI channels).
Key Features: Eight sequences running simultaneously, with 16 polyphonic channels per sequence; external sync to any MIDI source; dual keyboard inputs; music transcription with automatic beaming, stem direction and accidentals; editing resolution of approximately 1 millisecond.
SRP: $489.00.

SYNTECH CORPORATION
23958 Craftsman Road
Calabasas, CA 91302
(818) 704-8509

MUSIC DIGITAL STUDIO II

Runs on: Apple IIe, II+ and Commodore 64.
Interface: Passport Designs, Yamaha, Korg and company's own interface.
Tracks: Eight, with unlimited merging.
Key Features: 16 sequences with 6,300 MIDI events total capacity; four song placements per set of sequences; "Echo Thru" function enables playing of any MIDI-equipped keyboard or module from a master keyboard; "Track Shifting" provides digital delay to move entire track backwards or forwards because of processor delay in keyboard or drum machine (from 1/96 of quarter note to length of track or sequence); sync to off-tape signals.
SRP: $225.95; Apple interface with tape sync $199.95; Apple interface without tape sync $129.95.

IBM 48-TRACK

Runs on: IBM PC, or compatibles.
Interface: Company's own product (at present).
Tracks: 48, with unlimited merging.
Key Features: Similar editing capability as MDS II, plus editing of individual MIDI events; 32 sequences with total capacity of 20,000 notes; tempo entry via space bar, or as data; full on-screen autocorrect capability; color graphics.
SRP: approximately $425.00.

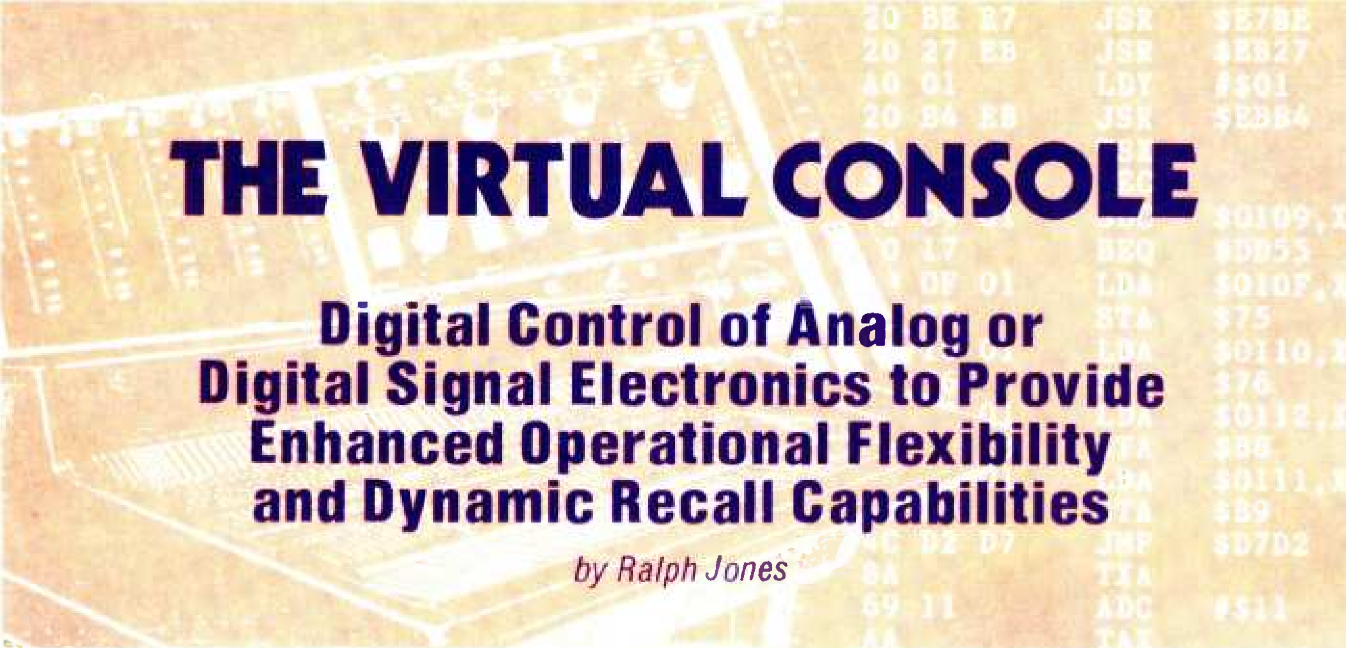
CHERRY LANE TECHNOLOGIES
P.O. Box 430
Port Chester, NY 10573
(914) 937-8601

CONNECTIONS

Runs on: Apple IIe (64 or 128 Kbyte).
Interface: Roland MPU-401.
Tracks: Eight.
Key Features: "Linear sequencer;" set up like an eight-track recorder, with similar "transport controls" plus manual/automatic punch in/out functions; 18,000-note storage on (128-Kbyte Apple); looping functions; "non-intimidating program."
SRP: $149.00.

TEXTURE

Runs on: IBM PC and Apple IIe (128 Kbyte).
Interface: Roland MPU-401.
Tracks: Eight.
Key Features: "Modular sequencer," in which 64 variable-length patterns are built up from beat information; 52,000 MIDI events available to create patterns; each pattern within a link can be repeated up to 255 times; efficient memory usage; individual note editing, plus compositional tools available.
SRP: $199.00.



THE VIRTUAL CONSOLE

Digital Control of Analog or Digital Signal Electronics to Provide Enhanced Operational Flexibility and Dynamic Recall Capabilities

by Ralph Jones

The concept of a virtual console has far-reaching implications for our industry, and to say that it is a "hot" subject at the moment is to understate the case. Mention the term to console manufacturers, and you are likely to receive responses that range all the way from dreamy predictions of the gleaming, all-digital "Studio of the Future," to a terse "No comment." The divergence of responses should come as no surprise: what is at stake here, after all, is no less than a fundamental redefinition of a studio's major creative tool. Consequently, manufacturers must confront an enormous complex of interrelated issues — some emotionally charged; all intellectually challenging — if they are to offer this new emerging technology.

By definition, a virtual console is capable of taking on the characteristics of a music-recording console, or a film-dubbing console, or a sound-reinforcement console — not only in terms of system architecture, but also in its control functions. In other words, the virtual console can be likened to a mirror: it reflects the functionality of the console topography best suited to the task at hand, which is to say that it forms an image of the task itself.

This specular feat is achieved by separating every manual control element from the audio signal path, and interposing digital control electronics. A computer system — which, in a practical implementation, usually requires the use of several interlinked microprocessors to control various functions — acts as a mediator between the controls and indicators placed in front of the operator, and the signal-processing circuitry. As a result, what the operator sees as the "console" becomes simply a fully-digitized interface, the sole function of which is communication with the controlling computer. In the emerging terminology, the switches, buttons, knobs, faders and indicators whose positions are constantly being scanned by the console's computer system are referred to as a "control surface."

The various in-depth interviews that *R-e/p* conducted with major console manufacturers indicate that, for many, the advisability of pursuing the virtual-console design concept is no longer in question. Indeed, at this writing, two manufacturers are beginning production of large-scale virtual consoles, and others are planning similar introductions within the next two years.

The first company to develop a virtual console was Rupert Neve, Inc., whose Digital Sound Processing (DSP) console features a totally digital signal path. A pair of DSPs have been in service at CTS Music Center and Tape One, London, for over a year now. Additional orders from The National Sound Archives, London, West Deutsche Rundfunk, West Germany, and others have also been announced. Now Harrison Systems, Inc. has joined the fray: the company plans to demonstrate its new Series 10, the first totally-automated, digital-controlled analog console to be offered as a production unit, at the upcoming AES Convention in New York during early October.

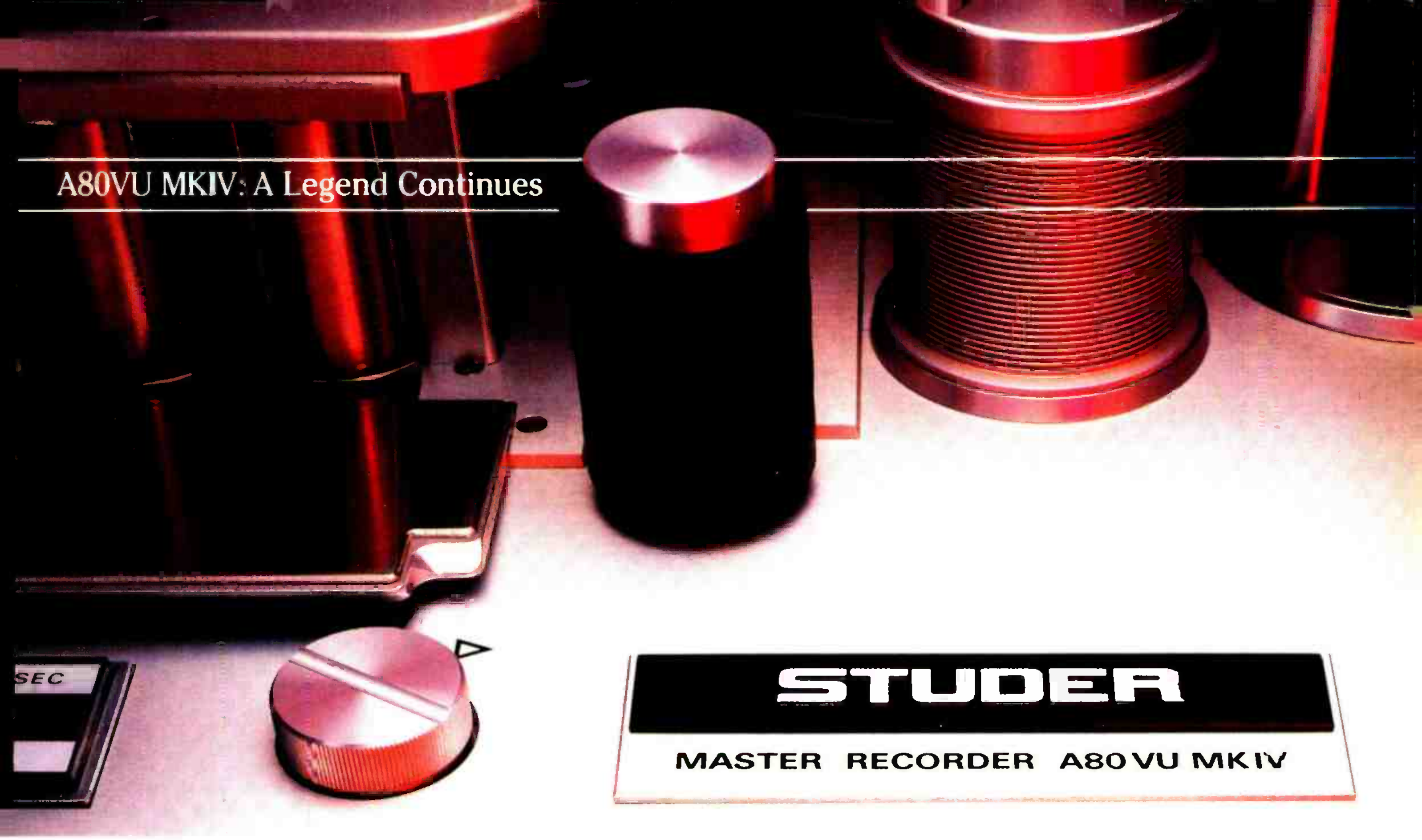
Among those companies with firm plans to produce a virtual console in the near future is Audio & Design/Calrec, Ltd., which currently is completing construction of a custom digitally controlled analog console slated for delivery to Thames Television, England, at the end of the year. The firm also plans to offer a production broadcast console in 1986, and a recording console system sometime in 1987. On another front, George Massenburg Laboratories and AMEK Systems and Controls, Ltd. have formed a new joint venture expressly to design, manufacture and distribute large-scale, automated virtual consoles. The company will be known as AML, Ltd. — an acronym for AMEK/-Massenburg Laboratories. (These developments are covered in greater detail in accompanying sidebars.)

The virtual console is clearly upon us, and we are likely to see some very interesting new approaches to console design over the next few years. Accordingly, it seems appropriate at this point to consider some of the questions that these and other manufacturers are currently addressing in their effort to develop practical implementations of the concept. Our discussion will include comments from those console manufacturers that consented to speak about the subject for the record.

The Control Surface

The digitizing of every control on a console surface opens up an extremely broad range of potential advantages and, as we shall see, some potential disadvantages as well. The key to maximizing the advantages lies predominantly in careful ergonomic design.

One of the more immediately obvious advantages of a digital control surface is the ability to scan, memorize, recall and reset every control function. Our industry has been creeping up on this capability (which George Massenburg predicts will be the "next flavor of the month" in



This is our newest multitrack. It is also the most affordable multitrack in Studer history.

For the fourth time since its inception, we've changed the A80VU. We've improved the sonic performance, tape handling, and durability. And we've substantially lowered the price.

Same outside, changes inside. In keeping with the Studer tradition, we made no superfluous cosmetic changes. We're not going to tell you this is an "all new" recorder. It isn't. It is a proven, legendary recorder incorporating several significant improvements.

Uh-oh, Something Is Missing. Yes. The transformers are gone. They've been replaced in the input and output stages with new high performance active balancing circuitry. Other MKIV improvements include a new master bias oscillator, extended record headroom, and a new record and bias driver compatible with all present and future high-bias requirements. Record electronics are now fully compatible with Dolby HX Pro* requirements.

Smoother Shuttling, Hardier Heads. The MKIV's new tape tension control system provides smoother tape handling, while a new extended wear alloy for record and play heads greatly increases head life.

Never Lower. The list price of the A80VU MKIV 24-track is lower than any of its predecessors. And that's in straight dollar figures, without adjusting for inflation. What's more, the A80VU MKIV now has a list price lower than most of its competition.

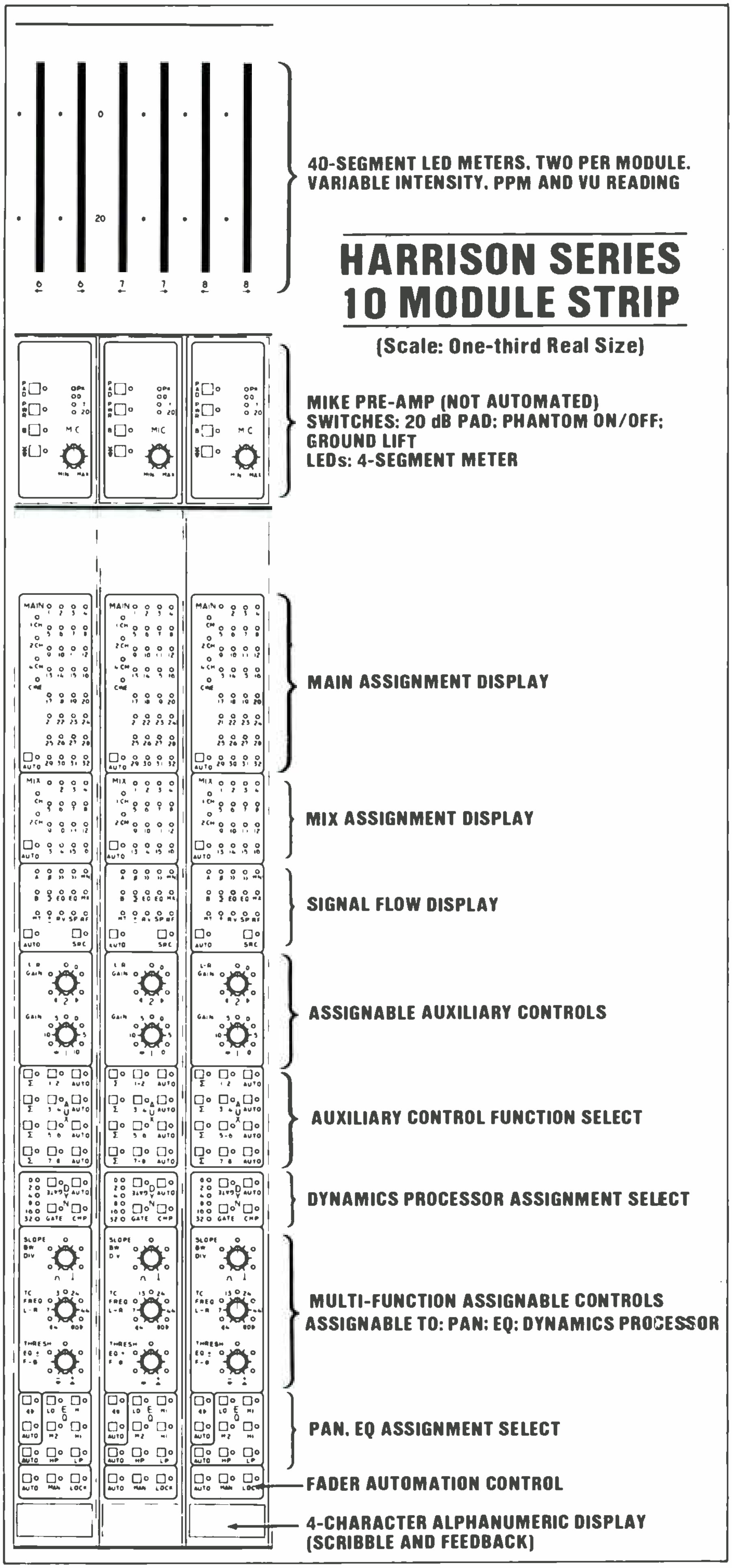
No Hocus-Pocus. How could we make the A80VU MKIV better and lower the price at the same time? Simple. We make it in Switzerland, and you pay for it in dollars. The favorable exchange rate does the trick. That means you get advanced electronics, Swiss precision, and low price. If you act now. This can't go on forever.

Your Time Has Come. If you've always wanted a new Studer multitrack but thought you couldn't afford one, your time has finally come. Call today and find out why the A80VU MKIV is one of the most advanced recorders available at any price. And then ask about our new lower prices. Be prepared for a pleasant surprise.

For more information, call or write: Studer Revox America, 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

*Dolby HX Pro is a trademark of Dolby Laboratories.

STUDER REVOX



HARRISON SERIES 10 MODULE STRIP

(Scale: One-third Real Size)

large consoles) for some time: a prominent example of a major step in this direction is represented by the Solid State Logic Total Recall system which, as an aid in manually resetting the console to its former status, stores front-panel knob and switch settings for subsequent display. When the console surface is under digital control, however, memorized settings may be transferred automatically, thereby reducing the time required to reset the console to just a few milliseconds.

But a digitized control surface implies far more capabilities than instant reset. For example, a single control may be assigned to any of several different functions. The same rotary control that serves as an auxiliary send level might, at the push of a button, become a pan control for that send. Three knobs that affect, respectively, frequency, bandwidth (Q) and boost/cut could be assigned in turn to different bands of a parametric equalizer. The number of controls on each strip may thus be reduced and, potentially, more functions included on each strip.

It goes without saying, of course, that such rotary controls will be continuous in operation — that is, without end stops — and that movement in a clockwise or counter-clockwise direction will simply provide a relative update instruction to the computer which, in turn, would alter the previous setting stored in memory. An obvious exception, however, would be servo-driven controls, which will function like conventional rotary and linear control elements, with the added ability to move the control instantly to a previously stored setting upon command.

Furthermore, grouping becomes a simple matter; any module could become a group master — or grand master — and every control on that module (or any combination of them) be made to affect every channel in the group. By extension, the total number of modules on the control surface may be reduced as well, with each module being assignable to any channel or group of channels. One major benefit of such a design is that it can make it easier for the engineer to handle a very large number of inputs: the modules closest at hand may be assigned to those channels that require attention, while other channels pass through the signal processor "invisibly," their settings having been previously defined and memorized.

The combined effect of such capabilities can reduce the physical size of a console. The ability to handle a

The work of these top producers speaks for itself.

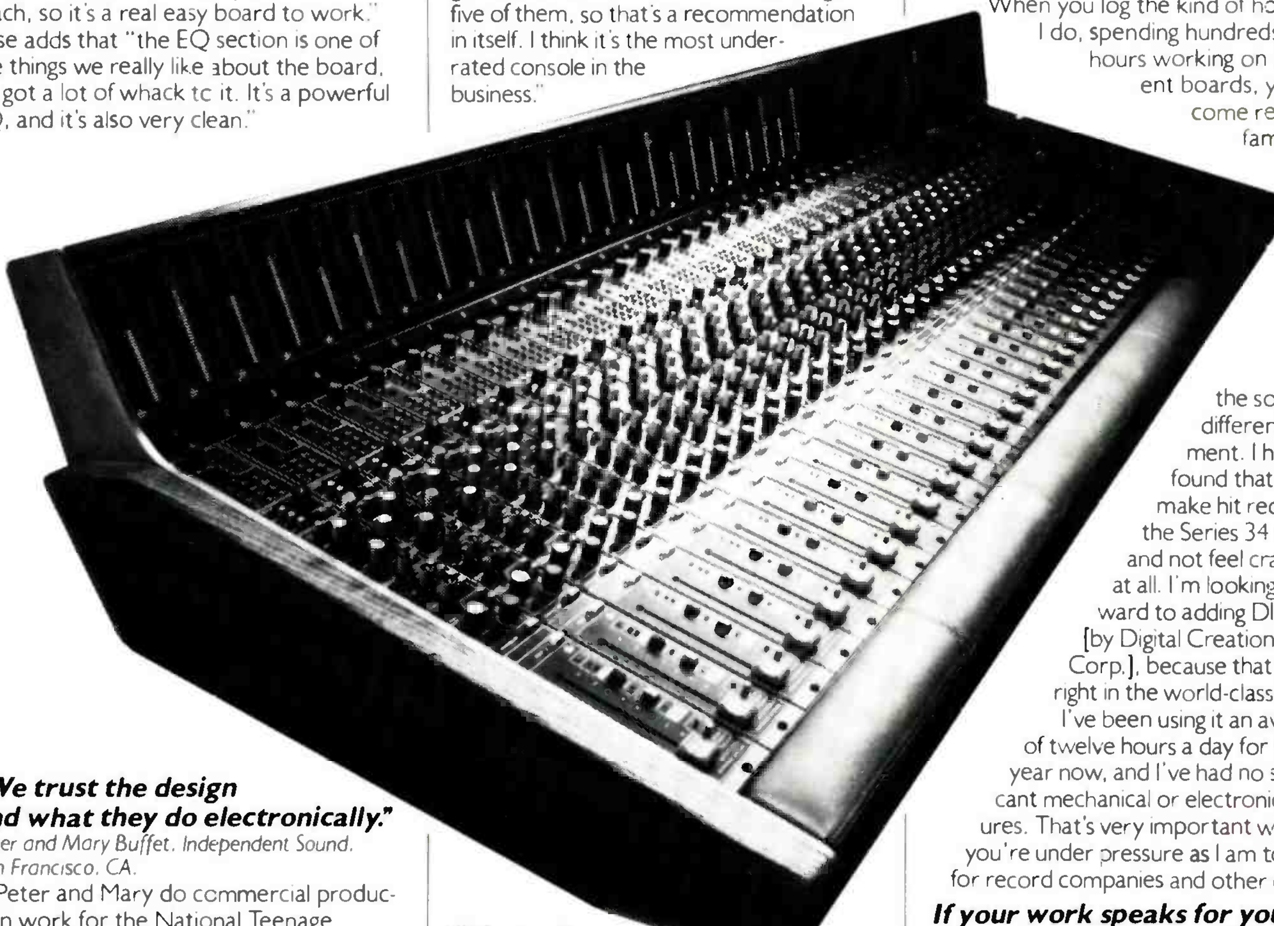
Now they speak for the Series 34.

"It works consistently. It's serviceable and simply laid out."
Bill Ryan and Jesse Henderson of Long View Farm, N. Brookfield, MA.

Long View Farm is the studio of choice for the J. Geils band and other major recording acts. "Our Series 34 helped us a lot on the Graham Nash solo album we did here recently," Bill recalls. "We used it for the vocal and guitar overdubs. The layout of the board is real nice. From the cushion to the meter bridge is a short reach, so it's a real easy board to work." Jesse adds that "the EQ section is one of the things we really like about the board, it's got a lot of whack to it. It's a powerful EQ, and it's also very clean."

"I'm too busy to use anything that gets in the way. The Series 34 helps me work fast and efficiently."
Denny Yeager, Denny Yeager Creative Services, San Francisco, CA.

Denny owns one of the world's largest Synclavier systems (32 megabytes). He uses his Series 34 for major film scores and national commercials. "I push my equipment to the limit every time I turn it on," Denny says, "and my Sound Workshop always gives me the results I'm after. I've bought five of them, so that's a recommendation in itself. I think it's the most underrated console in the business."



"We trust the design and what they do electronically."
Peter and Mary Buffet, Independent Sound, San Francisco, CA.

Peter and Mary do commercial production work for the National Teenage Drunk Driving campaign, the Milk Advisory Board and other demanding clients. Peter says the Series 34 helps them because "the board can be used by one person. It's fast, simple and very logically laid out. It doesn't get in the way of production. The ARMS console computer helps us on every job. It's part of the board, not an add-on. The company is fantastic to work with."

For additional information circle #253

"Musically, it's very transparent. It gives me back what I put into it."
Reggie Lucas, Quantum Sound Studios, Teaneck, NJ.

From his home studio, Reggie produces hit records and film soundtracks with major artists like Roberta Flack, Miles Davis, Stephanie Mills, Madonna, Randy Crawford and many others. "Basically I bought the Sound Workshop board because it sounded great," he says. "My engineer Joe Ferla and I had never worked on Sound Workshop consoles before, so we gave it a test, and we found that it sounded as good or better than many consoles for three, four or five times the price. I had worked on the ARMS system quite a bit at Sigma Sound, and had been real happy with that.

I think the Series 34 is as good as any of the most expensive consoles available. When you log the kind of hours that I do, spending hundreds of hours working on different boards, you become really familiar with the sounds of different equipment. I have found that I can make hit records on the Series 34 console and not feel cramped at all. I'm looking forward to adding DISKMIX [by Digital Creations Corp.], because that puts it right in the world-class league. I've been using it an average of twelve hours a day for about a year now, and I've had no significant mechanical or electronic failures. That's very important when you're under pressure as I am to deliver for record companies and other clients."

If your work speaks for you, depend on the Series 34 by Sound Workshop.

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It makes sense.

Sound Workshop Professional Audio Products, Inc.
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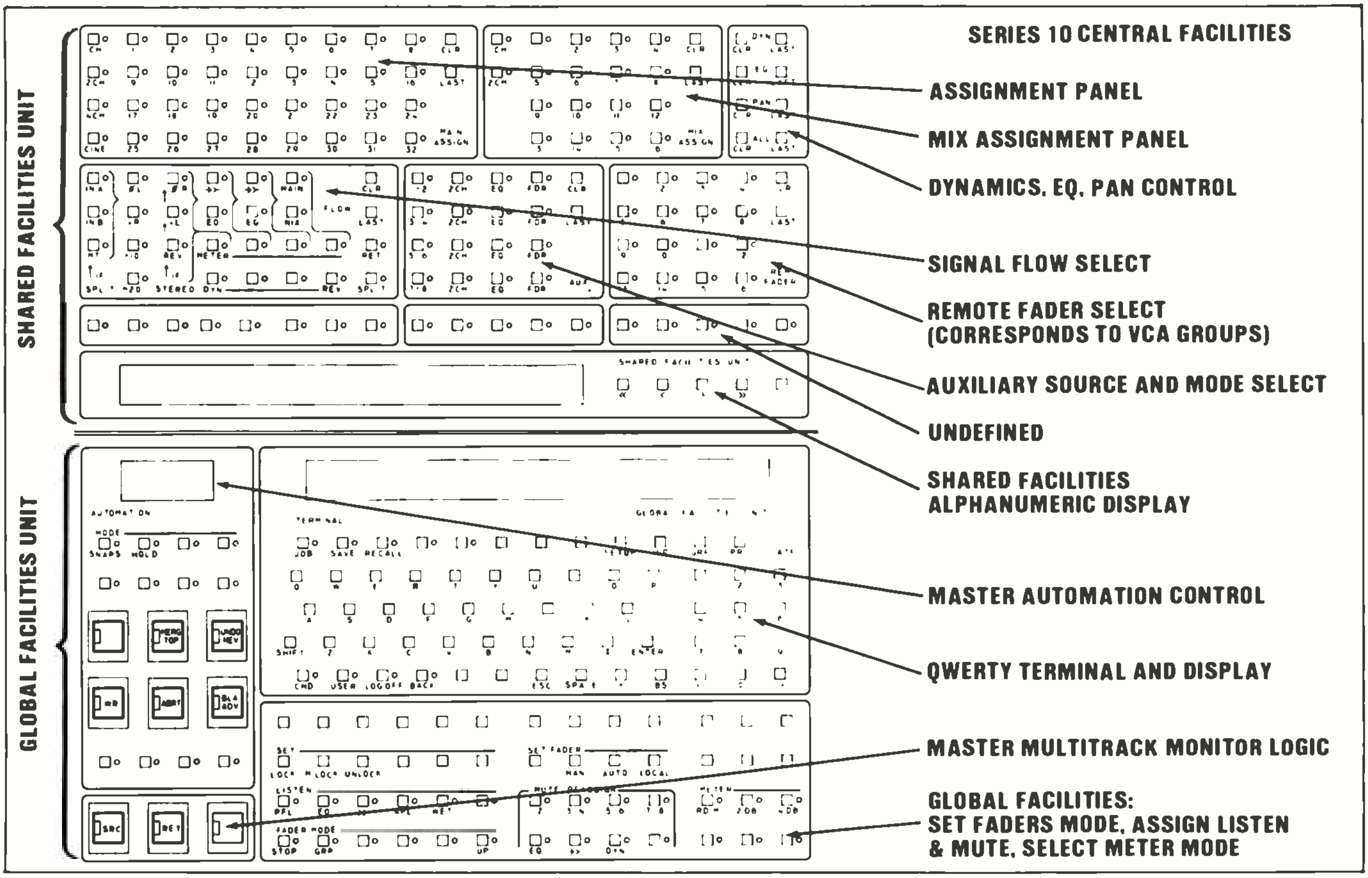


large number of inputs from a small control surface could be of major benefit in teleproduction studios, for example. Traditionally, audio has been allocated a fairly small amount of square footage in such facilities and, in many cases, that allotment is not likely to change for a number of years. The advent of Stereo TV broadcasting, however, poses greatly increased demands on television-audio facilities. Assignable consoles may make it possible for those demands to be met by studios in which, because of space limitations, a large, traditional console simply would not fit.

In music recording, reduced console size also may offer sonic advantages. If the console area is made significantly smaller, George Massenburg points out, "the acoustical surface which normally is present to great effect in front of any sort of control-room monitor is greatly reduced." A control surface of sufficiently small size and weight could easily be positioned for minimum acoustic reflection and, incidentally, would leave more space in the control room for direct-inject keyboards, drum machines, and similar instruments.

All of which may sound great in theory, but obtaining these advantages is not as simple as it might appear. Chris Jenkins, product development manager at Solid State Logic, explains: "The most obvious potential disadvantage of assignable architecture is the other side of the coin: because they are assignable, the controls become less immediately accessible. So, you have to be extremely careful how you implement the concept, or all you end up with is a smaller desk that's *harder* to use."

Obviously, manufacturers must strike some kind of balance between increased capacity and operational convenience. Some very careful analysis of the ways in which engineers interact with consoles is required. As

HARRISON SYSTEMS' VIRTUAL CONSOLE DESIGN PROPOSALS

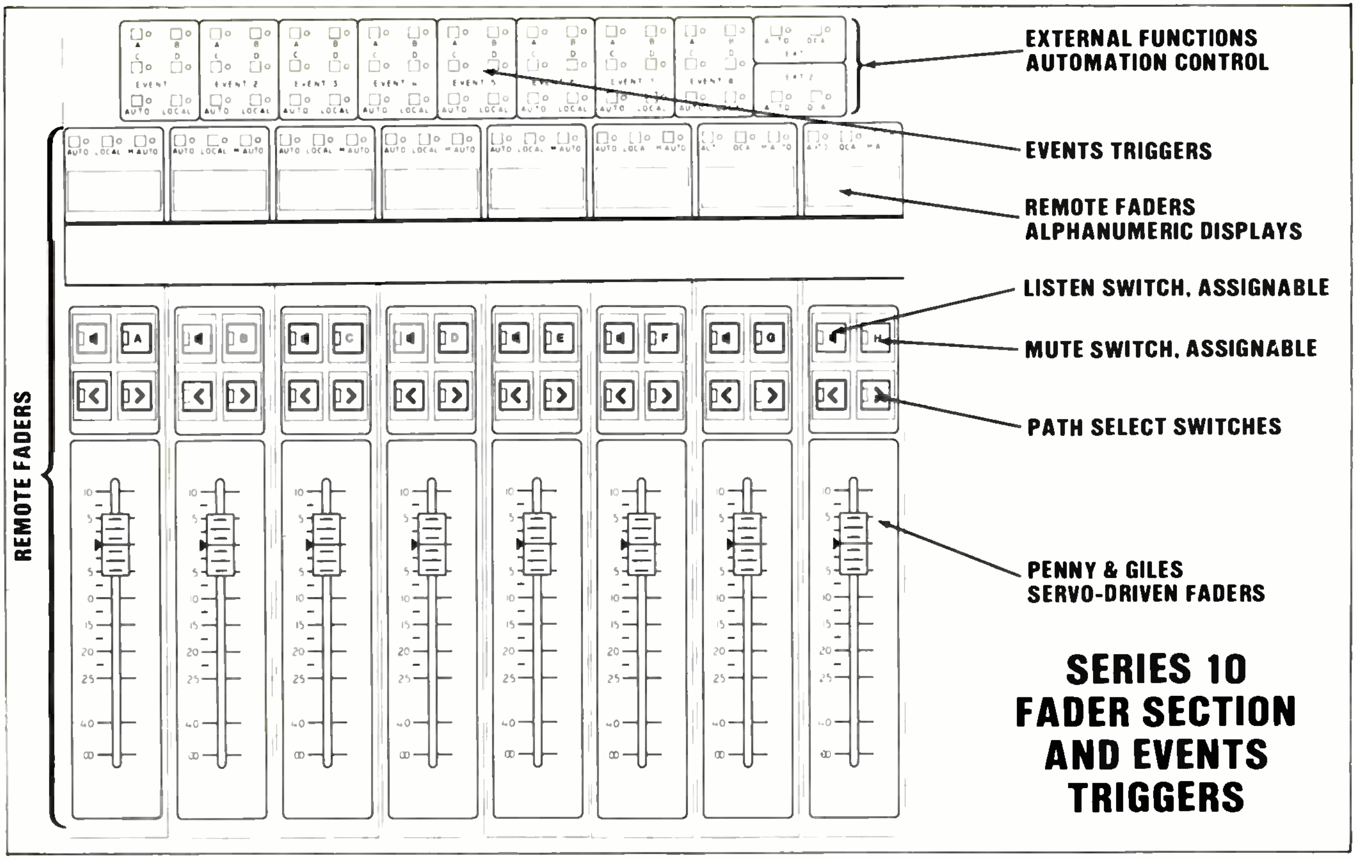
The Harrison Series 10, which debuts at the forthcoming AES Convention, is heralded as both the world's first production analog virtual console, and the first audio console to feature a totally-automated control surface.

In contrast to the majority of proposed virtual console designs, the audio path of the Series 10 is *not* separated from the control surface: both audio and digital control signals pass through each module. The system architecture employs distributed multiprocessing, comprising two microprocessors per module strip; one local automation slave processor for every 16 modules; and a central automation master processor located in an external rack. The Harrison 20-Mbyte hard disk automation system affords dynamic automation of all console functions with sub-frame accuracy, and the entire console configuration may be reset in less than one video frame.

Each Series 10 module controls two independent, completely automated signal paths, and modules can operate in a split mode or as a tracking stereo pair. In addition to a motorized Penny and Giles fader, five rotary controls are provided on each strip; these are assignable to equalization, pan, dynamics processing, and auxiliaries. Extensive local LED displays indicate signal flow, mix routing, main routing, and automation status of the selected path. A four-character alphanumeric display serves as a "scribble" panel, as well as providing precise feedback of control settings. Levels are indicated by 40-segment LED barmeters, which can be selected to be peak or VU reading.

The Series 10 central control panel is divided into two main sub-sections. The "Shared Facilities" section addresses modules one at a time, selecting module signal flow, main assignment, mix assignment, and auxiliary source and mode. This section also allows selection of "Remote Fader," the Harrison term for the equivalent of a VCA group. Provision is made for copying all or part of the settings from one module to another.

The "Global Facilities" section of the central control panel addresses the entire console at once, determining fader mode, metering mode, and "Listen" and "Mute" assignments. Master control for the automation is located in this section, including 32 external events triggers, and two external automated functions. The Global Facilities section also incorporates a standard keyboard terminal. □□□



Jenkins implies, often-used functions cannot be "buried" in such a way that the operator is constantly forced to go through intermediate steps to gain access to them. The design process thus demands that manufacturers consult with their clients, anticipate their demands, and perhaps resist attempts to go overboard in relying on assignable features.

An additional ergonomic question raised by assignable controls is that of control-status indication. If we press a button and thereby assign a particular rotary control to pan, for example, how should the current pan status be shown? (It should be remembered that, in a "conventional" console, a knob or fader fulfills two, complimentary functions: not only does it enable the user to alter a setting, the knob pointer or scale also provides a visual indication of the pan, cut/boost, level etc.)

A number of alternative solutions to providing an indication of previous control settings on a virtual console suggest themselves. Among these are alphanumeric displays of frequency, level, pan percent, etc; vertical rows of LEDs placed beside each control, with a scale marking; semicircular arrays of LEDs around a rotary control to show angular position; various types of CRT displays; or possibly a servo-driven potentiometer and knob.

Opinions among manufacturers differ greatly on this point. One way to arrive at a solution may be to examine how the operator visualizes a particular function — and one possible key to the visualization process is how we talk about console functions. Equalization, for example, is generally expressed as a particular amount of boost or cut (in decibels) at a particular frequency; bandwidth is rarely mentioned. It might be appropriate, then, to indicate frequency, cut/boost and Q via alphanumeric displays.

Similar reasoning could be applied to other functions. Though various manufacturers undoubtedly will arrive at different conclusions regarding the best means of display for a given function, there seems to be a consensus that the console should allow instant appraisal of, at the least, the status of most functions of selected channels.

Local versus Central Control

Assignability also raises the possibility of eliminating local control of certain functions, in favor of a centralized panel equipped with assignable controls. The important decision to be made, of course, is *which* controls may be centralized, and which are best left local to the module strip.

One function that everyone seems to agree is best handled centrally is signal routing. Indeed, computer-controlled routing has already been implemented with various degrees of sophistication in a number of currently available consoles, and has even reached mid-market boards. The Soundtracs CM4400 system, for example, employs a microprocessor-controlled switching matrix to determine subgroup and master assignments, and provides a means of memorizing patches. Similarly, the Allen and Heath/Brenell CMC Series of mixers employs a computer routing system, called CARS, which allows centralized control of input/output routing and mutes; this system also includes provision for memorizing assignments. In these and other computer routing implementations, assignments are made from a central facility, rather than from duplicate switch banks provided on each module.

However, it is the combination of computer-controlled routing with assignable controls that truly makes a virtual console, since this permits the construction of virtual architectures to conform to the operational requirements of different applications. One of the reasons that such a capability is attractive to console manufacturers is that it offers the possibility of selling a single piece of hardware in many different markets, which greatly simplifies the already complex manufacturing process, shifting the burden of customization from hardware to software. (The argument assumes, of course, that the manufacturer can find ergonomic solutions that will be acceptable to all markets.)

For an individual studio, flexible architecture can be a great advantage, since it may allow the studio to more easily broaden its own market base, booking sessions for just about everything across the spectrum from music recording to film and video post production. There is some evidence that this hope is realistic. Barry Roche, president of Rupert Neve, Inc., reports that "the CTS [digital] console has been used for film scoring, mixdown, sports programs, concert programs, disk mastering, tape transfer, you name it — all with great success."

While most agree that assignments can and should be made from a central location, there is less agreement about centralizing other functions. No one seems to be in favor of controlling everything from a single knob and button, for obvious reasons. But how far, in fact, can we go in distilling the control surface? The Neve DSP, for example, provides a total of two sets of equalization controls for the entire console; these may be assigned centrally to any of the inputs, outputs, monitors, auxiliary busses, etc. Similarly, while each input may be provided with a dynamics processor (which incorporates limiting, compression, expansion and gating), the dynamics parameters are controlled centrally — not on each strip.

QUAD EIGHT COMPUMIX IV HARD-DISK AUTOMATION SYSTEM: A FIRST STEP TOWARDS A VIRTUAL CONSOLE

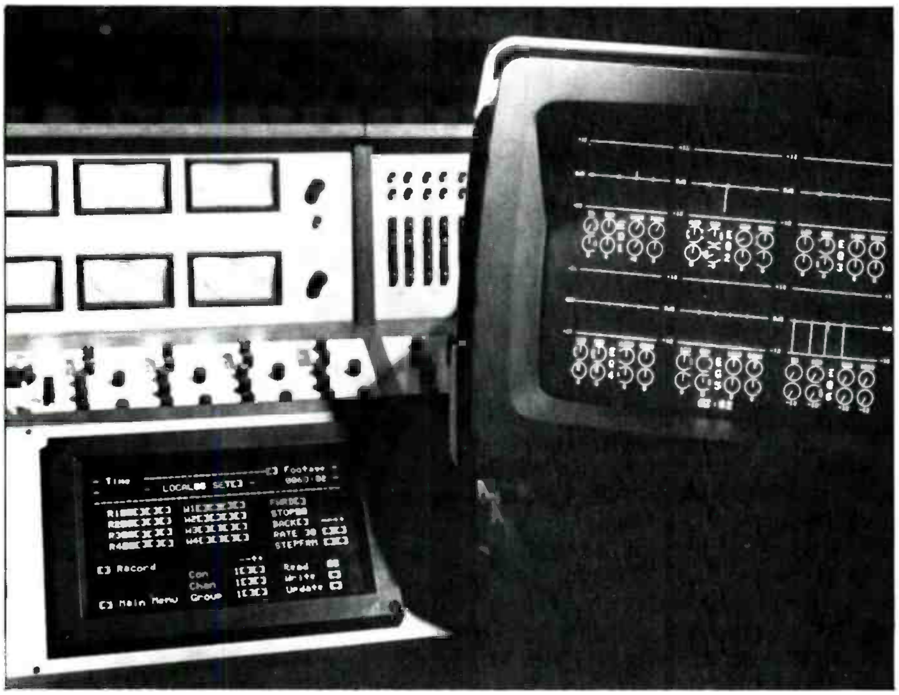
For many years, Quad Eight has been producing custom-designed consoles for the film re-recording industry. In pursuit of a virtual console best suited to its primary market, the company (now a subsidiary of the Mitsubishi Pro-Audio Group, and located in a new manufacturing facility in San Fernando, CA) has elected first to develop an automation system that is both tailored to the working practices of film re-recording, and capable of handling the data density of a large-scale virtual console.

The result of this development effort is Quad Eight's Compumix IV, a hard-disk automation system tha utilizes distributed multiprocessing. The system is designed around a 68000-based, 32-bit real-time host processor running a multi-user, multitasking version of the Forth language/operating system, with Smalltalk object program extensions written by Quad Eight.

In addition to performing the functions of data synchronization and operator interface, the host stores automation data on an 80-Mbyte Winchester hard-disk. The host communicates with up to three local console computers over an ARCNET local area network link (LAN), usng a one-megabit per second serial communication protocol. Each console computer, which serves as a central control point for automation functions, communicates in turn with individual 6805-based module computers via an eight-bit parallel bus.

To accommodate the needs of separate dialog, effects and music mixers, Compumix IV provides for three independent, concurrent automation environments. Each mixer controls his or her own automation using a touch-sensitive plasma display panel, and operator feedback is provided by a high-resolution color graphics display (managed by a graphics co-processor). At any time, each mixer has access to a maximum of four separate, real-time automation records that can be written to, read, updated, or selectively merged, independent of the actions of the other two mixers.

Currently, Compumix IV automates the functions of fader with mute; graphic and parametric equalization; joystick panners; multichannel film panners; A/B transfer; and device insert keys. The system data structure has been designed such that its capacity may expand as Quad Eight develops the hardware necessary to automate all the functions of a module strip. Compumix IV will be fitted to Quad Eight's new SuperStar console, which is scheduled to be unveiled at the forthcoming New York AES Convention, and features central automated routing assignment. ☐☐☐



Quad Eight Compumix IV touch-sensitive plasma input/output panel for film re-recording (*left*), and high-res CRT graphics display of automated parametric EQ.

Both AML and Audio & Design/Calrec are pursuing centralized control of all but fader and fader-associated functions. Harrison, however, has taken a different approach with the Series 10, which represents a relatively conservative solution. According to Claude Hill, VP of marketing at Harrison Systems, "Everybody's trying to decide what to control locally and what centrally. One of the options that we're making great use of is local display of centrally-controlled functions. For example, you determine routing centrally, but assignments then are indicated locally by LEDs on the module. Each and every module has a full set of assignment LEDs — 32 for the multitrack, and 16 for the groups. If you assign input 10 to track 10, then the LED on input 10 that [is labeled] 10 beside it comes on. You do not have to query the system to determine the routing: you can still look at the desk and see exactly what you see today on a conventional console."

Harrison also eschews central control of equalization or dynamics processing, providing instead assignable controls for these functions on each module, as can be seen from the accompanying front-panel layout diagrams of the Series 10.

Signal Processing

One of the most important aspects of virtual-console design is the challenge of developing a fully programmable audio processor. The main problem to be solved is finding a replacement for the variable resistor, which is a very inexpensive device with excellent audio capabilities. The problem extends far beyond the task of emulating a fader, since variable resistors are one of the fundamental building blocks of analog audio circuitry.

Two common approaches to the design of a programmable resistor employ control voltages to drive either a VCA (voltage-controlled amplifier) or a servo motor-driven potentiometer. Both solutions have gained acceptance as a replacement for channel, group and monitor faders, but neither is necessarily suitable for all console applications. Controlling every resistance-variable function in a single strip with servo-driven pots, for example, would be excessively costly, and requires an inordinate amount of space behind the panel; their use in several centrally assignable controls would be feasible.

On the other hand, despite the excellent specifications offered by present-generation VCAs, the thought of placing upwards of 20 such devices in the signal path would make many audio professionals blanch. George Massenburg offers the opinion that such objections are "religious in nature," pointing out that the use of VCAs for audio control is largely an emotional issue. While it is certainly true that VCA design has improved immensely since their introduction, practical circuits still require a fair number of parts and some adjustment to achieve good long-term tracking and linearity. Therefore, some manufacturers have sought alternatives.

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THE VIRTUAL CONSOLE

Harrison Systems is one such company. "The Series 10 has *no* VCAs," Claude Hill relates. "Levels are set by digitally-controlled attenuators — DCAs, if you will. The DCAs do all of the level-sets, pans and muting functions." And the advantages of using DCAs? "One, it is inherently and absolutely *stable*. Two, every one of them is the same — unit-to-unit match. Three, it does not have any control-voltage leak-through [which can cause problems with VCA designs]. DCAs are also much faster than VCAs, and the design that we have has the same apparent linearity as a linear control."

While Harrison naturally would not provide specifics of their DCA circuit, one common method of achieving such digital control involves the use of FET arrays or MDACs [multiplying digial-to-analog convertors] to switch ladders of fixed-value resistors. Some of the design factors that engineers must consider in implementing schemes of this type are standing noise, distortion, charge injection, resolution (particularly in the steep area of a logarithmic curve), and control speed.

Many have suggested that one of the advantages of the virtual console is that, since the front-panel controls are divorced from the signal path, the signal-processing circuitry may be physically isolated in a compact, rack-mounted unit. Electronics so mounted could require less service, since they are less vulnerable to physical shocks or tampering. Perhaps most importantly, in the case of digitally-controlled analog consoles, the audio signal path may be optimized without regard for the physical placement of controls. This offers the possibility of greatly improved audio quality, since the console designer no longer has to cope with, for example, routing busses that can be 26 feet or more in length.

Interestingly, Harrison has not chosen to separate the Series 10's signal path from the control surface: each module carries both audio *and* digital signals, the system being implemented by using distributed multiprocessing, with two microprocessors in each module strip. Claude Hill explains: "If you want to build a virtual console, one of the ways to do it is to take a desk and cover it with pushbuttons and LEDs, run a cable off it to a central computer, then run a cable off that to a rack of electronics. That's a great way to build systems. The only problem is, if that computer dies, you've got *no* console — you can't even do a voice-over!

"Our clients can't tolerate that, so we've built the system distributed in such a way that it's no less reliable than a standard console."

It's certainly true that reliability is a major concern in console design, and automation computers have been known to go down on occasion. Nonetheless, there are precedents for a separate signal path with computer mediation, and they are to be found in the reliability-conscious realm of teleproduction. Many video switching and special-effects desks consist of a control panel that is sampled digitally, the information then being passed to a rack of electronics that is usually located in the studio's clean-air room. This may, therefore, be yet another emotional issue, and it might be that virtual audio consoles so constructed will have to prove themselves over time in order to allay customers' fears.

Analog versus Digital

Yet another partially-emotional issue is the question of moving towards an all-digital signal path. While the strict definition of a virtual console deals only with the separation of the signal path from the control surface, and not with the nature of the signal path itself, nevertheless the digitizing of the control-surface functions at least implies the question of digitizing the audio as well. Because digital audio seems destined to ultimately gain hegemony over analog in the consumer marketplace (and, hence, in the professional markets), does it not make sense to go the full distance and digitize the signal path as well as the control path?

Some advocates of a totally-digital signal path are calling digitally-controlled analog consoles an "interim technology" — a distraction from the pure digital research which, they say, will change consoles completely, turning them into an entirely new instrument. On the other hand, digitally controlled analog could be portrayed as an evolutionary step, since a digital audio processor will require a digital control surface, too. Assignable analog consoles could, therefore, be viewed as a proving ground for sorting out the ergonomic questions.

Certainly, economic considerations are one major factor holding back most manufacturers from taking the full-digital plunge; the developmental costs associated with a fully-digital console are prodigious. Bob Budd of Quad Eight, a subsidiary of the Mitsubishi Pro-Audio Group, elaborates: "It seems to me that we're right about at a stage when the first dynamic RAMs were coming out: you could dream about megabytes of it, but when you had to do it 512 bits at a time, at $20.00 a chip, it was a little bit impractical! A lot of people just hung out with their core memory there for a while, because it did everything that they needed it to do."

AMEK/MASSENBURG LABORATORIES VIRTUAL CONSOLE DESIGN PROPOSALS

George Massenburg Labs, Inc. and AMEK Systems and Controls, Ltd. have announced the formation of a new co-venture to design, manufacture and distribute large-scale assignable audio production consoles. The new company, to be called AML, Ltd. (AMEK/Massenburg Laboratories), plans to develop a digitally-controlled analog console based upon four optimized methods of signal control. The unit will feature a symmetrical architecture, with no in-line tape monitoring; rather, tape returns and external processing will always enter a fully-featured input channel.

A very large input/output assign matrix is planned. Nominally 80 in by 64 out, the matrix will be expandable to 128 by 128, comprising up to 104 main mixes, 16 foldback mixes, eight "master" stereo mixes, and special-purpose outputs to accommodate multichannel film-sound formats.

Assignable, servo-driven channel faders will be provided, with soft switches for such functions as muting and assignment control. Local LED indicators will display EQ in/out, filter in/out, submaster assignment, graphics display assignment, dynamics in/out, and source (line in, mike in or group out). A four-character alphanumeric display will be provided to indicate channel number, track title or channel label.

It is planned that the system will accommodate two or more assignable master sections, containing motorized rotary controls for equalization, dynamics processing, trims, foldback levels and pan. Group and submaster assignments, as well as insert switching, will also be effected from the central panel.

Finally, the console will include a facilities section (for talkback and monitoring), plus a metering and display section. The latter will incorporate as many plasma bar graph displays as there are console group outputs, along with a high-resolution CRT graphics display of simultaneous parameters in groups of four channels. The graphics display will act as a "window" to be panned across the control surface.

AML asserts that the console will be provided with dynamic automation of every control. A monochrome video display terminal will be used for entry and display of automation parameters, and faders, of course, will be servo-driven. Additionally, the company has expressed a commitment to support MIDI data rates and interface specifications, as well as the proposed SMPTE/EBU specifications for external machine control. ◻◻◻

Quad Eight's soon-to-be-released SuperStar console, Budd reveals, will incorporate central automated assignment, and provision for automated equalization.

Market acceptance may be another factor that restrains most manufacturers from following Neve's DSP lead. After all, digital recorders are only now beginning to gain widespread acceptance in the professional market, and are by no means predominant — even in world-class studios. And what of the cost factor? As Michael Tapes of Sound Workshop puts it, "There's going to be a market for [digital signal-processing consoles], like the people who bought the first solid-state amplifiers. That's the way technology moves along. For a relatively small company like ours, however, digitally-controlled analog is more practical. We can achieve digital control very successfully, and finally optimize the analog circuitry without the physical constraints."

Some also say that the digital signal-processing console will only become practical when the console is integrated with the audio storage medium, and digital signal processors predominate. The argument goes something like this: It doesn't make sense to convert from the analog domain to the digital domain, do the processing, then come back to analog for insert points, convert back to digital for further processing, reconvert to analog and go to the storage media, and so on — until there's nothing left of the transient response and harmonic content of the musical information. When the process can be restricted to a single analog-to-digital conversion, and that conversion process can take place in a way that does not degrade the signal parameters, then we'll do it, the argument concludes.

Here we come to one of the more serious arguments about digital signal processing: The question of signal quality. Many contend that current digital signal-processing technology simply cannot equal the quality of contemporary analog consoles. The problem is complicated by the industry having standardized on 16-bit quantization. Bob Budd: "We have no particular interest in a 16-bit system. If you look at the dynamic range of our current analog system, it's better than 96 dB [offered by 16-bit digital]. So, we'd be looking at something more in the range of 20-bit, which means that you wind up doing your multiplications in more bits than that, and you can round off and throw away the LSB [least significant bits lost during digital amplification and data manipulation]." You also wind up with higher costs in the process.

Neve, of course, has gone the extra distance to 20-bit linear quantization within the DSP console, and 32-bit processing at critical mix points. As the acknowledged pioneers of such technology, Neve has demonstrated its commitment by making the considerable investment necessary to bring a fully-digital console to the marketplace. And Neve's Barry Roche has no reservations about sonic quality: "The benefits of the digital sound, of course, are paramount — that's what we're in the business for. We take that as being for granted; that we will have an improved audio signal throughout the entire process."

Console Function Automation

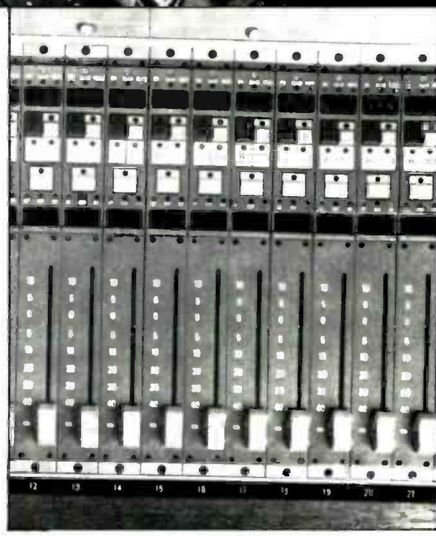
Perhaps the most immediately apparent benefit of a virtual console is total automation. As Claude Hill explains, "This is the one thing that our extensive market research has indicated people want and expect to be a benefit of a console that is either digital, or digitally-controlled analog. Total automation represents a quantum leap over conventional fader automation: until you've automated the entire console, you don't get the full benefit of automation."

Certainly, in some ways, assignable architecture makes it easier to automate the entire console. After all, once you've made a fully-assignable console, you've conquered the data-acquisition problem, which is half the battle: you're already sampling every control on the surface for remote or local control of audio electronics. Furthermore, a programmable audio processor obviates the need to worry about installing remote, optically-isolated relays, VCAs or servo motors, and other control circuitry. Finally, assignable architecture reduces the input requirements of the automation system, since there are fewer controls to sample.

As Solid State Logic's Chris Jenkins points out, however, "Assignability does nothing to reduce the output requirements of the automation computer, because each assignable control still has an audio processing counterpart for every channel, and the computer still has to look after all of these at once. That's *several* orders of magnitude more difficult than simple fader automation."

Input requirements notwithstanding, therefore, one still needs an automation system capable of very high data density if the aim is to provide dynamic control of console settings — a step beyond the static "snapshot"-style capability offered by present-day automation. Morever, a large number of changes must be

The Neve DSP features central assignment panels (*above*), multifunction "soft keys" for functional control, and servo-driven faders (*right*). Shown *above right* is the 64-channel DSP console delivered to the BBC, England, for installation in a remote truck.



effected over a large number of channels in a very short period of time. (It seems to be an accepted fact that dynamic updates would need to be generated at least every video frame, which is equivalent to about 33 milliseconds.) Typically, therefore, to achieve such high-speed data transfer and system updates, total automation requires that the system incorporates such features as multiprocessing architecture, a multitasking operating system, fast disk access, and a fair amount of on-board RAM (random-access memory). It's no surprise, perhaps, that the Harrison Hard Disk Automation System, which services the Series 10, took five years to develop.

Bob Budd of Quad Eight speculates that the sheer data density resulting from total automation may even require operators to do their own data management. "I could see how decisions regarding what actually is stored might be under the control of the mixer or the studio," he explains. "It may very well come down to the economics of data storage, just like buying hard disks for your PC: the system capacity may determine how much of a console [control settings] you could store in real time.

"If someone is assigning something hard left on a switch, for example," the designer continues, "or just wants to take a panner and put it hard left, I don't see any reason to log that piece of data 30 or 60 times a second, or whatever your scan rate is. If you're going to be slinging something back and forth all day long, then you do need to store [the dynamic changes]. So, I think that setting up a mix may demand some analysis of the task, and a little knowledge of data management — what you *want* stored, and what you *don't*."

In fact, some are still questioning the need for dynamic automation of every console function. Neve, for example, does not provide total dynamic automation for the DSP, relying instead on "snapshot" automation for all functions other than faders. Some designers regard this as a curious omission. But, as Barry Roche explains, "The real question at this point is: How useful is dynamic automation of EQ, pan, and other functions? That need has not pres-

NEVE DIGITAL SOUND PROCESSING CONSOLE: THE WORLD'S FIRST ALL-DIGITAL FULLY ASSIGNABLE CONSOLE

As is now well known, the Neve DSP was both the world's first virtual console, and the first audio console to employ an entirely digitalsignal path. In more ways than one, the DSP represents a radical departure from the conventional analog console.

The DSP makes extensive use of central, assignable controls. While in a "typical" configuration, 32 local faders with mute, PFL, solo, source selection and a "softkey" are provided, virtually every other console function is controlled from a limited number of assignable controls, and operator feedback is provided by a high-resolution, color video display.

All signal routing and processing configuration is performed from the "Assignment Panel," which also manages the console memory system. A 32-character display is provided for preview and, if necessary, correction of commands prior to execution.

One "Function Assignable Controls Panel" is provided for every four console modules. The FAC Panel is used to control levels and pan positions of the signal paths assigned to it, and two assignable rotary controls are provided per fader position for this purpose. Functions controlled by the FAC panel include mike sensitivity, channel pan, aux-send level and pan, Dynamic Range Control gain make-up, and multitrack-send level.

The "Equalizer Control Module," two of which are provided with a typical DSP, provides the controls for four bands of equalization, along with high-and low-pass filters. Frequency and bandwidth (Q) are stepped functions, selected via pushbuttons. Additional pushbuttons select boost or cut, and the amount thereof in stepped increments. The cutoff frequencies of the HP and LP filters are selected with rotary controls. The equalization function is selected by an "Access" button located on each fader module.

Two "Dynamic Range Control Modules," assignable to any channel, provide control over the functions of a limiter, compressor, expander, and noise gate for the selected channel. Parameters are set using individual assignable rotary controls with associated displays.

The signal processing electronics are physically isolated in racks separate from the control surface, and are intended to be installed in an air-conditioned machine room. The control surface communicates with these racks over a fiber-optic link. Both analog and direct digital (AES/EBU-format) outputs are provided. The DSP may be fitted with NECAM 96 servo-controlled automation for dynamic control of faders and mute functions; automation of all other functions is "snapshot." □□□

ented itself, actually — it may not exist. I've spoken with the people who have actually used the DSP, and we've explored the question of what the applications would actually be. When you get right down to it, they're extremely limited."

Some manufacturers agree with Neve. Audio & Design/Calrec, for example, plans on providing a total of only three instantly-accessible snapshots resident in RAM, plus 30 stored on disk, for its assignable broadcast console. Others feel that total dynamic automation is essential: George Massenburg calls it a "a major benefit of assignable architecture, the advantages of which in music recording, film post, and teleproduction are obvious and not at all trivial." As you might guess, AML, Ltd. plans to offer dynamic reset of every control on its assignable console from the point of introduction.

Another issue on which Massenburg is known to have strong opinions is the question of transporting automation data between systems of various manufacturers. Although GML has attempted to open the issue of standardization of list-management protocols and disk formats, to date there seems to be little sympathy for Massenburg's views among most makers of automation systems. Harrison's position is typical. "Like any other form of standardization," Claude Hill relates, "it's noble. But anything other than fader data in our system wouldn't mean anything to anybody else's, and their data wouldn't mean much to us. For example, we could probably read Total Recall data written with an SSL —since it is 3740 format — and reset our EQ close to it, but it *wouldn't* be exact."

Of course, assignability greatly complicates the question. If we can't agree on a standard format for simple fader level data, how can we deal with the far greater density of data generated by a totally-automated system? Massenburg is adamant, however: "Other industries, from automated circuit board parts insertion, to musical instrument Massenburg control, to PCB design automation, to video off/on-line automation have accepted transportability [of data]. With cooperation, it is certainly possible now, and inevitable in time."

Peripheral Considerations

In mid-June of this year, the European Broadcast Union (EBU) and the Society of Motion Picture and Television Engineers (SMPTE) carried out extensive tests of an interface protocol that the two organizations are jointly promulgating for control of broadcast-studio equipment. Designated the "EBU/SMPTE Digital Remote Control System," the protocol is a variant of the RS-422 standard.

Equipment developed by eight different manufacturing and broadcasting organizations — each according to its own interpretation of the specifications — was brought together for the tests on the premises of the Institute fur Rundfunkteknik (IRT) in Munich, Germany, and interconnected in a variety of configurations. The participants were Ampex, Bosch, BBC, IRT, Kudelski, Pro Bel, Solid State Logic, and Studer. Test support was provided by Dynair, the Grass Valley Group, and Sony.

According to a statement issued by the EBU, "The test successfully demonstrated the validity of the specifications, and the ability of the different kinds of equipment to pass control messages and return tallies between controlling and controlled devices, which includes video and audio tape recorders, a routing switcher, and a variety of control panels. Validation of the specification included investi-

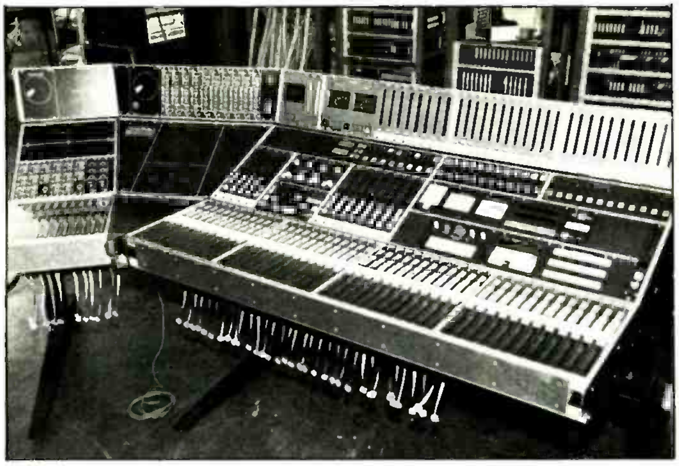
AUDIO & DESIGN/CALREC'S VIRTUAL CONSOLE DESIGN PROPOSALS

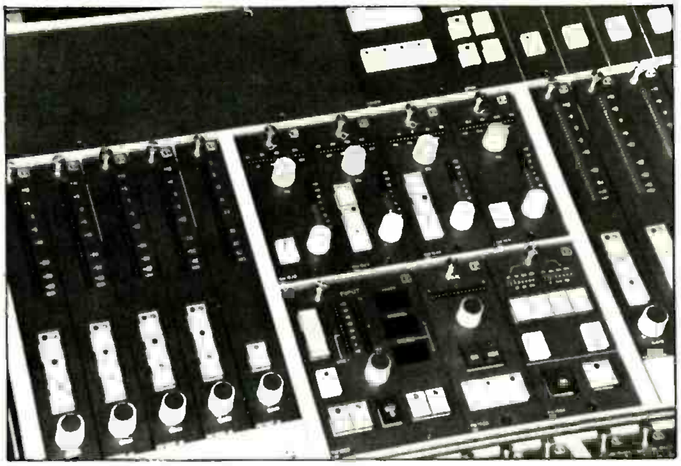
Calrec Audio, Ltd. is currently completing an 88-channel, digitally-controlled analog console under contract to Thames Television, London. Delivery is scheduled for the end of 1985. The project entails a single-control surface to be installed in one of three adjacent studios. The unit incorporates 84 faders and two assignable panels; centrally-controlled, assignable features include channel input, equalizer, and auxiliary functions, as well as routing assignments. The system provides for "floating" channels, which may be addressed by control surfaces to be installed in the other two studios of the complex at a later date.

Calrec plans to offer an assignable broadcast production console in 1986. The company projects that the basic unit will feature 48 channels or less (expandable to 96), with eight stereo groups, four stereo outputs, eight auxiliaries, and 24 recording groups.

Two assignable panels are planned, with controls for individual channel or group input levels, filters, equalizers, pan, and auxiliaries. Routing, memory and master status will also be centrally-controlled. Local controls will be restricted to fader, mute, solo, PFL, and VCA selection.

Automation for the console will allow for a total of three "snapshots" of the full console to be accessible instantly from RAM memory, with a maximum of 30 stored snapshots per disk. Faders will be servo-driven. □□□





The Audio & Design/Calrec 88-channel digitally-controlled analog console scheduled for delivery to Thames Television, London, England, by late-1985. The central control panel shown in close-up *right* features two assignable sections for EQ, pan, filters, and aux sends, plus assignment functions.

gation of the behavior of the bus when operating under the control of bus controllers from different manufacturing sources, under conditions of deliberately induced data errors, and with bus lengths up to 1.2 kilometers [¾-mile]."

Accordingly, the EBU Technical Committee has approved, in principle, the detailed specifications of control messages that will permit full implementation of the standard for VTR systems. SMPTE is still examining the standard, however, and plans final practical testing of a VTR dialect at a further joint test to be held in Redwood City, CA, in November of this year.

Formal approval of the standard will have considerable relevance to the design of virtual consoles and their companion automation systems. Teleproduction studios clearly represent a major potential market for console manufacturers, and the ability to offer integrated machine control can be a substantial selling point. Further, given that the lines separating "traditional" music recording studios from video post and film scoring facilities are constantly being eroded, equipment conforming to the standard will doubtless be appearing with increasing frequency in recording studios. That market will have substantial interest in the ability to trigger effects rolls, for example, from a totally-automated console. While this can be done presently with any of the various simple events-control schemes (relay closures, opto-isolators or open collectors, and so on), such circuitry offers only the most rudimentary transport control: cueing must still be done manually, which takes time.

Another interconnection protocol that some console manufacturers are considering is the Musical Instrument Digital Interface (MIDI). The advantages of such support are not trivial, particularly — but not exclusively — in music studios. Certainly, it would be interesting to envision a console automation system that might grow into musical-instrument recording and control. The potential for automating not only the console but also the entire control room is suggested, since some signal-processor manufacturers are already supporting MIDI control, and others are expressing interest in doing so. The ability to automate changes in delay or reverberation characteristics with video-frame accuracy — linked to the automation of not only faders and mutes, but also routing, sends, equalization, pan and dynamics processing — could open up new realms of creativity in both music and post-production for visual media.

Through the Looking Glass

"In a digitally-controlled analog console," George Massenburg reminds us, "the signal path is an abstraction rather than a physical set of links." In other words, we are no longer directly connected with the signal path: computer "intelligence" stands between, interpreting our communications and, in turn, commanding the processing circuitry. The virtual console presents a mediated experience.

A mediator can be extremely useful, as everyone who has worked with an agent or lawyer knows — provided that you can communicate with them, and that they accurately and effectively represent *you*. If it takes more time than it's worth to explain what you need — and how to get it — then you may as well just go and get it yourself. Similarly, a console that is difficult or confusing to operate won't be used, even if it offers excellent sound quality.

But not only does a good mediator not stand in your way, it also helps you grow. By the same token, a well-designed virtual console — which is to say one that is easily operated, highly flexible, and great sounding — will open new doors. That is the promise of this fledgling technology. The extent to which that promise is fulfilled will be determined by how well we answer the myriad questions that now confront us. ■■■

THE FUTURE OF THE VIRTUAL CONSOLE

Feedback from *R-e/p* Readers

As the world's leading operational pro-audio magazine, *R-e/p* prides itself of keeping its readers appraised of innovative developments in recording and production technology and systems. Without doubt, one of the major talking points throughout the industry is the potential offered by virtual or assignable consoles — either utilizing digital control of analog electronics, or total digital signal processing.

But what do you, the potential users, feel about the kind of technology that will be finding its way into recording and production studios over the next two to five years? *R-e/p* would be extremely interested in hearing from any reader that has a constructive point of view regarding the transition from working with conventional analog consoles, which offer duplicate features to enable simultaneous access to every control element — and with which we are the most familiar — to the potential offered by assignable controls, and the possibility of frame-accurate recall of the entire control surface for enhanced dynamic automation.

We will gather together your comments for possible inclusion in a follow-up article; please include a day-time telephone number so that we can contact you for additional, specific information.

Mail your letters, c/o the Editor, to the address on the Contents page of this issue. Alternatively, we can be reached via IMC EMail to REP-US, or via FAX to (213) 469-0513. □□□

The Directory

R-e/p's Product Listing of
DYNAMICS CONTROL, NOISE REDUCTION AND EFFECTS PROCESSORS

Coming in the next issue: Graphic and Parametric Equalizers

ANT TELECOMMUNICATIONS
U.S. Distributors: Solway, Inc.
P.O. Box 7647
Hollywood, FL 33081
Phone: (305) 962-8650

Model C4-DM/-M122/-M232

Channels: One, two, and two, respectively.
Effects Type(s): Noise reduction system.
Dynamics Parameters: Four-band; linear slope; peak indicators.
Operational Controls: N/A.

Selected Standard Features: -DM: replaces Dolby CAT. 22; 30 dB SNR improvement; no level alignments; no pumping or breathing. -M122 is a two-channel playback unit. -M232 is a two-channel ART or VTR unit.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: Less than 0.1%.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: -DM: $650; -M122: $2,030; -M232: $2,435

Model C4E/ESF-8/ESF-16/ESF-24

Channels: Two, 16, 32, and 48, respectively.
Effects Type(s): Noise reduction system.
Dynamics Parameters: Four-band; linear slope; peak indicators.
Operational Controls: N/A.
Selected Standard Features: Single-channel card for OEM use; multitrack eight-, 16-, 24-channel system, respectively; all offering 30 dB SNR improvement; no level alignments; no pumping or breathing.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: Less than 0.1%.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: C4E: $850; ESF-8: $9,640; ESF-16: $17,350; ESF-24: $25,050

For additional information circle #190

APHEX SYSTEMS, LTD.
13340 Saticoy Street
North Hollywood, CA 91605
Phone: (818) 765-2212

Aural Exciter Type-C

Inputs: Two.
Outputs: Two.
Effects Type(s): Psycho-acoustic audio enhancer.
Dynamics Parameters: N/A.
Operational Controls: Drive; tune; mix; in/out.
Selected Standard Features: Uses new Aphex MAX IC (Monolithic Aural Exciter).
Frequency Response (input/output): 10 Hz to 50 kHz, +0/-0.5 dB.
Distortion: THD less than 0.02%; IMD 0.1%.
S/N Ratio (input/output): Better than -100 dBv.
Pro-User Price Range: $295

Compellor

Inputs: Two.
Outputs: Two.
Effects Type(s): Compressor, levelor, peak limiter.
Dynamics Parameters: N/A.
Operational Controls: Input; process balance; output; silence; gate threshold; stereo enhance.
Selected Standard Features: Full automatic control of all operating parameters; program dependent.
Frequency Response (input/output): 5 Hz to 65 kHz.
Distortion: THD less than 0.01% at 20 dB compression.
S/N Ratio (input/output): Better than -95 dBm.
Pro-User Price Range: $1,195

Model CX-1

Inputs: One.
Outputs: One.
Effects Type(s): Compressor, expander module.
Dynamics Parameters: N/A.
Operational Controls: Display select; compression in/out; threshold and release time; expansion in/out; threshold and delay time in/out control defeat; input gain; expansion depth.
Selected Standard Features: Modular construction; expansion and compression in one package.
Frequency Response (input/output): 5 Hz to 50 kHz.
Distortion: THD less than 0.1%.
S/N Ratio (input/output): Better than -105 dBv.
Pro-User Price Range: $449

For additional information circle #191

ASHLY AUDIO
100 Fernwood Ave.
Rochester, NY 14621
(716) 544-5191

Model CL 52

Inputs: Two.
Outputs: Two.
Effects Type(s): Dual-channel compressor, limiter.
Dynamics Parameters: N/A.
Operational Controls: Gain; ratio; attack; release; output.
Selected Standard Features: Ten-level meters on each channel indicate gain reduction and output level.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion (input/output): Less than 0.05%.
S/N Ratio: Better than -90 dBv.
Pro-User Price Range: $659

Model SG33

Inputs: Two.
Outputs: Two.
Effects Type(s): Stereo noise gate.
Dynamics Parameters: N/A.
Operational Controls: Threshold; attack; hold; fade; floor.
Selected Standard Features: Key input; stereo "tie" point.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion (input/output): Less than 0.05%.
S/N Ratio: Better than -90 dBv.
Pro-User Price Range: $429

For additional information circle #192

AUDIO+DESIGN/CALREC
P.O. Box 786
Bremerton, WA 98310
Phone: (206) 275-5009

Filmex

Inputs: One.
Outputs: One.
Effects Type(s): Four-band, single-ended noise reduction system.
Dynamics Parameters: Four-band, single-ended noise reducing expansion with up to 60-dB control range.
Operational Controls: Ratio; threshold; release and range in each frequency band.
Selected Standard Features: Signal split into four bands; 12 dB per octave linear filter.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-0.5 dB.
Distortion: THD less than 0.01%.
S/N Ratio (input/output): Better than -90 dBm.
Pro-User Price Range: $2,660 per channel

Scamp S30

Inputs: One.
Outputs: One.
Effects Type(s): Noise reducing expander/gate.
Dynamics Parameters: Ratios 1:1.2 (soft); to 1:20 (gate).
Operational Controls: Attack; release; ratio; threshold; range; gate hold; side chain; pre-demphasis; in/out; key input; computer control note input.
Selected Standard Features: Hold facility; side chain equalizer; infinitely variable ratio; 60 dB control range.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-0.5 dB.
Distortion: 0.03%.
S/N Ratio (input/output): Better than -100 dB, reference to 80 dBm.
Pro-User Price Range: $445

Model F601

Inputs: Two.
Outputs: Two.
Effects Type(s): Fast limiter.
Dynamics Parameters: Peak limiter, clipper.
Operational Controls: Make-up gain; threshold; attack; release; clip in/out.
Selected Standard Features: Designed to protect digital PCM inputs; 100 dB dynamic range; dual mono/stereo.

Frequency Response (input/output): 20 Hz to 25 kHz, +0/-0.5 dB.
Distortion: 0.08% at 1 kHz, with 6 dB gain reduction.
S/N Ratio (input/output): Better than -90 dBm.
Pro-User Price Range: $1,490

Easy Rider

Inputs: Two.
Outputs: Two.
Effects Type(s): Dual-channel compressor, limiter.
Dynamics Parameters: Ratios variable from 1:1 to 20:1; attack time 500 microseconds to 5 milliseconds; release time 5 milliseconds to 4 seconds.
Operational Controls: Input level; output level (preset); ratio; attack time; release time; stereo couple.
Selected Standard Features: Thresholds automatically change with ratio; dynamic attack changes with level; 25 dB gain control range; 34 dB make-up gain; 1.5 dB stereo match over 20 dB of gain change.
Frequency Response (input/output): 20 Hz to 25 kHz, +0/-1 dB, at limit threshold reference of gain change.
Distortion: 0.15% at 1 kHz, with 12 dBm maximum limit level.
S/N Ratio (input/output): Better than -80 dB at 12 dBm.
Pro-User Price Range: $690

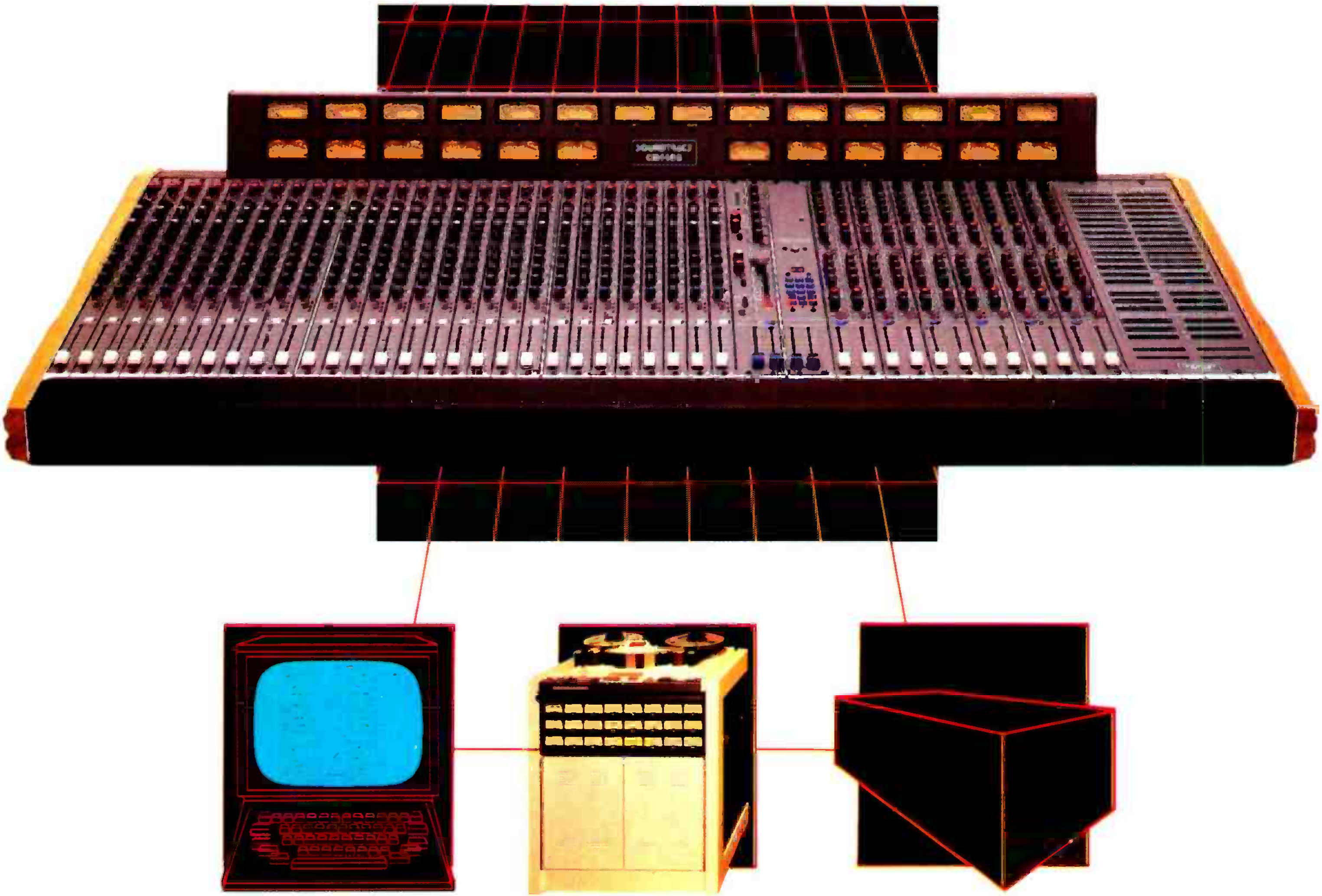
Express

Inputs: Two.
Outputs: Two.
Effects Type(s): Dual channel compressor, limiter, expander.
EQ Ranges: N/A.
Dynamics Parameters: Ratios switched from 1.5 to 20:1; attack time 500 microseconds to 5 milliseconds; release time 25 milliseconds to 3 seconds.
Operational Controls: Input level; output level; ratio; attack time; release time; expander threshold (preset) RMS or peak sensing; meter select.
Selected Standard Features: 25 dB gain control range; 1.0 dB stereo match over 20 dB of gain change.
Frequency Response (input/output): 20 Hz to 25 kHz, +0/-1 dB, at limit threshold reference 1 kHz.
Distortion: 0.15% at 1 kHz, with 12 dBm maximum limit level.
S/N Ratio (input/output): Better than -80 dB at 12 dBm.
Pro-User Price Range: $690

Scamp S31

Inputs: One.
Outputs: One.
Effects Type(s): Feed-forward compressor, limiter.
Dynamics Parameters: Ratios 1:1 to 20:1; threshold +12 dBm to -50 dBm; logarithmic/linear release.
Operational Controls: Make-up gain; compression ratio; compression threshold; limit threshold; release; attack; system in/out; side chain; computer control mute input.

70% OF?



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Attn: Al Lyons 714-592-2848

Allied Audio
4417 Bienville Ave.
New Orleans, LA 70119
Attn: Steve Chapman 504-488-1455

Studio-Sonics Corp
1165 Tower Rd.
Schaumburg, Ill 60195
Attn: Jim Stemke 312-843 7400

Hy James Enterprises
24166 Haggerty Rd
Farmington Hill, Michigan 48018
Attn: Henry Root 313-471 0027

Studio Services Inc.
P.O. Box 2338
Dallas, Tx 75221-2338
Attn: Ron Lagerlof 214-828-4075

Dealer list and brochure from: **Soundtracs Inc.** 745 One Hundred and Ninth Street, Arlington, Texas, 76011. Tel: (817) 460 5519
MCI Intertek Inc 745 One Hundred and Ninth Street, Arlington, Texas, 76011. Tel: (817) 469 1600
In Canada: **Omnimedia Corporation Ltd** 9653 Côte de Liesse, Dorval, Quebec H9P 1A3. (514) 636 9971

For additional information circle #260

Selected Standard Features: Zero to 60 dB control range; separate peak limiter and compressor threshold.
Frequency Response (input/output): 20 Hz to 25 kHz, +0/-0.5 dB, at limit threshold reference 1 kHz.
Distortion: 0.05% at 1 kHz, with 12 dBm maximum limit level.
S/N Ratio (input/output): Better than -100 dB at +8 dBm.
Pro-User Price Range: $445

F760X

Inputs: Two.
Outputs: Two.
Effects Type(s): Compressor, limiter, expander, gate.
Dynamics Parameters: Ratios 1:1 to 20:1; threshold +14 dBm to -20 dBm.
Operational Controls: Input/output; limit in/out; compressor attack; release; threshold release; expander/gate attack; release; threshold; range; system in/out.
Selected Standard Features: Only one VCA used for all four separate functions of compression, limiting, expansion, and gating.
Frequency Response (input/output): 30 Hz to 20 kHz, +0/-0.5 dB.
Distortion: 0.1%.
S/N Ratio (input/output): Better than -89 dB.
Pro-User Price Range: $1,890

F769X

Inputs: Two.
Outputs: Two.
Effects Type(s): Integral compressor, limiter, expander, gate, plus side-chain parametric equalizer.
EQ Ranges: 40 Hz to 1.4 kHz; 80 Hz to 1.6 kHz; 400 Hz to 14 kHz; 800 Hz to 16 kHz.
Dynamics Parameters: Ratios 1:1 to 20:1; threshold +14 dBm to -20 dBm.
Operational Controls: Input/output; peak limit in/out; compressor attack, release threshold, ratio; expander/gate in/out, attack, release, threshold, range; EQ frequency boost, cut, input gain.
Selected Standard Features: Compressor and EQ section may be used together or separately; equalizer may also be routed pre- or post-compressor or to side chain.
Frequency Response (input/output): 30 Hz to 20 kHz, +0/-0.5 dB.
Distortion: 0.1%.
S/N Ratio (input/output): Better than -89 dB.
Pro-User Price Range: $1,775

Compex 2

Inputs: One.
Outputs: One.
Effects Type(s): Compressor, limiter, expander, gate.
Dynamics Parameters: Ratios 1:1 to 20:1; threshold down to -60 dB.
Operational Controls: Make-up gain; system in/out; compressor ratio; threshold; release and attack; limiter in/out and threshold; expander/gate ratio; threshold; range; hold; attack.
Selected Standard Features: Choice of logarithmic or linear release; infinitely variable expander from soft 1:1.2 (hard gate); side chain access.
Frequency Response (input/output): 20 Hz to 25 kHz, +0/-1 dB.
Distortion: Better than 0.05%, 0 Hz to 100 kHz.
S/N Ratio (input/output): Better than -100 dB, reference to 8 dBm.
Pro-User Price Range: $990
For additional information circle #193

BIAMP SYSTEMS, INC.
P.O. Box 2160
Portland, OR 92708-2160
Phone: (503) 641-7287

Model LG-2/-4

Effects Type(s): Two-channel or four-channel (respectively) limiter, compressor, noise gate.
Dynamics Parameters: N/A.

Operational Controls: Front panel threshold control; individual release time; switchable selection for limiting or noise gate.
Selected Standard Features: Individual compression-noise indicators; balanced XLRs; unbalanced inputs and outputs; series patchable.
Frequency Response (input/output): 20 Hz to 25 kHz, ±0.5 dB.
Distortion: THD less than 0.1%, at +15 dBv output.
S/N Ratio (input/output): Maximum threshold is better than -103 dB, A weighted.
Pro-User Price Range: LG-2: $349; LG-4: $599
For additional information circle #194

BROOKE SIREN SYSTEMS, INC.
U.S. Distributor: Klark Teknik
262a Eastern Parkway
Farmingdale, NY 11735
Phone: (516) 249-3660

Model DPR402

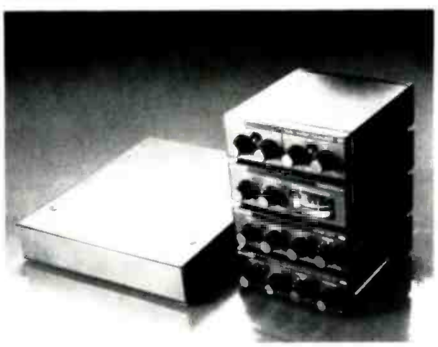
Inputs: Two.
Outputs: Two.
Effects Type(s): Compressor, expander, peak limiting.
Dynamics Parameters: N/A.
Operational Controls: Ratio, attack, release, and threshold; output level; tuneable filter; gain reduction.

Selected Standard Features: Input/output metering; gain reduction; below threshold; bypass switch; stereo to reconfigure either section as expander or compressor; frequency dependent.
Frequency Response (input/output): 25 Hz to 20 kHz, ±1 dB.
Distortion: Less than 0.03%.
S/N Ratio (input/output): Better than -106 dBm rated output.
Pro-User Price Range: $995
For additional information circle #195

CONNECTRONICS CORP.
652 Glenbrook Road
Stamford, CT 06906
Phone: (800) 322-2537

Accessit Compressor

Inputs: One.
Outputs: One.
Effects Type(s): Compressor.
Dynamics Parameters: Ratio 6:1.
Operational Controls: Input release from 0.1 to 2 seconds.



Selected Standard Features: Dynamic range reduction meter.
Frequency Response (input/output): N/A.
Distortion: Less than 0.01%.
S/N Ratio (input/output): Better than -65 dBm.
Pro-User Price Range: $149

Model RBS2

Inputs: One.
Outputs: One.
Effects Type(s): Noise gate.
Dynamics Parameters: -60 dBm to 200 dBm; attack: 0.500 microseconds; release: 100 milliseconds to 10 seconds.
Selected Standard Features: Input for external signal to trip gate; operation in mike or line input.
Frequency Response (input/output): N/A.
Distortion: THD less than 0.1%; IMD 0.1%.
S/N Ratio (input/output): Better than -75 dBm.
Pro-User Price Range: $149
For additional information circle #196

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dbx
71 Chapel Street
Newton, MA 02195
Phone: (617) 964-3210

Models 941A/942A

Channels: Two-channel encoder (941A), and decoder (924A) type II noise reduction system.
Dynamics Parameters: +40 dB effective noise reduction.
Operational Controls: Separate level adjusters for both channels.
Selected Standard Features: Up to 40 dB increase in dynamic range; modular construction; differential input; compatible with all type II NR systems.
Frequency Response (input/output): 40 Hz to 20 kHz, ±0.5 dB.
Distortion (input/output): THD less than 0.1% from 100 Hz to 20 kHz.
S/N Ratio (input/output): Better than -110 dB dynamic range.
Pro-User Price Range: 941A: $269; 942A: $279

Models 140A/180A

Channels: Two-channel type I (140A), and type II (180A) NR systems.
Dynamics Parameters: +40 dB effective noise reduction.
Operational Controls: Separate channel level adjusters; balanced inputs; unbalanced outputs; provision for output balancing tranformers.
Selected Standard Features: Up to 40 dB increase in dynamic range.
Frequency Response (input/output): 30 Hz to 20 kHz, ±0.5 dB; -1 dB for 20 Hz.
Distortion (input/output): THD less than 0.1% from 100 Hz to 20 kHz.
S/N Ratio (input/output): Better than -115 dB dynamic range.
Pro-User Price Range: 140A: $589; 180A: $589

Model 150

Channels: Two-channel type I NR system.
Dynamics Parameters: +40 dB effective noise reduction.
Operational Controls: Simultaneous encode and decode of noise reduction.
Selected Standard Features: Gold-plated RCA jacks; compatible with all type I units; hard-wire bypass.
Frequency Response (input/output): 30 Hz to 20 kHz, +0.5 dB; -1 dB at 20 Hz.
Distortion (input/output): THD less than 0.1% from 100 Hz to 20 kHz.
S/N Ratio (input/output): Better than -105 dB dynamic range.
Pro-User Price Range: $249

Model 911

Channels: Single-channel type I NR system.
Dynamics Parameters: +40 dB effective noise reduction.
Operational Controls: Separate record/play level adjusters; NR in/out.
Selected Standard Features: Up to 40 dB increase in dynamic range; compatible with all type I units; modular construction; differential input.
Frequency Response (input/output): 30 Hz to 20 kHz, ±0.5 dB; -1 dB for 20 Hz.
Distortion (input/output): THD less than 0.1% from 100 Hz to 20 kHz.
S/N Ratio (input/output): Better than -110 dB dynamic range.
Pro-User Price Range: $310

Model 903

Effects Type(s): Single-channel Overeasy compressor, limiter.
Dynamics Parameters: Compression ratio varies from 1:1 to infinity:1 "and beyond to dynamic inversion."
Operational Controls: Threshold; compression ratio; output gain.
Selected Standard Features: Modular; slaves to 907 for stereo.
Frequency Response (input/output): 30 Hz to 20 kHz, ±1 dB.
Distortion (input/output): 2nd harmonic: 0.07%; 3rd harmonic: 0.2%.
S/N Ratio (input/output): EIN better than -85 dBm; maximum output level +20 dB.
Pro-User Price Range: $359

Model 166

Effects Type(s): Dual-channel compressor, noise gate.
Dynamics Parameters: Compression ratio varies from 1:1 to infinity:1.
Operational Controls: Threshold; peak-stop level; output gain; noise-gate threshold.
Selected Standard Features: Unbalanced quarter-inch jacks; side-chain input/monitor.
Frequency Response (input/output): 20 Hz to 20

kHz, ±0.5 dB.
Distortion (input/output): THD less that 0.1% at max compression, 0 dBm.
S/N Ratio (input/output): Output noise better than -85 dBv A weighted.
Pro-User Price Range: $549

Models 160X/163X/165A
Effects Type(s): Single-channel Overeasy compressor, limiters.
Dynamics Parameters: Compression ratio varies from 1:1 to infinity:1; dynamic inversion mode (160X).
Operational Controls: Threshold (-40 to +10 dB); attack/release controls, peak-stop, "soft clipper" and stereo-strappable (165A).
Selected Standard Features: Quarter-inch balanced in/outs; terminal-strip connectors (165A).
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-1 dB (160X and 165A); 20 Hz to 20 kHz, ±1 dB (163X).
Distortion (input/output): 2nd harmonic 0.05%, and 3rd 0.07% (160X and 165A); THD less that 0.2% (163X).
S/N Ratio (input/output): Output noise better than -89 dBm, 20 Hz to 20 kHz (160X); EIN -85 dBv unweighted (163X); EIN -90 dBm unweighted (165A).
Pro-User Price Range: 160X: $429; 163X: $149; 165A: $699

Model 904
Effects Type(s): Single-channel modular noise gate with Overeasy expansion.
Dynamics Parameters: Attenuation 0 to 60 dB; threshold -40 to +10 dB; expansion ratio 1.5:1 to 5:1; attack times 500 to 2.5 dB/mS; release 2.5 dB/mS to 22 dB/second.
Operational Controls: Limit; ratio; threshold; attack; release.
Selected Standard Features: Modular; balanced in/out; LED column meter.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-1 dB.
Distortion (input/output): THD less that 0.02% at 1 kHz.
S/N Ratio (input/output): EIN -82 dBm 20 Hz to 20 kHz, unweighted.
Pro-User Price Range: N/A.
For additional information circle #222

DOD ELECTRONICS

5639 South Riley Lane
Salt Lake City, UT 6906
Phone: (801) 268-8400

Model R-825 and MT-828
Inputs: One and two, respectively.
Outputs: One and two, respectively.
Effects Type(s): Compressor, limiter and stereo compressor limiter, respectively.
Dynamics Parameters: Gain reduction; 828 includes noise gate.
Operational Controls: Input level; output level; compression ratio; attack time; switchable side chain; de-essing circuitry. 828: gate; threshold; ratio; attack time; release time; input level; output level; compress in/out; link in/out.
Selected Standard Features: Simultaneous de-essing circuitry; adjustable attack and release time; switchable side chain; LED bargraph. 828: ratio variable from 1:1 to infinity:1; independent noise on each channel; gain reduction LED side chain connections.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.5 dB.
Distortion: THD 0.02% at 0 dB gain reduction.
S/N Ratio (input/output): Better than -85 dB.
Pro-User Price Range: 825: $249.95; 828: $299.95
For additional information circle #197

DOLBY LABORATORIES, INC.

731 Sansome Street
San Francisco, CA 94111
Phone: (415) 392-0300

Models 360/361/362/372
Channels: 360 and 361: one; 362 and 372: two.
Effects Type(s): Dolby A-type noise reduction system.
Operational Controls: 360: record/play, NR in/out, Dolby tone; 361/362/372: monitor signal, check tape.
Selected Standard Features: 360/361: three XLR input/output connectors; 362: quarter-inch socket; 372: seven-pin Tuchel/Binder; all have level setting meters.
Frequency Response (input/output): 30 Hz to 20 kHz, 1± dB.
Distortion: Less than 0.1%, +4 dBm.
S/N Ratio (input/output): N/A.
Pro-User Price Range: 360: $1,100; 361: $1,350; 362: $1,850; 372: $2,500

Models SP8/16/24/32/48
Channels: Eight, 16, 24, 32 and 48, respectively.
Effects Type(s): Dolby A noise reduction.
Operational Controls: Individual track controls: NR standby, bypass, calibration, monitor, check tape.
Selected Standard Features: LED indicators; balanced inputs; switch-selectable outputs.
Frequency Response (input/output): 20 Hz to 20 kHz, ±1 dB.
Distortion: Less than 0.004%, at 1 kHz.
S/N Ratio (input/output): Better than -80 dB, unweighted.
Pro-User Price Range: 8: $9,400; 16: $15,400; 24: $21,400; 32: $29,100; 48: $41,100

Models 221B/234/380/226
Channels: Two; 226: three.
Effects Type(s): Dolby A-type noise reduction; module-type system.
Operational Controls: Two switches control NR in/out for tracks #1 and #2 independently; a third activates Dolby tone oscillator for calibration; 226: provides output level controls; and mike/line time-code selector.
Selected Standard Features: 221B: NR for Sony BVH-1100 and -1000; 234: NR for Sony BVH-2000; 380: NR for Ampex VPR-3 and RCA TR-900; 226: Ampex VPR-2 and 2B.
Frequency Response (input/output): 20 Hz to 20 kHz, ±1 dB.
Distortion: Less than 0.004%.
S/N Ratio (input/output): N/A.
Pro-User Price Range: 221B: $2,000; 234: $2,100; 380: $2,000; 226: $2,300

Models 43/330/330L
Channels: 43: one; 330/330L: two.
Effects Type(s): Dolby B-type noise reduction; module-type system.
Operational Controls: Remote control; Dolby tone; NR in/out.
Selected Standard Features: Balanced transformer; five-pin XLR connector; lowpass filter; metering.
Frequency Response (input/output): 30 Hz to 15 kHz, +1 dB.
Distortion: Less than 0.1%.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: 43: $825; 330: $1,900; 330L: $2,050
For additional information circle #198

DRAWMER

U.S. Distributor: Harris Sound, Inc.
6640 Sunset Boulevard Suite #110
Hollywood, CA 90028
Phone: (213) 469-3500

Effects Type(s): Dual-audio gate system.
Operational Controls: Two-channel operation with separate threshold, attack, hold, decay, range, low- and high-key filter controls.
Selected Standard Features: Frequency conscious keying; two-stage release control; stereo link; gating; and ducking.
Frequency Response (input/output): 25 Hz to 40 kHz, +1 dB.
Distortion: Variable between 0.03% and 0.05%.
S/N Ratio (input/output): Better than -90 dBm.
Pro-User Price Range: $595

1960 Tube Compressor
Effects Type(s): Dual-channel vacuum-tube compressor.
Dynamics Parameters: "Soft knee" ratio 1:5 to 20:1.
Operational Controls: Two-channel operation with threshold, attack, release output level, mike gain, auxiliary gain and bass, and treble controls.
Selected Standard Features: Balanced outputs; mike input with phantom power available; auxiliary input with equalizer; stereo link.
Frequency Response (input/output): 22 Hz to 22 kHz.
Distortion: N/A.
S/N Ratio (input/output): Better than -88 dBm.
Pro-User Price Range: $1,395
For additional information circle #199

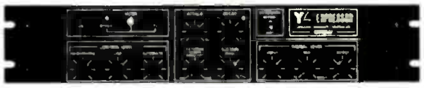
DREW ENGINEERING CO.

35 Indiana Street
Rochester, NY 14609
Phone: (716) 544-3337

Y-Expressor D58 and Y-Processor D55
Inputs: Three and four, respectively.

The Directory

Outputs: Four and two, respectively.
Effects Type(s): Dynamic Sound Shaper and envelope controller, respectively.
Dynamics Parameters: D58: detector and processor calculates envelope shapes.
Operational Controls: D58: mode; control level: microphone, line, automatic. Signal level: input-output; mix; expression; attack; decay; sustain pedal. D55: mode: attack; sustain; control level: automatic.



Selected Standard Features: Real-time envelope control; sound transformation; sound combining; synchronization; rythmic effects; voice processing; foot-pedal control; MIDI CV optional.
Frequency Response (input/output): 5 Hz to 50 kHz, -1 dB. D55: 10 Hz to 30 kHz.
Distortion: THD less than 0.02%; D55: 0.05% typical.
S/N Ratio (input/output): Better than -94 dB. D55: -84 dB.
Pro-User Price Range: N/A.

UEX-4

Inputs: 12.
Outputs: Four.
Effects Type(s): Four-channel processor.
EQ Ranges: N/A.
Dynamics Parameters: N/A.
Operational Controls: N/A.
Selected Standard Features: "Integrated system per customer specifications."
Frequency Response (input/output): 5 Hz to 50 kHz, -1 dB.
Distortion: THD less than 0.02%.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: N/A.
For additional information circle #200

EVENTIDE, INC.
One Alsan Way
Little Ferry, NJ 07643
Phone: (201) 641-1200

Model 2830 Omnipressor

Inputs: One.
Outputs: One.
Effects Type(s): Compression, expansion, gain linearity.
Dynamics Parameters: 1:1 to infinity:1 compression; 1:1 to 10:1 expansion; ±1 dB for 60-dB change in input level.
Operational Controls: Threshold, attack time; release time; attenuation limit; gain limit; dynamic reversal effects.
Selected Standard Features: Metering system employs a logarithmic amplifier to generate information on input, output and gain.
Frequency Response (input/output): 15 Hz to 20 kHz, +0/-1 dB.
Distortion: Variable between 0.02% and 0.5%.
S/N Ratio (input/output): Output noise level better than -90 dBm, at unity gain.
Pro-User Price Range: $700
For additional information circle #201

FOSTEX 3070
15431 Blackburn Avenue
Norwalk, CA 90650
Phone: (213) 921-1112

Model 3070

Effects Type(s): Two-channel limiter, compressor, noise gate.
Dynamics Parameters: 32-dB maximum limit; compression ratio of 1:1 to infinity:1; release time 20 milliseconds to two seconds.
Operational Controls: For each channel input: output, attack release, ratio, gate sensitivity, in/out switch, and stereo interlock for each channel.
Selected Standard Features: Gate sensitivity indicator; five-segment LED display indicating gain and reduction; reduction created by pulse width modulation; detector loop access for frequency gain-reduction.
Frequency Response (input/output): N/A.

Distortion: THD less than 0.03% below threshold.
S/N Ratio (input/output): Better than -80 dB, unweighted.
Pro-User Price Range: $400
For additional information circle #202

FURMAN SOUND, INC.
30 Rich Street
Greenbrae, CA 94904
Phone: (415) 927-1225

Model LC-3/-X

Inputs: One.
Outputs: One.
Effects Type(s): Limiter, compressor; -X features built-in expander.
EQ Ranges: N/A.
Dynamics Parameters: Expander from 1:1 to 5:1; compression 1:1 to 20:1; limiter: 30:1.
Operational Controls: Attack time of 100 microseconds; release time 50 microseconds to 1.1 seconds; compression ratio of 2:1 to 50:1. -X: attack; release; expand threshold and ratio; compression threshold and ratio; output gain and limit.
Selected Standard Features: De-ess and side chain modes; gain reduction meter. -X: 10-segment bypass switch.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-0.5 dB.
Distortion: Less than 0.012%, THD at 0 dBv.
S/N Ratio (input/output): Better than -102 dB; -3: better than -102 dB.
Pro-User Price Range: $342; -X: $449

Model QN-4

Effects Type(s): Quad noise gate.
Dynamics Parameters: Release rate 0.005 to 5 seconds; threshold infinity to +10 dBv.
Operational Controls: Threshold (×4); fade time (release time ×4).
Selected Standard Features: LED indicator; unit modulation for low noise and distortion.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.5 dB.
Distortion: Less than 0.01% THD at 0 dBv.
S/N Ratio (input/output): Better than -111 dB.
Pro-User Price Range: $395
For additional information circle #203

GOTHAM AUDIO CORP.
741 Washington Street
New York, NY 10014
Phone: (212) 741-7411

Neumann U473A

Inputs: One.
Outputs: One.
Effects Type(s): Compressor, limiter, expander.
Operational Controls: Compressor: ratio, release with auto attack, output, gain, compressor gain. Expander: threshold, recovery time; stopped controls for repeatable settings.
Selected Standard Features: LED gain reduction meter; limiter and expander LED indication; bypass and expander in/out switches; stereo control link.
Frequency Response (input/output): 40 Hz to 15 kHz, ±0.3 dB.
Distortion: Less than 0.03%.
S/N Ratio (input/output): Better than -100 dB, unweighted +6 dB out.
Pro-User Price Range: $945

NTP 179-170

Inputs: Two.
Outputs: Two.
Effects Type(s): Compressor, limiter, expander, gate.
Operational Controls: Gain: 0 to 20 dB with constant output level; hold level is -10 to -50 dB; gate level is 0 to -50 dB.
Selected Standard Features: Automatic release option on all front-panel adjustments; gain/attenuation LED metering with limiter LED; external ganging of limiter/expander control signal.
Frequency Response (input/output): 40 Hz to 15 kHz, ±0.3 dB.
Distortion: Less than 0.01%, +15 dBu out.
S/N Ratio (input/output): Better than -89 dB, unweighted.
Pro-User Price Range: $2,310

EMT 257

Inputs: One.
Outputs: One.
Effects Type(s): Limiter.
Dynamics Parameters: "Soft" limiting with optional pre-emphasis compensation.
Operational Controls: Attack is 50 to 500 microseconds; threshold is -2 to +10 dB; release is 0.25 to 10 seconds and automatic, pre-emphasis compensa-

tion switch.
Selected Standard Features: Gain meter; in/out level adjustment from front panel; automatic release proportional to loudness.
Frequency Response (input/output): 40 Hz to 15 kHz, ±0.3 dB.
Distortion: Less than 0.05%, auto release.
S/N Ratio (input/output): Better than -75 dB, unweighted.
Pro-User Price Range: $1,290

EMT 260

Inputs: One.
Outputs: One.
Effects Type(s): High-frequency filter limiter.
Operational Controls: Attack is 50 to 500 microseconds; treble reduction is -1 to -4 dB; release is 0.25 to 10 seconds and automatic, selectable 60 and 180 microseconds control weighting.
Selected Standard Features: Gain meter; in/out level adjustment from front panel; automatic release proportional to loudness.
Frequency Response (input/output): 40 Hz to 15 kHz, ±0.3 dB.
Distortion: Less than 0.05%, auto release.
S/N Ratio (input/output): Better than -75 dB, unweighted.
Pro-User Price Range: $1,255

EMT 261

Inputs: One.
Outputs: One.
Effects Type(s): Compressor, limiter.
Operational Controls: Gain is 0 to 12 dB; ratio is 2:1 to 20:1, release is 0.25 to 10 seconds and automatic; switchable expander and limiter in/out.
Selected Standard Features: Gain meter; in/out level adjustment from front panel; automatic release proportional to loudness.
Frequency Response (input/output): 40 Hz to 15 kHz, ±0.3 dB.
Distortion: Less than 0.05%, auto release.
S/N Ratio (input/output): Better than -73 dB, unweighted.
Pro-User Price Range: $1,330

266STAP

Inputs: Two.
Outputs: Two.
Effects Type(s): Delay-based transient limiter with adaptive pre-emphasis.
Dynamics Parameters: Limiting without overshoot; pre-emphasis without excessive limiting.
Operational Controls: Input is -20 to +15 dB, output is -20 to +15 dB hard relay bypass; automatic or three second release time; compressor in/out switch; pre-emphasis weighting in/out switch.
Selected Standard Features: LED indication for limiter action; -6 dB nominal level; adaptive pre-emphasis action; channel coupling for stereo.
Frequency Response (input/output): 30 Hz to 15 kHz, +0.3/-0.5 dB.
Distortion: Less than 0.02%, for 60 Hz to 16 kHz.
S/N Ratio (input/output): Better than -92 dB, unweighted +8 dB out.
Pro-User Price Range: $4,565

For additional information circle #204

JBL PROFESSIONAL
8500 Balboa Boulevard
Northridge, CA 91329
Phone: (818) 893-8411

UREI 1178/1176LN

Inputs: Two and one, respectively.
Outputs: As above.
Effects Type(s): Peak limiters.
Dynamics Parameters: 4:1 to 20:1.
Operational Controls: Input, output, attack time; release time; ratio select.
Selected Standard Features: Continuously variable attack and release time; independent of degree of limiting.
Frequency Response (input/output): 20 Hz to 20 kHz, +1 dB.
Distortion: THD less than 0.5%.
S/N Ratio (input/output): Better than -81 dB.
Pro-User Price Range: 1178: $896; 1176: $596

UREI LA-4

Inputs: One.
Outputs: One.
Effects Type(s): Compressor, limiter.
Dynamics Parameters: 2:1 to 20:1 compression ratio.
Operational Controls: Threshold level; ratio, output level; meter level.
Selected Standard Features: RMS responding, standard VU indicator; stereo coupling.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.5 dB.
Distortion: THD less than 0.25%.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: $496

For additional information circle #205

LT SOUND
P.O. Box 338
Stone Mountain, GA 30086
Phone: (404) 493-1258

Model NR-2/-4/-8

Channels: Two, four, and eight, respectively.
Effects Type(s): Tape noise reduction; compatible with dbx type-I.
Dynamics Parameters: 30 dB tape noise reduction with 10 dB headroom improvement.
Operational Controls: Individual in/bypass on each channel as well as record and play calibration for +4 or -10 operation.
Frequency Response (input/output): 20 Hz to 20 kHz, ±1.5 dB.
Distortion (input/output): 0.078%, THD at 1 kHz.
S/N Ratio (input/output): -94 dBv, A weighted; 116 dB dynamic range.
Pro-User Price Range: -2: $249; -4: $475; -8: $795.

SL-2 Stereo Limiter

Effects Type(s): Compressor, limiter; with de-essing and noise gating capability.
Operational Controls: Threshold, release, de-ess amount; expander threshold, gate release, output gain.

Selected Standard Features: Limiting, de-essing, and compression can all be done simultaneously. Stereo or independent tracking with a push of a switch.
Frequency Response (input/output): 20 Hz to 20 kHz +1.5 dB.
Distortion (input/output): Less than 0.06% THD at 15 dB limiting, fastest release.
S/N Ratio: Better than -92 dBv, A weighted.

The Directory

Pro-User Price Range: $295
For additional information circle #206

MISTUBISHI PRO AUDIO GROUP
225 Parkside Drive
San Fernando, CA 91340
Phone: (818) 898-2341

Model CL-22

Inputs: One.
Outputs: One.
Effects Type(s): Compressor, limiter.
Dynamics Parameters: Variable attack 0.002 to 5 milliseconds at 10 dB; release: 100 to 5 seconds at 5 dB.
Operational Controls: Expand: gain: ratio: attack release: CL threshold pots: DS and CL in-switch with indicator LED meter.
Selected Standard Features: Feed-forward VCA design with a range of -40 dB to +26 dB adjusted from 2:1 to 20:1, expansion from -60 to +200 dB.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.25 dB.
Distortion: THD less than 0.05% clout.
S/N Ratio (input/output): Better than -94 dBm, reference +4 dB where 0 is 0.775V.
Pro-User Price Range: $475

Model NSD 120

Inputs: One.
Outputs: One.
Effects Type(s): Noise gate.
Dynamics Parameters: Attack is less than 25 microseconds: adjustable release from 0.03 to 5 seconds.
Operational Controls: Release: threshold: attenuation pots: auxiliary input keying switch: selectable decay.
Selected Standard Features: Unity gain device, 0 to 50 dB attenuation: sensitivity is 31 dBm at 0.03-second release: LED indicator.
Frequency Response (input/output): 30 Hz to 20 kHz, ±0.5 dB.
Distortion: THD less than 0.25%: rated output at threshold.
S/N Ratio (input/output): Better than -90 dBm, reference +4 dB where 0 is 0.775V.
Pro-User Price Range: $295
For additional information circle #207

RUPERT NEVE, INC.
Berkshire Industrial Park
Bethel, CT 06801
Phone: (203) 744-6230

Series 33609

Inputs: One or two.
Outputs: One or two.
Effects Type(s): Compressor, limiter.
Dynamics Parameters: Limiting ratio is 100:1; compression ratios is 1.5:1 to 6:1.
Operational Controls: Limit threshold and recovery: limit in/out: fast/slow attack: compression threshold, recovery and ratio: gain make-up: compression in/out: bypass in/out: quad/stereo link: PPM gain reduction meter(s).
Selected Standard Features: Stereo and four-channel switchable linking: all rotary controls are switches.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.5 dB.
Distortion: 0.2% at 6:1 compression, 27 dB above threshold.
S/N Ratio (input/output): Better than -75 dB at 0 dB with 20 dB gain make up.
Pro-User Price Range: Starting from $1,465
For additional information circle #208

OMNI CRAFT, INC.
P.O. Box 1069
Palatine, IL 60078
Phone: (800) 562-5872

Model GT4A

Inputs: Four.
Outputs: Four.
Effects Type(s): Noise gate.
Operational Controls: Threshold: range: and release.
Selected Standard Features: Key switch for LEDs.
Frequency Response (input/output): 0 Hz to 100 kHz, +1/-0.5 dB.
Distortion: "Not measurable."
S/N Ratio (input/output): As above.
Pro-User Price Range: $395

Model GTX (D) or (K)

Inputs: One.
Outputs: One.
Effects Type(s): Noise gate.
Operational Controls: Switchable high- and low-pass filters: switchable "ducking" circuits.
Selected Standard Features: Designed to fit two self-powered, rack-frame sizes.

Frequency Response (input/output): 10 Hz to 100 kHz, ±0.5 dB.
Distortion: 20 Hz to 20 kHz, 0.002%.
S/N Ratio (input/output): Less than -110 dB, reference +22dBv.
Pro-User Price Range: $275
For additional information circle #209

ORBAN ASSOCIATES, INC.
645 Bryant Street
San Francisco, CA 94107
Phone: (818) 843-7567

Models 412A/414A/422A/424A

Inputs: One, one, two and two, respectively.
Outputs: As above.
Effects Type(s): Compressor, limiter. 424A: gated compressor, limiter de-esser.
Dynamics Parameters: 35 dB gain reduction: 424: 25 dB gain reduction.
Operational Controls: Input/output attenuation: threshold: attack and release time: compression ratio: hard-wire system bypass: 424 has de-esser sensitivity.
Selected Standard Features: Control interaction: output adjustments automatically change to attack time: phone jacks and barrier strip.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.25 dB.
Distortion: Less than 0.04% at 100 Hz: 424 and 424A: less than 0.03% at 1 kHz.
S/N Ratio (input/output): Better than -90 dB, typical.
Pro-User Price Range: 412A: $425: 414A: $799: 424A: $989: 422A (mono): $629
For additional information circle #210

PEARL INTERNATIONAL, INC.
406 Harding Industrial Drive
Nashville, TN 37222-1240
Phone: (615) 833-4477

Model CO-04

Inputs: One.
Outputs: One.
Effects Type(s): Compressor.
Operational Controls: Attack: tone: level: sustain.
Selected Standard Features: Attack and tone adjustments.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): N/A.
Pro-User Price Range: $115

Model SU-19

Inputs: Two.
Outputs: Two.
Effects Type(s): Noise gate.
Operational Controls: Threshold: decay: gate indicator.
Selected Standard Features: Stereo capable.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): N/A.
Pro-User Price Range: $119.50
For additional information circle #211

ROCKTRON CORP.
2146 Avon Industrial Drive
Auburn Heights, MI 48057
Phone: (313) 853-3055

Model HUSH II/IIB/IIC

Channels: One; IIC: two.
Effects Type(s): Single-ended noise reduction system.
Dynamics Parameters: Expander 30 dB; dynamic filter 1.5 kHz to 30 kHz.
Operational Controls: In/out switch; line/instrument switch; expander/threshold control: IIC has

fast/release switch.
Selected Standard Features: 30 dB effective noise reduction program dependent attack release; IIC is stereo model.

Frequency Response (input/output): 30 Hz to 20 kHz, ±0.5 dB.
Distortion: THD less than 0.2%.
S/N Ratio (input/output): Better than -110 dB dynamic range.
Pro-User Price Range: II: $200; IIB: $250; IIC: $330

Model 120A/140A/180A

Channels: Four, eight and 16, respectively.
Effects Type(s): System One encode/decode noise reduction.
Dynamics Parameters: 2:1 compression to 1:2 expansion.
Operational Controls: Record/playback levels; bypass switches.
Selected Standard Features: Two, four, and eight channels (respectively) of simultaneous encode-decode; RCA connectors; 40 dB effective noise reduction.
Frequency Response (input/output): 30 Hz to 20 kHz, ±0.5 dB.
Distortion: THD less than 0.05%.
S/N Ratio (input/output): EIN better than -95 dB, A weighted.
Pro-User Price Range: 120A: $319; 140A: $545; 180A: $950

Model 300/310

Inputs: One.
Outputs: One.
Effects Type(s): Compressor, peak limiter, single-ended noise reduction; and compressor, respectively.
Dynamics Parameters: 25 dB compression and limiting; 30 dB expansion; 310 has 25 dB compression.
Operational Controls: 300: compression, attack, release, limiter threshold, bypass, side chain in/out meter, expander threshold, output gain; 310: input gain switch, compression, output, output in/out switch, compression switch.
Selected Standard Features: Logarithmic compression, simultaneous peak limiting and 30 dB noise reduction, single rack space; 310: 10 LED gain reduction meter, program dependent ratio, attack release.
Frequency Response (input/output): 30 Hz to 20 kHz, +0/-1 dB.
Distortion: 0.1 , maximum.
S/N Ratio (input/output): Better than -115 dB dynamic range; 310: 95 dB dynamic range.
Pro-User Price Range: 300: $429; 310: $239.

Powerplay Deluxe/Basic

Inputs: Two.
Outputs: Seven.
Effects Type(s): Compressor, EQ, distortion, HUSH II, echo, stereo chorus, exciter; Basic features stereo ambient chorus.
Operational Controls: Membrane touch switching; echo: regenerate; ambience: chorus; 2, 4, 6 delay tape; exciter; output gain loop.
Selected Standard Features: Program dependent compressor; automatic Hush II, digitally controlled; fully foot switchable.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): N/A.
Pro-User Price Range: Deluxe: $1,270; Basic: $699

RX-1/-2H

Inputs: Two.
Outputs: Two.
Effects Type(s): Exciter/image; 2H features built-in Hush II single-ended noise reduction.

Dynamics Parameters: -2H: 30 dB expansion, dynamic filter 1.5 kHz to 30 kHz.
Operational Controls: Process level; phase mix; bypass switches.
Selected Standard Features: Process level LED meters; stereo; -2H has 30 dB effective noise reduction and program-dependent attack release.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-0.5 dB.
Distortion: THD less than 0.5%.
S/N Ratio (input/output): 100 and 110 dB dynamic range, respectively.
Pro-User Price Range: RX1: $375; -2H: $550
For additional information circle #213

PHOENIX AUDIO LAB, INC.
P.O. Box 127
Manchester, CT 06040
(203) 649-1199

Loft Model 400B

Effects Type(s): Quad gate, limiter
Dynamics Parameters: Gain reduction 30 dB; attenuation 27 dB.
Operational Controls: Gate; limiter; threshold, attack/release; input/output impedance.
Selected Standard Features: Four independent limiters with noise gates; feed-forward; side-channel processing; front-panel LED indicators; balanced inputs and outputs.
Frequency Response (input/output): N/A.
Distortion (input/output): Less than 0.2%.
S/N Ratio: -95 dBv, A weighted; reference 0 dB
Pro-User Price Range: $602

Loft Model 410

Inputs: Two.
Outputs: Two.
Effects Type(s): Compressor, limiter, expander, gate.
Dynamics Parameters: Expander — attenuation: switch selectable to -3 dB, -15 dB, -40 dB, slope: switch selectable to 1:12, 1:5, 1:20; threshold: -60 dBv to 0 dBv; attack time: 5 milliseconds; release time: 10 milliseconds to 5 seconds; control input: switchable: normal, external "key." Compressor — slope: 1:1 to 00:1; threshold: -10 dBv to +10 dBv; attack time: 2 to 200 milliseconds; release time: 5 milliseconds to 5 seconds. Limiter — attack time: 4 dB per millisecond; release time: 2 dB per millisecond.
Selected Standard Features: Ability to compress, expand, and limit simultaneously.
Frequency Response (input/output): N/A.
Distortion (input/output): Less than 0.03%.
S/N Ratio: -92 dBv, A weighted.

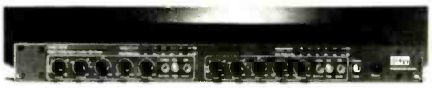
The Directory

Pro-User Price Range: $699
For additional information circle #212

SCV, INC.
414 North Sparks Street
Burbank, CA 91506
Phone: (818) 843-7567

Model NGS-2
Effects Type(s): Stereo noise gate with frequency-shaped gate trigger.
Dynamics Parameters: +20 dBv; HF and LF gate settings.
Operational Controls: Balanced and transformer-less balanced outputs.
Selected Standard Features: -990 dBv noise capacity.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-1 dB.
Distortion: 0.01%.
S/N Ratio (input/output): Better than -84 dB.
Pro-User Price Range: $697

Model SLR-2
Effects Type(s): Stereo limiter, compressor, de-esser.
Dynamics Parameters: 2:1 to 20:1 compression, +20 dBv input.
Operational Controls: Ratio; attack; release; and bypass.



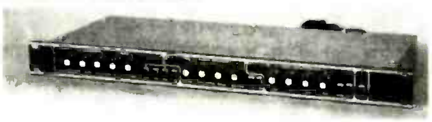
Selected Standard Features: LED meter reading gain reduction on output level.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-0.25 dB.
Distortion: 1 kHz, 10 dBv compression, 0.15%.
S/N Ratio (input/output): Better than -73 dB.
Pro-User Price Range: $695
For additional information circle #214

SHURE BROTHERS, INC.
222 Hartley Avenue
Evanston, IL 60202-3696
Phone: (312) 866-2200

Model SE-30
Inputs: Three.
Outputs: One.
Effects Type(s): Compressor, mixer.
Dynamics Parameters: N/A.
Operational Controls: Response rate; input/output controls; VU/compression meter; compressor defeat; gated hold.
Selected Standard Features: AC or internal batteries.
Frequency Response (input/output): 30 Hz to 20 kHz.
Distortion: THD less than 0.5%.
S/N Ratio (input/output): Better than -70 dB.
Pro-User Price Range: $750
For additional information circle #215

SOUND PERFORMANCE LABORATORY
11 Burlingame Avenue, Suite 5
Burlingame, CA 94010
Phone: (415) 344-8787

Model USM-4
Inputs: One.
Outputs: One.
Effects Type(s): Compressor/limiter, exciter, noise gate.
Dynamics Parameters: Noise gate release time of 6 milliseconds to 3 seconds; attack from 4 microseconds to 7 milliseconds.



Operational Controls: Variable filter control; high, mid and low filter; EXARHY psycho-acoustic processor; noise gate processor; attack; threshold; release.
Selected Standard Features: XLR connectors; LED level indicator.
Frequency Response (input/output): N/A.
Distortion: Less than 0.01%.
S/N Ratio (input/output): Better than -93 dB.
Pro-User Price Range: $995
for additional information circle #216

SPECTRA SONICS
3750 Airport Road
Ogden, UT 84403
Phone: (801) 392-7531

Model 610/601
Inputs: One.
Outputs: One.
Effects Type(s): Compressor, limiter.
Dynamics Parameters: Compression ratio of 1.1:1 to 100:1; attack time is variable 100 nanoseconds to 2 microseconds on limiter; compressor is 100 nanoseconds to 1.2 milliseconds; release time is less than 90 nanoseconds on limiter; compressor is variable from 50 milliseconds to 10 seconds.
Selected Standard Features: Two units allow stereo tracking; 601 modular plug-in card allows daisy-chaining of "unlimited" number.
Frequency Response (input/output): 20 Hz to 20 kHz, +0.1 dB.
Distortion: Less than 0.1%, 30 Hz to 20 kHz full output.
S/N Ratio (input/output): Better than -80 dB, +4 dBm, unweighted.
Pro-User Price Range: 610: $699; 601: $142
For additional information circle #217

SUMMIT AUDIO, INC.
P.O. Box 1678
Los Gatos, CA 95031
(408) 395-2448

Tube Compressor Limiter
Inputs: One.
Outputs: One.
Effects Type(s): Vacuum-tube compressor, limiter.
Dynamics Parameters: Up to 30 dB gain reduction.
Operational Controls: Gain; gain reduction; attack-release, meter and bypass.
Selected Standard Features: Stereo linkable, side-chain access.
Frequency Response (input/output): N/A.
Distortion (input/output): N/A.
S/N Ratio: N/A.
Pro-User Price Range: $1,000
For additional information circle #218

SYMETRIX, INC.
4211 24th Avenue West
Seattle, WA 98199
Phone: (206) 282-2555

Model CL-150/501
Inputs: Three.
Outputs: Three.
Effects Type(s): Compressor, limiter.
Dynamics Parameters: 0.4:1 to infinite compression.
Operational Controls: Threshold; attack; release; auto/manual; in/out; stereo slave; output gain.
Selected Standard Features: XLR or quarter-inch inputs; automatic or manual operation; side chain insertion capability; 501 has peak limiting.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-1 dB.
Distortion: THD less than 0.025%, 0 dBm at 1 kHz.
S/N Ratio (input/output): Better than -100 dB at 0 dBm, 1 kHz.
Pro-User Price Range: 150: $335; 501: $425

Model 525
Inputs: Three.
Outputs: Three.
Effects Type(s): Gated compressor, limiter.
Dynamics Parameters: 1:1 to 20:1; gated 4:1 dynamic range.
Operational Controls: Expander/gate threshold; compressor/limiter threshold and ratio; output gain.
Selected Standard Features: Simultaneous compression and gating; two channel or stereo linked operation.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-1 dB.
Distortion: THD less than 0.25%, 0 dBm at 1 kHz.
S/N Ratio (input/output): Better than -100 dB.
Pro-User Price Range: $495

Model 522
Inputs: Four.
Outputs: Four.
Effects Type(s): Compressor, limiter, expander, gate, ducker.
Dynamics Parameters: 1:1 compression; 60 dB expansion capable.
Operational Controls: Threshold; attack release; range; ratio; channel in/out; mode select; internal-external key control; stereo slave select.
Selected Standard Features: Balanced inputs and outputs; stereo or two channel operation.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-1 dB.
Distortion: THD less than 0.03%, 0 dBm at 1 kHz.
S/N Ratio (input/output): Better than -100 dB.
Pro-User Price Range: $595

Model 511
Channels: Two.
Effects Type(s): Noise reduction unit.
Dynamics Parameters: 30 dB of noise reduction.
Operational Controls: Expander threshold; dynamic filter threshold; expander in/out; filter in/out; stereo link switch.
Selected Standard Features: Choice of blanced XLR or unbalanced, quarter-inch inputs and outputs; two-channel stereo capability.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-1 dB.
Distortion: THD less than 0.025%, 0 dBm at 1 kHz.
S/N Ratio (input/output): Better than -100 dB.
Pro-User Price Range: $575

Comp 2240
Inputs: Two.
Outputs: One.
Effects Type(s): Compressor and peak limiting.
Operational Controls: Compression: threshold; ratio; attack and release time; limit threshold and release; output level.
Selected Standard Features: Track mode; side chain in 10-segment meter.
Frequency Response (input/output): 20 Hz to 30 kHz, +0.25 dB.
Distortion: THD less than 0.01%, +20 dBm.
S/N Ratio (input/output): Better than -85 dBu.
Pro-User Price Range: $360
For additional information circle #219

U.S. AUDIO, INC.
U.S Distributor: Valley People, Inc.
P.O. Box 40878
Nashville, TN 37204

Gatex/904
Inputs: Four and one, respectively.
Outputs: Four and one, respectively.
Effects Type(s): Noise gate, expander; 904 can be housed in dbx F-900 powered frame.
Dynamics Parameters: Range of attenuation is 0 dB to 80 dB; maximum input level before clipping is +24 dB; maximum output level is +21 dBm (600 ohm or greater).
Operational Controls: Variable threshold, release and range controls; mode switch allows switching of the unit in or out of the audio path, or to be "triggerred" by an external or "keying" signal; mode select switch places each channel in the noise gating mode or 1:2 expansion mode, or 2:3 noise reduction mode.
Selected Standard Features: Three LED "stop light" metering; green LED indicated a "full on" or unity gain condition; yellow LED provides visual indication of ongoing expansion; while red LED shows maximum attenuation as determined by the range control.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.5 dB.
Distortion: Output — less than or equal to 0.015% at THD at unity gain; less than or equal to 0.04% SMPTE IMD at unity gain.
S/N Ratio (input/output): Better than -82 dB, reference +4, unweighted.
Pro-User Price Range: $435; 904: $250
For additional information circle #220

VALLEY PEOPLE, INC.
P.O. Box 40306
Nashville, TN 37204
Phone: (615) 383-4737

Model 430
Inputs: Two.
Outputs: Two.
Effects Type(s): Noise gate, expander, limiter, de-esser, ducker (voice-over device).
Dynamics Parameters: Range of attenuation is 0 dB to 60 dB; maximum input level before clipping is +24 dB; maximum output level is +21 dBm (600 ohm).
Operational Controls: Variable threshold, release, range and output control for each channel; source switch determines the source of the signal which is fed to the detector; mode switch; detector switch.
Selected Standard Features: Eight LED gain reduction meter; clipping warning indicator; external input connector; remote meter/control input connector.
Frequency Response (input/output): 20 Hz to 20

kHz, ±0.5 dB.
Distortion: Output — quiescent distortion at +10 dB; input — less than 0.04% at 1 kHz THD at unity gain; less than or equal to 0.3%, SMPTE IMD at unity gain (typically 0.1%).
S/N Ratio (input/output): Better than -88 dB, reference +4, unweighted.
Pro-User Price Range: $560

Model 811

Inputs: One.
Outputs: One.
Effects Type(s): Limiter, compressor, ducker (voice-over device).
Dynamics Parameters: Gain reduction range is 0 dB to 48 dB; maximum input level before clipping is +24 dB; maximum output level is +21 dBm (600 ohm).
Operational Controls: Variable threshold, release, range/gain, ratio and threshold control; linear-logarithmic release shape switch; mode switch to allow switching of the unit in or out to the audio path, to be controlled by an external signal appearing at the side-chain input.
Selected Standard Features: 13 LED gain reduction meter.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.5 dB.
Distortion: Output — quiescent distortion at +10 dB; input — less than 0.01% at 1 kHz THD at unity gain; less than or equal to 0.025%, SMPTE IMD at unity gain.
S/N Ratio (input/output): Better than -87 dB, reference +4, unweighted.
Pro-User Price Range: $400

Kepex II

Inputs: One.
Outputs: One.
Effects Type(s): Noise gate, expander.
Dynamics Parameters: Range of attenuation is 0 dB to 80 dB; maximum input level before clipping is +21 dB; maximum output level is +21 dBm (600 ohm).
Operational Controls: Variable threshold, release, range, ratio and threshold controls; linear/logarithmic release shape switch; mode switch allows switch of the unit in or out of the audio path, or to be "triggerred" by an external or "keying" signal appearing at the side chain-input.
Selected Standard Features: 13 LED gain reduction meter.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.5 dB.
Distortion: Output — quiescent distortion at +10 dB; input — less than 0.015% at 1 kHz THD at unity gain; less than or equal to 0.05%, SMPTE IMD at unity gain.
S/N Ratio (input/output): Better than -87 dB, reference +4, unweighted.
Pro-User Price Range: $400

Model 440

Inputs: One.
Outputs: One.
Effects Type(s): Compression, expanded compression, peak limiting, FM limiting, AGC, dynamic sibilance processing, peak clipping.
Dynamics Parameters: Gain reduction range (compressor) is 0 dB to 40 dB; limiter 0 dB to 20 dB; maximum input level at 1 kHz is +24 dB balanced, +21 dB unbalanced; maximum output level is +24 dBm, +21 dBm unblalanced (600 ohm).
Operational Controls: Variable threshold, release and attack controls; variable expander threshold; variable limiter/clipper threshold; variable limiter release; variable output control.
Selected Standard Features: FM pre-emphasis, limiting, and compression; auto attack and release mode for compressor and expander; AGC mode; peak clipper; linking for two unit operation as either master, slave or stereo coupled configuration; selectable VU meter mode for input or output reading; hardwire bypass; electronically balanced inputs and outputs.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.5 dB.
Distortion: Output — less than or equal to 0.01% at 1 kHz, THD at unity gain; less than or equal to 0.025%, SMPTE IMD at unity gain.
S/N Ratio (input/output): Better than -84 dB, reference +4, unweighted.
Pro-User Price Range: $599

Model 610

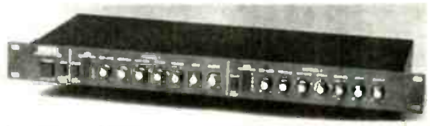
Inputs: Two.
Outputs: Two.
Effects Type(s): Compression, expanded compression, peak limiting, noise gating, FM pre-emphasis limiting, voice-over, expansion, limited expansion.
Dynamics Parameters: Range of gain reduction is 0 dB to 60 dB; maximum input level at #2 at 1 kHz is less than or equal to +24.5 dB balanced; maximum output level is +24 dBm, +21 dBm unblalanced (600 ohm).
Operational Controls: Variable compressor/limiter threshold; variable compressor/limiter ratio; variable expander/noise gate threshold; variable range, release and gain; switch selectable compressor-limiter attack time; selectable expansion slope; stereo couple switch; hardwire bypass.
Selected Standard Features: Auto release mode; eight LED gain reduction meter; electronically balanced inputs and outputs; external input connector; peak reversion correction circuitry; threshold, ratio, output coupling circuitry; gain recovery computation circuitry.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.5 dB.
Distortion: Output — less than 0.01% at 1 kHz, THD at unity gain; less than or equal to 0.2%, SMPTE IMD at unity gain.
S/N Ratio (input/output): Better than -87 dB, reference +4, unweighted.
Pro-User Price Range: $995

For additional information circle #220

YAMAHA INTERNATIONAL CORP.
6600 Orangethorpe Avenue
Buena Park, CA 90620
Phone: (714) 522-9011

Model GCZ020

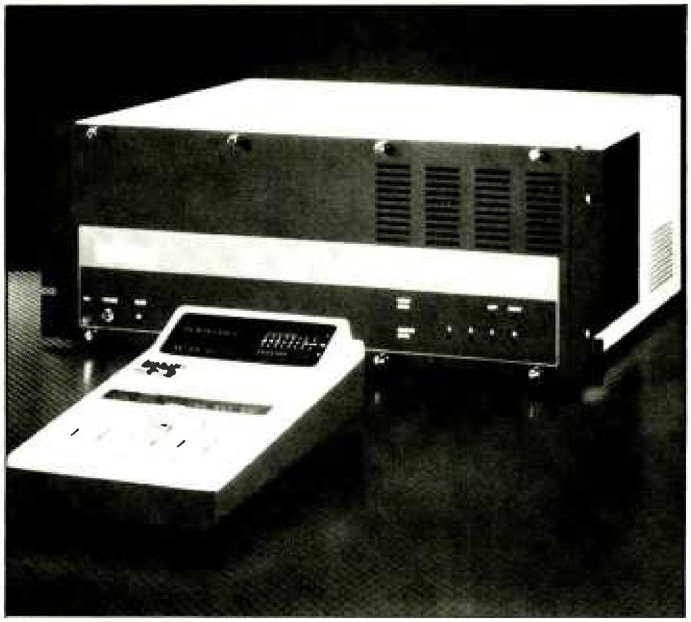
Inputs: Four.
Outputs: Four.
Effects Type(s): Compressor, limiter.
Dynamics Parameters: 1:1 to infinity: 1; 0.2 to 20 milliseconds attack; 50 millisecond to 2 seconds release.
Operational Controls: Expander gate; threshold; compression ratio; attack; release; compressor in/out.
Selected Standard Features: 24 dB gain reduction meter for each channel; detector loop; stereo or dual-mono modes 32 dB limiting.



Frequency Response (input/output): 20 Hz to 20 kHz, +2/-2 dB.
Distortion: Less than 0.03%.
S/N Ratio (input/output): Better than -87 dB.
Pro-User Price Range: $295

For additional information circle #221

...ICON 224XL DIGITAL REVERB AND EFFECTS PROCESSOR WITH LARC REMOTE AND SOFTWARE UPGRADE



Reviewed by Bob Hodas

Lexicon has been manufacturing digital reverberation systems for many years now, and has probably placed product in every major recording studio in the world. The company began with the 224, improved on that design with the 224X, and then brought out a very user-friendly control head, the LARC (Lexicon Alphanumeric Remote Console). While it may seem unusual to review a unit that has already proved to be so popular, Lexicon recently introduced a new software upgrade with additional reverb and special effects programs that will surely be of interest to current 224X owners.

System Configuration

The 224XL comprises a rack-mountable mainframe unit and the LARC remote control head, which connects to the processor via a 50-foot flexible cable. The mainframe unit is fan cooled, and should be kept in as cool a spot as possible. Since all control information is displayed on the LARC, the mainframe can be stored almost anywhere in the studio control room or live-sound equipment rack. The review unit came with the standard 50-foot cable, although runs of up to 1,000 feet are possible with an optional remote power supply. While I didn't verify this with the factory, it seemed to me that the mainframe's fan was much quieter than on previous models. Normally, I would move the unit out of the control room for mixdown, but this one sat on the floor in front of the console and was pretty quiet in operation.

The device is a two-input, four-output processor that can be run either in a balanced or unbalanced configuration. Inputs may be either mono or stereo, while the outputs operate as mono, stereo, dual-stereo or quad, depending on the program selected. Front-panel adjustment pots are provided for input and output level adjustment, which is a simple procedure using one of two delay-line diagnostic programs.

The LARC is a clean, well laid out piece of work that is simple to use yet delivers quick, powerful control capabilities into the engineer's hands. Since a description of the LARC front panel would be wordy and rather boring, take a look at Figure 1 for a simplified description. As can be seen, the dedicated sliders and keys allow for fast, sure operation, and keys are sized to allow easy operation with fingers. By using the main and parameter display windows, control functions can quickly and easily be learned without continually referring back to the user's manual. With the cost of studio time being what it is, not to mention shrunken recording budgets, the LARC can prove to be a speedy tool both for a first-time user and the experienced pro. All displays are quite legible, and programs are both named and numbered for easy access. Parameter descriptions are abbreviated yet logical.

Reverb and Effects Programs

Accessing programs via the LARC follows a logical path. Programs are grouped according to acoustic nature in five categories called *Banks*. Within each Bank are stored up to six *Programs*, each of which holds up to eight *Variations*. (See Table 1 for a complete listing of the 224XL's Hall, Room, Plate, Effects and Split Programs.) Any of these variations may be altered by the operator, simply by calling up the different *Pages*. *Parameters* within each page are adjusted using the six control sliders. Most Programs have four or five different Pages (all have at least one), and each Page contains up to six Parameters. As you can see, not only does the 224XL come with 83 factory-loaded Programs, but each Program has many Variables that can be altered to fit the user's particular needs. Flexibility is a key word here, and the 224XL allows a great deal of creative control.

Personalized program alterations may be stored for future use in one of 36 non-volatile *Registers*. These altered programs may be assigned to any of 10 Banks, each of which holds up to 10 Registers. Both Banks and Registers may be labeled with an alphabetic name and, for easy recall, each storage location is assigned a number that also displays the Program from which it was derived. I found the memory storage and labeling process to be easy to learn, and quick to implement. The LARC will even tell you when the memory Registers are full.

If 36 non-volatile memories aren't sufficient, the LARC will download as many programs as you need onto an external cassette deck, which may then be used to upload 36 at a time. At any one time you can have up to 119 programs at your fingertips, with many more waiting in the wings. Tape storage is a relatively simple process; Lexicon supplies the necessary cables so all you need is a good quality cassette machine and a good audio tape. The LARC inserts a tone prior to downloading the Registers, so that you can set proper record level to ensure a good transfer. LARC will also verify that your transfer is good or bad by comparing the data recorded on tape to that in the Registers. Recall from tape to memory can also be verified.

Independent engineers and mixers

will find tape storage a very attractive feature. One has only to carry a cassette tape of your favorite Programs to a studio equipped with a 224XL in order to recreate that "signature sound." Also, all effects developed for an album project, for example, can be easily reproduced in the live-concert situation with a very short setup time. (I recall watching the house-sound mixer for The Police painstakingly programing pages of data by hand into a 224X at a concert in Portland; LARC will make his future concerts much simpler!)

Software Updates

Lexicon is making a statement with its new software package, by reaffirming a commitment to improving quality and holding a competitive place in the market of creative programming. I found several improvements in this unit that I believe make the 224XL distinctly superior to past models. The new system handled transient material much better than did the 224X of my past experiences. I was able to input zero to +6 VU in most programs without noticeable weirdness on the top-end (although it is better to play it a little safer with all digital devices). This also meant that with higher input levels, output levels could be brought down much lower than previously, resulting in greatly reduced output noise and hiss. The 224XL also handled bass frequencies much more naturally, which came in handy while doing Pop-Dance mixes calling for effects such as placing the kick in reverb. Some Programs — usually those with exaggerated high-end response — still got rather "sizzly" on top, but this could be controlled by reducing the input level or rolling off the high-frequency crossover point.

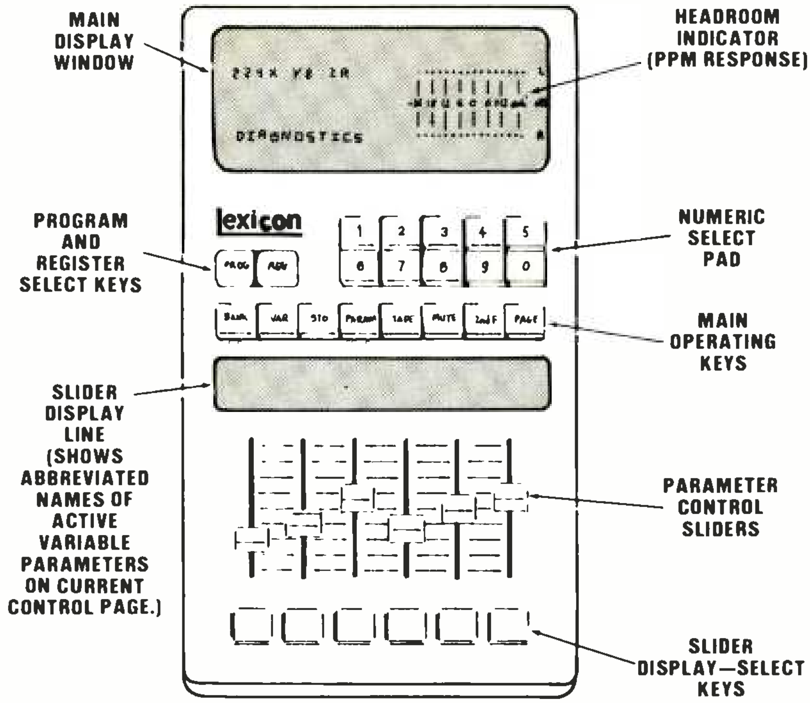


FIGURE 1: LEXICON 224 LARC REMOTE CONTROL

The manufacturer has also done its homework on Programs, and has put something for everyone into the 224XL. The inclusion of 83 factory-loaded Programs certainly provides a good starting point for program modification and, in fact, quite a few of them are very attractive as shipped by the factory. Lexicon has provided many concepts from past models, adding to and improving on ones that work. Also included are several nice programs from the competition, with some sonic improvements. I tried to take enough time to run through all the programs, yet found that some attracted more attention than others. From a glance at Table 1, it will be easy to see how Lexicon groups its banks of Programs, a factor that makes it easy to quickly find a middle ground when searching for a particular effect.

Session Applications

For this review, I worked with the 224XL mostly on rock material involving instrumentation of sax, drums, bass, guitar, synthesizers of many varied types, and both male and female vocalists. I did not use any real string or horn sections which would have been nice in evaluating the larger halls.

•The sound of the **Concert Halls** were really too big for the test material, but I can give a few impressions. Initially, I noticed that the Programs tended to load up, just as halls with extremely long reverb times tend to do. Synthesizers sounded really huge, and the Programs were very effective when using more reverb than original signal.

. . . continued overleaf —

TABLE 1: LEXICON 224XL SOFTWARE PROGRAMS

Programs	Halls	Rooms	Plates	Effects	Splits
1	Concert Hall 7 Variations	Room 4 Variations	Plate 6 Variations	Chorus & Echo 4 Variations	Hall/Hall 1 Variation
2	Bright Hall 5 Variations	Small Room 4 Variations	Small Plate 6 Variations	Resonant Chords 1 Variation	Plate/Plate 2 Variations
3	Dark Hall 7 Variations	Chamber 1 Variation	Constant Density Plate A 1 Variation	Multiband Delay 1 Variation	Plate/Hall 1 Variation
4		Rich Chamber 8 Variations	Constant Density Plate B 3 Variations		Plate/Chorus 1 Variation
5		Dark Chamber 8 Variations	Rich Plate 8 Variations		Rich Split 1 Variation
6		Inverse Room 3 Variations			

•The **Bright Halls** are described as similar yet much brighter than the Concert Halls. I found this to be true, and very nice for creating reverb effects. Variation #3 (V3) worked well to create a modern drum sound, and sounded very crisp and fast with a large ambience. V4 had a great low-end and good midrange presence that sounded really good on synthesized cellos.

•The **Dark Halls** are designed to more realistically portray natural halls than the other two programs. This is achieved with a more drastic high-frequency rolloff, simulating natural reverb times. V3 was very effective with synthesizers when featuring low-end information, or creating a dark or ominous mood. V4 gave a natural, very full sound to stacked background vocals. I had some problem with V5, which manifested as a low feedback that built slowly even after input was removed — it sounded just like a close miked gong.

All Hall Programs incorporate adjustable pre-echo delays that can add to the realism by emulating the slapback we're used to hearing in most concert arenas or large halls.

•**Room** is similar to the Hall Programs in that it uses similar parameters, yet creates the ambience of a smaller space — one feels much more of a natural sense of walls and dimension with this program. V4 demonstrated some interesting vocal presence effects and ambience. I also found a great kick drum ambience when manipulating parameters.

•**Small Room** recreates a space about ¼th the volume of Room. Walls feel as though they are much closer, and I found many different instruments and voices to sound quite nice with V4. There are definite applications in Automatic Dialog Replacement (ADR) with this Program.

TECHNICAL SPECIFICATIONS FOR LEXICON 224XL DIGITAL REVERBERATOR

Programs: 18 programs, 59 preset variations, expandable through software updates.
Register storage: 36 nonvolatile registers divided into 10 user-labeled banks with one to 10 registers per bank.
Reverberation time: Adjustable in two bands from approximately 0.2 to 70 seconds (program-dependent).
Additional controls: Four mode-select buttons (BANK, PROG, VAR, REG) used with 10 numeric-select buttons (1 to zero); tape storage and register control buttons (TAPE, STO); a page-select button (PAGE); three auxiliary control buttons (MUTE, PARAM, 2nd F); six sliders for control of up to 42 parameters per program, with associated display-select buttons.
LARC display: Two lines of 12 alphanumeric LEDs for interactive menu-driven display; additional line of 24 alphanumeric LEDs (six groups of four for each slider); dual 16-position LED headroom indicator (calibrated -24 to +12 dBm plus overload).
Mainframe controls: Power and indicator light; system reset; left and right input level adjustments; A, B, C, D output level adjustments.
Frequency response: 20 Hz to 15 kHz, ±1.5 dB; 20 Hz to 12 kHz, ±0.5 dB.
Dynamic range: *Reverberant mode:* 84 dB typical, 81 dB minimum relative to reference level; 20 Hz to 20 kHz noise bandwidth for all reverb times from zero to 10 seconds. *Nonreverberant mode:* 90 dB typical, 86 dB minimum; 20 Hz to 20 kHz noise bandwidth.
Total Harmonic Distortion (THD and noise): 0.04% typical, 0.07% maximum at reference level for all reverberation times between zero and 35 seconds.
Interchannel Crosstalk: -55 dB at 1 kHz.
Inputs: Two, balanced and transformer isolated; impedance: 20 kohm; maximum level adjustable: +8 to +18 dBm.
Outputs: Four, balanced and transformer isolated; impedance: 90 ohm; maximum level adjustable: +8 to +18 dBm; power-on muting.
LARC cable: 50-foot extra-flexible cable; cables can be linked — up to 1,000 feet possible with optional remote power source.
Power: *Mainframe:* nominal is 100, 120, 220, 240 VAC (-10%, +5%) switch-selectable; 50 to 60 Hz; 150 watts. *LARC:* normally powered through 224X mainframe; miniature jack accepts optional remote power supply for distances greater than 100 ft — 10 to 20 VDC or 10 to 20 VAC, 6.25 watts.
Diagnostic programs: Control and display via LARC; automatic at power-up or reset.
Size: *Mainframe:* Standard 19-inch rack mount: 19 by 7 by 15 inches (W×H×D), (483×178×381 mm). *LARC:* 5.9 by 9.5 by 3.2 inches (W×H×D), (150×242×82 mm).
Weight: *Mainframe:* 34 pounds (15.5 kg); 40 pounds (19 kg) shipping. *LARC:* 1.9 pounds (0.9 kg); 7 pounds (3.2 kg) shipping.
Automation interface: Optional RS-232C serial interface.
Suggested Pro-User Prices: 224X with LARC: $12,500; LARC retrofit kit: $800; V8.1 software: $95.

Lexicon Inc. 60 Turner Street Waltham, MA 02154 (617) 891-6790

□□□

•**Chamber** is really a very good Program. My favorite chambers were at Wally Heiders in San Francisco (now Hyde Street Studios). It is hard to build a good acoustic chamber, but Lexicon has done a fine job digitally. It is a realistic chamber with beautiful diffusion. I no longer yearn for Heiders!

•**Rich Chamber** has a very even diffusion that eliminates the sense of walls to a good degree. Dimension may be added to the program by using the pre-echo delays and, combined with a related pre-delay, can make some brilliant spaces. Many variations were impressive, with V4 being a good all-around program. V2 and V5 were especially nice for keyboards, while V3 worked well for drums. V6 and V8 emulated a modern gated drum reverb, with V8 having a faster closing gate. Rich Chamber was one of my favorite 224XL Programs.

•**Dark Chamber**, like Dark Hall, has a sharper high-frequency rolloff to more naturally simulate the sound of real acoustics. The variations are effective, and good acoustic environments are a strong point. This will be another good Program for film work. V1 made a good showing on keyboards, while V5 was very nice for vocals. I found V7 and V8 to be very interesting and created some wild metallic rooms with the parameters.

•**Inverse Room** creates distinctly different effects in its three variations. V1 is your basic "Phil Collins-style" gated-drum/room-mike sound — it is a good sounding program and a great deal of tailoring can be accomplished. The extensive frequency control comes in very handy for emulating the competition's non-linear programs, or for creating extended frequency response versions. V2 is a backwards reverb with a reverse envelope, to give the impression of reverb building as opposed to decaying. I liked this a lot too, and utilized the delays to time a snare reverb so that it built up when leading into the kick-drum attack. V3 is an enhancement program that is effective for adding increased presence to a voice without actually adding volume.

•**Plate** is a very good representation of a true plate reverb. I especially liked V5, and found it to be realistic. Some variations are much more flexible than a real plate though, since you can perform frequency tailoring and add as many as six pre-echo delays for a more acoustic reverb feeling. Diffusion is also adjustable, and one can find just the right density to compliment different instruments.

•**Small Plate** didn't do a lot for me. I got kind of a hollow feeling from some variations, and I guess it just didn't suit my taste. Because my first impression was not so strong, I didn't spend any time manipulating the Program parameters.

•**Constant Density Plate A** is an original program from the 224. It simulates a plate with high initial diffusion, and maintains that density of reflections. This capability differentiates the Program from normal plates in which echo density decreases with time. I didn't utilize this program as it was not effective with the test material.

•**Constant Density Plate B**, on the other hand, was nice and bright and very impressive with a kick drum. I used V2 and was happy with the thick type of reverb, without getting in the way of input material.

•**Rich Plate** was another dense, "tight" reverb that had many applications. V2 and V5 sounded good with almost any input material from sax to drums to guitar, while V6 was hot for drums and, especially, kick. V4 was used on vocals, producing a rich, fat, bright sound.

All the Reverb Programs discussed so far possess two Parameters that are unique to Lexicon. Low Frequency Stop Decay and Mid Stop Decay proved to be very useful, and really improved the quality of the Programs. These two adjustment Parameters allow you to set the reverb time differently in the absence of input than when input is present. In this way, you have the ability to create long, flowing reverb tails that fill silent spaces, while maintaining a short reverb time during program material and thus a high degree of clarity. You can achieve the effect by setting the Stop Decay Parameters to longer times, and working with shorter decay-time Parameters. All in all, it's very effective and well thought out.

Another nice touch is that there is a gate parameter on most Hall, Room and Plate Programs, a feature that provides control for creating many different gated-reverb Programs. Gated reverb is certainly in vogue at the present time, and Lexicon is offering the engineer many options for its use.

Aside from the varied effects possible with the Reverb Programs, the 224XL offers several strictly effect-type programs:

•**Chorus & Echo** can create doubling, flanging, chorused echo and many other special effects. The Pro-

TABLE 2: EXAMPLE OF CONTROL PAGES AND VARIABLE PARAMETERS FOR RICH CHAMBER

	Variable Parameters and Adjustable Ranges					
Page	Sliders: 1	2	3	4	5	6
1	*LF Decay* 0.1 to 83 sec*	*Mid Decay* 0.1 to 83 sec*	*Crossover* 170 Hz to 19.0 kHz*	*Treble Decay* 170 Hz to 19.0 kHz*	*Attack* Zero to 99	*Pre-delay* Zero to 834 mS
2	*LF Stop Decay* 0.1 to 83 sec*	*Mid Stop Decay* 0.1 to 83 sec*	*Chorus* Zero to 99	*HF Bandwidth* 170 Hz to 19 kHz*	*Diffusion* Zero to 99	*Definition* Zero to 99
3	*Pre-echo Level 1* L greater than AD Zero to 99	*Pre-echo Level 2* R greater than CB Zero to 99	*Pre-echo Level 3* R greater than AD Zero to 99	*Pre-echo Level 4* L greater than CB Zero to 99	*Pre-echo Level 5* L greater than AD Zero to 99	*Pre-echo Level 6* R greater than CB Zero to 125 mS
4	*Pre-echo Delay 1* L greater than AD Zero to 125mS	*Pre-echo Delay 2* R greater than CB Zero to 125 mS	*Pre-echo Delay 3* R greater than AD Zero to 125 mS	*Pre-echo Delay 4* L greater than CB Zero to 125 mS	*Pre-echo Delay 5* L greater than AD Zero to 125 ms	*Pre-echo Delay 6* R greater than CB Zero to 125 ms
5	*Size* 8 to 87 meters	*Inactive*	*Reverb Stop Delay* (Gate) Zero to 1.26 seconds	*Inactive*	*Inactive*	*Inactive*

*Can also be set to infinite.

gram generates three voices per input channel (six total), and each voice can be assigned a separate level, delay, feedback amount, and left-to-right pan position. A chorus control determines the pitch shift for all voices. You can make some seriously "massive" vocal effects with this program, and the potential also exists for guitar and keyboards. V4 was quite unusual, and created a "mutron-type" effect when fed bass guitar. I took three tracks of background vocals — each with four singers on them — modified V1 for full-field panning and depth perception using the level and delay controls, and came out with a really primo sound.

•**Resonant Chords** is a unique program limited only by the engineer's level of weirdness. An input (which does not have to be of musical nature) excites six voices with control parameters of level, pitch, duration of ring, and an overtone lowpass filter. The best thing I can say about this Program is to put it up and see what appeals to you. It certainly is capable of some *interesting* sounds.

•**Multiband Delay** is just what you would expect. The Program comprises six separate delay lines with high-and low-pass filters, and full left to right panning — I used it for sax with spectacular results, and created a combination of center slapback and left-to-right movement with ease.

I believe the **Split Programs** to be a very strong sales point for the 224XL. With these five setups you can run two completely separate Programs, each with mono input and stereo output. Although control parameters are not always as extensive as those provided for the single Programs, having *two* Programs in place of one surely makes up for this minor limitation. Splits offered are *Hall/Hall* (based on Concert Hall V1), *Plate/Plate* (based on Plate V1), *Plate/Hall* (as above), *Plate/Chorus*, and *Rich Split* (based on Rich Chamber).

Due to the many adjustable parameters available, I would like to stress the extreme amount of program modification that can be performed by the user. Table 2 lists the Control Pages and Variable Parameters for the Rich Chamber program; a glance at this chart will better help the reader understand the 224XL's versatility.

Upgrade Flexibility

Technical engineers will be happy to know that Lexicon supports its customers with 14 pages of maintenance information in 224XL user's manual, including a complete description of each module plus charts showing module and fuse locations. On power-up the 224XL runs a 25-second major component test. If errors are found, two extensive diagnostic programs — one LARC test and one mainframe test — can then be run to identify the exact problem.

Lexicon also supports its users with update kits, and the 224X was designed to allow software changes with ROM (read-only memory) circuits. Anyone with a 224X can request the updates from the factory. In fact, some updates are provided free, yet Lexicon says that many owners don't register their units and so can't be notified when updates are available. The LARC remote-control head may also be retrofitted to any 224X for those who wish to take advantage of its tape storage and superior control features.

The Lexicon 224XL is the most versatile digital reverb and special-effects processor that I have worked with to date. It combines good reverb programs with some nice special effects in a very user-friendly package. The internal programs are easy to operate, and the sounds offered cover a lot of territory. ■■■

I would like to thank Studio D, Sausalito, CA, for donating the session time needed to perform this in-use operational evaluation.

KURZWEIL 250 DIGITAL SYNTHESIZER WITH 50-kHz SAMPLING AND STORAGE OPTIONS

Reviewed by Bobby Nathan

Of all the current sampling-technology keyboards, none has caused more controversy than the Kurzweil 250. When I first saw the 250 two years ago during the New York AES Convention at the Hilton Hotel, I was blown away! Not only was the demonstration most impressive, the sound of the 250 was *electrifying*. The unit shown then was still in its early development, and the microprocessor had not yet been compacted to fit into the keyboard's casing as we know it today. During the AES booth demonstration, the sounds were coming from a mainframe computer, which also was handling all of the programming. The promise from Kurzweil was that all the sounds we had heard would be put into a stand-alone unit. But, when the 250 was again shown at the 1984 New York AES as a stand-alone unit, the list price had been upped and the design changed to incorporate the use of an Apple Macintosh computer for sound-sample storage, which meant yet another additional price increase.

These changes to the 250 had disappointed many potential users. When the sampling option was finally released, again many were disappointed with the 25-kHz sampling rate, and its resultant limited bandwidth. This *R-e/p* review was written after working with the new 2.2 REV operating software and the 2.0 REV Digitizer software; I was anything but disappointed after hearing the 250's new sampling quality.

To start with, the new sampling rate has been increased to 50 kHz, and is variable over other different preset rates. At a 50-kHz sample rate, the total sampling time is 10 seconds; at 41 kHz the time is 11 seconds; at 37.5 kHz it is 13 seconds; at 31.25 kHz it is 15 seconds; and at 25k it is extended to 20 seconds. (If you are not unfamiliar with sampling bandwidth versus sampling time, this is the normal trade-off of shorter sampling time for higher frequency response.)

Sampling sounds into the 250 is quite easy; the user has the ability to divide up the sampling time to multi-sample most instrument sounds. There are five different sampling types, ranging from *Quick Take* (which functions as the name implies), to more complex schemes. Quick Take has a pre-emphasis equalization curve that brightens the sample, while the second type of sampling, *De-emphasis*, samples the sound normally. The other three types use compression (slow, normal or fast decay) to reduce digital aliasing and noise. The only trade-off of these latter sampling types is that they require additional processing time, but for a good sample it's all worth it.

The 250 also has another interesting feature: an automatic split determinator between two multisamples. For instance, if you sample a piano at middle-C (as the lower sample), and sample again at C one octave above middle-C, the microprocessor will choose whether the split between the two samples should occur on the F or F# above middle-C. The auto-split function also has an override feature that allows you to set the range by striking the lower and upper notes. A computerized filter-adjust feature is also provided that will automatically adjust how the filter closes as the sample gets transposed during multi-sampling.

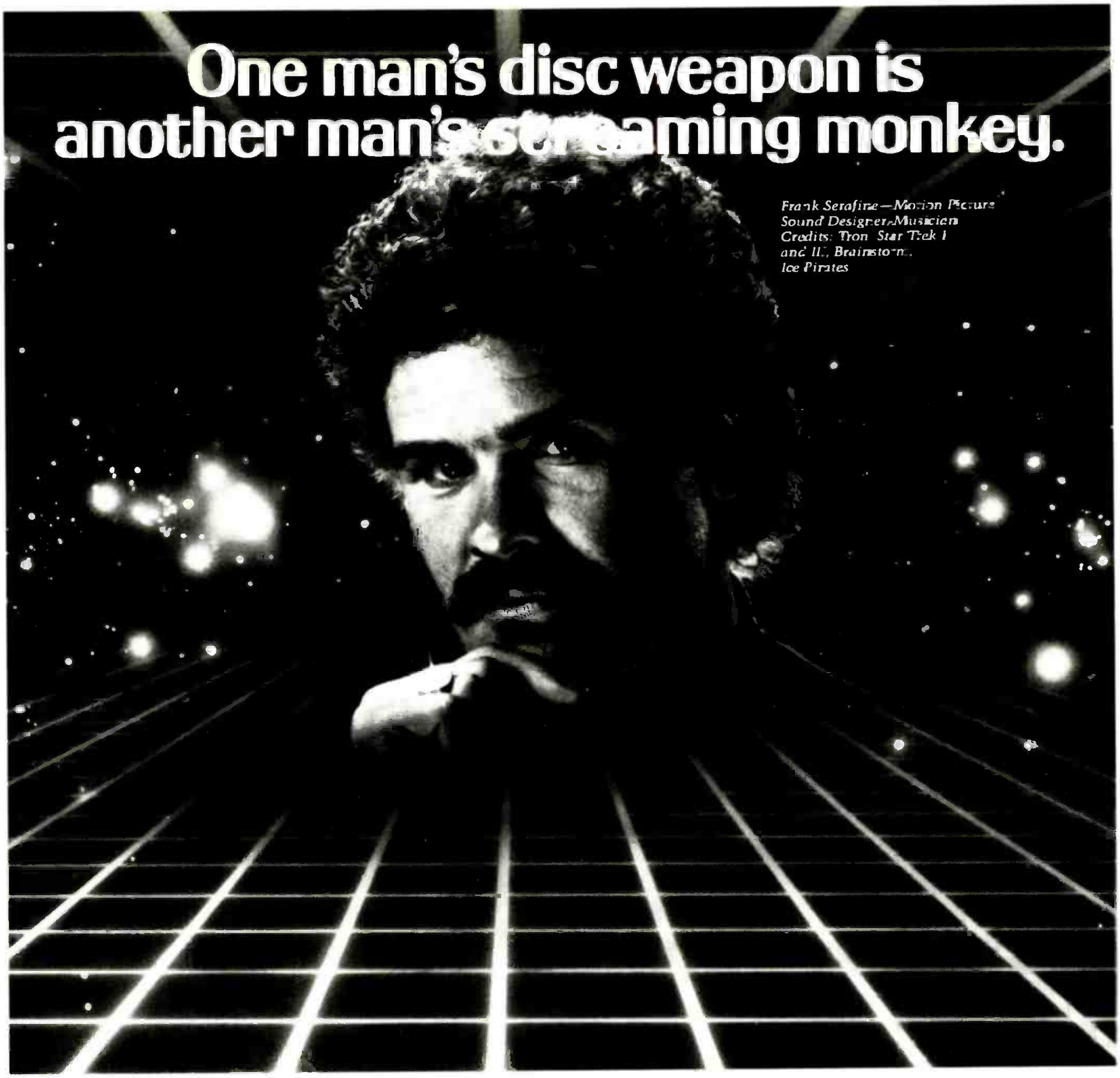
If you are unfamiliar with the way in which timbre changes as the pitch of a sample is transposed, I'll try to explain. The phenomenon, which occurs in all sampling machines, means that if you sample a piano at middle-C and transpose it up to G above, you will notice its timbre (or tone) becoming brighter. For every half-step you transpose a sample upwards in pitch, the timbre keeps getting brighter; as you transpose a sample downwards in pitch the timbre becomes duller. While multisampling at middle-C and C above, at F# (the split point between the two samples) you will notice the upper sample at F# above middle C is very bright, whereas at F the sample is dull. The computer within the 250 compensates for this effect, and dulls the lower sample as it gets transposed upwards in pitch. By doing so, it smoothes out the keyboard and makes the sound seem more even.

Looping samples on any sampling keyboard has never been a favorite pastime for me, but nevertheless has to be done to make certain samples usable. Even though no on-screen display is provided on the 250 for viewing prime edit points, the unit does have another interesting feature to improve the sound quality of loops, called *Crossfade*. The crossfade feature allows you to fade the end of a loop into its start point; you simply loop the sample as close as possible and then adjust the amount of crossfade. In many cases the crossfade seems to remove the annoying glitch that occurs during the looping of difficult waveforms, such as strings, voice and piano samples.

System Options

Since the Kurzweil 250, as an instrument, is available in several different levels, I should stop here and describe just what they are. The "basic" unit comes with 40 internal sounds, including the famous Kurzweil grand piano sound. Also included are the other favorites, such as strings (fast, slow and bowed), acoustic upright bass, acoustic guitar, and drums.

The next level of sophistication is to add *Sound Bloc A*, which increases the internal sampled library to 125 sounds. Although many of the new sounds are variations of the original 40, they will be well worth the addi-



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SUMMARY OF KURZWEIL 250 DIGITAL SYNTHESIZER SPECIFICATIONS

Keyboard: 88 notes, velocity sensitive.
Channels: 12
Dimensions: keyboard 57×27×9 inches (L×W×H); pedal pod 17¾×11⅛×4⅛ inches.
Power consumption: AC 110V, 50/60 Hz, 380 W (220V option available).
MIDI: (In, Out, Thru): 16 channels, user-assignable. Each sequencer track can be assigned to a separate MIDI channel. Special MIDI mode slaves one Kurzweil 250 to another.
Inputs: mike/line input (user-sampling optional); two ¼-inch, assignable volume-type pedal jacks; computer port.
Stereo audio output levels: balanced XLR-type 600 ohm, 10V p-p nominal; hi-level, ¼-inch 600 ohm, 10V p-p nominal; low-level, ¼-inch 600 ohm, 1V p-p nominal; Headphone, stereo ¼-inch, 8 to 600 ohm.
Dynamic Range: over 90 dB.
Resident Voices: Concert Grand Piano, Violin Section, Viola Section, Cello Section, Bass Section, Plucked Acoustic Bass, Snare Drum, Bass Drum, Tom-tom (two-octave chromatic), Hi-hat open, Hi-hat closing, Hi-hat closed, Crash Cymbal, Cowbell, Sandpaper, Hammond B-3 Organ (three settings without percussion, one setting with percussion), Trumpet, Baritone Horn, Valve Trombone, Sine Wave, "Endless Glissando," nylon-stringed Acoustic Guitar, Hand Claps, Finger Snaps, Temple Blocks, Grater up, Grater down.
Keyboard Setups: the base unit contains 40 factory-installed keyboard setups, with up to 40 user-definable keyboard setups available. Factory instruments total 30, with 48 user-definable instruments available.
Programmable Functions: variable 256-segment envelope generator; 87-way keyboard split with up to six instrument layers; 24 LFOs; four wave shapes (ramp up, ramp down, square and triangle); continuously variable tremolo/vibrato/amplitude parameters; variable brightness levels, including velocity-to-brightness mapping; variable pitch modulation; five modes transposition (octave pitch shift, chromatic pitch shift, octave transpose, chromatic transpose, timbre shift); stereo chorus parameters, doubling, flanging, echo, full chorusing; variable delay time (up to 30 seconds), variable detuning (+1,200 cents, -6,000 cents); keyboard dynamics table with 11 different settings available.
Assignable Controls: two assignable levers, three assignable sliders, two assignable on/off foot switches (pod), two assignable external pedal jacks. . . *continued overleaf —*

tional cost. Sound Bloc A also adds choirs (men only, men and women, and cathedral), woodwinds, orchestral percussion, mallet instruments, and electric and slap basses.

If enhanced sampling capability is what you're after, you will need to add the *Sound Modeling* option. The previous version provided only 25-kHz sampling; future versions, including the 2.0 REV Digitizer software and companion PCB reviewed here, provides a 50-kHz sampling frequency. The Sound Modeling package, which costs $1,995 for the 2.0 REV version (and $250 as an update to previous owners of 1.0 REV), gives you the ability to sample your own sounds into the 250; if you then want to save those samples to floppy disk, you'll also need an Apple Macintosh computer, Kurzweil's *MacAttach* software, and connection cable. (While a Mac with 128 Kbytes of memory works, for increased speed a 512-Kbyte version is recommended; in either case a second, external disk drive also is required to run MacAttach.) Since the 250 samples sounds into volatile memory, upon power down they will be lost. If you need the speed of recalling sound samples from disk, the extra cost of a Macintosh computer is the only way out.

The Macintosh computer also allows you to store instrument files which, in reality, are effects settings. A total of 125 instrument files are stored in the 250's internal ROM (read-only memory), with provision to add 100 user-defined instrument file setups. The 250 also enables approximately 40 additional user-defined keyboard setups (depending on their complexity) to be stored internally in RAM (random-access memory), in addition to the existing 125 setups held in ROM. An instrument setup file includes envelope parameters, chorusing type and parameters, vibrato, output grouping assignments, and modulation parameters. One can have preset envelopes that can instantly be assigned to any existing sound in the 250, or to any user sample. (For example, a favorite chorus or vibrato and stereo panning can be assigned to any sound.) Alternate sustain pedal settings can also be stored as an instrument file.

A good example of how to use an instrument file is as follows: start with the acoustic piano sound, and let's say you'd like the chorusing and stereo panning of the stereo vibes provided in the 250. You could use the stereo vibe's filter, envelope, chorus and output grouping settings found in instrument file list, capture them into what is called the current region (the

full or partial range on the keyboard), and then the grand piano would have all those settings. You could also simply assign those settings to just one note of the piano, or split the keyboard and copy them to only the upper half.

The 250 also has three banks of 10 *Storage Bins* that can be used for storing variations of any of the internal presets, or for just calling up your favorite ones in a studio or live-performance situation. The bins are stored in non-volatile RAM, and can also be saved to floppy disk with the Macintosh as a "Keyboard" setup file. Each bin has 10 presets; by touching any number from 0 thru 9 on the 250's numeric keypad, sounds change instantly.

The 250 has *Split* and *Layer* functions that allow you to assign strings on the upper portion of the keyboard, and piano on the lower. In Layer mode the strings can be layered on top of the piano across the entire 88 keys, or over just the middle section. Two different or identical instruments can be assigned into a Split or Layer keyboard, and then assigned to either output group A or B. By using the separate outputs for group A or B, the 250 can be routed to separate channels of the recording console, and then equalized and enhanced separately.

I found the *Timbre Shift* function very useful for changing the 250's preset keyboard sounds. By simply hitting *Edit* and then both *Transpose* buttons, you can down-arrow (scroll) through the transpose options and choose *Timbre Shift*. If you take the original Kurzweil Grand Piano sound and shift its timbre to a setting of "3," the result sounds like a Baldwin, while at "6" it sounds bright like a Yamaha Grand. The timbre shift also works great on the brass presets, or on any user samples.

I was also quite amazed at what the *Instrument Editor* can do. The 250 has a most powerful envelope editor that can edit any parameter of the envelope's attack, decay and release. While working with any these parameters they are displayed numerically on the 250's backlit liquid-

KURZWEIL SPECIFICATIONS. . . continued —

Sequencer: 12-track, polyphonic, 7,900-note storage capability (battery-backed RAM). Complete software includes: sequence editing, individual track editing, individual track volume control, looping, quantization on playback, individual note editing/insertion, variable-rate external synchronization, click-track output, simultaneous access to all on-board sounds.

Sound Modeling Program
Variable sampling rate (5 to 50 kHz); total sampling time (10 to 100 seconds), depending upon sampling rates; compression; adjustable loop decay and release rate; automatic natural amplitude envelope extraction and artificial envelope generation; level check meter, clip indicator; trimming and looping functions; multiple samples on each key; pitch and amplitude adjustment; multiple-and single-key multisampling; internal storage (64 sounds, eight keyboard setups in volatile, non-battery-backed memory); external storage capability on Apple Macintosh diskettes via MacAttach™ software.

Sound Block A
Resident sound module containing 15 new voices, plus 84 factory-defined keyboard setups using these sounds along and in combination with the 30 resident voices in the base unit. New voices include: choir; flute; electric bass (open); electric bass (slap); clarinet; oboe; harp arpeggios; harp glissando; marimba; conga (slap); chimes; vibes; timpani. Factory keyboard setups include: choir, cathedral choir; harp/slow choir; timpani/ choir; timpani/ harp; oboe/chimes; digital chimes; flute; woodwinds & reeds; marimba; conga & marimba; vibes; clarinet; clarinet & oboe; electric bass/slap bass; dual electric bass; alien harp; piano/flute; guitar & flute; strings & flute; strings & oboe; dual electric bass/rock piano; piano & marimba; piano & vibes; rock and roll piano; cow piano; choir & percussion, and more.

MacAttach
Off-line storage and editing of sound files, keyboard and instrument setups, sequences. Apple Macintosh interconnection cable; (transfer rate: 5,670 cps); 3½-inch hardcase disk.

Suggested list price: basic K250 $12,970; Sound Modeling Program $1,995; Sequencer Memory Expansion (adds 4,600-note capacity to the base unit's 3,400-note capacity) $450; Sound Block Module A $1,995; MacAttach software and interface $195; stand $195; plexiglass music rack $75.

An Expander system is also available, and comprises a K250 without keyboard unit. Three versions can be supplied: a basic system ($9,980); base system plus enhanced instrument voices ($11,975); and a base system plus voices, sampling, Sound Modeling and Macintosh software ($13,970).

Kurzweil Music Systems, Inc.,
411 Waveley Oaks Road, Waltham, MA 02154.
(617) 893-5900.

crystal display. During the editing of an envelope, the value slider's on/off button just has to be activated for you to use the slider to adjust each segment's values. You can also type in numerical values via the keypad. The envelope can be adjusted further by inserting up to 255 segments.

To best understand the 250's envelope, I find it helpful to compare it to that of the Yamaha DX-7. Like the DX-7, the 250 has a separate cutoff level and rate adjust for each segment. There are also 12 different low-frequency oscillator (LFO) shapes for both vibrato and tremolo. When editing the chorus options you can choose chorus, echo, delay, flanging and full chorus effects, and then adjust the detune and/or delay amounts. Again, they can be typed in or adjusted via the front-panel sliders. There is also provision to adjust the 250's velocity sensing, and the filter for varying degrees of brightness. The velocity can be set to open the brightness of the filter, an effect that is very useful on brass and percussive instruments.

MIDI Control

MIDI (Musical Instrument Digital Interface) is well implemented on the 250. A front-panel switch labelled *Mode 1* turns the MIDI-control capability on and off. By pressing first *Edit* and then the *Mode 1* buttons, you can down-arrow through the other MIDI modes, such as Mono/Poly, Cycle Mode, Channel Assign, etc. (When used with the recently announced Expander System — in essence a 250 without keyboard unit —, the Cycle function enables cycling through the 250's 12 voices and the Expander's 12 voices for a 24-voice piano). The 250 comes with MIDI-In, -Out and -Thru jacks on the rear panel. MIDI control works exceptionally well, and makes the 250's 88-note, wooden-weighted keyboard ideal as a master MIDI controller. Triggering the 250 via MIDI input works equally as well. For those that don't care for a piano-type action, a DX-7 could be used as a master keyboard to trigger the 250.

The 250's *Sequencer* is configured as a 12-track sequencer with a total capacity of 8,000 notes of memory, each track holding a maximum of 12 voices. (Since only 12-voice polyphony is provided in the 250, a little careful planning is necessary.) The sequencer features its built-in mixer, which is useful for mixing the level of each track, so that the 250 could be recorded directly to a digital tape machine. The sequencer has a separate click output located on the rear panel, and can be synchronized to external sources such as 96, 64, 48, or 24 pulses-per-quarter-note clocks. It can also generate and retrieve its own sync tone to and from tape. All the MIDI provisions are available on the sequencer, meaning you can assign a track to a certain MIDI channel and have it play an external MIDI-equipped synthesizer.

One of the unit's nicer features is the *Loop in Record* capability, which enables you to pre-determine the length of a sequence, and then loop it to record like a drum machine. This feature is most useful in conjunction with the 250's drum samples, or when using your own with the sequencer. Each track can be quantized individually, the quantization being post performance. I found the quantization to be excellent, and capable of preserving the live feel by moving the start time of a note while keeping its duration the same — a function called "quantizing the event." The quantization range was from a half-note all the way down to 1/256th of a beat; tempo is adjustable from 10 to 600 beats per minute. Full step editing is implemented, making possible step recording and erasing. Other modes of step editing include individual note velocity, pitch correction, and editing each beat and fraction thereof for each event.

The 250's sequencer editor can be used to go in on any track and modify that track's instrument setup file to produce great effects. For instance, on the piano track you could punch in and change the piano's decay envelope to be very staccato for the desired bars, and then punch out. The sustain pedal or pitch could also be modified at any desired section, or for the whole track. With this flexibility you can create amazing sequenced performances, the simplest of which would be a drum sequence for improvising or composing.

Kurzweil has contracted Southworth Music Systems, Inc. (creators of Total Music, a MIDI recorder software package for the Apple Macintosh) to write a special version for the Kurzweil 250. The soon-to-be-released software package will allow on-screen editing of sequences written graphically; notes on staves editing; music scoring and printing; and the ability to read external sync.

I like the Kurzweil 250 digital synthesizer. In its fully-blown state, with all options and Macintosh computer included, the 250 represents a medium-priced sampling keyboard and a most powerful studio instrument. It is good to see that the manufacturer has supported the current updates, and comforting to know that there will future updates as well.

■■■

SANKEN CU-41 DUAL CAPSULE STUDIO CONDENSER MICROPHONE

Reviewed by
Lowell Cross
Professor of Music, University of Iowa School of Music

The most distinctive feature of the Sanken CU-41 studio condenser microphone is its use of two capsules mounted one above the other, whose outputs are combined electrically in a cardioid-only configuration. The Japanese manufacturer's design premise is an innovative approach to the problems and tradeoffs encountered in the choice of capsule size. In theory, large-diaphragm condenser microphones have superior output versus noise characteristics, at the expense of high-end "colorations" caused by frequency-and phase-response aberrations in the region defined by the wavelength of sound corresponding to the capsule diameter. Smaller capsules, while allowing these resonances to be placed out of the audible range, have less output for a given sound-pressure level. Greater demands are therefore imposed on the system electronics for low-noise performance, both within and following the microphone.

By utilizing the high-output, low-noise capabilities of the larger of the two diaphragms over most of the audio spectrum, and by crossing over to the smaller element for the extreme high-end, Sanken engineers appear to have successfully accomplished their design goal. Also, to judge from published specifications, calibration curves supplied with the microphones, and our own listening tests at the University of Iowa School of Music, the engineers seem to have dealt with the potential problem of response deviations occurring at the wavelength determined by the distance between the transducers.

The CU-41 is an elegant example of mechanical craftsmanship. The finish of its satin nickel-plated brass case and stainless-steel mesh screen is directly comparable to that found on the well-known German and Austrian studio microphones. Though physically large — 180mm (7.1 inches) long by 50mm (2 inches) in diameter — the unit's shape and general appearance are aesthetically pleasing. The three-pin XLR-type output terminals are gold plated; standard DIN 45 596 48-volt powering is employed. Optional accessories include dual power supplies, elastic shock mounts or conventional stand adapters, and a range of cables. The CU-41 specification sheet states that cable runs as long as 400m (1,200 feet) are possible without signal degradation up to 20 kHz.

This single-pattern microphone is clearly intended for "purist" applications. One can appreciate the difficulties in providing variable patterns and capsule attenuator pads in the two-element design; the engineers also have chosen to dispense with any of the low-frequency equalization capabilities commonly found in competing studio microphones equipped with directional patterns.

Given these obvious and deliberate restraints, we are left with a first-class microphone with excellent sonic and dynamic-range characteristics. In our most recent evaluation session, recorded on May 13, 1985, a stereo pair of the Sanken CU-41s was recorded simultaneously along with pairs of many fascinating and prized "vintage" tube microphones (AKG, Neumann, and Schoeps), contemporary FET models from these same manufacturers, as well as Milab VIP-50 (FET) and Coles 4038 (ribbon) models. A full report on all of these microphones, old and new, is being prepared for a future issue of *R-e/p;* for purposes of this present review, the CU-41 has been compared directly to three of its closest rivals: the AKG C414EB/P48, the Milab VIP-50, and the Neumann TLM170.

The recording and listening sessions for this report have followed the procedures described in the April 1984, December 1984, and February 1985 issues of *R-e/p:* the microphones were set up as identically as possible in a reverberant concert hall; a near-coincident cardioid technique was used to record vocal and piano music in stereo (each pair occupying two tracks on a 24-channel, 15 ips ANT Telcom noise reduction master); and the microphones were directly compared during playback sessions using control room monitor loudspeakers and AKG K-141 stereo headphones. However, two significant changes were made over previous evaluation sessions: a greater variety of music was recorded and subsequently auditioned, and a pair of Klein and Hummel O92 monitor loudspeakers has replaced the previous JBL Model 4320s.

Conditions of temperature, humidity, and atmospheric pressure in Clapp Recital Hall on May 13 of this year were well within "normal" limits: 20°C (68°F) ±2°, 50 to 60% humidity, and 29.86 inches of barametric pressure. Each of the four types of microphones required about the same amount of pre-amplifier gain on the Neve console used throughout the sessions: 40 to 45 dB. The C414EB-P48, VIP-50, TLM170, and CU-41 microphones all exhibited extremely low noise and wide dynamic-range characteristics.

In addition to the Mozart and Gershwin songs faithfully performed again for us by Carol Meyer, soprano, and Patricia Cahalan, pianist, we recorded other musical combinations and works graciously provided for us by faculty and students at The University of Iowa School of Music: Beethoven: *Sonata No. 9 in A for violin and piano* ("*Kreutzer*"), op. 47; first movement, Professors Leopold LaFosse and Kenneth Amada; Beethoven: *Sonata No. 21 in C* ("*Wald-*

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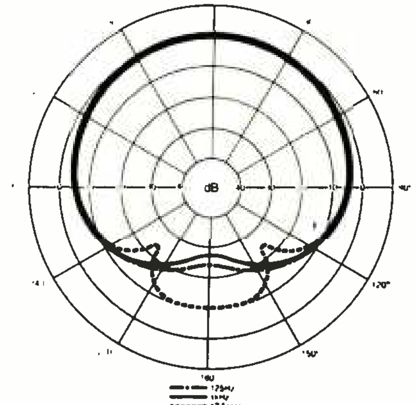
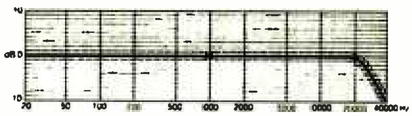
SUMMARY OF SANKEN CU-41 STUDIO MICROPHONE SPECIFICATIONS

Transducer Type: Two-way, condenser capsules.
Directional Pattern: Cardioid.
Frequency Response: 20Hz to 20 kHz, ±1 dB.
Sensitivity: at 1 kHz for 74 dB SPL, 0.7 mV.
Nominal Impedance: 150 ohm or less.
Recommended Load Impedance: 600 ohm or higher.
Equivalent Noise Level (A weighted RMS, IEC 179): 15 dB or less (0 dB = 0.0002 dynes/cm^2).
Maximum SPL for 0.5% THD at 1 kHz: 134 dB; 1.0% THD at 1 kHz: 140 dB.
Connector: Gold plated three-pin XLR-type: pin #1 is ground; pin #2 is audio (positive); pin #3 is audio.
Supply Voltage: 48, ±6 V phantom.
Current Consumption: 4.2 mA.
Dimensions: 180 mm by 50 mm (7.1×2.0 inches).
Weight: 582 grams (1.3 pounds).
Optional Accessories: S-41 shock absorbing stand adaptor for use with floor stand or microphone boom (recommended for recording low frequencies); AD-41 stand adaptor; P-41 power supply; SC-F or SC-M microphone cable assembly in varying lengths of cable.
Price: Suggested list price is $1,495, complete with S41 Shock Mount.
Manufacturer: Sanken Microphone Co., Ltd., 2-8-8 Ogikubo, Suginami-ku, Tokyo 167, Japan.
Export Agent: Pan Communications, Inc., 5-72-6 Asakusa, Taito-Ku, Tokyo 111, Japan. 03-871-1370.
U.S. Distributors: Martin Audio Video Corp., 423 West 55th Street, New York, NY 10019. (212) 541-5900.
Studio Supply Co., Inc., 1717 Elm Hill Pike, Suite B-9, Nashville, TN 37210. (615) 366-1890.



Polar Pattern



Frequency Response

stein"), op. 53; first movement Prof. Amada; and J.S. Bach: "*Contrapunctus IX*" from *The Art of the Fugue*, arranged by John Glasel for brass quintet Student Ensemble.

The comments and opinions that follow are entirely those of the reviewer, and, since they can never be completely avoided, subjective elements must be acknowledged in this process. (Results of the "ratings" from a panel of listeners as they relate to these microphones will be included in a subsequent report.) The Sanken CU-41 acquitted itself admirably against the formidable competition from AKG, Milab, and Neumann. With noise and dynamic-range specifications approaching theoretical limits in all four microphones, remaining factors relate to cost, versatility, and the elusive properties of "sound quality" that result from frequency response, phase response, pattern integrity, off-axis response, etc. Each of the four studio microphones exhibits some form of individual "coloration," to a greater or lesser degree. The eventual selection of microphones from this group will most likely come down to a determination of which quality of "coloration" can benefit or enhance a particular musical situation or acoustical environment.

The CU-41 is just slightly "crisper" in the extreme high-frequency region than the other three; the contribution of the smaller second capsule seems evident here. These impressions were formed from the reproduced quality of vocal sibilants, upper string harmonics, and the attack transients of the nine-foot Steinway piano. Also, the CU-41's handling of hall reverberation revealed this same "crispness" a bit more than the other microphones, indicating good off-axis and pattern characteristics in the high-end of the spectrum.

On the other hand, its low frequencies (as heard in certain piano passages and from the tuba in the brass quintet) are somewhat less "full" than those of the Neumann TLM170, for example. To continue the comparison of the CU-41 to the TLM170, one could say that the latter has a slightly more "velvety" quality in the midrange and highs. The "crisp" versus "velvety" qualities of these two microphones was evident while auditioning vocal and violin sounds, but the very slight differences were subtle indeed. These perceptions would seem to tip the scales by a very minute amount to the TLM170 in the "warmth" department, for what that is worth. But, expressed in a different way, the TLM170 could be characterized as having more low-frequency output than the CU-41, with a slightly recessed or withdrawn upper-midrange response, and perhaps offering a bit less extended extreme high-frequency response. Yet the CU-41 and the TLM170 sounded more alike than any of the microphones reviewed here.

The AKG C414EB/P48 is a strong competitor. Again, the main differences between its sound properties and those of the CU-41 are found in the higher frequency regions. The C414's upper-midrange response is more pronounced than that of the CU-41, while its extreme top-end does not appear to be as extended. By comparison, the C414 has a "confined" or "contained" quality. My perception of "containment" here is not necessarily a critical one — the C414EB/P48 could be a good choice for enhancing a given recording situation where a "tighter"-effect is desired. With its four polar patterns (including hypercardioid), attenuation and equalization capabilities, and relatively unobtrusive and elegant appearance, the C414EB/P48 microphone is quite versatile — and not nearly so "bright" as many earlier AKG models.

Since the Milab VIP-50 is even more expensive than the Sanken, one should expect great things from this Swedish import. The VIP-50 is a worthy competitor in all respects except for three: cost, physical appearance, and a rather destractingly "bright" sonic quality. (I have been assured by the importer of Milab microphones that our evaluation models were pre-production units of a design that is likely to evolve further, but I do feel entitled to expect a finish other than black crinkle paint and even perhaps a *pleasing* — yet distinctive — shape for a microphone as expensive as this one.)

It was difficult to judge just how extended the high-frequency response of the VIP-50 may actually be, owing to the emphasis in the 5 to 10 kHz region. I suggest that this "bright" characteristic should be brought under greater control before full production is undertaken. These early examples of the VIP-50 simply have too much of a "condenser sound" for my liking, especially since they are being introduced at a time when most other condenser-microphone manufacturers are striving for as uniform frequency response as possible.

In one category, the CU-41 and the VIP-50 are directly comparable: they are quite expensive (around $1,500 each). The TLM170 costs about one-third less: and *two* C414s can be bought for about the price of *one* CU-41 or VIP-50. As we have noted, the Sanken CU-41 has no switchable pattern, attenuation, or equalization settings; the other three microphones are all very versatile in these categories. Like the C414EB/P48, the CU-41 apparently contains an output transformer, while the VIP-50 and TLM170 are transformerless. The VIP-50 has an additional valuable feature: it may be switched between a microphone-level or a line-level output circuit, both electronically balanced.

The Sanken CU-41 microphone represents an uncompromising concept based on a single unidirectional pattern, and therefore it should appeal to "purists" and "minimalists" of the cardioid persuasion, just as the Bruel and Kjaer 4000 (omnidirectional) and Schoeps MK41 (hypercardioid) designs have attracted their respective loyal followings.

I am indebted to Jan Hebel of Martin Audio Video Corporation, in New York, for his assistance in arranging the loan of the Sanken microphones for this evaluation.

dbx MODEL 166 DUAL-CHANNEL DYNAMICS PROCESSOR

Reviewed by Roman Olearczuk

The newest addition to the company's line of compressors/limiters, the Model 166 Dual Channel Dynamics Processor, combines stereo or dual-mono operation with some familiar features, such as OverEasy Compression, PeakStop Limiting, and Sidechain control, in an aesthetically-pleasing and low-cost package. In addition, an adjustable Noise Gate has been incorporated into each channel as a new function.

This hands-on report discusses the operation and usefulness of these features, and hopefully will provide the reader with insight towards the benefits and cost trade-offs designed into this device.

Input/Output Characteristics

Unfortunately, the Model 166 does not provide XLR-type input/output terminations. Instead, PCB-mounted, three-circuit (or stereo) ¼-inch phone jacks are used, apparently to reduce manufacturing costs. Inputs are balanced electronically, with differential amplifiers having an input impedance of 25 kohms. Outputs are unbalanced, and use line amplifiers to drive loads greater than 600 ohms. (Oddly, neither the unit's specification sheet nor operational manual provides a value for actual output impedance; both sources only state that the output impedance is "low.")

Each channel also has a Sidechain Input leading to a signal-detector circuit for external control of the compressor and gating. The Sidechain Input is unbalanced and has an impedance of 6.8 kohms; termination here is a two-circuit (or mono) ¼-inch phone jack. To help avoid any damaging mistakes in wiring, dbx wisely screened circuit connection symbols on the rear panel, next to the audio and sidechain jacks, which identify the tip, ring and sleeve designations.

The operational manual provides a thorough explanation of balanced and unbalanced hookups. For maximum hum rejection, dbx recommends that the user avoids common grounding of the Model 166's inputs and outputs. Instead, the manual suggests: "The best starting point is to ground the shield of the input cable and the source device (leaving it unconnected to the 166) and to ground the shield of the output cable to the ground of the 166 (leaving it unconnected at the receiving device)." For balanced sidechain hookups, the company states that most balanced sources will work properly without having to connect the low or minus side of balanced signal to the circuit ground (sleeve) of the detector circuit. However, it does offer one word of caution: "Some sources require the dotted connections shown (see Figure 1) — 'transformer-isolated' balanced outputs. We recommend making this connection *only if necessary* for your installation, because some active balanced and ground-referenced outputs may be damaged by doing so."

The quoted maximum input level to

Figure 1: Balanced/unbalanced connection scheme.

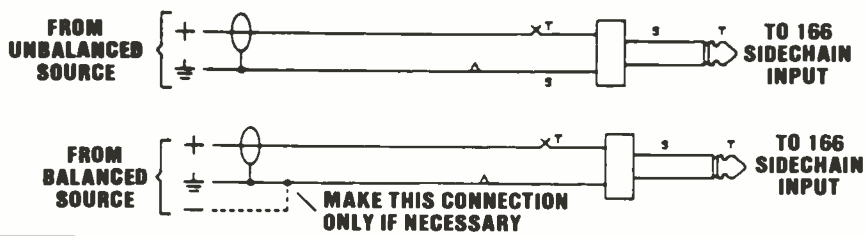
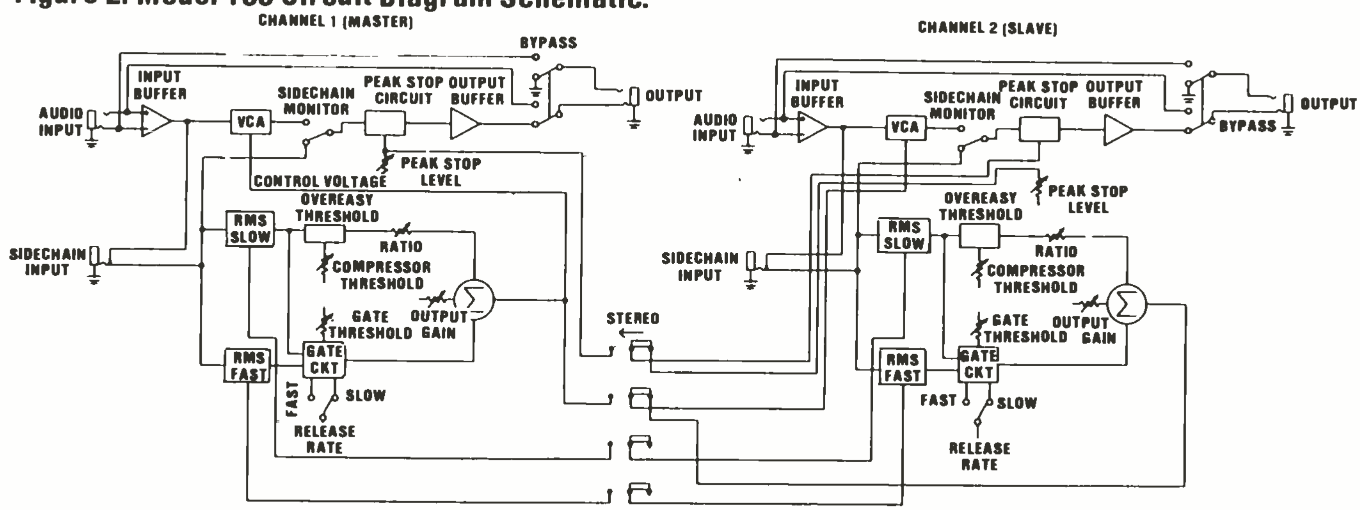


Figure 2: Model 166 Circuit Diagram Schematic.



In the early morning hours of November 15, 1984 tragedy struck the Bethany Lutheran Church of Cherry Hills, Colorado. A faulty electric organ was blamed for a multiple alarm fire that claimed much of the structure. Thankfully no one was injured in the blaze that caused over one million dollars in damage.

In the ensuing clean-up operation a Crown amplifier was discovered under charred timbers. Owing to the intense heat of the fire the chassis had warped and the AC cord was a puddle of wire and rubber.

The amplifier found its way to John Sego at Listen Up, Inc. of Denver. Armed with insatiable curiosity and a knowledge of Crown dependability John installed a new AC cord and proceeded to verify operation on the test bench. The amplifier met factory specifications in all functions.

In the photo above we offer you another glowing report of Crown durability.

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For additional information circle #282

the device is +24 dBm, and the maximum output level from the device +21 dBm. Each channel features a hardwire bypass switch for easy comparison of the processed signal to the original sound.

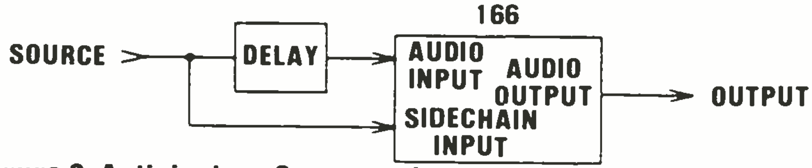
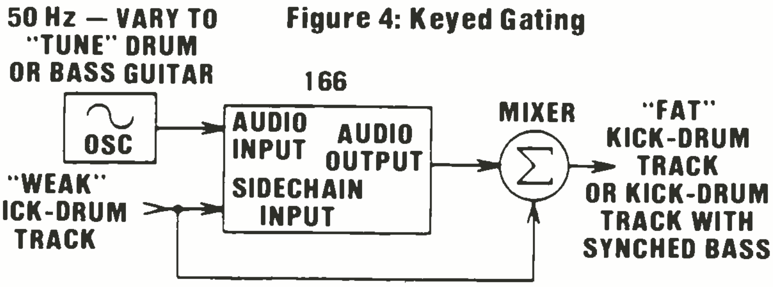


Figure 3: Anticipatory Compression.

Figure 4: Keyed Gating



Controls Description

Each channel has the following process controls per channel on the front panel: Gate Threshold control with Off position, and Gate Fast/Slow release rate (labelled Rel Rate) switch; Overeasy Compressor Threshold and Ratio controls; Peakstop Level control; Sidechain Monitor in/out switch; Output Gain control; and a Bypass in/out switch. Indicators include: Gate On, Gain Reduction (via an eight-segment LED display), Sidechain Monitor On, and Bypass On. A Stereo Couple in/out switch enables the controls for channel #1 to serve as the master for both sections stereo mode; in this situation the slave channel #2 controls, except for Sidechain Monitor and Bypass, are disabled. An LED indicator provides visual confirmation of this configuration.

The noise gate features two front-panel controls: Threshold and Rel Rate. Gate Threshold is variable from -60 to +10 dBm, and the user can disable the noise gate by turning this control counterclockwise to the Off position; an LED lights whenever the Peakstop Level control sets an absolute limit on final output peaks, and is user-adjustable from 0 to +22 dBm. The manufacturer recommends that the control be set one to two dB below the chosen maximum level, to allow some headroom for signal "rounding," a process that reduces the higher-order harmonics found in conventional clipper circuits. An intensity-calibrated LED glows from dim to bright as the output signal further exceeds desired Peakstop Level settings.

The Sidechain Monitor switch (when enabled) directly connects the Sidechain Input to the Audio Out, a feature that allows the user to monitor the sidechain signal during setup. An LED verifies selection of this mode. The Bypass switch and confirming LED provide a hardware bypass, connecting Audio-In directly to Audio-Out. Balanced circuits will follow through this switch even in the absence of AC power.

Operational Comments

The dbx 166 Dual-Channel Dynamics Processor is packed with a lot of features, especially when one realizes the recommended retail price is only $549. Technology like the OverEasy Compressor and PeakStop Limiter, originally developed for the dbx Model 165 models, has been incorporated in this efficient and economical package. All controls were smooth and easy to read, even in a dimly lit studio control room. Control voltages, not audio signals, pass through these front-panel potentiometers, thereby eliminating any potential "noisy pot" problems in the future. The unit was found to be quiet in operation, and did not add any appreciable noise when inserted in a console module signal path.

As an application, the Model 166 was used to compress several mono noise gate shuts the signal off. The Rel Rate switch provides a fixed 10 dB per second release rate in the Slow mode, and a fixed 1,000 dB per second rate in the Fast mode. The gate attack time is internally set to two milliseconds for 28 dB signal rise, while the gate attenuation is constant at 40 dB.

The OverEasy compressor section is provided with six front-panel controls: Threshold, Ratio, Output Gain, Peakstop Level, Sidechain Monitor, and Bypass. The Threshold range is adjustable from -40 to +20 dBm, while compression Ratio can be varied from unity (1:1) to infinity-to-one (output then being constant, irrespective of input dynamics). Maximum compression is greater than 60 dB. Attack and release times are program dependent, and have been factory set.

The Output Gain control ranges from -20 to +20 dBm and occurs prior to the Peakstop Level circuit (see circuiting diagram of Figure 2). The mixes during audio layback from an Ampex ATR-124 to a Sony BVH-1100 one-inch videotape machine. The noise-gate feature worked quite well, considering the fact that only the threshold level is adjustable and just two fixed release rates are available. For some reason, a section of new Scotch 250 tape (which normally provides satisfactory results) developed a nagging print-through problem, a factor that was very noticeable at the start of each mix. The Model 166's gate immediately cleaned up this objectionable noise, with only a minor threshold adjustment.

The unit's OverEasy Compressor section worked just as expected; audi-

SUMMARY OF dbx 166 STEREO DYNAMICS PROCESSOR SPECIFICATIONS

Frequency Response: 20 Hz to 20 kHz, ±0.5 dB.
Total Harmonic Distortion (THD): 0.1% at maximum compression, 1 kHz at 0 dBm.
Output Noise: -85 dBv A weighted, 20-kHz bandwidth, unity gain (-87.2 dBm).
Crosstalk: 70 dB.
Maximum Input: +24 dBm.
Maximum Output: +21dBm.
Input Impedances: 25 kohms differential; 18.5 kohms unbalanced; detector: 6.8 kohms, unbalanced.
Output Impedance: Low, single-ended, designed to drive 600 ohms.
Output Gain: -20 to +20 dB.
Threshold Range: Compressor: +20 to -40 dBm; gate: -10 to -70 dBm; peak stop: 0 to +22 dBm.
Release Rates: Compressor: 125 dB per second; gate: 10 dB per second slow, 1,000 dB per second fast.
Gate Attack Time: 4 milliseconds for 66%.
Gate Attenuation: 40 dB.
Power Requirements: 90 to 135 volts (117-V model); 200 to 260 volts (220-V model) 50 to 60 Hz at 15 watts.
Dimensions: 1¾ × 19 × 8 inches (H × W × D).

dbx Professional Products Division, 71 Chappel Street, Newton, MA 02195. (617) 964-3210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When you hear the fidelity and accuracy of the AKG K 240DF Studio Monitor Headphones, you'll know why it's become a standard for recording engineers and professional musicians around the world.
This latest version of our well known K 240 (now K 240M) has been created to meet a recently proposed IRT (Institute for Broadcast Technology) international standard. The K 240DF establishes a uniform sound quality free from environmental variables. As opposed to sound from loudspeaker monitors, that is colored by variations in control room design, the K 240DF is unchanging and reliable.
Each K 240DF is tested in a diffused sound field to arrive at a headphone design with a flat frequency response (±2dB) and matched sensitivity. This professional headphone is close to perfection—without coloration or distortion. The self-adjusting headband supports the circumaural ear cups, each containing hand selected, large dynamic moving-coil transducers and acoustic filters. Minimum weight is well distributed for maximum comfort over longtime wear.
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ble results were just as smooth sounding as the output from the more expensive Model 165A (mono $699; stereo $1,398 — *Editor*). The Gain Reduction meter presented a good indication of gain-reduction action, although it is a poor substitute for the accurate three-mode VU meter featured on the Model 165A. It takes some time to get used to its readings, since the eight LED segments are arranged in sort of a descending logarithmic pattern, from 1 dB to 30 dB. The same meter also doubles as in indicator of noise-gate attenuation. Therefore, to properly measure compressor gain reduction, the noise gate must be turned off; otherwise, a combination reading of gating and compression is always present.

Sorely missed are the three-segment Threshold LEDs present on both the Model 160X and 165A — these useful indicators would provide a visual representation of the action below and above the compression knee. (Perhaps they had to be eliminated for lack of front-panel space.)

The Output Gain and Peakstop Level controls were quite accurate, and performed as expected. The sound of the peak limit or clipper circuit was free of harshness, and its indicator represented the limiting action correctly. Setting Peakstop to its maximum setting effectively put it out of the signal path; the loudest transients did not once trigger the limiter on.

Although the side-chain function was not tried, the unit's Operational Manual does provide some useful application ideas, including de-essing, broadcasting, anticipatory compression, keyed gating, and selective gating. For de-essing application, the manual recommends: "In the absence of a de-esser, small amounts of high-frequency (6 to 10 kHz) boost in the side-chain path frequently will help in the processing of vocals that may have been brightly equalized beforehand, or that may suffer from prominent sibilance ('ess' sounds)." For broadcasting: "A pre-emphasis filter network placed in the sidechain of 166 processing pre-emphasized audio permits higher average signal levels to be run within the headroom limits of the broadcast chain."

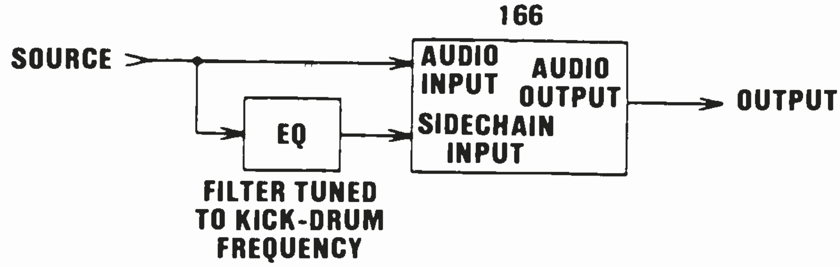


Figure 5: Selective Gating.

For anticipatory compression (shown in Figure 3), the manual states: "If you feed the program directly into the sidechain and send the audio signal through a delay before the 166 audio input, the 166 can 'anticipate' the need for gain change . . . Such a special effect sounds similar to the dynamic-envelope inversion you may be familiar with from reverse tape playback."

For keyed gating (Figure 4): "Controlling the gating of one signal by another permits perfectly in-sync playing and overdubbing among individual instruments or precise sonic augmentation — 'fattening' — of a weak solo . . . An example of the latter would be using the drum signal to key an oscillator which is set at an appropriate frequency to 'tune' and 'punch up' the drum sound."

For selective gating (Figure 5): "You can also do frequency-sensitive gating [to] tune the response of the gating action. If you're gating a kick drum, for example, in a track with lots of leakage, you can tune in to the frequency of the kick with an EQ and the gate will respond only to the drum."

In spite of all the favorable features included in this compact unit, there are some packaging drawbacks. Curiously, no power on/off switch of indicator is provided. Perhaps dbx felt that since the unit only draws 15 watts (and post-oil-embargo power is again cheap) Model 166 owners wouldn't mind a little inconvenience for some additional product-cost savings. Also, there are no plug-in replaceable fuses (external or internal). A close examination of the easily accessible circuit board, however, did reveal soldered fused resistors leading from the +15 VDC power-supply rails to the remaining circuitry. Finally, the top and bottom panels flex easily under certain placement pressures. While for rack-mount applications this flexing does not present a problem, stacking several external devices on top of the Model 166 could possibly crush some of the vulnerable circuit components mounted inside.

Summary

The dbx 166 Dual-Channel Dynamics Processor offers excellent value for the price. Overall the unit performed as promised, equaling the performance found in more expensive products. Hopefully, the manufacturer will take into consideration some of the criticisms mentioned here when they design future new products. The provision of ¼-inch phone plugs might be all satisfactory for "semi-pro" equipment, but XLR-type connectors should be offered as an option for professional audio users. Also, the cost of a fuse holder and power on/off switch would not add much to the bottom line, yet the convenience to the new user would be greatly increased. ■■■

MANUFACTURER'S COMMENTS

from: **Scott Berdell, Director of sales and marketing, Professional Products Division, dbx, Inc.**

In the design process for the dbx 166, every effort was made to both maintain the highest possible sonic quality with necessary control features, *and* to offer a *very* cost-effective tool for musicians, sound-reinforcement and recording engineers, and home recordists in a simple, easy-to-use package.

Given the design goals of the 166, the queries raised by the reviewer may be easier to understand. The lack of XLR connectors is based on our desire to appeal to the widest possible range of users. We employed ¼-inch "middle ground," since XLR-to-¼-inch and RCA-to-¼-inch adaptors and cables are readily available.

The absence of a power switch and the internal fusing are related to our power supply design. The 166 is designed to be rack-mounted and connected to the rack's main power strip; since the 166 does not have a turn-on "thump" or transient, a power switch is not necessary in this case.

Basically, there are two reasons for fuses: protection of the user (safety); and protection of the unit. The power supply of the 166 is designed to absorb very large voltage swings in the main supply. The transformer is equipped with a thermal fuse for user safety. One should be looking for problems in the power line, not the 166, if the thermal fuse in the transformer blows.

The absence of metering is due to a simple constraint — lack of space on the front panel. It should be noted that the combination of LEDs on the front panel allow for a complete set-up of the 166.

The noted flexing of the top panel is due to the choice of cosmetic construction materials for the chassis. The top and bottom are made of vinyl-clad rolled aluminum, and are reasonably costly. Since the unit is designed to be rack-mounted, we did not consider extensive use of the unit in stacked equipment to be a common application. We therefore felt that the very limited flexing was not a major problem in the design of the unit. We will be monitoring this possible use closely for future modifications.

Overall, dbx believes that the design goals for the 166 were successfully achieved to provide users with limited budgets a low-cost compressor/limiter in the dbx tradition for sonic excellence. □□□

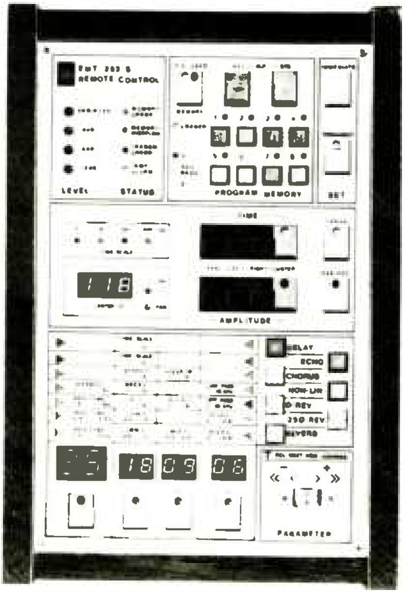
New Products

EMT 252 DIGITAL REVERBERATION SYSTEM AVAILABLE FROM GOTHAM AUDIO

The new system is said to provide very natural reverberation through digital processing using high-resolution 16-bit analog/digital conversions and a 32-kHz sampling frequency. In addition to generating reverberation effects, the unit provides three delay-based effects: straight delay, loop echo and chorusing effects.

The reverberation program provides up to nine individual reflections before reverberant signal, and individually adjustable time and amplitude of reflections. Frequency response of the system is adjustable in four separate bands. Reveberation is generated using the entire audio frequency range of the source signal, resulting in a very natural sounding output. Four reverberation algorithms are implemented in the Model 252: a main reverberation program; a reduced bandwidth (EMT 250) program; a Doppler-shifted program; and a non linear decay program.

The delay mode provides up to 480 milliseconds of delay for three individual taps, plus a "cluster tap" of six individual taps in fixed relation to one another. Echo mode provides four "loops" of up to 440 milliseconds of delay, with feedback amplitude adjustable on each. The chorus mode provides up to four voices from one input source, with control of depth and rate of the chorus effect.



The processing system is housed in a 19-inch rack mounting enclosure, with control of all functions and display of all settings appearing on a separate eight- by 12-inch remote, which can be placed up to 300 feet from the processor.

The remote provides 128 memory presets for each of the reverberation and effects programs: half are preset at the factory, and half programmable by the user.

Presets are stored in battery-backed memory: 16 fixed and 16 user presets for the main reverberation program, and eight fixed and eight user presets for each of the six other reverberation and effects programs. Single-button recall of random or sequential presets is provided, as well as means to copy settings form one preset to another with desired modifications. Factory-fixed settings can be converted to user-modifiable settings, optionally allowing all 128 memory locations for user presets.

The EMT 252 digital reverb system has a professional net user price of $16,500.

GOTHAM AUDIO CORPORATION

For additional information circle #287

SYMETRIX MODEL 544 FOUR-CHANNEL NOISE GATE AND EXPANDER

Designed specifically for profesisonal studio and live-performance applications where distortion-free gating is mandatory, the 544 is said to offer a maximum amount of processing power. The new unit ecncloses four channels in 1¾ inches of rack space while, at the same time, providing a full complement of user-variable expander/gate controls including attack time, release time, range/ratio, and threshold.

Each channel can be set to trigger internally, or keyed from external input signals for special effects. Gate mode response has been optimized for highly transient material such as drums and percussion. The downward expander is described as being exceptionally linear, and doubles as an expander and noise reducer. Intelligent automatic time control circuitry works in conjunction with the manual release time control to eliminate low-frequency distortion in both gate and expand modes.

In addition to its variable parameter and mode-select controls, each channel provides the user with a five-segment LED gain-reduction display for visual indication of the unit's performance.

Suggested retail price of the Model 544 is $549.

SYMETRIX

For additional information circle #288

NEW STUDIO DOMINATOR THREE-BAND LIMITER FROM APHEX

The Studio Dominator™ is an intelligent three-band limiter with a proprietary circuit that varies the threshold for limiting, unlike traditional "dumb over-threshold" devices. A unique Transient Enhancement Circuit is said to actually increase the perception of transients, while maintaining absolute peak limiting.

Tuneable crossover frequencies, plus high- and low-frequency drive controls allow the

"We're Committed To Building A Quality Product That Really Works..." "We're Committed To Being Number One."

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Vice President/Engineering,
QSC Audio.

Barry Andrews
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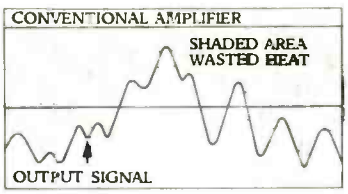
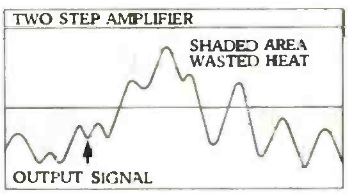
QSC Audio.

Commitment runs deep at QSC. We're dedicated to continually improving our products and our company. For us, building a better product and backing it up with top-notch customer support is the key to success. It's as simple as that. The QSC linear output circuit is one outcome of our commitment to design excellence. Its three-stage signal path optimizes the sonic advantages of traditional push-pull amplifier circuits. By combining a multiple level DC power supply with conventional power transformers and rectifiers, we've improved on previous efforts at increasing heat efficiency–anticipating the benefits of "Class D" and "smart power supply" amplifiers, without relying on unproven technology. This has enabled us to build a power amp that is more compact and reliable, and which delivers unmatched audio performance. The diligent research that went into our Series Three paid extra dividends in the development of our economical Series One amplifiers. Both series feature our patented Output Averaging™ short circuit protection, dual isolated power supplies, calibrated gain controls, premium components throughout, and complete rear panel connection facilities that include balanced XLR and 1/4" jacks, octal sockets for active and passive input modules and a full selection of output connectors. Our dedication to design excellence goes hand-in-hand with our commitment to providing full-service support on all our products. When you put it all together, QSC amplifiers reflect the commitment to leadership, service and design innovation that has guided us since we were established in 1968. For more information contact: QSC Audio Products, 1926 Placentia Avenue, Costa Mesa, CA 92627, (714) 645-2540.





QSC Quality Service Commitment

For additional information circle #289

New Products

user to create different effects. Limiting can be preshaped to match the medium's saturation characteristics for maximum signal-to-noise performance.

Because of its unique design, the new unit is described as being ideal for use in any situation where clipping is a problem, such as digital audio, disk mastering, video post production, and optical film. According to Marvin Caesar, Aphex President, the device could effectively eliminate watching the clash lights when mixing in Dolby Stereo for optical release, and will still allow a "transparent sound with the perception of natural transients."

Caesar also stated that the Studio Dominator is the perfect companion to the company's Compellor compressor/levelor/limiter, which maintains steady average level, without coloration. The Dominator is a peak processor that can be used to achieve different sounds and effects. The combination of the two units "provides the ultimate flexibility in dynamics control," Caesar offers.

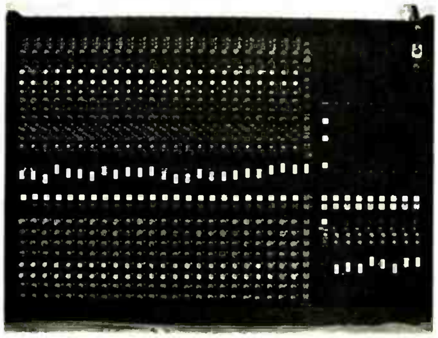
APHEX SYSTEMS LIMITED

For additional information circle #292

MEYER DISTRIBUTING JAPANESE ATL STAGE MONITORING CONSOLE

Meyer Sound Laboratories, Inc. will be distributing a mid-sized stage-monitor console that is a combined effort of Meyer and its Japanese distributor, Acoustic Technical Laboratory (ATL). The console will be available in limited quantities for users interested in high-fidelity stage sound.

The console configuration is 24-by-8, with an additional four auxiliary mixes. All 12 outputs have large LED metering that may be switched to VU or peak reading. Any of the 12 mixes may be re-assigned in any order to the eight main outputs via a fast electronic matrix assignment system.



Each transformerless input channel has switchable phantom power, a highpass filter, and four-band "true-complimentary" EQ. A switching system allows a master fader to control the send to the matrix. Insert points and direct outs are furnished for each input.

Monitoring solo points are at input, summing, and output stages, and peak indicators at each of these stages is said to provide distortion-free operation. Talkback can be assigned to individual outputs for improved musician-mixer communications. Two auxiliary inputs can be used to route effects to any output.

MEYER SOUNDS LABORATORIES

For additional information circle #293

BLACK AUDIO DEVICES LAUNCHES MICROPHONE STAND REPAIR AND REPLACEMENT PARTS

Swivel Levers replace the "dumbbells" on AKG-type booms. Original parts can be easily lost, leaving the boom virtually useless; yet factory replacements are virtually impossible to find. The Swivel Levers are a precision-machined replacement that, according to Black Audio, will neither fall off nor bend, like the factory parts.

Thread Strips enable the loose fit to be taken out of the threads on mike stands, booms, and accessories, and restore a like-new snug fit to threads that have become loose or stripped from age, long-term use, or damage. They come in packages of 12, and are available in four precision thicknesses.

Suggested list price of the Swivel Lever is

$6.00, and Thread Strips $2.00 per pack.

BLACK AUDIO DEVICES

For additional information circle #294

"HUM-KILLER" FROM HAASE NOW AVAILABLE IN U.S. FROM ESL

Until now, it has been customary to use notch filters or equalizers to tackle the annoying hum interferences that originate from ground loops, defective equipment, light dimmers, and the like. However, the use of such filters does not reduce the interference harmonics, while phase distortion and insufficiently narrow bandwidth negatively influences the sonic quality.

The Haase Hum Killer uses very narrow bandwidth filters (±3% of center frequency), not only for the bass frequencies but up to and inclusive of the 13th harmonic. Attenuation of the fundamental, even and odd frequencies is independently adjustable. The stereo unit is available for 60 Hz or 50 Hz fundamental frequencies.

Attentuation ranges are: 40 dB at 50 Hz to 25 dB at 650 Hz; or 40 dB at 60 Hz to 22 dB at 780 Hz. Phase shift is quoted to be less than 10 degrees between channels and frequency response throughout the passband ± 1.5 dB, 20 Hz to 20 kHz.

Front-panel functions include linear/filter selector (for A/B comparison); mute switch for even or odd harmonics filter; and attenuation controls for F1, F-even, and F-odd.

ELECTRONIC SYSTEMS LABORATORIES INC.

For additional information circle #295

RPG DIFFUSOR SYSTEMS UNVEIL QRD-734 MODULAR ACOUSTICAL DIFFUSORS

The new low-cost, space-saving model measures 23⅝ by 47⅝ by 8¼ inches, and weighs 30 pounds. The diffusors can be wall-mounted in clusters, providing horizontal and vertical diffusion, or ceiling-mounted in standard suspended grid systems.

The RPG Diffusor System is a new, formerly unavailable reflection phase grating room treatment that is said to enhance the acoustics of any critical listening or performing environment, by uniformly diffusing sound, without absorption or attenuation.

Each QRD-734 panel diffusor has a suggested pro-user price of $245; prices for a complete control-room installation, requiring 48 square feet or more, begin at $1,470.

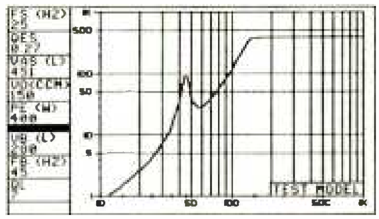
RPG DIFFUSOR SYSTEMS, INC.

For additional information circle #296

SOUND DESIGNER SOFTWARE FOR MACINTOSH CONTROL OF EMULATOR II SYNTHESIZER

The computer music system includes an interface that allows sampled sounds to be transferred between the Apple Macintosh and E-mu systems Emulator II at 500 kbits per second. Sound waveforms are displayed on the Mac's high-resolution screen. The software provides extensive sound editing capabilities, including Macintosh-style "cut-and-paste" editing.

The waveform display can be magnified to show extremely fine detail, with editing accuracy to 33 microseconds. Calibration scales provide exact readouts of time and amplitude values at any location in the waveform, and the waveform display can be horizontally and vertically scrolled.



Sound Designer also includes Fast Fourier Transform-based frequency analysis and modification of sounds; digital equalization; enveloping; digital mixing and digital compression; as well as a variety of other digital waveform processing functions for modifying sampled sounds and creating unique sounds.

Direct digital synthesis (including FM and waveshaping) can be performed on the Mac, and the resulting sounds transferred to the Emulator II for playback.

DIGIDESIGN, INC.

For additional information circle #297

New Products

NEW MODEL 450 EIGHT-INPUT MIXER FROM FOSTEX

Designed for both multitrack recording and sound reinforcement/live recording, the Model 450 features eight balanced mike inputs, each with phantom powering; in-line monitoring (so that tape returns do not occupy line inputs); solo on all inputs and four program busses; three-band parametric-type equalization; a dedicated stereo master bus; and accessible patch points on the top panel.

The eight input channels, each with a direct output, feed four main program busses which, in turn, are mixed to stereo on the master buss. The two auxiliary busses (one stereo, the other mono) are independent, and may be used in a number of pre-EQ, post-fader configurations. The monitor/headphone bus is also independent.

Suggested retail price of the Model 450 is $995.

FOSTEX CORP. OF AMERICA

For additional information circle #290

SANKEN UNVEILS FOUR NEW STUDIO CONDENSER MICROPHONES

• The CU-31 is a low-impedance, cardioid-pattern condenser model with a quoted dynamic range of 129 dB, frequency 0 Hz to 18 kHz, and self-noise of 19 dB or less. In incorporates a "Push-pull" DC-bias transducer element and a titanium membrane diaphragm. Quoted sensititity is 0.355 mV/0.1 Pa, and maximum SPL for 1.0% THD 148 dB. Supply voltage is 48V phantom. Dimensions are 113mm in length by 20.5mm in diameter.

• CU-32 is a right-angle cardioid model with a quoted dynamic range of 129 dB, frequency range of 20 Hz, and 19 dB or less self-noise. Apart from its length (117mm) and capsule orientation, the CU-32 is electrically and performance identical to the CU-31.

• The CMS-6 MS stereo condenser has a quoted dynamic range of 108 dB, frequency range 18 kHz, and a 19 dB or less self-noise. With the companion CMS-MBB battery PSU and switchable matrix, the output can be altered from MS to L-R. The mike utilizes "push-pull" DC-bias transdensers, and a titanium membrane diaphragm. Maximum SPL for 1.0% THD is a quoted 127 dB, and nominal impedance 150 ohms. Dimensions are 170mm in length by 40.5mm in diameter.

• The CMS-2 is a MS-type, single-point stereo condenser microphone that has a quoted dynamic range of 129 dB, frequency range of 20 Hz to 18 kHz, and self-noise of 16 dB or less. Like other Sanken models, the CMS-2 utilizes a push-pull DC-Bias condenser element with a titanium membrane diaphragm. Dimensions are 176mm in length by 43mm in diameter.

SANKEN MICROPHONE COMPANY

For additional information circle #302

SOUNDCRAFT TV24 BROADCAST PRODUCTION CONSOLE

Scheduled for unveiling to the U.S. market at the forthcoming New York AES Convention, the TV24 is an "in-line" master recording console that provides live stereo and mono mix with routing to eight stereo audio sub-groups, and is intended for TV and video production. A completely independent 24-track recording and monitoring facility also is provided.

Standard features include fader start on every channel; stereo equalizers for the audio sub-groups; fast status control that reconfigures the whole console at the touch of a button; and a comprehensive monitoring selection.

SOUNDCRAFT ELECTRONICS, INC.

For additional information circle #301

BROOKE SIREN SYSTEMS ADDS SOFT LIMITING TO DPR402 COMPRESSOR-LIMITER

The DPR402 normally provides a hard-knee compressor and de-esser transfer function. To reconfigure the unit for Soft-Knee simply requires the addition of a resistor on the rear barrier strip. This change can be made to one or both channels of the DPR402 independently, and does not affect the unit's other functions. Full technical details are available by contacting Jim Jacobelli at BSS: (516) 249-3660.

Suggested retail price in the U.S. for the DPR402 is now $1,095.

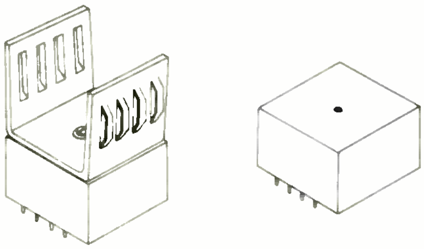
BROOKE SIREN SYSTEMS

For additional information circle #303

For additional information circle #298

SUMMIT AUDIO ANNOUNCES HIGH-RELIABILITY 990 OP-AMP

Better heat dissipation is said to be achieved with an aluminum radiating surface, tapped to accept a standard heat sink or to conduct to an outer surface.



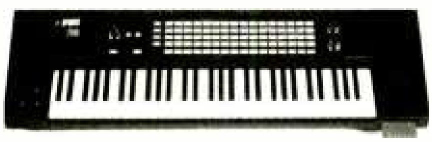
The op-amp comes supplied with gold pins in either ±15 or ±24 volt configuration.

SUMMIT AUDIO

For additional information circle #304

NEW PROPHET 2000 DIGITAL SAMPLING KEYBOARD FROM SEQUENTIAL

The new eight-voice sampling instrument enables the user to sample any sound using features that include a variable input-level control, complex sample editing (reverse, mix, truncate), and automated looping functions such as computer-assisted zero crossover and zero slope selection, to help find the best possible loop points. A built-in 3½-inch disk drive provides for fast and easy storage of custom sounds and programs. A large selection of pre-recorded sounds are available from the company's library of factory disks.



The Prophet 2000 also features multiple wavetables stored in on-board memory for building "traditional" synthesizer sounds. Such sounds can be played alone, or in conjunction with sampled sounds by splitting the keyboard or layering sounds on top of each other.

A velocity-sensing, five-octave keyboard is said to provide precise control over loudness, modulation amount, timbre, sample start points and crossfading between two separate sounds; its weighted action is described as responding positively to every nuance of the player's individual technique. The keyboard can be split, with different sounds assigned to each half, or with multiple sounds layered on top of each other. Up to 12 keyboard combinations can be created, with instant access of up to 16 sound variations. The Prophet 2000 also features pitch and programmable modulation wheels, as well as arpeggiation capabilities that include programmable up, down, assign, extend, auto-latch, and transpose modes.

MIDI implementation includes full support of Modes 1 (poly) and 3 (omni), receivable in Mode 4 (mono), Clock In and Out, pitch and modulation wheels, main volume, switchable MIDI Thru and second MIDI Out, complete wavetable access/programming via an external computer, and the ability to transmit and receive sound samples over MIDI.

The sampling capability is based on 12-bit digital resolution. Samples with a duration of 16 seconds have a bandwidth of 8 kHz, with eight seconds having 15 kHz, and six seconds having 20 kHz. The user may sample at rates of 15.625 kHz, 31.250 kHz, and 41.667 kHz, up to 16.

SEQUENTIAL

For additional information circle #305

DENECKE ANNOUNCES DCODE TC-1 TIMECODE READER

The Dcode™ TC-1 is designed as a low-cost timecode reader for general film and video applications. In film, using timecoded film dailies, editors can use the unit to assist them in syncing dailies, making high-speed searches, logging, and keeping accurate time-date records of the actual production.

The TC-1 reades SMPTE or EBU timecode from 0.1 to 15 times speed, in both forward and reverse, from VTRs, VCRs, film editing machines and film synchronizers.

For transferring ¼-inch tape to mag film, the unit reshapes codes for film-to-tape transfers, and simultaneously displays code while generating 60 Hz sync pulse at 24 and 30 fps.

DENECKE, INC.

For additional information circle #306

New Products

NEW WESTREX FILM RECORDER PRODUCTS

Following Mitsubishi's acquisition of Quad Eight/Westrex, the company is focusing on the technical marriage of film studios with digital technology. The new Westrex ST-6000 six-channel film recorder is now capable of running in slave mode to an X-800 digital 32-track. The film recorder can be driven by film pulses generated by hardware designed to convert timecode to this standard. Such a system enables slaving a large number of dubbing transports, projectors, and film recorders to numerous digital machines, allowing for replacement of analog dubbers with digital tracks, thus improving the overall performance of the standard film chain presently in use.

Currently in development is a new film Synchronizer/Master Controller that will enable lock-up of film transports requiring pulse rates of 2400 Quadriture down — this is reported to be the first time that studios will not have to worry about conversion-rate devices to control the transports of different manufacturers. To simplify the marriage of the film and digital worlds, the new controller will generate timecode from pulse information without having to print it, therefore saving the use of an audio track.

Other features will include RS422 and parallel interfaces, keyboard control of timecode, preset, reset, and freeze-frame control. Autolocator features with Go-To capabilities and Return-to-Zero will be standard.

MITUBISHI PRO-AUDIO GROUP

For additional information circle #310

TASCAM STUDIO COMBINED MIXER AND EIGHT-TRACK RECORDER

Comprising and eight-track open-reel recorder and an eight-channel fully assignable mixer, combined SMPTE/EBU interface, and a microprocessor-controlled Load function the Studio 8 is said to provide a production system of tremendous power and flexibility. Once a seven-inch reel of tape is threaded on the unit, the Load function ensures that the tape will never run off the reels, no matter what transport mode is in use. Thus, the tape can be manipulated with cassette-like ease while the fidelity and editing flexibility of the open-reel is retained.

SMPTE/EBU synchronizers and controllers are plug-compatible with an accessory jack fitted to the Studio 8, making it an ideal system to introduce high-quality eight-track audio to the sync-video market, at a very low cost. Also, composers working in the film or video industry, and musicians using electronic systems based on MIDI/SMPTE, will also find such a feature essential to their work.

The integral eight-channel mixer features eight program busses, and eight-channel monitor capability. The mixer also has a bus during multitrack work. Any channel of the mixer may be recorded on any or all tracks of the recorder at any time, through a unique combination of Assign and Record Function switches. An auxiliary-bus system can be used as an additional cue or effects mix, and the stereo effects mix system accommodates a wide range of signal processors. The mixer's equalizer is a three-band, sweep-type parametric system, with range 50 Hz to 15 kHz.

Return-To-Zero, Search-To-Cue, and Real-Time Counter featured on the section, which utilizes three motors with full servo-control. The unit's dbx noise-reduction system has a separate dbx defeat switch, so that SMPTE/EBU or other timecode can be recorded on track #8. Even with timecode on track #8, track #7 may be used for audio without bleed into the adjacent track.

Suggested retail price of the Studio 8 is $3,495.

TASCAM

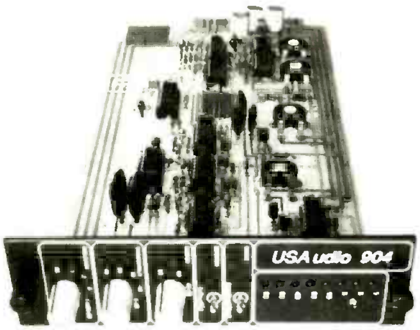
For additional information circle #311

US AUDIO UNVEILS SINGLE-CHANNEL GATEX NOISE-GATE/EXPANDER

The new single-channel version is designed to be housed in and powered by the dbx

For additional information circle #307

F-900 powered frame. In its gating mode the Model 904 employs program-dependent attack to eliminate turn-on "pop," while maintaining attack times sufficiently short to accommodate all percussion instruments. Program-controlled sustain automatically lengthens the release time as dictated by program content, thereby reducing distortion when using shorter release times.



The unit offers two expansion modes, and features an expanded eight-segment LED gain-reduction meter.

List price of the Gatex Model 904 is $250.

US AUDIO, INC.

For additional information circle #312

GOLD LINE MAD SERIES OF MULTIWAY DIRECT BOXES

Up to four active direct boxes are provided in a single-space, AC powered, rack-mountable unit, or in single- or two-channel combinations.

The MAD Series is offered in three models:

- MAD-4 comprises has four independent active direct boxes, each with a ¼-inch input, a balanced low-impedance output (male XLR), and an unbalanced, actively buffered ¼-inch output. Each channel has its own gain control for matching of system-signal levels, and a ground-life switch.

- MAD-2 is an active two-channel direct box, which is either phantom or 9-volt battery powered. A battery-status LED indicator is provided for each channel.
- MAD-1 is a single-channel active direct box identical in circuitry and features to the MAD-2.

Suggested retail prices of the MAD-1, -2 and -3 are $89.95, $174.95 and $349.95, respectively.

GOLD LINE

For additional information circle #313

MUSICWORKS UNVEILS MACMIDI SERIES OF MIDI-TO-MACINTOSH INTERFACES

Using available music-composition programs on the Apple Macintosh, the MacMIDI series of interfaces allows Mac-composed songs to be played on synthesizers and other

s. MacMIDI devices
ınect to audio, video,
ment.
vorking with MIDI·
..e, also enables the
..ed for automatically transcrip-
..on of music played on synthesizers or composed on the Macintosh into standard music notation.

The series consists of four devices:

• *MacMIDI Star* can connect a Mac to 16 MIDI channels on a star network of three synthesizers, each of which can connect to other synthesizers via MIDI-thru connections. Suggested retail price is $79.

• *MacMIDI 32* allows one Mac to play up to 32 independent MIDI channels, which can be assigned to an unlimited number of synthesizers or other MIDI devices. MacMIDI 32 also allows for input from two MIDI devices into the Mac allowing for simultaneous recording of two synthesizers or other MIDI devices. SRP is $149.

MacMIDI 32 can be upgraded to a MacMIDI Sync or MacMIDI SMPTE device by plugging in a *MacMIDI Sync/Up* board, priced at $149, or *MacMIDI SMPTE/Up* board, priced at $279.

• *MacMIDI Sync* includes all MacMIDI's features, plus it allows the Macintosh to synchronize (in/out) with drum machines, professional FSK studio equipment, and unmodified 35mm Kodak Carousel slide projectors. MacMIDI Sync is now available, SRP is $249.

• *MacMIDI SMPTE* includes all the features of MacMIDI Sync, plus it generates SMPTE timecode, allowing a Mac to be connected to external recorders for laying down music and special effects tracks on videotapes and films. SRP is $329.

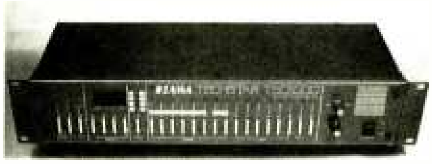
MUSICWORKS

For additional information circle #317

TAMA UNVEILS NEW INTERFACES FOR TECHSTAR ELECTRONIC DRUMS

The TTB1000 Trigger Bank enables any Techstar Voice Module to be triggered from various sources, such as acoustic drums and drum machines. The TSQ1000 is a six-channel programmable drum sequencer that lets the user create and store drum patterns, and play on top of the patterns in real time.

The TSQ1000 is compatible with the Techstar TS305, TS306, and TS200 or other voice modules, and will control up to six different



voices. Specification includes: program pattern of eight Banks by four Patchs to provide 32 Patterns, one pattern maximum four bars program; maximum 64 steps per single pattern; six-channel sequence control; Run/Stop Key, Clear Key, Mode Key (Play/Write), LED display (Bank, Patch, Inst), Step Key and LED (12/16, 24/32, 36/48, 48/64); Tempo control (40 to 300), Trigger output level switch (15/5V); and sync connector.

The TTB1000 triggers any Voice Module from acoustic drums, drum machine, keyboard, tape machine, or other audio or trigger source. Each channel accepts inputs from XLR or phone jack sources, and creates a suitable trigger output. Specifications include six-channel interface; input sensitivity 50 to +10 dBm, or -20 to +20 dBm; and output trigger level 15V to 5 V.

CHESBRO MUSIC COMPANY

For additional information circle #318

GML ANNOUNCES MIX EDITOR FOR AUTOMATION USERS

The principal feature of the new version 3.1 software is a program, Mix Editor, which provides the mixing engineer with the capability of manipulating mixes much the same as a word processor enables a writer to manipulate text.

It operates by presenting the mixer with a short menu from which editing operations may be selected. Such operations include:

• Merge: Combine portions of any two mixes.
• Splice: Copy a portion of one mix to a later or earlier part of the same mix. (A typical application of this would be copying the "perfect mix" for one chorus of a song to the second and third choruses, etc.)
• Copy: Copy the data on one channel to any other channel.
• Clear: Erase selected data from one or more channels.
• Swap: Swap the data between any two channels. (This feature is said to be extremely useful in complex mixing sessions where one module of the console develops sudden audio problems. The swap command allows the user to simply move all of the mix data from the bad module to any other working, unused module.)
• Trim: Add or subtract gain (specified in dB) to selected channel faders, for any portion of a mix.
• Extract: Extract all desired data from a mix, erasing the rest (the inverse operation of "clear").
• Mix shift: Move the entire mix backwards or forwards in time (as related to the SMPTE timecode on tape).
• Channel shift: Move the data from one or more channels backwards or forwards in time. (This feature is said to have been requested by a client who frequently mixes certain tracks by "riding" the faders while listening, and desired a method of correcting for the slight time lag caused by mixing "on-the-fly" in this manner.)

All commands (except Mix Shift) operate by asking the user to specify parameters of edit operations in a simple and logical syntax. The parameters are: the list of channels to be processed in the edit operation; SMPTE timecode numbers representing the portion of a mix to be edited; and the automated functions to be processed, such as faders, mutes (and other switches), or others.

The Editor also includes a "help" facility, whereby the user can hit the Help key at any time, and receive concise on-screen information explaining the current operation.

DAVID SMITH

The update to Version 3.1 is provided free of charge to all existing GML installations.

GEORGE MASSENBURG LABS

For additional information circle #319

EMILAR ANNOUNCES MODEL EL-15J AND -12J BASS DRIVERS

The new 15-inch EL-15J will handle 500 watts of continuous program material, and offers an average sensitivity rating of 104.5 dB SPL, 1 watt/1 meter over the frequency band of 200 to 800 Hz; overall frequency range is 20 Hz to 2 kHz.

Intended for use in live-performance sound reinforcement systems and high-level music playback systems, the EL-15J is said to provide sound-system designers and users with a new standard in quality sound reproduction.

The new EL-12J 12-inch driver has a rated power capacity of 400 watts continuous program material, and a 100.5 dB SPL average sensitivity over the 200 to 800 Hz bandwidth, 1 watt measured at 1 meter. Usable frequency range is 20 Hz to 3 kHz.

EMILAR

For additional information circle #320

SOUND TECHNOLOGY 3000 SERIES AUDIO TEST EQUIPMENT

The 3000 Series consists of two separate components: the 3100A generator and the 3200A analyzer. Both microprocessor-based instruments feature front-panel programmability that allows storage of extensive automated test sequences. Both also feature the ability to communicate test data through the audio line being tested via an exclusive Frequency Shift Keying (FSK) technique, allowing unmanned, automated remote transmission-line testing without the need for external computers and modems.

Test results are achieved in less than 60 seconds, and may be graphed on a standard printer or plotter. If desired, the 3000 Series functions may also be controlled via RS-232C or GPIB (IEEE-488) communications ports.

The two-channel, electronically balanced and floating 3100A generator outputs sinewaves, squarewaves, IMD, toneburst and sine-step waveforms. The two-channel companion 3200A analyzer will measure level, noise, frequency, harmonic distortion, inter-

New Products

modulation distortion, phase error, channel separation and quantizing noise (digital data).

Pro-user prices are: 3000A (single main-frame); $8,950; 3100A audio generator $3,950; and 3200A audio analyzer $5,350.

SOUND TECHNOLOGY

For additional information circle #324

COMTRAN ANALOG FILTER/ AMPLIFIER DESIGN PROGRAM FROM JENSEN NOW RUNS ON HP-300 AND HP-217 COMPUTERS

The Software division of Jensen Tranformers has announced that its COMTRAN program, which includes advanced AC circuit analysis with optimization, group delay, time domain and integrated measurement capability, has been ported to the Hewlett-Packard 300 Series running BASIC 4.0 and to the Model 217, and also to other HP machines running BASIC 3.01.

The program, which consists of four modules, is intended for circuit modeling with desktop computers, and enables an engineer to quickly and thoroughly simulate and optimize the design of analog circuits. COMTRAN now runs on HP 300, 217, 9836, 9816, 9920, 9845, and 9020 computers; and it is described as being 50 times faster than the HP AC Circuits program.

Features include: optimization of active or passive circuits with up to 98 nodes; calculation of group delay and relative phase; calculation of output waveform given any circuit and any input waveform; study of circuit behavior without building a prototype; combination of measurements with circuit models; and synthesis active filter circuits using fewer op-amps.

JENSEN TRANSFORMERS, INC.

For additional information circle #325

VALLEY PEOPLE MODEL 440 COMBINED LIMITER, COMPRESSOR AND DYNAMIC SIBILANCE PROCESSOR

The Model 440 is a single-channel device offering the convenience of a peak limiter, a high quality compressor/expander package, and a Dynamic Sibilance Processor section, each controlling a common VCA. Intercoupling of the control circuitry used for each function allows the device to simultaneously limit, compress, expand, and eliminate high-frequency components in sibilance.

The unit's compressor control section features continuously adjustable threshold, attack time, ratio and release time. In addition, an interactive expander control is integrated with the compressor control circuitry to reduce residual noise that otherwise would be "pumped up" or accentuated by the compression process. Special release coupling is said to make the transition from compression to expansion imperceptible, thus eliminating problems associated with the use of separate single-function units.

The limiter control section exhibits extremely fast attack characteristics — typically 1 microsecond per dB or less — continuously variable threshold, a fixed 60:1 ratio, and variable release time.

VALLEY PEOPLE, INC.

For additional information circle #326

AURATONE RT5-S AND RTS-6 RACK-MOUNTABLE MONITORS

Both systems have baffles with components mounted on the center axis, so that one unit can be inverted to form a symmetrical stereo pair. The modified polypropylene woofers are shielded to prevent flux leakage that might distort the image of an adjacent CRT.

The RT5-S features a ¾-inch wide dispersion, polyamide dome tweeter and a 12 dB per octave crossover network. Frequency response is a quoted ±3 dB from 70 Hz to 20 kHz, with power handling of 40 watts RMS. Impedance is 6 ohms. The enclosure occupies 5.25 inches of vertical rack space: width is 16.5 inches and depth 8.5 inches.

The RT6-S acoustic design is derived from the T6 Sub Compact Two Way monitor, with the same one inch soft-dome tweeter, 12 or 8 dB per octave crossover network and nearly identical Thiele-Small woofer parameters, except that the magnet structure is heavier and shielded against flux leakage. Frequency response is a quoted ±2.5 dB from 60 Hz to 20 kHz, with 50 watts RMS power handling. Impedance is 8 ohms, and enclosure measurements 8.75 by 16.5 by 9.5 inches (H×W×D).

Suggested pro-net pricing is $165 each for the RT5-S, and $180 each for the RT6-S.

AURATONE CORPORATION

For additional information circle #327

LAKE EXPANDS CAPABILITIES

LAKE'S audio systems engineers Dennis Smyers (foreground) and Steve Blake analyze data on the TEF System 10

What's new at LAKE? Besides the influx of new people . . . A host of new computer systems. Computers that assist in the design, engineering, drafting, and service of audio/video systems. One of the most exciting new computer systems is the audio departments Tecron TEF System 10. A portable audio spectrum analyzer that can be used in the field and the data brought back to the office for further analysis.

LAKE is involved in the design and building of television stations, recording studios, post production editing systems, and sound reinforcement systems worldwide. A computer system that could quickly analyze the acoustic parameters of any space was very important to the engineering department. They are currently using the TEF 10 to help expedite the engineering requirements of an expanding customer base.

An example of its value was recently discussed at a meeting I attended. It seems that microphones placed at a specific area on stage were experiencing excessive feedback. The client had tried a number of corrective measures to no avail. LAKE's engineers, using the TEF 10 were able to pinpoint the problem, something that at first glance seemed insignificant, a steam pipe located near the speaker cluster was causing a strong reflection into the problem area. Covering the pipe with absorbent material, eliminated the problem.

Without a doubt, this type of commitment on the part of LAKE in R & D, positions them as the systems company of choice in the audio field. Contact them at (617) 244-6881.

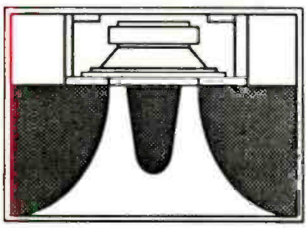
For additional information circle #322

Another In A Series Of Application Notes On Sound System Design From EAW.

Lower Midrange Reproducers

Subject: The EAW MR Series: a complete range of mid-bass horns.

The Eastern Acoustic Works MR Series Mid-Bass Horns are the only systems to offer increased output capabilities without sacrificing distortion, coverage or response linearity. Since the introduction of the MR102 in 1978 it has become the standard on six continents for high quality music reproduction systems.



Features:

- Flat frequency response +- 2 dB on axis over entire operating range (no other mid-bass horn comes near).
- Flat power response less than -6 dB deviation from on axis response up to 45 degrees off axis, over the entire operating range.
- Complex construction using high density polyurathane foam reinforcing cross-grain-laminated birch , for absolute acoustical integrity and freedom from output robbing resonances.
- Kenton Forsythe designed throat displacement plug enables unmatched accuracy.
- Complete range includes units for the following bandwidths: 150 Hz to 1.8k Hz, 250 to 1.8k Hz, and 275 Hz to 2.2k Hz.
- Each system is complete with specially designed mid-bass RCF driver, road enclosure, handles, and hardware.

Price:

MR142L $810
Extended low frequency mid-bass system with RCF LAB L12/P11W 300mm driver, 150 Hz to 1.8k Hz, 109 dB SPL 1w @ 1m, 13.5 x 24.6 x 19.75

MR102L $665
High output horn loaded mid-bass system with RCF LAB L12/P11W 300mm driver, 250 Hz to 1.8k Hz, 109 dB SPL 1w @ 1m, 18.5 x 29.75 x 24.6

MR101L *NEW* $525
Extended high frequency mid-bass system with RCF L10/750 250mm driver, 275 to 2.2k Hz, 108 dB SPL 1w @ 1m, 13.5 x 24.6 x 19.75

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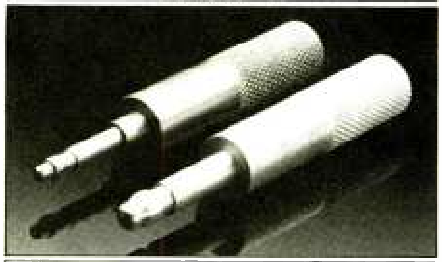

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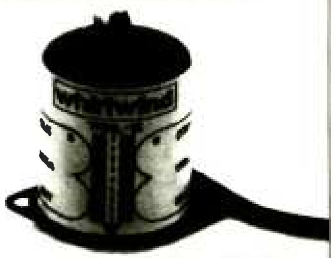

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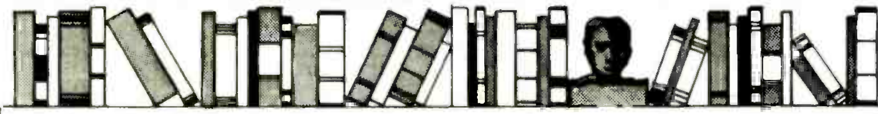

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— continued from page 37 . . .

duct. Moderator was Richard Vincent, sales engineer, Eastern Region, ElectroSound, Inc., and panel members included Joe Ciccone, assistant sales manager, King Instrument, Bob Coningsby III, sales executive, Apex Machine Company, John Arnold, general manager, specialty tape products, W.H. Brady Company, and Tim Wilsey, sales and marketing, Polymatrix. Ciccone suggested the use of bar-coding for cassette identification as a more aesthetic process. When asked what the customer looks for when buying prerecorded cassettes, Wilsey replied that part of the whole marketing package was how good the package looked. He called for a "need to work on prettiness" and not taking anything away from the artist; "if the package looks good, it will sound good." When the panel was asked what changes, if any, would result in future cost savings, Coningsby stated that Apex was working on a new ultraviolet dryer that would increase the speed of the printing process and, over the long term, cut costs. It was also stated that more efficient auto-assembly equipment could contribute to a long-range savings.

Mike Jones, audio consultant from Mike Jones & Associates, was session chairman for day three. The first presentation, "Elements of a Basic Quality Assurance Plan," was given by John Matarazzo, national technical manager of Agfa-Gevaert. The first panel discussion, "Recording Standards and References," was moderated by David Bowman, senior VP of ElectroSound; panel members included Daniel Gravereaux, associate director, CBS Technology Center, Rick Wartzok, audio engineer, RCA Records, Jay McKnight, president, Magnetic Reference Laboratories, and Walter Derendorf of BASF. Questions were fielded from the audience regarding current standards.

"Where's the Gap?", the afternoon presentation by John French Jr., president of JRF Magnetic Sciences, comprised a detailed look at the design and construction of recording and playback heads, and was followed by a panel discussion on achieving high quality from them. David Bowman moderated, while panel members John French Jr., Gene Sakasegawa, president of Saki Magnetics, and Bob Reiss, VP of technical operations at Sprague Magnetics, discussed head wear, maintenance, and answered a variety of questions from the floor.

All in all, the ElectroSound seminar can be considered a huge success, if only in providing an excellent forum for all parties involved in the duplication process — from producers and artists, to tape manufacturers — to come together and discuss their mutual requirements for high quality pre-recorded cassettes.

According to David C. Bowman, VP of ElectroSound, "we are gratified by the enormously positive comments that we received during the planning stages.

From the feedback received during the proceedings, the time and effort expended was well worth it." The company plans to hold a similar meeting some time next year; dates and location will be published in *R-e/p* as soon as they become available.

News

OPERA THEATER OF ST. LOUIS BROADCASTS IN TWO-CHANNEL AMBISONIC SURROUND SOUND

Four operas have been recorded Ambisonically by KWMU-FM, the local NPR affiliate, for national distribution via satellite to the NPR Network throughout the U.S. Taped digitally before a live audience this past June, the programs feature world premiere performances of Minoru Miki's *Joruri* and Stephen Paulus' *The Woodlanders* (based on the novel by Thomas Hardy), as well as *The Barber of Seville* by Rossini, and Mozart's *Idomeneo*.

According to production director, Barry Hufker, "Opera Theatre is very creative in the use of their performing space with action — not only on the stage but also in and around the audience. We wanted to recreate that same experience for our listeners, and Ambisonics is the perfect way to achieve this."

MITSUBISHI X-80 DIGITAL TWO-TRACK NOW AVAILABLE FOR RENT FROM GERR ELECTRO-ACOUSTICS

The Canadian head office for Digital Entertainment Corporation, the North American representative of the Mitsubishi Pro-Audio Group, has already rented the X-80 to several well-known Canadian bands, including Rush, Triumph, Parachute Club, Rational Youth and New Regime. Recording studios such as Sounds Interchange, Manta, Metalworks, McClear Place, ESP, Round Sound, Capitol Records and CBS Records have also rented the X-80.

Gerry Eschweiler, sales manager for GERR, says: "The Mitsubishi X-80 has become an industry standard in the U.S. and Europe for the digital mixing of master tapes. At GERR, we want to make that standard available to the Canadian recording industry."

NEW COLOSSUS FOUR-TRACK DIGITAL RECORDER FROM BY THE NUMBERS TO BE UNVEILED AT NEW YORK AES

The new recorder is said to incorporated a significant development in multichannel digital recording technology, and which can be implemented at costs far less than those associated with present digital standards. By The Numbers is a joint venture between Louis Dorren, and associates, and Brad Miller. Dorren is the inventor of the Dorren/Quadracast four-channel FM broadcasting standard approved by the FCC. Miller is well known as a producer of the Mystic Moods Orchestra series, and numerous environmental recordings for record and film. The company has secured the services of John Eargle and JME Consulting Corporation in the areas of market planning, development, and product licensing.

The proprietary code developed by Dorren will first be embodied in a product, trade-named Colossus™ — a portable unit capable of recording four channels of 16-bit audio with bandwidth in excess of 25 kHz. Dorren emphasizes that the proprietary PCM code makes no use whatever of data compression.

JOINT VENTURE ANNOUNCED BY AMEK AND GML

GML, Inc. of Los Angeles and AMEK, Ltd. of Manchester, England, have formed a new corporation to research, develop, manufacture and market a large architecture, fully automated Virtual Master Recording and Mixdown Console.

The new company will be incorporated in England as AML, Ltd. (AMEK/Massenburg Labs), with design and manufacturing responsibilities performed in England, and design, programming and prototyping responsibilities performed in Los Angeles.

. . . continued overleaf —

News

Introduction for the new console is set for the NAB Convention in spring of 1986. First deliveries are scheduled the following month.

OCEAN AUDIO PLACES TWO NEW CONSOLE RENTAL PACKAGES; ANNOUNCES SALES

• Debernedetto Recording, Bridgeport, CT, has rented with an option to purchase a 56-input Neve 8108 console, formerly in use at Abbey Road Studios, London. The new facility offers both scoring and music-recording capabilities.

• Cougar Run Studio, Incline, NV, a resort studio located in the Tahoe basin, has rented with the option to purchase a 48-input Neve 8108 with NECAM I automation.

• Clinton Recording, New York City, has purchased a 40-input Neve Model 8078 with NECAM II, formerly located at Studio 301, Sydney, Australia.

• House of Music, Clifton, NJ has purchase a 40-input Neve 8078 console formerly as Nova Studio, London.

• Smoketree Ranch, Chatsworth, CA, has purchased a 49-input 8078 formerly at Sound Holland.

• London Bridge Studio, Seattle, WA, recently took delivery a 32-input Neve 8048 formerly at Polygon Studio, France.

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NEW COMPANY FORMED TO MARKET DISKMIX CONSOLE AUTOMATION

Michael Tapes, president of Sound Workshop Professional Audio Products, has announced the formation of Digital Creations Corporation (DCC) to develop and market computer-based audio-production products. DISKMIX (Release 2.0) is the first such product, being introduced at the New York AES Convention.

All DCC product will be developed in partnership with Paul Galburt, president of SWI Engineering. The Tapes/Galburt team was responsible for developing the series of Sound Workshop consoles.

The new release of DISMIX typifies the future product direction for Digital Creations, Tapes says. A custom-designed, "computer-on-a-card" that mounts inside of an IBM PC or most compatibles, DISKMIX handles only system-specific functions such as time-code and console communications. the more generic user-interface and disk processing is handled by the PC and its industry-standard operating system.

Full product information is being released at the NY AES, with demos at booth #410. Digital Creations Corp. is located at 1324 Motor Parkway, Hauppauge, NY. (516) 582-6229.

— People on the Move —

• **LEE POMERANTZ** has been promoted to the position of sales manager at **SOUND WORKSHOP PROFESSIONAL AUDIO PRODUCTS, INC.** Over the past three years he has held the positions of quality control/customer service manager, and most recently served as technical product manager. Prior to joining the company, he was chief engineer at The Workshoppe Recording studio, Douglaston, New York. In addition, **MICHAEL CUNEO** has been promoted to operations manager, from his previous position as head of material control and purchasing for console production.

• **JIM RONDINELLI** has joined **SOUND GENESIS**, the San Francisco-based pro-audio dealer and service center, as sales representative for accounts in the corporate studio market. He joins the company from the University of Iowa, where he was awarded a BA in business administration.

• **WALTER J. KELLEY** has been named vice-president of audio/video sales at **LAKE SYSTEMS CORPORATION**, a builder of audio/video systems based in Newton, MA. Kelley has been with the company since 1972, and most recently served as audio/ video sales manager.

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.

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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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For additional information circle #338

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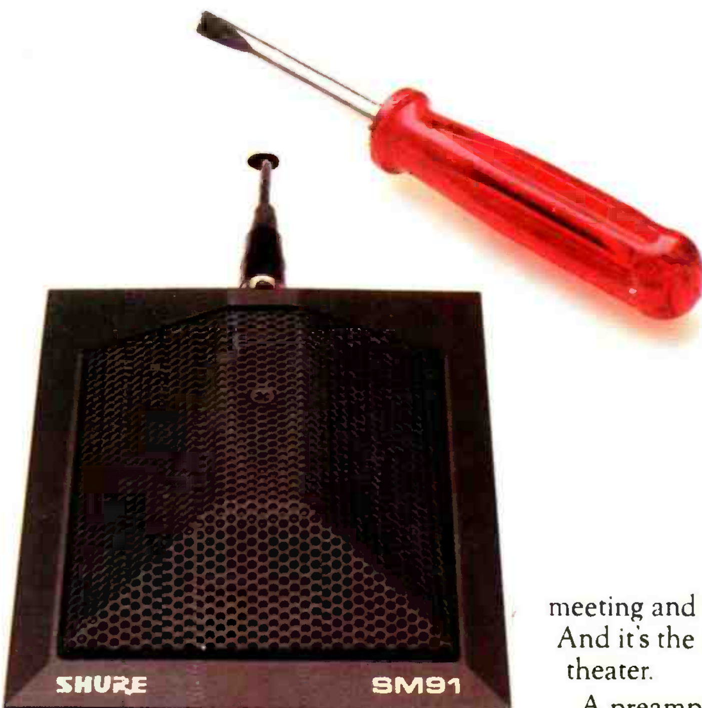

No carpet strips or plastic baffles needed. Until now, all surface-mounted mics have been omnidirectional. Trying to add directionality has required a lot of busy work. The new SM91 brings the big advantages of unidirectionality to boundary effect microphones by incorporating a condenser cartridge with a half-cardioid pattern that isolates the speaker from surrounding noises.

The new smoothie. The sleek SM91 delivers wide-band, smooth response throughout the audio spectrum, while greatly reducing the problems of feedback, low-frequency noise and phase cancellation.

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A preamp ahead of its time. The ultra-low noise preamplifier provides switch-selectable flat or low-cut response, excellent signal-to-noise ratio and a high output clipping level. A low-frequency cutoff filter minimizes low-end rumble—especially on large surfaces.

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